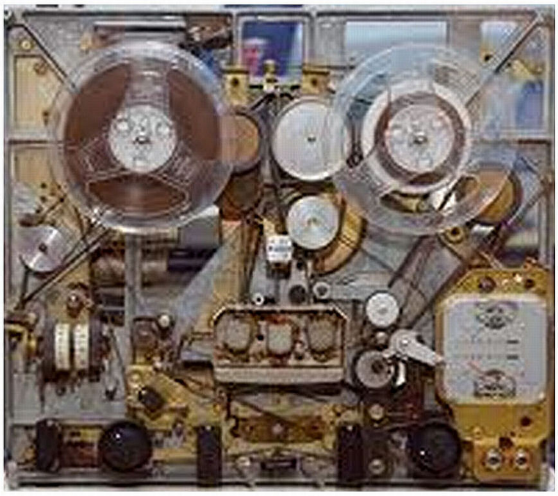
Tape Recording Techniques

*Articles from **Audio** and **Audio Engineering** 1950 -1990*



NAB Recording Standards Meeting

THE NATIONAL ASSOCIATION of Broadcasters' Committee on Recording and Reproducing Standards, quiescent since 1942, met in Atlantic City on September 16 to reorganize for postwar activity. It will be recalled that this Committee, originally organized in 1941, had by 1942 adopted sixteen industry-wide standards for disc recording.

In response to industry requests Royal V. Howard, NAB Director of Engineering, called for a reorganization meeting, held during the NAB Convention. About fifty attended, including a number of foreign delegates to the I.T.C. meetings. R. M. Morris, Chairman of the Executive Committee before the war, agreed to remain active long enough to reorganize the executive and sub-committees. This reorganization was necessitated by the fact that many company representatives had shifted to non-recording work in the postwar era and were no longer available for committee work.

Mr. A. E. Barratt of the BBC discussed the results of a meeting of British magnetic recorder manufacturers and the BBC engineers last May. These preliminary standards are tied to those of the German magnetophon, because of the number of such machines already in use on the Continent. He gave details, which were as follows:

Tape width = 0.245" + .005", − .000"
Tape thickness, coated = .002" ± .0001"
Tape stretch = 1.08 to 1.27% at 250 grams load
Tape output variation = not over +3 db on any one spool or group of spools
Spools = center boss of 4" minimum diameter, with one flange, is recommended. The magnetophon spool, with no side plates, is not satisfactory.
Tape speed = 77 cm per second
Speed constancy = ± .5% over long period
= ± .1% instantaneous variation
Rewind time = 2½ minutes, for 21-minute spool

The Committee voted to have Mr. Morris organize a new subcommittee, on tape standards.

There was some discussion of disc standards adopted in 1942. Mr. Barratt said that the NAB pre-emphasis characteristic was more extreme than the BBC could use with its present equipment, for the tendency to overload on higher frequencies was too great. They use a curve with 10 to 12 db rise at 8000 cycles.

Mr. Theodore W. Lindenberg felt that judging from the sound of present transcriptions, 100-microsecond pre-emphasis was too much, and tracking distortion was excessive. Mr. Miller said that 35 to 40 microsecond pre-emphasis had been tried, and found good. Perhaps NAB pre-emphasis was set too high; it might be preferable to reduce it, with a resultant increase in surface noise of perhaps 2 db. Two db more surface noise was negligible, beside the benefit from the resulting reduction in tracking distortion. There was considerable discussion of 50, 75, and 100 microsecond pre-emphasis values, and of the resulting cost if each station had to purchase a new equalizer. Mr. Lawrence Ruddell said that the cost would be of negligible importance if better results could thereby be secured. He would favor the change if significant improvement could be shown.

Mr. C. C. Rieskind said that he was using the same crossover frequency for phonograph records as specified in the NAB characteristic. A recent conference of visiting EMI engineers with a group of leading American commercial recording studios had shown crossovers ranging from 250 to 600 cycles in use in this country. A single crossover for all phonograph records would give the stations much more uniform reproduction.

Considering all these remarks, the Committee decided to study both 78 and 33 1/3 rpm recording characteristics. Mr. Howard hoped that the subcommittees could get their work done in time for a full Committee meeting at the IRE Midwinter Meeting in March, 1948.

The meeting then adjourned.

Members Participating

Members participating in the Recording and Reproducing Standards Committee meeting were: F. L. Allman, President and General Manager, Radio Station WSVA, Harrisonburg, Va.; L. J. Anderson, RCA, Camden, N. J.; Ross Beville, Chief Engineer, Radio Station WWDC, Washington, D. C.; Warren L. Braum, Chief Engineer, Radio Station WSVA, Harrisonburg, Va.; Daken K. Broadhead, Manager, Allied Record Manufacturing Company, Hollywood, California; W. J. Brown, Bell Telephone Laboratories, New York City; J. H. Capp, General Electric Company, Syracuse, New York; E. P. Carter, Poinsettia Company, Pitman, N. J.; Robert J. Coar, Coordinator, Joint Senate and House Recording, Washington, D. C.; John D. Colvin, Audio Facilities Engineer, American Broadcasting Company, New York City; A. N. Curtiss, RCA, Camden, N. J.; W. H. Deacy, Jr., Staff Engineer, American Standards Association, New York City; O. B. Hanson, Vice-President and Chief Engineer, National Broadcasting Company, New York City; Fred. de Jaager, Empire Broadcasting Corporation, New York City; C. M. Jansky, Jr., Jansky & Bailey, Washington, D. C.; J. G. Lawrence, Radio Division, Western Electric, New York City; C. J. LeBel, Vice-President, Audio Devices, New York City; Theodore Lindenberg, Fairchild Camera and Instrument Corporation, Jamaica, New York; William B. Lodge, Director of General Engineering, Columbia .Broadcasting System, New York City; Frank Marx, Director of General Engineering, American Broadcasting Company, New York City; Carl Mayfield and H. P. Meisinger, U. S. Recording Company, Washington, D. C.; R. A. Miller, Bell Telephone Laboratories, Murray Hill, N. J.; George M. Nixon, National Broadcasting Company, New York City; Robert Z. Morrison, National Broadcasting Company, New York City; J. F. Palmquist, Chief Audio Engineer, Radio Corporation of America, Camden, N. J.; R. H. Ranger, President, Rangertone, Inc., Newark, N. J.; Oscar W. B. Reed, Jr., Jansky and Bailey, Washington, D. C.; H. I. Reiskind, Chief Engineer, Record Department, RCA, Camden, N. J.; Arthur F. Rekart, Chief Engineer, Radio Station KXOK, St. Louis, Missouri; H. E. Roys, Advanced Development Engineering Section, RCA, Camden, N. J.; L. A. Ruddell, American Broadcasting Company, New York City; G. J. Saliba, President & Chief Engineer, Presto Recording Corporation, New York City; C. R. Sawyer, ERD Division, Western Electric, New York City; H. F. Scarr, Radio Division, Western Electric, New York City; R. A. Schlegel, WOR Recording Studios, New York City; H. H.

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A report on the first postwar meeting of this NAB committee

NAB Recording Standards

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Scott, H. H. Scott, Inc., Cambridge, Mass.; Alex Sherwood, Standard Radio Transcriptions, New York City; K. R. Smith, Vice-President, Muzak Corp., New York City; H. F. Tank, Radio Station WWJ, Detroit, Mich.; O. W. Towner, Chief Engineer, Radio Station WHAS, Louisville, Ky.; L. Vieth, Bell Telephone Laboratories, New York City; A. E. Barrett, BBC, New York City; Angles d'Auriac, International Broadcast Organization, Brussels; Roy Cahoon, Canadian Broadcasting Corp., Montreal, Canada; W. T. C. Dowding, Engineer, Toronto Studios, RCA Victor, Toronto, Canada; Raul Lopes Duarte, Chief Engineer, National Portuguese Broadcasting System; L. D. Headley, Mgr., Radio-Recording Division, RCA Victor, Toronto, Canada; W. Hilarius, Chief Engineer, South African Broadcasting Co.; Ing. Raoul Karman, RHC Network, Havana, Cuba; Leonid A. Kopytin, Department Chief Engineer, Ministry for Postal and Electrical Communications, USSR, Moscow, Russia; and M. L. Sastry, Station Engineer, All India Radio.

Limitations of Magnetic Tape

W. S. LATHAM*

While thoroughly satisfactory for ordinary musical and program recording, magnetic tape often has minor defects which become apparent when used for certain scientific applications.

THE RAPIDLY EXPANDING use of magnetic tape as a convenient and efficient medium for recording and storing sound energy is familiar to nearly everyone associated with the sound recording industry. Commercial recording organizations, broadcasting companies, and the film industry have recognized the economy, adaptability, and versatility of magnetic tape. Consequently, its development has been accelerated during the past few years in an attempt to keep pace with the growing use of this medium.

Because of the acceptance of their product and the subsequent increased demand, the manufacturers of magnetic recording tape have been faced with several problems. One of these has been the necessity to supply a better product which would be available in sufficient quantity at an acceptable price. The production of a uniform tape capable of high-fidelity recording and reproduction of voice and music has been achieved. Over a very brief period of time, great advances have been made in the development of oxides, binders, and backing materials used in magnetic tape.

From the beginning of its development, the versatility of magnetic tape indicated that its use would not long be confined to strictly commercial applications. Here was a new tool available for scientific applications in fields of research almost completely unrelated to the original purpose of the medium. Soon new requirements were placed on both the physical and magnetic properties of recording tape. These requirements exceeded the capabilities of existing tapes and could only be satisfied by modifications in the manufacturing process. Such modifications would necessitate additional capital expeditures on the part of tape manufacturers, and this capital could be recovered only by increasing the price of an item already acceptable to the majority of users. Quite understandably, the manufacturers have been unwilling to resort to this procedure. Consequently, until the critical users of tape constitute a substantial market for a superior product, they are forced to accept the available tape and cope with the limitations of the medium in its present state of development.

Such limitations confront the Recording Branch of the U. S. Navy Underwater Sound Laboratory whenever it is requested to supply special instrumentation involving the design and development of unique tape recording and reproducing devices. For economic reasons these devices normally employ commercial magnetic tapes. Because the Laboratory places demands on these tapes in excess of those established by the recording industry generally, the Recording Branch has found it necessary to undertake a continuing investigation of all currently available magnetic tapes in order to determine their performance limitations. Significant phases of the investigation are described in this article.[1]

One of the major problems encountered in the use of commercial tapes is the appearance of "holes" or nodules in the envelope of the signal energy recorded on the tape. Although these holes are of no great concern when complex wave structure is being recorded, they become an important factor whenever a carrier-modulated signal is recorded. Contrary to a general belief that such holes are confined to the high-frequency portion of the tape-recorded spectrum, the Laboratory's investigation has proved that the holes are present on the tape regardless of the spectrum content involved. In fact, their presence may be observed by viewing through an oscilloscope the playback of a completely erased tape.

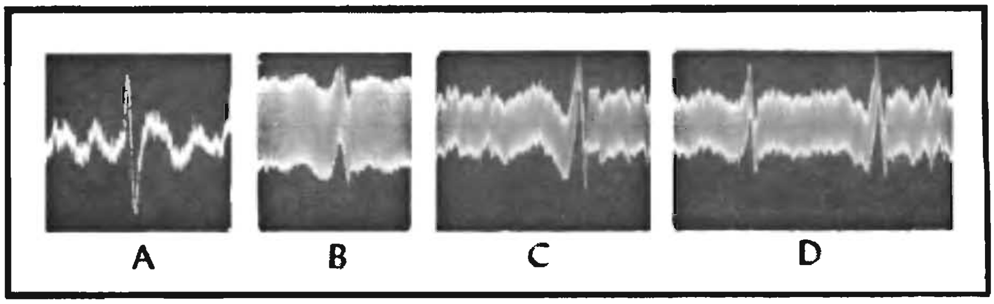

Fig. 2. Oscillograms of transients: (A) Transient as it appeared on erased tape after five plays; (B) effect of this transient on a 15-kc envelope; (C) effect of the same transient on a 2-kc carrier; (D) the transient on the left was produced by a tape hole—the one on the right was the desired signal.

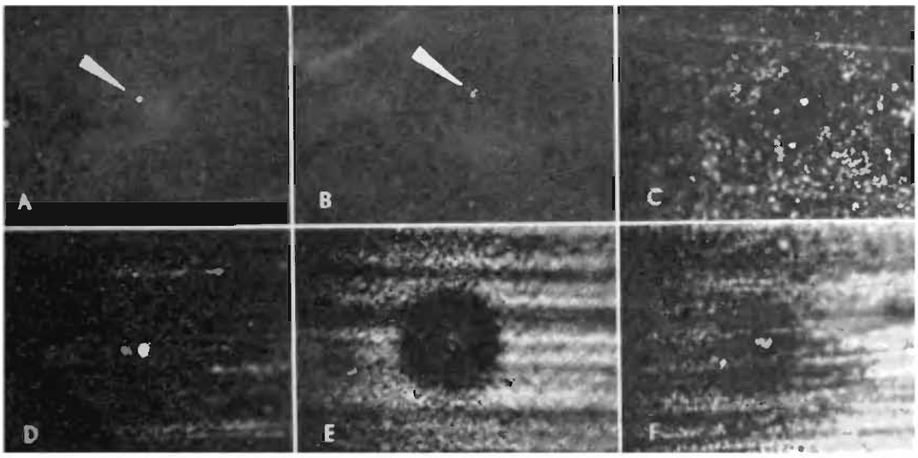

Fig. 1. Microphotographs showing the development of a typical nodule in magnetic tape: (A) Pinhole in virgin tape; (B) tape sample including the hole after one passage over the head assembly; (C) same sample after five passages; (D) after 12 passages; (E) after 1000 passages; and (F) after 1800 passages.

Change of Pin-hole Characteristic

A sample of virgin tape when first transported over the heads of a tape machine was found to exhibit fewer holes than those observed in tape previously used. When the same sample was erased and again passed over the

* *Recording Branch, General Engineering Division, U. S. Navy Underwater Sound Laboratory, Fort Trumbull, New London, Conn.*

[1] The author acknowledges the competent assistance of Mr. C. R. Turner, of the Recording Branch, in conducting many of the tests described and in compiling the associated data, and the painstaking care exercised by the Laboratory's Photography Laboratory Section in making microphotographs and oscillograms.

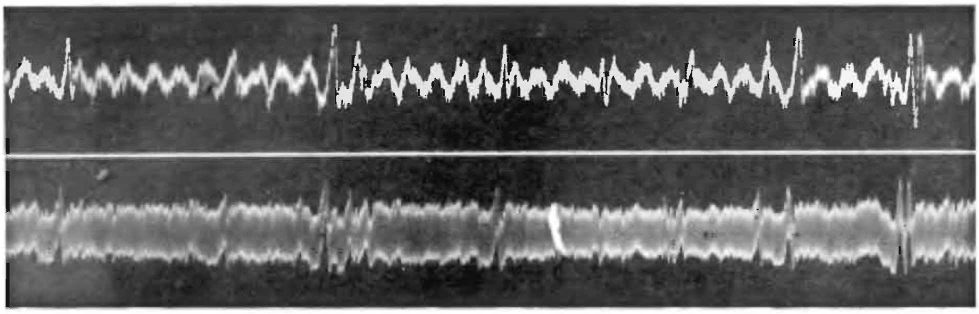

Fig. 3. Effect of tape holes on signal energy: (A) above, inherent noise on erased tape—the high peaks represent holes; (B) below, noise on the same strip of tape modulating a 15-kc carrier.

same heads, an increase in the number of nodules was noted. This brief test indicated the possibility that the supersonic erase and bias energy might have some effect on the phenomenon, but variations in the frequency and amplitude of these energies produced no effect upon the number of holes present in the sample tape. Similarly, variations in the amplitude of the signal energy applied to the medium appeared to be of little consequence in establishing these holes. The possibility that the physical configuration and the surface geometry of a particular recording head might have a bearing on the nodule count of the tape under investigation was eliminated when comparable results were observed on a recorder which used unlaminated head pole-piece construction. Thus, it was apparent that the hole phenomenon is a function of the medium itself. In other words, the holes are present in the virgin tape, but their number increases and their configuration changes with re-use of the tape. This condition is exaggerated so long as the tape remains in motional, frictional contact with a surface, regardless of its state of polish. However, the process reaches a limit beyond which little change is observed in the resulting effect, even though the configuration continues to enlarge.

The development of a nodule is demonstrated in *Fig.* 1. A reel of virgin tape taken at random from stock was examined under a microscope until an appreciable pinhole was found in the oxide coating. A sample including this hole was cut from the unused tape. The abrasive marks which may be seen at (A) resulted from scoring introduced as the tape passed over rollers, probably during the drying process. At (B) the same tape sample has passed once over the head assembly. Although no apparent change has occurred in the configuration of the hole structure, the presence of more lines on the oxide surface indicates the abrasive effect of tape-head contact. This effect is important in the later stages of development of the hole.

At (C) may be seen the same tape sample after it passed over the head assembly five times. Now, in addition to the presence of increased abrasive action, an accumulation or "clumping" of oxide is noticeable in the area surrounding the original ! ole in the magnetic coating. After twelve successive passages of the tape over the head assembly, the hole structure developed to the extent shown at (D). Repeated transport of the tape over the head was continued until, at the conclusion of 1000 passages, the same hole had developed to the extent indicated at (E), which illustrates effectively the progress of the "clumping" process. Excessive abrasion has taken place around the nodule, which now resembles a small crater when viewed from above. The accumulation of the loose oxide particles about

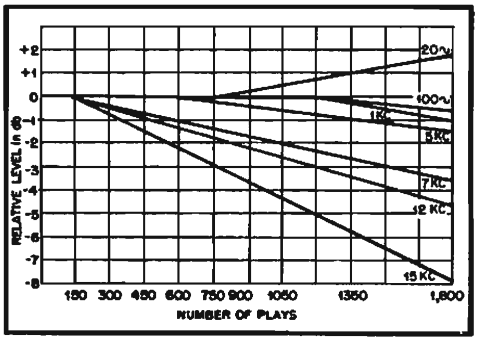

Fig. 4. Behavior of playback level of tape samples containing individual sine-wave components during 1800 replays.

the original irregularity in the oxide surface has reached a point which might be termed saturated. Continuous passage of the tape over the heads beyond this point led to a gradual deterioration of the oxide ring until the condition illustrated at (F) was reached after 1800 successive replays. Although the tape surface exhibits extreme physical wear and the oxide ring has been vastly reduced, the pinhole is still visible. Its presence, moreover, was still apparent through its ability to produce a pronounced transient.

Effect on Playback

An oscillogram of the transient as it appeared on erased tape after the first five plays is shown at (A) in *Fig.* 2, and the effect of this nodule on a 15-kc envelope is illustrated at (B). It should be noted that the amplitudes of the transient excursions are virtually equal. The irregularities in the unadulterated portions of the carrier envelope resulted from poor head alignment in azimuth on the particular machine used. At (C) is shown a 2-kc carrier being affected by the same tape hole.

It is of interest to consider how the conditions exhibited at (B) and (C) would affect the interpretation of the results if these same carriers were being modulated by strain gauges or by other transient-producing pickup devices. To distinguish between the desired signal and the undesired pulse, in this case, would be a difficult task, as can be seen from (D). Here, the transient on the left was produced by a tape hole; the disturbance on the right was the desired information in the form of a pulse modulating a 15-kc carrier. Under uncontrolled conditions, recognition would be practically impossible, since both the hole and the signal modulation occur at irregular intervals. The frustrating effect on signal recognition produced by holes in the tape is further illustrated by *Fig.* 3. Here can be seen the same length of tape first as erased and then as recorded with a signal of 15 kc. Even the minor nodules create an unwanted effect.

A spectrum analysis of the transients created by these holes in the magnetic structure of the tape revealed that the predominant energy was confined to the region of 100 cps. The amplitude of the excursions of these pulses averaged approximately 5 db above the steady-state condition, with occasional peaks approaching 15 db. Under certain conditions, these levels are sufficient to mask the carrier frequencies completely.

All the various types of magnetic recording tapes which are currently available (red and black oxides on plastic and paper) exhibit the hole effect. This also holds true for impregnated tapes as well as coated tapes. Hand selection of tapes does not seem to be the complete answer since, as has been illustrated, holes which are not readily apparent in virgin tape will develop through replay and re-use of the tape.

While the tape tests were being conducted, samples of early as well as current production runs of tape were examined. A tape sample of the type first produced and distributed by one manufacturer displayed noticeably fewer holes per foot than did any of the other samples tested; samples from the same manufacturer's current production were also included. This manufacturer was contacted in an effort to determine the factors which might account for this distinction. A significant factor seemed to be that the oxide for the original tape was in the mixing barrel or bonderizing process for a period of 10 days prior to application to the base material, while under the present practice this operation was continued for only 24 hours. Also, if the pre-coating and coating processes are not completely enclosed and prevented from absorbing foreign substances from the surrounding atmosphere, minute particles will enter the mixtures. Although these particles may easily be dislodged later, their removal will leave pinholes in the oxide coating. Finer filtering of the oxide coating immediately prior to application should greatly improve the surface uniformity.

Effect of Continued Replays

In connection with the tests conducted to demonstrate the development of a tape hole, the behavior of the playback

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MAGNETIC TAPE

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level of tape samples containing individual sine-wave components recorded at normal level was observed during the course of 1800 replays of each sample. The results of these observations are illustrated in *Fig.* 4. Since certain applications of the magnetic medium in the fields of analysis and autocorrelation subject tape to many passages over the heads, information of this nature is vital. Actual values of the level changes shown are qualitative, inasmuch as slight variations will occur depending upon the tape tension factors involved. Nevertheless, the general trend is indicative of the behavior of the several varieties of commercial tapes currently available. For the first 125 replays, no observable deterioration of signal intensity occurred at any frequency. As expected, however, the higher frequencies, which are recorded with a minimum of flux penetration into the oxide, were the first to suffer. This deterioration is a direct result of the continuous abrasive action, which removes surface particles of oxide.

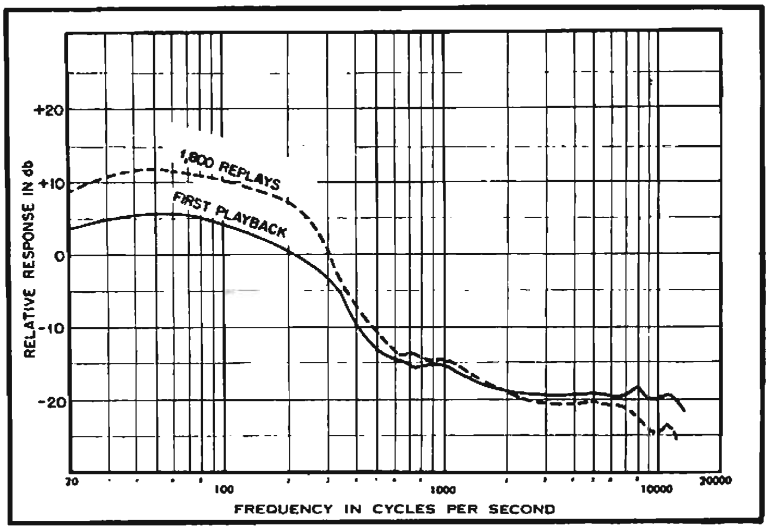

Fig. 5. Frequency analysis of over-all tape noise before and after 1800 replays.

All frequencies above 100 cps suffered losses at some period during the 1800 consecutive replays, but frequencies below approximately 50 cps displayed an increase in energy at the conclusion of the tests. This is explained by the fact that the greater remanent flux in the tape at the longer wavelengths influenced the residual oxide particles which were being removed from the surface by the abrasive action between the tape and reproduce head. These particles formed a gradually accumulating deposit of sludge about the head gap.

The same effect was noted when spectrum analyses of a sample of blank tape were made prior to and at the conclusion of 1800 replays. As indicated in *Fig.* 5, the low-frequency content of over-all tape noise was enhanced during the 1800 passages over the head, but the upper portion of the spectrum suffered. Because of the physical appearance of the tape samples at the conclusion of 1800 replays, there was some doubt as to whether any coating remained on certain portions of the tape surface. The samples, however, were erased and re-recorded with very satisfactory results.

Conclusions to be drawn from this investigation indicate that certain applications of magnetic tape, in the fields of research and development as well as in industry, are at present restricted by limitations of the medium itself. Although these limitations may appear to be insignificant when considered from the viewpoint of commercial recording concerns and broadcasters, the specialized requirements of scientists and researchers must nevertheless be appreciated and respected by those who control magnetic tape production if the recording of information by means of the magnetic process is to be exploited fully.

Correction of
Frequency Response Variations Caused by Magnetic Head Wear

KURT SINGER* and MICHAEL RETTINGER*

Increased gap reluctance due to wear causes a lowering of head inductance and a consequent variation in performance which can be restored to normal by an adjustment of bias current.

It has been noticed in the past that wear on a magnetic recording head results in a decrease of high-frequency response of the over-all magnetic recording/reproducing system and also in a change of head sensitivity. The information and data contained in this article explain the reasons for the change in frequency characteristic and offer a simple expedient for correcting the losses and thereby extending the useful life of magnetic heads.

While the benefits of a high-frequency bias current employed in magnetic recordings have been described in numerous publications, it is not frequently noted that the use of too much bias entails the loss of recorded high frequencies. This is due to an erase action produced by the bias flux at the front gap of the recording head. As the recording medium moves past the gap, it is subjected to a rapidly alternating

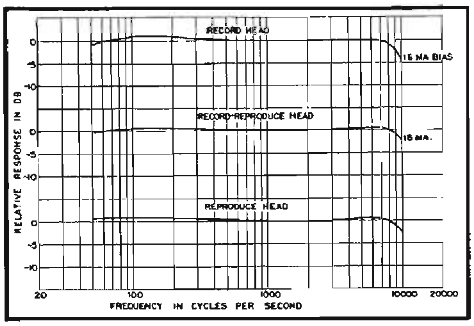

Fig. 1. Frequency characteristic at initial bias current with a head inductance of 4.9 mh; 45 ft. per minute

magnetic field, which tends to restore the medium to its neutral or virginal state, wherein the magnetic dipoles are oriented heterogeneously. This effect is more pronounced for the high frequencies than for the lows and appears to be associated with the recorded wave length.

Wear on a magnetic recording head reduces the front-gap pole-face depth and thereby produces an increase of the gap reluctance. This in turn produces a higher effective bias flux which has, as noted above, an erase action and thus tends to attenuate the high frequencies as they are being recorded on

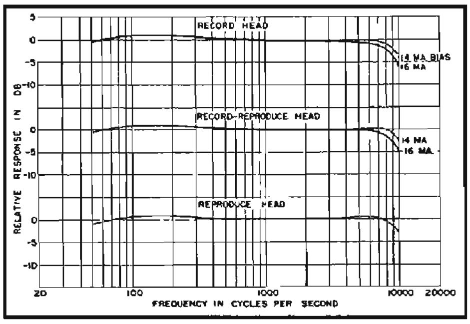

Fig. 2. Frequency characteristic vs. initial and optimum bias current for a head inductance of 4.5 mh at 45 ft./min.

the recording medium. It should be noted that this higher front-gap reluctance is due only to the decrease in front-gap pole-face depth and not to any widening of the gap, which with our type of magnetic head construction remains constant.

It is the purpose of the following to present these performance variations as a function of the lowered inductance associated with head wear and to show how, simply through a correction of bias current, proper performance can be restored.

To permit a ready evaluation of the test results, it is desirable to describe the method of testing. First, a frequency recording was made with an MI-10795-1 Head hereinafter called the test head. The film speed was 45 ft. per minute (9 in. per sec.) and the initial bias current 16 ma at 68 kc. The recording was then reproduced on a similar head and the properly equalized output from it was taken as an indication of the performance of the test head as a record head. Next, the recording was reproduced on the test head and the output from it was taken as an indication of the performance of the test head as a combination record-reproduce head.

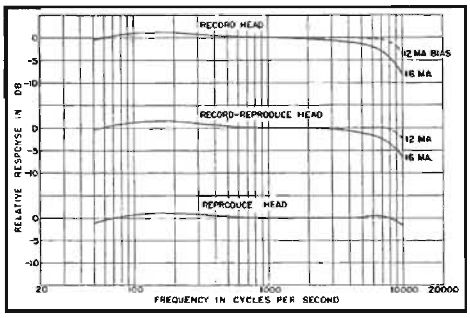

Fig. 3. Frequency characteristic vs. initial and optimum bias current for a head inductance of 4.2 mh at 45 ft./min.

A frequency film which had been made previously was then reproduced on the test head and the output from it was considered an indication of the performance of the test head as a reproduce head. The three frequency characteristics thus obtained are shown on the curves of *Fig.* 1. The top, center, and bottom curves show the initial test head performance as a record, record-reproduce, and reproduce head respectively.

The test head was then removed from the recorder, lapped until its inductance was lowered by 0.4 mh, that is, reduced from an initial 4.9 mh to 4.5 mh. The entire test was then repeated, thereby obtaining a new set of performance data on the test head as a record, record-reproduce, and reproduce head. It was noticed that the change in frequency response (loss of highs) resulting from the lowered inductance was greater when the head was used as a record head than when it was used as a repro-

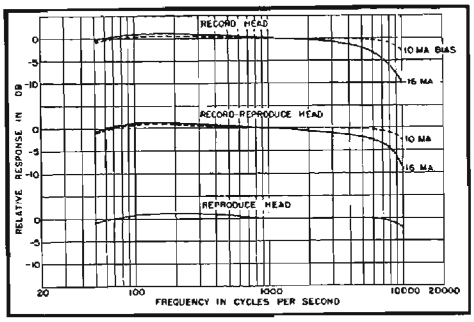

Fig. 4. Frequency characteristic vs. initial and optimum bias current for a head inductance of 3.85 mh at 45 ft./min.

duce head. To restore the frequency response of the record head to normal, the bias current had to be reduced to 14 ma. The frequency characteristics obtained from the test head with its inductance reduced to 4.5 mh are shown on the curves of *Fig.* 2. The upper and center curves show head performance as a record and record-reproduce head with the initial bias of 16 ma and the reduced bias of 14 ma (dashed line). The test head was then removed again from its mount, lapped so that its inductance was lowered again by a certain amount—in this case from 4.5 to 4.2 mh—and the tests were repeated. The frequency characteristics obtained from this series of tests are shown on the curves of *Fig.* 3. Again it should be noted that the reduction of bias current to 12 ma for

(*Continued on page 46*)

* *RCA Victor Division, Radio Corporation of America, Hollywood, California.*

Presented on May 1, 1953 at the SMPTE Convention in Los Angeles, California. This article will also appear in the SMPTE Journal.

MAGNETIC HEAD WEAR

(*from page 29*)

this head inductance of 4.2 mh restored the head performance to normal. The curves of *Figs.* 4, 5, and 6 depict the head performance for inductances of 3.85, 3.5, and 3.1 mh. These curves also show the change in bias current required to regain proper frequency characteristics.

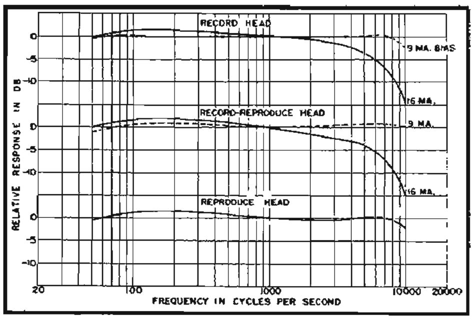

Fig. 5. Frequency characteristic vs. initial and optimum bias current for a head inductance of 3.5 mh at 45 ft./min.

Figure 7 shows the gradual loss in high frequencies as the recording head inductance drops from 4.9 to 3.1 mh at a constant bias current of 16 ma.

Exploring the region of maximum sensitivity bias of the record head over the range of inductances from 4.9 to 3.1 mh, it was noticed that the initial bias current of 16 ma and the reduced optimal bias currents in all cases represented bias currents corresponding to a value either equal to or slightly lower than maximum sensitivity bias. However, this statement should not be construed to mean that it is only necessary

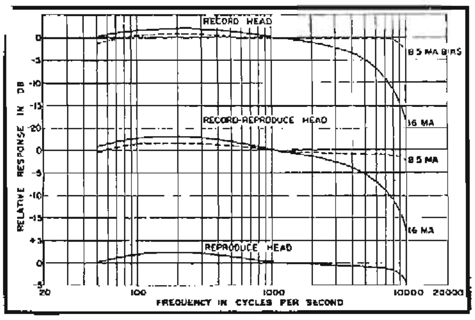

Fig. 6. Frequency characteristic vs. initial and optimum bias current for a head inductance of 3.1 mh at 45 ft./min.

to adjust the bias current to maximum sensitivity bias to recover the lost high frequencies. This procedure would only result in an approximately normal performance. In order to compensate for

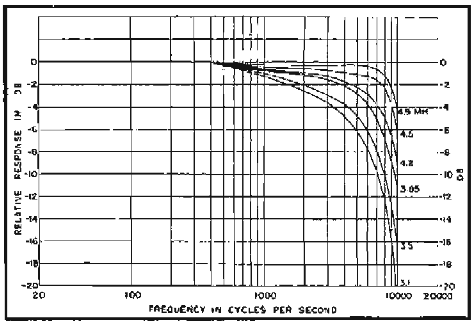

Fig. 7. Frequency response vs. inductance of recording head measured at a constant bias of 16 ma at 45 ft./min.

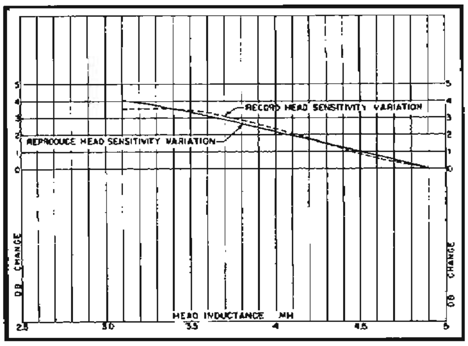

Fig. 8. Head inductance vs. sensitivity change, 45 ft./min., 400 cps.

head wear accurately, it is necessary to reduce the bias current experimentally to a value which will produce the initial frequency characteristic.

During these tests it was also noticed that a sensitivity change occurred in the test head. The sensitivity variations are shown on the curves of *Fig.* 8. Zero sensitivity of the test head as a record head corresponds to the sensitivity of the head with its full inductance of 4.9 mh operating at a bias current of 16 ma. It should be noted that as the head inductance is decreased and the head sensitivity increased it is necessary, in order to obtain 100 per cent modulation (approximately 2.5 per cent distortion at 400 cps), that the signal input to the head (signal current) be reduced by the amount shown on this curve. In exploring the performance of the test head as a reproduce head, zero sensitivity was assumed as the sensitivity of the head with an inductance of 4.9 mh. As the head inductance was lowered, the output from the head increased by the amount shown on this curve.

The curve of *Fig.* 9 shows the change in the 100 per cent modulation level that was noted as the head inductance was decreased and the bias current readjusted for satisfactory high-frequency performance.

Figure 10 has been included to show approximate values of optimum bias currents which can be used in an initial attempt at correcting for high-frequency loss by the record head when the head inductance has been reduced due to head wear. It must be understood that this curve can only be offered as an approximation toward the desired optimal bias current. Minor deviations from it may exist in individual cases.

All the above-described tests were made at a film speed of 45 ft. per min., or 9 in. per sec. The attenuation of recorded high frequencies due to magnetic

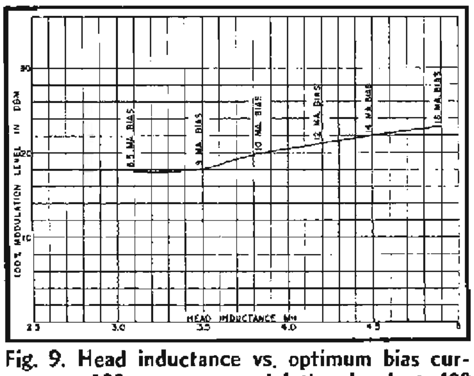

Fig. 9. Head inductance vs. optimum bias current vs. 100 per cent modulation level at 400 cps, 45 ft./min.

head wear at other film or tape speeds will have the same trend, although it does not necessarily follow that the same pattern obtained with a 45 ft. per min. film speed will result. However, practice has shown that in all cases it has been possible to regain lost high frequencies through a reduction of bias current.

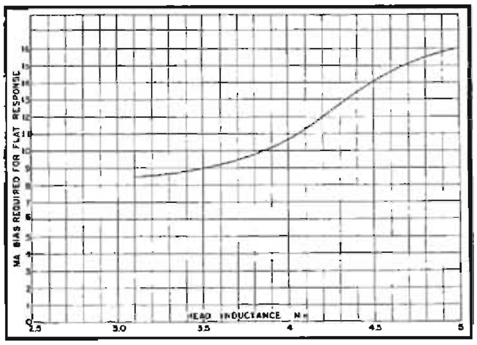

Fig. 10. Head inductance vs. optimum bias current, 45 ft./min.

We would like to inject a note of caution: Each change in bias current necessitates the re-establishment of the 100 per cent modulation recording level to avoid overloads and resultant increase in distortion.

Problems Involved in Magnetic Tape Recording*

NORMAN E. GIBBS†

Results of a program of research into defects of magnetic tape and their applications to pulse reproduction when tape recorders are used as computer memory elements.

MAGNETIC RECORDING PROCESSES are rapidly being applied in many phases of scientific research, industrial measurement and control, as well as for storage of information in large-scale computers. In many of these applications it is essential that the pulsed data recorded be completely and accurately recovered.

A number of factors affect the accuracy of the storage processes. Chief among these are the design features of the tape-handling mechanism including recording heads, the characteristics and stability of the magnetic medium, and the reliability of associated circuitry.

It is the purpose of this paper to examine in particular the contribution of the medium itself, especially currently available tapes, to the over-all reliability of the magnetic recording process. Heretofore, chief attention in most applications has been focused upon design of equipment, and the currently "best available" tapes have been utilized.

In observing the cathode-ray patterns produced during playback of a continuous train of pulses from a magnetic tape, it has often been noted that the amplitude of some of the pulses became quite small or practically disappeared from the otherwise nearly constant signal envelope. These apparent losses of pulses have been termed "pulse drop-outs." A program of investigation which has been pursued at Raytheon for nearly two years is summarized in the following brief outline of the causes and effects of "drop-outs."

Continued research has established conclusively that the causes of drop-outs are (a) defects inherent in the tape itself, (b) foreign matter, such as dust, temporarily on the tape surface, and (c) mechanical flutter in tape drive and accumulation of static charge.

Inherent Tape Defects

At the start of the investigations, the high-power microscope revealed that the surface of the magnetic tape contained zones varying in color which departed from the usual flat and smooth texture.

Before examining the causes of these abnormal areas, consideration of the construction of a typical tape lends perspective. In general, a thin film of magnetic material approximately .00025 to .0004 inch in thickness is deposited upon a cellulose acetate base which is itself only about .002 inch thick. Fabrication is complicated by several factors. The magnetic particles must be sub-micron size in order to provide favorable magnetic properties. Colloidal suspensions of these minute particles must be utilized, and contamination by traces of impurities or dust may cause flocculation of particles at local zones, resulting in lumps of adhering material. It is imperative that filters and agitators be used, that all ingredients be checked as to purity, and a high degree of cleanliness maintained in all operations.

Traces of impurities, dust, or vapor, may affect the surface tension of bonding agents, causing pin-holes where surfaces were not perfectly wetted. Great precautions must be taken during drying operations to avoid dust falling on the still tacky surface of the tape. Yet the over-all thickness of the applied layer must be maintained within close limits or response characteristics will suffer. Fabrication of good tapes is not an easy task.

It was found that defective areas in tapes were due chiefly to:

1. Particles of metal or other foreign matter embedded in the tape surface.
2. Bubbles formed during the drying of the tape which left a ring of magnetic material in the center of which the film had collapsed and dried in folds and crevices.
3. Pin-holes in the magnetic layer, easily visible by holding the tape up to a light source.
4. Droplets of magnetic material were rolled out over the tape and attached stickily to the magnetic layer like chewing gum on a floor.
5. Smaller particles of magnetic material, evidently not very tacky since a few passes of tape under the heads wipes them free.
6. Clumps, formed by a process of which the exact nature is unknown, but which entails the apparent growth of the clump at the expense of the magnetic material in the surrounding area.
7. Areas in which the bonding agent had failed to keep magnetic layer intact and the latter had flaked off.
8. Imperfections in the acetate base such as wrinkles and roughness somewhat resembling orange-peel in appearance.

It has been gratifying to find during the course of time that the tape manufacturers have succeeded largely in eliminating many of these causes of defects. Tapes are now much cleaner and are minus the metal particles, "globs" resembling gum, and usually the pin-holes.

The sketch of *Fig.* 1 shows a cross section of the chief offender remaining unconquered. The zone surrounding a clump is marked by its rough appearance. Apparently clumps are formed after the magnetic material has been applied to the tape and is still in a moist condition. Some factor causes adherence of the particles in this local zone to form a clump while sufficient migration of others in the suspension causes the recession in the surface level in the surrounding area.

With elimination of metal particles embedded during tape fabrication, dust and dirt constitute the chief sources of foreign material on tape surfaces. Magnetic mud, consisting of small particles sloughed off the tape and mixed with minute particles of core material and dust, frequently becomes rolled into small lumps which stick to the tape. Elimination of all of these is highly desirable.

Mechanical flutter of the tape as it passes under the heads results in variation of contact between tape and head surfaces which produces serious shifts in signal amplitude if not actual loss of information. These variations in signal level are due to the additional gap be-

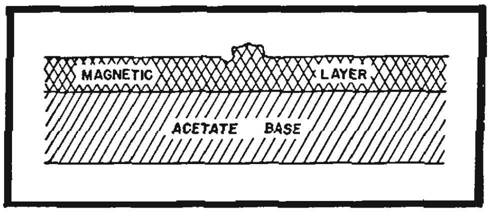

Fig. 1. Cross-section of a clump, showing the effect on the surface of the tape.

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*From a paper delivered at the Symposium on Electronic Computers, sponsored by the Los Angeles Professional Group on Electronic Computers, University of California, April 30th, May 1st, 1952. Reproduced by permission from the *Symposium Proceedings.*

† *Computer Equipment Section, Raytheon Mfg. Co., Waltham, Mass.*

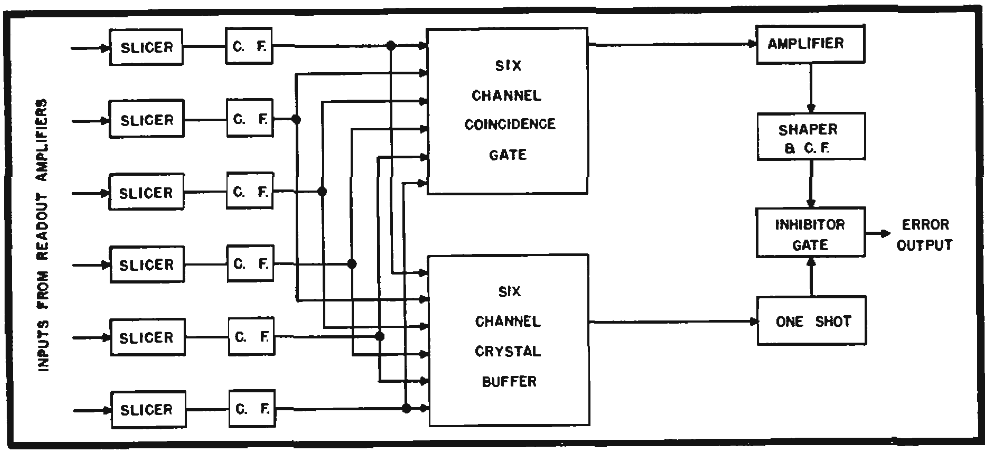

Fig. 2. Block diagram of the 6-channel coincidence circuit used in the investigations.

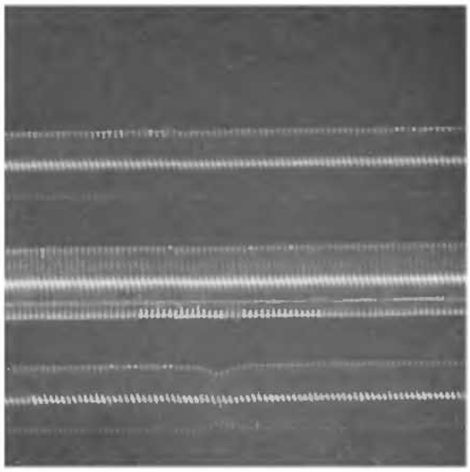

Fig. 3. Photograph of recovered pulses as they appear on an oscilloscope shows a typical minor defect appearing on tape using permanent magnetic bias.

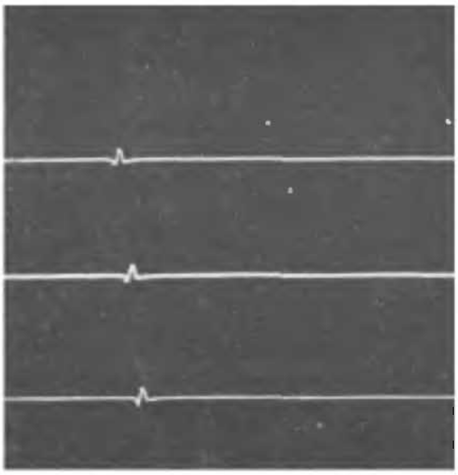

Fig. 7. Playback of tape with butted splice shows this pattern after permanent-magnet erasure.

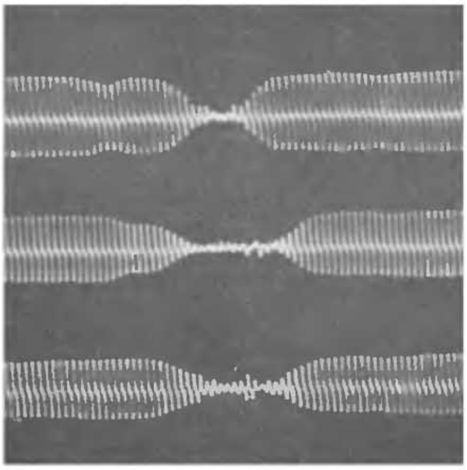

Fig. 8. Playback of writing over a lapped patch.

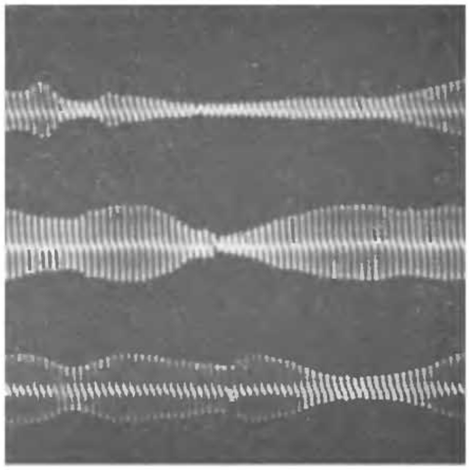

Fig. 9. This is the result of an attempt to cut and splice tape containing a continuous train of pulses.

tween the head and the magnetic surface.

All of these losses may not be due to the tape mechanism, however, since wrinkles in the tape, scalloped edges, etc., may cause similar effects. Stretching in local zones may cause curling of the tape and hence a variation in tension as these zones travel under the head. Attention must be paid at all stages of handling tapes to factors such as alignment of guides and idlers, use of proper tension in rewinding, and so on, to avoid these difficulties.

In rooms where the relative humidity is quite low the tape may acquire a static charge. Unless this is dissipated it may cause undesirable transients in playback circuits and every vary the tape tension due to adherence of adjacent layers of tape.

Having ascertained that defects existed in the magnetic medium which probably were responsible for pulse drop-outs, several problems remained. Methods of detecting tape defects had to be devised in order to determine the number of defects per roll of tape and reality of correlation between defects counted and presence of actual bad areas on the tape. As a defect passes under the playback heads actual effects taking place had to be determined. What could be done with tapes containing defects to make them usable in computer service?

Methods of Detecting and Counting Defects

Microscopic inspection of tapes would be difficult to achieve. With illumination at a grazing angle, however, microscopic studies were valuable in establishing size and contour of clumps.

Observations of cathode-ray patterns are informative but it is usually difficult to stop the tape upon sighting a drop-out. High-speed photography of multichannel patterns appearing simultaneously on cathode-ray tubes is highly effective in establishing the effects produced by clumps.

Comparison bridge circuits proved promising when checking playback signals from two channels in which continuous trains of pulses had been written.

The most satisfactory unit, however, was designed to permit simultaneous examination of all six channels on the tape. As shown in the block diagram, *Fig. 2*, the playback signals from the six-channel preamplifiers are fed through slicers and cathode followers. A steady stream of pulses reaches the six-channel coincidence gate simultaneously from all channels. As long as this condition exists, these pulses are fed through the amplifier, shaper, and cathode-follower to the inhibitor gate.

Meanwhile, however, pulses have been fed simultaneously to the six-channel crystal buffer causing operation of the one-shot. The output of the latter is led to the other side of the inhibitor gate. A positive pulse is emitted by the latter only when a pulse is missed in one or more of the channels. Controls can be set so that if the amplitude of the playback voltage for one or more pulses drops below 50 per cent of average value, a count results, hence the threshold is set at 50 per cent. The threshold may be varied. An electromagnetic counter can be tripped by the pulse from the inhibitor gate.

A more convenient arrangement is to use this pulse to stop a Raytheon Tape Mechanism involving a tape travel of about 3/16 in. so that the area of the tape immediately under the playback head may be carefully inspected.

This detector, with requisite precautions, yielded consistent counts on repeat runs of the same reel of tape.

Test Procedures and Results

Test procedures were as follows:

1. A tape was cleaned of dust by wiping it with a lens tissue pad during writing and playback operations.
2. Using the coincidence detector, the number of defects in a given length of tape was determined for all six channels.
3. This section of tape was then played back and the cathode-ray patterns of three channels were photographed simultaneously with a high-speed camera. A repeat run provided data on the remaining three channels.

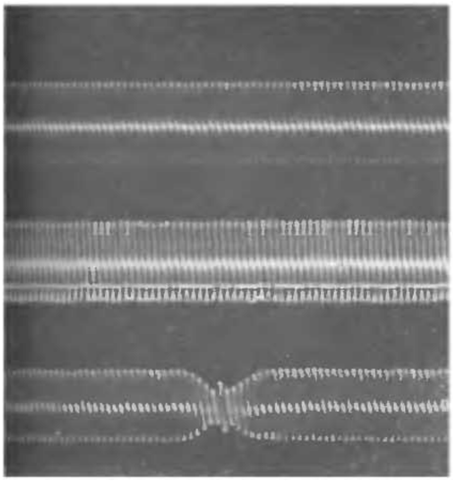

Fig. 4. A serious defect on tape with permanent-magnet bias. Note the trend to partial center recovery.

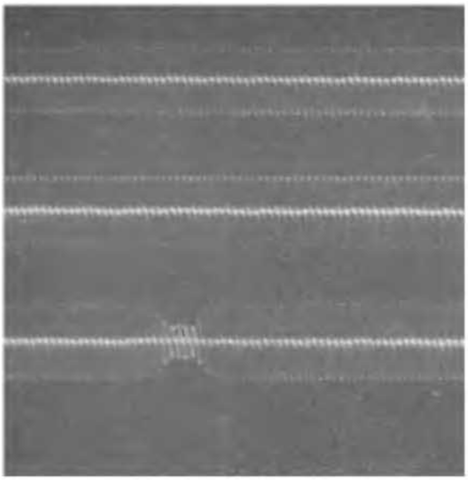

Fig. 5. A serious defect without any center recovery.

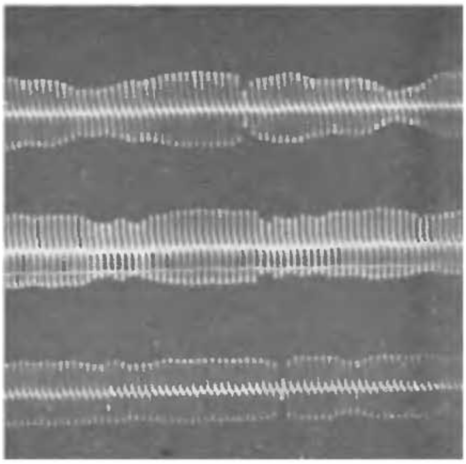

Fig. 6. This oscillogram shows the effect of writing over a butted splice on magnetically neutral tape.

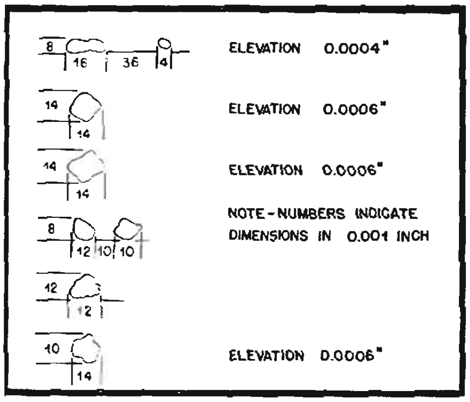

Fig. 10. These were among the defects manually removed from the tape. The drawings show the general shapes and sizes. Dimensions are given in thousandths of an inch.

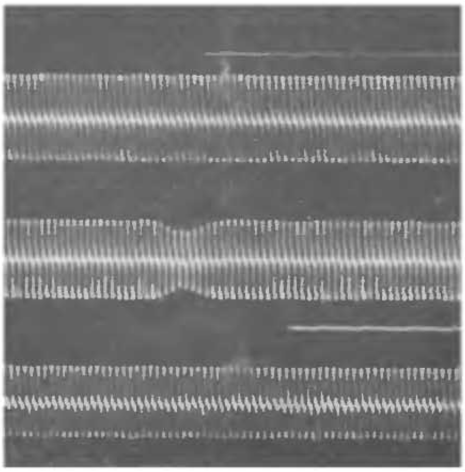

Fig. 11. After removal of defects from the tape, this pattern is indicative of about the worst remaining fault.

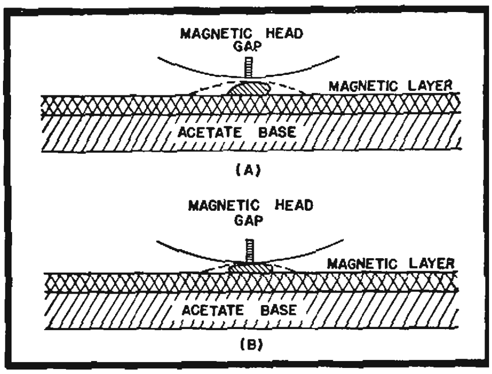

Fig. 12. The action of the tape and head when a defect appears. In A a condition like that of Fig. 5 is produced. In B some center recovery is possible, as in Fig. 4.

4. The tape was erased, then rewritten.
5. Recounts were made on the coincidence detector, which checked with originals.

Figure 3 shows the effect produced by a typical minor defect. About a half dozen pulses are affected. Pulse density is 100 per inch in all the illustrations. Counting pulses therefore gives an idea of the area affected by the tape defect.

In *Fig.* 4 can be seen the typical effect produced by an average clump. *Figure* 5 shows a somewhat different effect caused by a clump in which no center recovery occurs.

Since a patch in the tape introduces a defect, some photographs were made of these also. *Figure* 6 shows the effects caused by a typical butted splice held by splicing tape and *Fig.* 7 shows the effect produced by the same splice when tape was erased with a permanent magnet.

A lapped patch causes greater disturbance of patterns as shown in *Fig.* 8. *Figure* 9 shows the effect produced by attempting to cut and splice a tape bearing a continuously recorded train of pulses.

It is apparent in many of the photographs that practically no variation of signal amplitude occurs except at areas of defects. This indicates that mechanical flutter effects are quite small. Some effects could be observed in the case of the patches.

Using the techniques described, it was possible to examine small local areas, rather precisely located by the stopping of the tape drive, for defects. In nearly every case where the drive was stopped due to action of the coincidence detector a defect was found by microscopic examination.

Figure 10 shows the general shape and gives measurements for six typical defective areas. Elevation of the clumps was measured also. Some of the photographs correspond to these defects.

In some cases no defect was found on the tape. In the huge majority of these, however, a small dust particle or bit of lint was found adhering to the tape.

At the marked locations of the errors removal operations were attempted to determine whether unmarred magnetic material lay below most of the clumps. By very dextrous and delicate use of small scalpels under a microscope it was found possible to scrape away all of these defects leaving a relatively unmarred, normal surface of the tape. In removing one such defect an actual pit was dug through the magnetic layer. This was dexterously filled with some of the material removed. Utmost cleanliness and care were used during these processes.

The tape was now erased, rewritten and passed through the coincidence detector. The area of tape formerly giving a count of 7 now gave a count of 1. A pattern of one of the worst effects on this section of tape now appears in *Fig. 11.* Effects are much less than the approximate 50 per cent shift in level formerly obtained.

Mechanics of a Drop-out

Figure 12 portrays the mechanical lifting of the head away from intimate contact with the tape surface under two conditions. In B center recovery is possible since the center flat portion of the "glob" permits laying down of at least one magnet.

R. L. Wallace Jr. of the Bell Laboratories has made experimental determinations of the spacing losses involved when a playback head loses contact with the magnetic surface. As reported in the October, 1951, issue of the *Bell System Technical Journal,* this loss corresponds in db to 55 times the ratio of the separation distance to the wavelength of the magnets involved.

The elevation of clumps was found to be about .0006 in., as shown in *Fig.* 10, and the wavelength of the magnets is close to .005 inch giving a ratio of .012. Fifty-five times this ratio gives a figure of 6.6 db as the separation loss. This agrees closely with the operation of the detection circuit and the photographic data.

(Continued on page 52)

MAGNETIC TAPE PROBLEMS

(from page 21)

The Role of Dust

The importance of dust and dirt on tapes was at first underestimated because it was believed that the wiping action of the heads would remove much of it. Once the coincidence detector had been developed so that stopping of the tape mechanism permitted easy examination of a small local zone of the tape, it was discovered that stopping was frequently caused by dust and dirt particles rather than by actual defects on the tape.

How appalling this situation was can be realized by considering early total counts for 6 channels of nearly 1000 per 900-foot reel of tape and the fact that the computer cannot miss one pulse during normal operation without creating an error.

Installation of the lens tissue wiping pads over which the tape gently glided during reading operations reduced the count by about 60 per cent. Since dust might have interfered with the original writing operation the tapes were erased and rewritten; the counts, repeatable consistently, then fell to about 200 for the same tape. Removal of the wiping pads for a few moments raised the count by 20 to 30 per cent.

While these investigations were in progress, active interest and cooperation of a number of tape manufacturers was obtained. Tapes in recent months show counts running from 3 or 4 to about a dozen per 1000-foot roll, using the 6-channel coincidence detector. The reduction in count also reflects efforts to keep dust from the tapes. Not only were wiper pads used to remove dust which had reached tape surfaces, but tapes were kept in metal cans, the tape mechanisms were provided with dust covers completely enclosing all the tape and caution was used in handling tapes. Greasy hands can coat tapes with a small amount of oil which later accumulates dust. Gloves with as little lint as possible were used until personnel handling tapes learned to let the mechanisms do most of the actual moving of tape and touched it only with very clean hands.

In an atmosphere with very low relative humidity the tapes tend to become electrically charged and hence to collect dust. Air conditioning was a great assistance in this respect, probably because filtering of the air removed considerable dust. Efforts at lubrication of tapes to minimize static charge have not proven very effective and wire brushes and other devices have given conflicting results.

However, with some degree of humidity control and the use of dust covers and wiping pads, together with care in handling tapes and storing them, the dust menace has been largely eliminated. All the precautions are vital, however, and must be maintained.

What about Remaining Defects?

The tape manufacturers have succeeded in ridding their product of the chewing-gum type of defect, embedded metal particles, the deflated and dried bubbles, and many of the other defects. Clumps are the main cause of defects sufficiently large to cause drop-outs. The orange-peel roughness in the base material apparently results from too rapid evaporation of solvent during its preparation and tape manufacturers—now that they are cognizant of the demands of computer and telemetering service—are bending even greater efforts toward improving the quality of plastic tapes.

But such programs consume time, and while they give us a feeling of optimism for the future, do not allow us to use tapes containing defects.

In the Raytheon computer, information is written on the tape in blocks, each of which is about 2¼ in. long. Normally a free zone of about ⅜ in. is allowed between blocks.

Utilizing the detectors developed, it has been found possible to indicate on the tape the locations of defective areas. These indicators are picked up photoelectrically by the tape-marking apparatus which places the printed photo marks on the rear of the tape. These not only number the blocks consecutively along the tape, but also designate the beginning and end of each. On encountering an indicator designating a defective area, the tape printing equipment automatically allots spacing for the succeeding block so that the defective area is always left in a free zone between blocks. Thus, defects never are present within a designated block and all writing and playback operations may be undertaken with reasonable assurance that errors will not arise from pulse drop-outs. Precautions against dust must still be used since a grain of dust beneath the gap of a writing head may cause a number of pulses to be ineffective.

This procedure, on the whole, has been very effective. Together with Raytheon parallel-channel recording head assemblies, it has made possible relatively high pulse densities and effective use of tapes as storage media.

The Future?

As has already been pointed out, tape manufacturers are now acquainted with the demands of computer service and are actively engaged in improving the quality of their products. The strides they have made in the past two years are striking and of inestimable value.

Not only are they attacking the problems of eliminating defects but also of improving the base material. Cellulose acetate suffers because of its absorption of moisture which affects its dimensional stability and tensile strength.

With techniques already developed and assurance of better magnetic media to come, full realization of the potentialities of magnetic tape memory devices is closer to our grasp. They are here to stay and to perform their tasks in the computer field.

Acknowledgement

It is both a pleasure and a duty to acknowledge the painstaking dexterity of Dr. A. J. Devaud in the microscopic work on tapes, the suggestions of K. Rehler on circuits, and the rugged patience of J. Kent who made most of the tape measurements.

This work was undertaken as a part of Contract No. N70NR-38902 for the Office of Naval Research.

Compression and Dialog Equalization in Motion Picture Sound Recording

EDWARD P. ANCONA, JR.*

Intelligibility of dialog recorded for motion pictures is strongly affected by acoustic conditions during recording and reproduction. The usefulness of compression and dialog equalization in assisting intelligibility is shown and typical circuits of compressors and dialog equalizers are discussed.

SOUND MOTION PICTURES are today an important means of communication, embracing the diverse fields of art, entertainment and education. In most films the spoken word, as recorded on the sound track, is of paramount importance; especially in educational, training and documentary films the effective communication of ideas by the voice of the narrator or actors is a major factor in the success of the film. It is the purpose of this article to describe some of the methods used to improve the transmission of the words and increase the effectiviness of the communication channel between a film producer and his audience.

Motion picture sound recording equipment today can produce recordings whose low distortion, frequency response, and dynamic range are more than adequate for faithful and natural reproduction of dialog and music. Operation of the equipment, and quality control of the recording from stage microphone to theatre loudspeaker involves well established techniques, and with reasonably qualified technicians all along the line high-quality recordings can easily be made.

The basic steps in transmission of narrators' and actors' words to the ears of a motion picture audience can be summarized—and idealized—as follows:

1. Record under controlled acoustic conditions on good equipment maintained in good condition.
2. Edit and rerecord to achieve dramatic continuity and to smooth out previous technical imperfections.
3. Process with care to get best prints with the least distortion.
4. Project on good equipment which is maintained in good condition.
5. Surround the audience with desirable and controlled acoustics.

Steps 1, 2, and 3 are, for the most part, well under the control of the recording technicians. With good equipment and qualified personnel, the recordings will be good mechanically and electrically. To the sad experience of many producers, however, steps 4 and 5 often depart from the ideal, and recording engineers—and directors and producers—must take these departures into account to insure the effective presentation of their film to the audience. The audience have only one chance (the one showing they attend) to abstract a maximum of intelligence from the recorded sounds which are being reproduced in a room whose acoustics profoundly influence these sounds, but acoustics over which they, the audience, have no control. The producer, director, and sound men concerned with the production of a given film, however, practically always hear the recorded sound under optimum acoustic conditions and on well maintained equipment. Therefore, those involved in producing films should take cognizance of the fact that their audience will often be seeing and hearing the film under less than optimum conditions, and during recording and rerecording, the sound should be so modified as to assist that audience in gaining the most intelligibility from the reproduced sounds.

Degradation Factors

What are some of the specific factors which tend to degrade the intelligibility of the sound the audience hears? First, perhaps, is the monaural recording. (The obvious exceptions will not be discussed here.) With recordings made on a single channel and reproduced on a single speaker the audience has lost the power of binaural discrimination against interfering sounds occurring on the stage with the actor. *Good microphone technique* and control of stage acoustics play a large part in helping the audience over this hurdle.

Second, the apparent frequency content of the recorded material is changed by certain physiological and acoustic factors. The well known Fletcher-Munson curves of equal loudness show the variation in sensitivity of the ear to different frequencies as a function of the intensity of the sound. Since motion picture dialog recordings are nearly always played back at a level higher than the actor's normal speaking level (in order to cover a large audience) the result is an apparent increase in the low-frequency content of the recorded material. Also, the voice level employed by an actor on a quiet set is usually lower than normal speech levels, because a person in quiet surroundings will involuntarily lower his voice. Studies of spectral energy content of speaking voices have shown that in these circumstances the voice characteristic shows a relative increase in low-frequency content. Furthermore, because most acoustic materials are less absorbent at low frequencies and increasingly absorbent at higher frequencies, recording stages and auditoriums will have longer reverberation periods at the lower frequencies. The combined result of these factors—the Fletcher Munson effect, the speaking level of the actor, and the reverberation characteristics of stages and auditoriums—is to make recorded dialog sound heavy and boomy when reproduced at a high level. Correction for this effect, called *dialog equalization*, is commonly used in making dialog recordings for motion pictures.[1]

A third factor affecting the recorded dialog is perhaps more subtle, due to its transient nature, although in its cumulative effect it certainly results in as much intelligibility degradation and audience fatigue as the other factors mentioned above. Most speech sounds encompass a volume range which is greater than that which can advantageously be reproduced in the average theatre. When faithful volume range reproduction of such material is attempted, either the loud passages are too loud, or the low-level passages too low, depending on the gain of the reproducing system. In the first case the loud passages seem to be exaggerated in loudness, producing a false staccato effect or "bounciness;" while in the latter case the low-level sounds will be lost, due to being within or below ambient theatre noise level.

Speech sounds are transient in nature,

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* *RCA Film Recording, 411 Fifth Ave, New York 16, N. Y.*

[1] M. Rettinger and K. Singer, "Factors governing the frequency response of a variable-area film recording channel," *JSMPTE*, Vol. 47, No. 4, October 1946, p. 299.

and low-level syllables or consonants can follow quickly after high level syllables or vowels. Thus, in the previous paragraph, the term, "ambient theatre noise level," could be taken to include the reverberation of a high-level syllable or vowel. This reverberation can mask a low-level syllable or consonant which immediately follows it. Particularly in 16-mm motion pictures, which are so often shown under unfavorable conditions—high ambient noise level, little or no sound proofing, highly reverberant rooms, the projector in the same room with the audience, and using a recording medium of inherently low dynamic range—particularly under those conditions are effects of masking of low-level sounds by ambient noise and reverberation most telling in their deterioration of intelligibility of the recorded material and in the production of listening fatigue. This transient deterioration of intelligibility is not easily apparent upon casual listening, especially to those familiar with the recorded script, but it can interfere with the effective communication potential of the film to the same or greater extent than the other factors discussed previously.

It is desirable, therefore, to compress the volume range of dialog recordings to make the high-level sounds relatively less loud, and the low-level sounds relatively louder. The physical limitations of any attempt manually to compress the volume range of speech are obvious, and electronic means have been developed to accomplish this task. Such an electronic device is known as a compressor, and is sometimes also referred to as an "electronic mixer."

At this point, the necessity, or at least, the desirability of certain practices in the recording of motion picture dialog has been shown. These are: good microphone technique, dialog equalization, and compression. The subject of microphone handling on the recording stage is an extensive one and offers material enough for a separate paper (or book) by itself and will not be dealt with here. The remainder of this article will be devoted to a discussion of typical circuits used in compressor amplifiers and dialog equalizers and some of the practical aspects of their use.

The Compressor

The basic components of a typical compressor amplifier are shown in the simplified schematic diagram, *Fig.* 1. The main signal path is from the input through the variable-gain stage and the output stage, to the output. A portion of the output is fed back through the side amplifier to a rectifier where a voltage is developed which is a function of the peak level of the output signal. This voltage is applied as a bias to the grids of the variable mu tubes to control the gain of the amplifier in a predetermined manner. Depending on the polarity and proportions of this control voltage, the amplifier will act as a compressor, a limiter, or an expandor. If the input of the side amplifier is connected to the output of the main amplifier, as shown, the device is said to be "backward acting." If the side amplifier input is connected to the main amplifier input, it is said to be "forward acting." In general, compressors will be "backward acting," and expandors will be "forward acting."

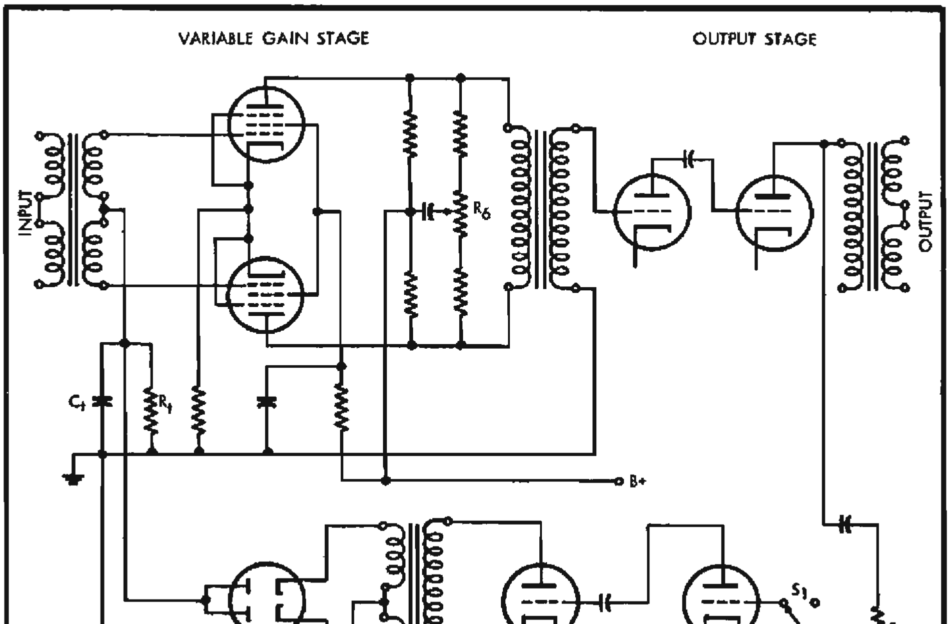


Fig. 1. Simplified schematic of a typical compressor amplifier.

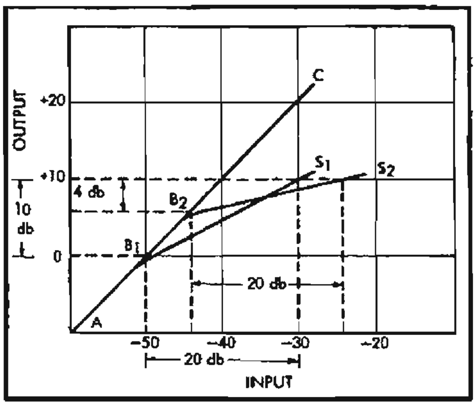
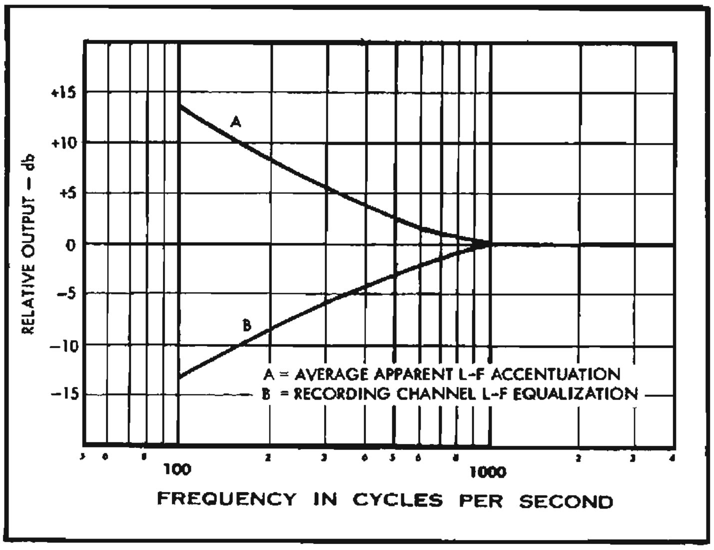


Fig. 2. Input-output characteristic of a compressor. A-C, linear (uncompressed). A-B_1-S_1, compression with 2:1 slope. A-B_2-S_2, limiting with 5:1 slope.

The operating characteristics of an amplifier with automatic gain control are shown in *Fig.* 2. A normal amplifier will have an input-output characteristic represented by the straight line, A-C, of 45 deg. slope; a change of 1 db in input results in a 1-db change in output. When operating as a compressor (*Sw_1* in *Fig.* 1 closed), the amplifier will cause its input-output characteristic to depart from the straight line, A-C, at some point, B_1 and assume a new slope, B_1-S_1. The particular point at which the characteristic breaks away from the straight line is known as the "breakaway point" and can be set up by the rectifier bias control, *R_2* in *Fig.* 1. The particular slope of the new line can be set with the side amplifier gain control, *R_1* in *Fig.* 1. Thus, B_1-S_1 represents a condition of breakaway at 0-dbm output level, with a 2:1 slope; a change of input of 2 db results in a change of output of 1 db. For B_2-S_2 the slope and breakaway controls have been set for a breakaway at +6 dbm, and a 5:1 slope. The slope and breakaway controls are somewhat interacting and several successive adjustments are usually necessary to obtain a desired characteristic.

As a matter of practice in motion picture recording work, the 2:1 slope is most often used when compression is desired, and the recording system is so adjusted that the range from 100 per cent to 10 db below 100 per cent is compressed. Under these conditions the system is said to be using "20 into 10 compression;" that is, the top 20-db range of microphone output is compressed into the top 10 db of the recorded track. Other amounts of compression, such as "30 into 15" or "10 into 5" are often used.[2]

[2] J. G. Frayne and H. Wolfe, *Elements of Sound Recording*, John Wiley & Sons, New York, 1949, Chapter 10, pp. 173–183.



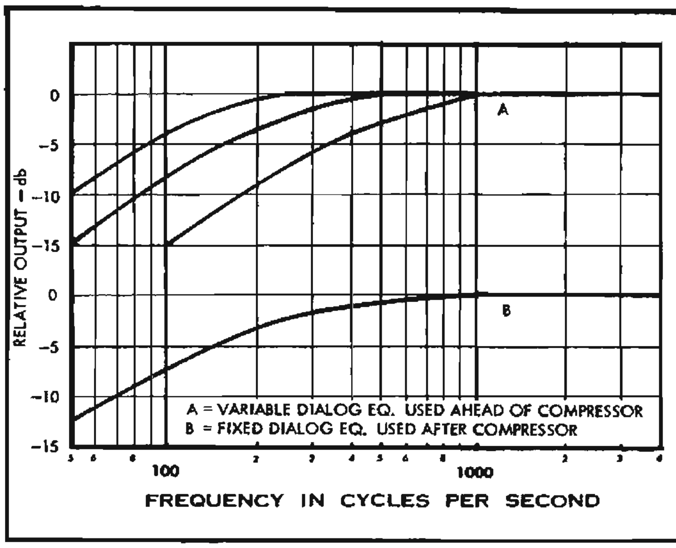


Fig. 4 (left). Characteristic curves of compressors. Fig. 5 (right). Effect of dialog equalizer.

When the amplifier is set for a slope ratio of 5:1 or greater, it is said to be acting as a limiter. The limiting characteristic has different advantages from the compression slope, as will be discussed later. Limiters used in radio broadcast work have ratios as high as 20:1, where the input-output characteristic is practically horizontal.

Two other important operating characteristics of a compressor in addition to its input-output curves are its attack time and release time. The attack time is the length of time required for the amplifier to reduce its gain when a signal suddenly appears at a level above the breakaway point—in other words, the time required to charge the timing capacitor, C_t, through the side amplifier rectifier. Critical listening tests show the desirability of very fast attack time, on the order of a millisecond or less. *Figure* 3 shows a recording of a timing test on the RCA MI-10234 compressor. The test was made by sending a 5000-cps signal into the compressor at a level just below breakaway and then suddenly increasing the level of the signal 20 db. The compressor immediately acted to reduce its gain to limit the change in output level to 10 db. It is apparent from *Fig.* 3 that the attack time was between 0.4 and 0.6 milliseconds.

The release time is the time required for the compressor gain to return to normal when the input signal falls below the breakaway point—in other words, the time required for the timing capacitor to discharge through R_t. The release time for dialog recording is usually set to about 100 milliseconds. (This is equivalent to a release *time constant* of about 25 milliseconds.)

An important consideration in the operation of a compressor is the matter of *balance*. There are very few electronic devices which will change gain without changing their d.c. operating point. This change in d.c. potential must be balanced out or it will appear at the output as an undesirable addition to the program, usually as a low-frequency transient which is termed "compressor thump." In the compressor circuit of *Fig.* 1, it will be noted that the gain change potential from C_t is applied to the 6K7 grids in parallel. As a consequence, the plates will change potential in parallel, resulting in no current flow through the interstage transformer primary. (The program signal, of course, is fed to the 6K7 grids in push-pull, and the amplifier works in the usual manner of any push-pull amplifier for this signal.) In order for the d.c. change to be completely balanced out, however, the two 6K7's must have exactly similar E_g-I_p characteristics and identical plate loads. The first requirement is met by using a matched pair of tubes, and the second by adjustment of the balance potentiometer, R_b.

A number of gain control devices other than the variable mu tube have been used, and with few exceptions all require some sort of arrangement to balance out "thump" generated by the gain change potential. Among these are the variable load tube—one tube acts as a variable plate load resistor for another tube; the "dynastat"—a closely coupled microphone and speaker, with gain control obtained through variation of speaker field current; pulsed triodes, with gain control obtained by variation of pulse width; and pentagrid tubes. Of these, only the variable mu tubes and the variable load circuit are capable of fast acting time, high signal-to-thump ratio,

Fig. 3. Attack time oscillogram of RCA MI-10234 compressor.

low distortion, and easily maintained balance.

In a previous section were discussed the various factors contributing to an increase in low frequencies heard during reproduction of motion picture dialog. These were the Fletcher-Munson effect, the actor's voice level, and the reverberation characteristics of sets and auditoriums. If the average amounts of low-frequency accentuation due to these causes are added together and a smoothed out average curve drawn, we have the graph of (A) in *Fig.* 4. It is apparent that for natural reproduction of motion picture dialog, we should insert in the recording channel an equalizer with a characteristic inverse to that of (A). This is shown at (B).

A compressor tends to smooth out any difference in levels in program material which it handles. Thus, if the low-frequency attenuation described above were placed in the channel ahead of the compressor, the compressor would tend to erase some of the effect of the equalizer. This action is not in the manner of a tone control but is a transient action which changes the relative levels between syllables or vowels of different low-frequency content. Because of this effect it has been found desirable to split the total amount of recommended dialog equalization roughly in half, placing part ahead of the compressor and part after. The part after the compressor is fixed while that ahead of the compressor is located in the console and is made adjustable so that the mixer can compensate for such variations as different actors, different speaking levels, and different acoustic conditions. *Figure* 5 shows, at (A), a representative set of dialog equalization curves used in a recording console, and (B) shows the fixed equalizer which completes the total amount of dialog equalization used. The console equalizers can be simple *R-C*

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equalizers using series capacitors; the fixed equalizer is usually of the constant resistance type.

Operation

Figure 6 is a block schematic showing the basic components of a typical motion picture recording channel. The input section of the system has been split into two parts to permit compression of dialog independently of other sound tracks. This system is quite flexible and can be "patched" for use in several ways to obtain maximum benefits from use of the compressor:

1. For narration or stage dialog recording the microphone signal is fed to mixer pot #1 as shown and the compressor is in use. Mixer pots 2, 3, and 4 remain idle.
2. For use in a "direct mix," where a live narration is mixed with music and sound effects tracks, the microphone signal is fed to pot #1 and up to three film phonographs or other sources are fed to pots 2, 3, and 4. The narration is compressed, and the combined music and effects bypass the compressor.
3. For rerecording up to 4 tracks, including a dialog track which has been previously compressed, pots 2, 3, and 4 are used. (The output of pot #1 can be patched into the unused COMBINING NETWORK IN if needed.) The output of the master pot bypasses the compressor.
4. For use in rerecording, when limiting is desired on the combined mix, the inputs can be patched as in paragraph 3 above, and the compressor can be patched to replace linear booster #2. The compressor slope and breakaway controls would be readjusted to produce a limiting characteristic, for example, "20 into 4."

Note that dialog equalizers equivalent to those shown in *Fig.* 5 have been used in this channel in the console and after the compressor, respectively. Quite often more elaborate equalizers are used in all four mixing positions, but those in positions handling dialog will always include the characteristics of (A) in *Fig.* 5.

What are the advantages of the channel configurations described above, and what exactly are the audible effects obtained by use of the compressor?

Paragraph 1, of course, is the basic circuit—a single microphone and a single gain control, with dialog equalization and compression. Probably the first effect noticed by the mixer is that the dialog sounds louder when using compression, even though the volume indicator peaks to the same level. Furthermore, an increased intelligibility and smoothness becomes apparent, due not simply to the sound being "louder," but due to the transient rearrangement of levels within words and syllables effected by the compressor action. Loud or explosive vowel sounds which would produce a disagreeable and distracting blasting effect are reduced in intensity, and lower level consonant sounds which contribute so much to individual word meaning are raised in level. These effects of increased intelligibility and smoothness are most apparent when the dialog material is reproduced in an auditorium (or 16-mm classroom) and in combination with other sounds, either from the sound track (music, sound effects) or ambient in the theatre (audience noise, outside traffic, reverberation). The amount of dialog equalization used is a matter of judgement on the part of the mixer, based on his experience in the sound of his recordings when reproduced in the

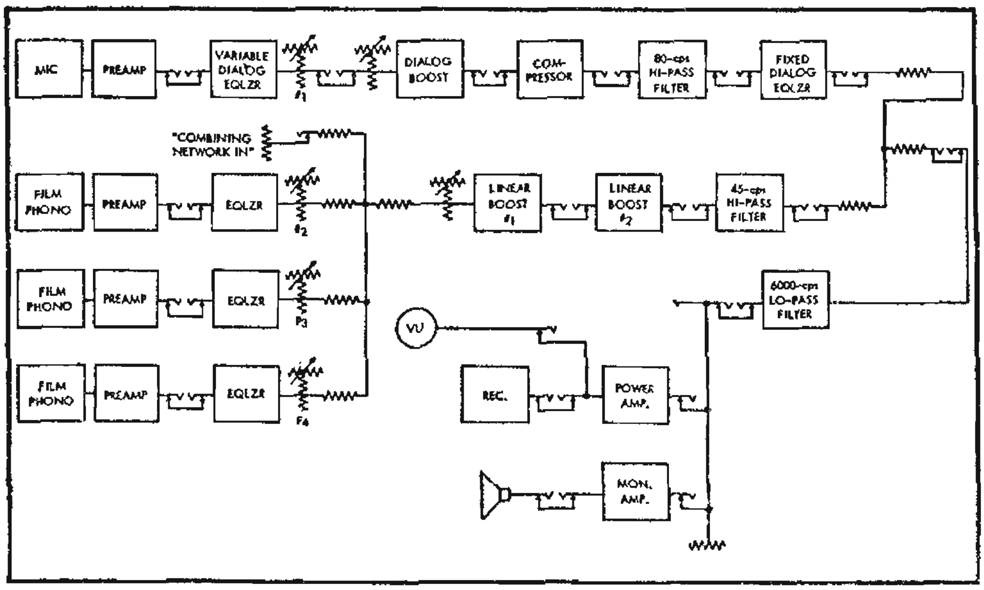

Fig. 6. Simplified block schematic of a typical motion picture recording and rerecording system using a compressor.

average theatre and his knowledge of the characteristics of his own monitor system. The amount used will vary with different voices and acoustics.

The circuit of paragraph 1 is also used when transferring original location recordings to film for editing. Since portable recording equipment usually will not include a compressor, compression can be added during transfer dialog. Additional dialog equalization can also be added at this time, if necessary.

In the circuit of paragraph 2 the use of compression on the voice will make the narration less liable to be covered up by high level sound effects or music. The smoothness and more consistent level of the compressed voice will give greater freedom to the mixer in his handling of the music and effects tracks, since the voice will maintain its intelligibility over a greater range and variety of background sounds. The circuit of paragraph 3 would, of course, be identical to that of paragraph 2 in its results if one of the tracks being mixed was a narration track which had been compressed previously.[3]

The circuit of paragraph 4, in which the compressor amplifier is set up as a limiter, is often used to protect the recording device from serious overmodulation. It is similar to the usual broadcast audio circuit in which a limiter (of more severe slope) is used to prevent overmodulation of the transmitter. The general effect of the limiter is to permit the mixer to mix his program "louder" without fear of overmodulation. The results are not the same as those when compression is used, since in *limiting* the automatic gain control feature of the amplifier is only occasionally used, whereas in *compression* the automatic gain control action is almost continuous.

Disadvantages in Use of Compressor

It might be expected that all the benefits described above are not without a certain penalty and extra effort on the part of the recording crew. This is quite true, and in addition to the obviously increased complexity of adjustment and maintenance of a channel which uses a compressor, there arise certain difficulties through its use in stage recording. With a compressor set for "20 into 10" compression, the channel has 10 db more gain for low-level signals than it has for 100 per cent level signals. This means that all background set noises—air conditioning, outside traffic, noisy lights, etc.—are 10 db louder than they would be without compression. This can present real problems, and it sometimes falls to the judgement of the mixer to decide whether the advantages of compression are outweighted by the distraction of the increased background noises brought about by its use. However, the advantages are considerable enough that it is worth the attempt to avoid or silence the interfering sounds on the set before abandoning compression. Because of the greater freedom in choice of acoustics and microphone position in narration recording, the use of compression here seldom introduces problems of the type described above.

Conclusion

Dialog equalization and compression have been described in this article in relation to their use in motion pictures. However, insofar as they are used to overcome fundamental difficulties arising from inherent characteristics of the human ear and auditorium acoustic conditions, their use would be advantageous regardless of the recording medium, be it tape, film, or disc; or even a live program.

This would indicate that their advantages could apply in recording film strip narration or in public address system work. Conversely, motion pictures made for television use will be reproduced at lower levels in small rooms, and many of the arguments would not apply. However, compression gives commercial announcements more apparent volume and carrying power—a sound preferred by all leading advertising agencies—and the "woompy' quality of many radio consoles and the characteristically close microphone technique of radio announcers often makes some dialog equalization desirable.

It might be well to make a few remarks about the relationship of these techniques to "hi-fi." Offhand it would appear that absolutely faithful reproduction of speech in terms of frequency response and dynamic range would be preferable to the attenuated frequency characteristic and restricted dynamic range recommended here. Unfortunately, "hi-fi" is a loosely defined term; it does not specify the acoustic conditions surrounding the recording microphone or the reproducing loudspeaker, nor does it specify the volume level at which recorded material is to be reproduced. It it just these conditions, of course, which have made dialog equalization and compression desirable. The recording engineer has a large fund of techniques at his disposal, but when making a recording, he must place these techniques in their proper perspective depending on their relative utility to the particular audience for which the recording is intended and depending on the recording medium (35- or 16-mm film, disc, tape, etc.) to be used. When making recordings for motion pictures, slide films, or radio or television commercials, the engineer might re-define "high fidelity" to mean "fidelity in transmission of ideas." In these applications the *meaning* of the recorded words is the important thing. The extremes of frequency and dynamic range actually contribute little or nothing to intelligibility, and the recording engineer must use his techniques to emphasize the useful speech components and to uncover those that would be lost under adverse listening conditions.

Reference

In addition to the citations in footnotes, the following reference contains interesting information on intelligibility factors, together with methods for determining the degree of intelligibility in certain specific acoustic situations ("calculation of articulation index"):

L. L. Beranek, *Acoustics*, McGraw-Hill, New York, 1954, pp 406-417.

[3] J. O. Aalberg and J. G. Stewart, "Application of non-linear volume characteristics to dialog recording; *JSMPTE*, Vol. 31, p 248, Sept., 1938.

Distortion in Tape Recording

Common sense, careful thinking, and a set of accurate measurements will enable anyone to choose an operating point which will give the best over-all quality from his tape recorder. The author tells you how.

HERMAN BURSTEIN* and HENRY C. POLLAK

More and more audio fans, especially in areas having one or more "good music" FM stations, are making off-the-air tape recordings. Often the program source is live—symphony, chamber music, instrumentalist, singer, or choral group—while at other times the source consists of a first rate disc or tape recording. In either case, many owners of tape recorders have numerous opportunities to capture musical moments worth preserving, either indefinitely or until a better rendition comes along. Moreover, some recordists make tapes of their own singing or instrumental playing, which they are eager to hear for pleasure or improvement.

Unfortunately, the recording does not always sound "clean" in playback. It may lack the effortless, silky quality of the original source. Due to distortion, it may have a more or less grating quality, either constantly or only during loud passages. This situation is not confined to amateur recordings. Sometimes professional recordings contain objectionable distortion.

Distortion, presuming none in the source, may be due either to a fault in the tape recorder or to an excessive amount of signal applied to the tape. The latter is of concern here, that is, distortion resulting from high signal levels, and it shall be assumed that the tape recorder heads and electronics (amplifiers and bias oscillator) are in proper condition.

Although in a direct sense over-recording—that is, the desire for a high signal to noise ratio—may be blamed for distortion, in a basic sense the desire for wide frequency range, perhaps unnecessarily wide, may also be partly at fault. This can be true in two ways. First, in order to maintain good response out to 15,000 cps or so at a speed as low as 7.5 ips, the amount of high-frequency preemphasis required in recording may be sufficient to cause tape overload at treble frequencies. Above 7,500 cps, where most of the boost occurs, there would be virtually no audible harmonic distortion inasmuch as the harmonics fall outside most persons' hearing range as well as outside the recorder's pass band, which cuts off sharply beyond 15,000 cps or earlier. However, in any nonlinear system there would still be intermodulation products generated by interaction between two high frequencies or between a low and a high frequency; many of these products would be within range of the ear and the recorder.

The desire for extended high-frequency response can also be responsible for distortion by virtue of the required bias setting. Over the bias range customarily used, an increase in bias generally causes distortion to fall, while a decrease in bias generally causes distortion to rise. However, increased bias also results in greater attenuation of high-frequency response. The desire to maintain high-frequency response well beyond 10,000 cps at low tape speed may lead to bias reduction, thereby resulting in greater distortion at a given recording level.

The following discussion seeks to throw light on:

1. The relative changes in harmonic and intermodulation distortion as input level is varied.
2. The relative changes in harmonic and intermodulation distortion as bias is varied; determination of bias for minimum distortion.
3. Variation among tapes with respect to intermodulation distortion.
4. Method of setting bias so as to yield the optimum combination of high signal-to-noise ratio, wide frequency range, and low distortion.

It should be made clear that the measurements described in the following discussion are not definitive in the sense of providing exact values under given recording conditions. Rather, they are broadly indicative of what happens. The values may fluctuate as the test is repeated at a different time, on a different machine, with a different tape, at different temperature or humidity, and so on. However, the tests have been repeated sufficiently to indicate reliably the general nature of the observed phenomena.

The measurements underlying the following discussion were made on two professional tape recorders in the $2,000 class, operating at 15 or 7.5 ips, and using a commercial high quality tape. The machines have separate record and playback heads, permitting immediate plotting of results. Test equipment consisted of an audio oscillator, an oscilloscope, a sensitive a.c. VTVM, a harmonic distortion tester which measures the total signal content after the fundamental has been filtered out, and an SMPTE type IM tester which, using 60 and 6,000 cps respectively in 4:1 ratio, measures the extent to which the high frequency is modulated by the low frequency.

Variation of Distortion With Input Level

Invariably, tape recorder specifications make no mention of IM distortion, referring only to harmonic distortion. Tape recorders have a VU meter or other type of recording level indicator to show when recording level is such as to produce 1 or 2 or 3 per cent harmonic distortion. However, as *Fig.* 1 reveals, when harmonic distortion is still at relatively innocuous levels, below 3 per cent or so, IM distortion can be disruptive—20 or 30 per cent or more.

The measurements in *Fig.* 1 were made on a machine operating at 15 ips with bias set approximately at optimum, in the manner described later. The 0 db reference input level for measuring IM distortion was equated to that for harmonic distortion by adjusting these input levels for equal peak-to-peak readings on an oscilloscope.

Figure 1 indicates that IM distortion begins to rise much earlier than harmonic distortion, and that the rate of increase is far greater for IM distortion. After IM distortion has reached about 4 or 5 per cent, it rises very precipitously. It may be observed, therefore, that in the effort to add a few db to signal-to-noise ratio, the recordist runs the risk of trading a slight decrease in noise for a large increase in IM distortion.

For the purposes of the measurements underlying this discussion, the recorder was adjusted so that its VU meter indicated 0 when IM distortion was approximately at the maximum level considered tolerable for high fidelity purposes, say about 2 or 3 per cent.

In actual use, however, the recorder should be adjusted so that the VU meter indicates 0 for a signal perhaps 8 or 10 db *below* that which causes maximum

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allowable distortion, because on transients the pointer of the VU meter may lag 8 db or more behind peak signal level. If in actual use, the meter were calibrated to read 0 for a steady-state signal which produces 2 or 3 per cent IM distortion, allowing the needle to hit 0 when recording program material would often bring the reading into the region of extreme distortion, albeit briefly. Therefore, it is necessary to allow a margin in adjusting the VU meter. Even so, unless the recordist uses discretion, based on the nature of the music he is recording, fortissimo portions of a musical work, or at least the attacks, can be marred by the breakup and fuzziness symptomatic of distortion, even though the VU meter indicates only 0.

The recordist is forced into a choice among three alternatives: (1) to accept occasional high distortion in exchange for an improved signal-to-noise ratio; (2) to make some sacrifice in signal-to-noise ratio (which means relatively more hum, tube noise, and tape hiss) in exchange for low distortion throughout the recording; (3) to ride gain, reducing input level during loud passages, which means exchanging dynamic range for low distortion throughout a recording. The last alternative implies ability and willingness to compare the program source against a score and accurately anticipate changes in level.

The recordist's decision on the course to follow will be influenced by the tape recorder he is using and purposes for which it is employed. If it is a quality machine with a high signal-to-noise ratio, he may well follow the expedient yet satisfactory course of setting recording level just low enough so that peak passages are recorded at a level of distortion which, at least for a brief period, has no appreciable effect upon the listener. On the other hand, if the machine's signal-to-noise ratio is inferior, the preferable course may be to accept some obvious distortion during peaks for the sake of keeping background noise comfortably low throughout the recording. The program source can also influence the decision. For example, a relatively high input level might be used to record the spoken voice because in this instance a considerable amount of distortion can usually go unnoticed. On the other hand, one might have to exercise considerable more restraint in setting gain for an organ or piano in order to obtain a pleasing similarity to the original.

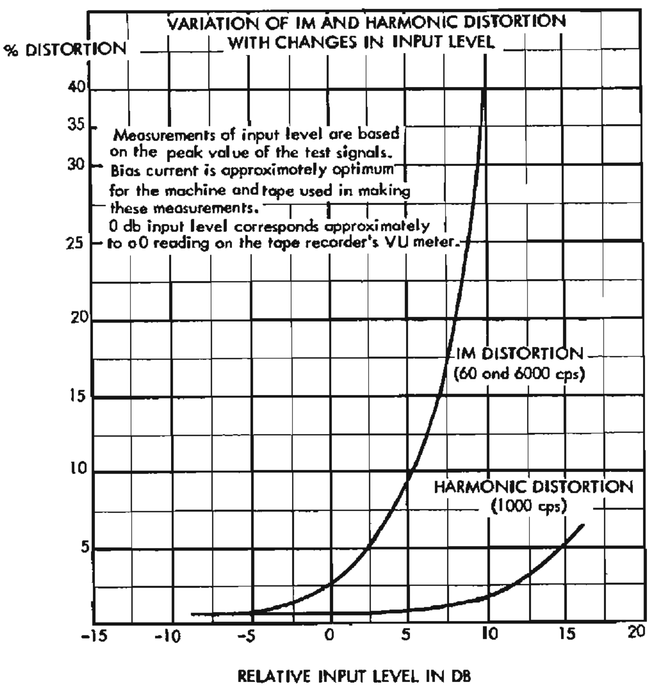
Variation of Distortion With Bias

Figure 2 indicates the effect of bias current on distortion, using two relatively high input levels. It must be taken into account that as bias varies so does the amount of signal recorded on the tape. In short, tape output as well as distortion varies with bias. However, we are only interested here in how distortion varies with bias. Therefore it is necessary to hold tape output constant. For this reason, the input level was constantly adjusted to maintain a fixed indication on the VU meter *in playback.* Curves 1 and 2 are based on a playback indication of 0 db on the VU meter. Curves 3 and 4 result from levels 3 db higher. At the 0 VU playback level, with bias set for minimum IM distortion, the harmonic distortion test signal was matched to the IM test signal by comparing peak-to-peak playback amplitudes on the oscilloscope.

Figure 2 reveals that: (1) IM distortion once again varies much more than harmonic distortion; (2) Distortion does not indefinitely continue to decline as bias is increased, but rises again, and this rise is sharper in the case of IM distortion; (3) The higher the input level, the more critical is the bias setting for minimum distortion; thus, in order to find the minimum-distortion bias with ease, it is merely necessary to use a very high input level. (4) A rise in input signal level produces the least increase in distortion when bias is set for minimum distortion.

From the above it can be concluded that to the extent the recordist seeks to maximize signal-to-noise ratio by turning up gain, the more important it becomes that he adjust bias properly for the particular tape he is using. Otherwise he may get much more distortion, especially IM, than is acceptable.

(An interesting phenomenon is displayed by the left portion of the curves in *Fig.* 2. If bias current is reduced enough below the normal working range, distortion drops again. Inasmuch as a reduction in bias current serves to improve high frequency response, it might seem that one might profitably operate in the area of extremely low bias current. However, there is good reason for not doing so. The reduction in distortion achieved by using very low bias current is most striking for high input levels. At low input levels, however, distortion re-



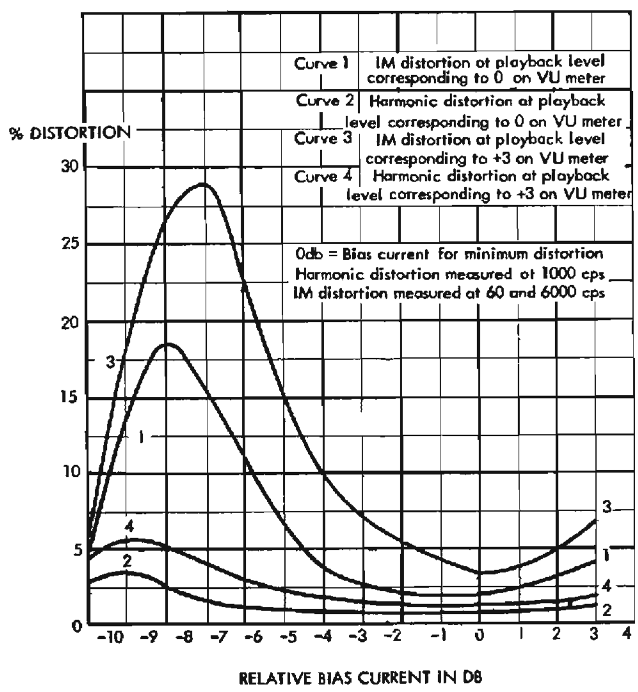


Fig. 1 (left). Variation of IM and harmonic distortion with changes in input level. Fig. 2 (right). Variation of IM and harmonic distortion with changes in bias current.

mains higher than when operating in the normal bias range. Furthermore, the amount of recorded signal drops at low bias values, so that to maintain the same amount of tape output requires considerably greater power from the output stage supplying the record head.)

Variation in Distortion Among Tapes

Using a relatively high recording signal, several popular brands of tape were compared with respect to IM distortion. Input level was varied so that each tape produced the same output level as read on the VU meter during playback. Bias was adjusted for each tape until minimum distortion was obtained. Following were the results.

Tape	Minimum IM Distortion	Relative Bias Setting
A (reference)	7.6%	0.00 db
B	9.0	.75
C	11.0	−.50
D	10.0	0.00
E	3.5	−1.00

It is interesting to note that the bias setting for minimum distortion varied only moderately from tape to tape, while the amount of distortion varied considerably more. However, these findings would not be sufficient on which to base the choice of a tape. It would be further necessary to consider the tape's frequency characteristics at the bias current resulting in minimum distortion, the *shape* of its output versus bias curves for different frequencies, its noise properties, and so on.

Determination of Optimum Bias Current

Let us assume that on the basis of curves such as in *Fig.* 2, the bias current for minimum distortion has been ascertained, using a given machine and a particular tape. However, depending upon the tape speed and upon the brand and kind of tape (regular, high output, long-play, etc.), high-frequency response may be inadequate at this bias current.

As previously stated, treble response goes down as bias is increased. This is a wavelength effect. Inasmuch as a given frequency results in a shorter wavelength at reduced tape speed, the problem of poor treble response due to high bias current is most serious at the lower speeds such as 7.5 and 3.75 ips. Consequently at these speeds, in order to maintain satisfactory response, it is probably necessary to use less bias than the amount permitting minimum distortion. This means greater distortion for a given amount of tape output, or less output for the same distortion (lower signal-to-noise ratio), or a compromise between the two.

Figure 3 indicates the procedure to be used in determining optimum bias current. It is assumed that the tape recorder provides ready means for varying bias current and for varying treble preemphasis in recording. It is further assumed that playback equalization is fixed (in accordance with the NARTB standard for 15 ips). Curves 1 and 2 in *Fig.* 3, representing variation of IM distortion with bias, have been redrawn from *Fig.* 2. 0 db bias represents bias current for least distortion.

When the tape recorder represented in *Fig.* 3 is operating at 15 ips, Curves 3 and 4 respectively show how response at 400 cps and at 15,000 cps varies with bias; input level was kept low enough to avoid any possibility of saturation. 400 cps is used as a reference frequency, not being affected by equalization used in the record preamplifier. When 0 db (minimum distortion) bias current is used, response at 15,000 cps is 1.5 db higher than at 400 cps. In order for frequency response to be perfectly flat at 15,000 cps, it is necessary either to increase the amount of bias current to 1.4 db or reduce the amount of treble preemphasis. Since a rise in bias current would increase distortion, the desirable step is to lower the treble boost.

Thus it can be seen that at a speed as high as 15 ips, at least for the machine and tape represented in *Fig.* 3, one can set bias for minimum distortion and yet maintain response out to 15,000 cps. (It should be noted that a final determination of the amount of treble preemphasis required would depend upon a frequency-response run. Possibly, if response at 15,000 cps is kept flat, there would be excessive boost at lower treble frequencies. Thus in order to achieve the flattest possible response over the treble range as a whole, it may be necessary to accept response which is a few db down at 15,000 cps.)

Now let us consider the situation where the tape recorder represented in *Fig.* 3 operates at 7.5 ips. Curve 5 shows the 15,000-cps response at 7.5 ips as bias is varied. At minimum distortion bias, 15,000 cps response is about 10 db below 400 cps. Possibly this situation can be improved by increasing the amount of treble boost in the record amplifier. On the other hand, increasing the treble boost may cause appreciably greater tape overload in the upper treble range. Let us therefore assume that Curve 5 is based on the maximum amount of treble boost which may be safely used, taking into account the typical distribution of musical energy over the frequency range;[1] any additional treble boost would increase the likelihood of distortion.

Consequently, in order to maintain response out to 15,000 cycles at 7.5 ips, it is necessary to reduce bias. Curves 3 and 5 intersect at approximately −3.6 db bias; at this reduced bias, flat response out to 15,000 cps can be had. However, as bias is reduced to −3.6 db, IM distortion rises from 3.5 to 8.5 per cent for the signal level represented by Curve 1. On the other hand, by sacrificing 3 db in signal-to-noise ratio—that is, reducing signal level to the proportions represented by Curve 2—IM distortion can be kept at only 3 per cent when bias is −3.6 db.

(*Continued on page 81*)

[1] See the article by Herman Burstein, "Tape Recording Equalization," *Radio & Television News*, February 1956.

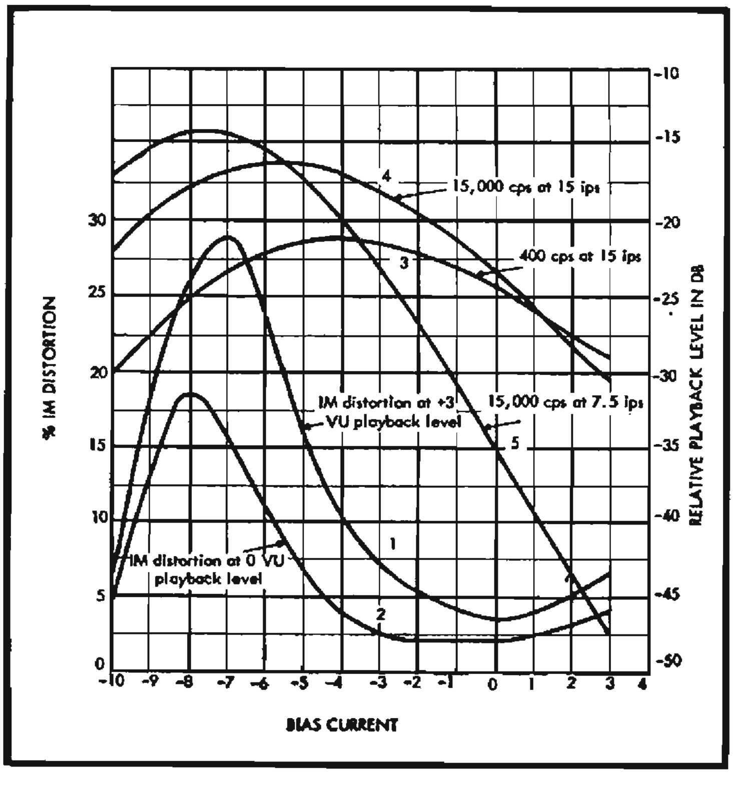


Fig. 3. Determination of optimum bias current.

TAPE DISTORTION

(from page 34)

It would seem that a reduction of only 3 db in signal-to-noise ratio is little enough to exchange for frequency response good to 15,000 instead of 7,500 cps. However, there are two counter views: (1) Few if any tape recorders have decibels to spare in the matter of signal-to-noise ratio. Whereas ratios of 70 db, 80 db, and better are commonly found in preamplifiers and power amplifiers, a tape recorder is doing extremely well if it gets up to 55 db. The designer of such a tape recorder fights hard for every last decibel or two in striving for a figure of 55 db, and a sacrifice of 3 db is consequently not unimportant. (2) Operating at –3.6 db bias puts the tape recording process into a region where a slight miscalculation as to input level produces a large difference in IM distortion. On the basis of *Fig.* 2 (or 3) at 0 db bias a 3 db miscalculation in level increases IM distortion only 1.5 per cent, but at –3.6 db bias the same miscalculation raises distortion by 5.5 per cent.

In view of the above two considerations, a recordist or tape machine designer equipped with the necessary test instruments might decide that at 7.5 ips he cannot afford, in terms of distortion and/or noise, the luxury of response more or less flat to 15,000 cps. Instead he may decide on a compromise course, shifting to a bias current intermediate between 0 and –3.6 db. Thus, for example, his choice might cost him only a 1 db reduction in signal-to-noise ratio and a reduction in flat response from 15,000 cps to 10,000 or 12,000 cps. At the same time he would have better protection against the consequences of over-recording than if he used –3.6 db bias.

In order to find this optimum bias point, it would be necessary to draw a number of curves similar to Curve 5 in Figure 3, showing the effect of bias current variations on several frequencies such as 9,000, 10,000, 12,000 cps, and so on. Input level should be kept 20 to 30 db below maximum recording level to avoid saturation. Then for each frequency curve one can evaluate, along the lines indicated in Fig. 3, what flat response out to this frequency signifies in terms of increase in distortion and/or reduction in signal-to-noise ratio because of departure from 0 db current. Based on these evaluations, the bias current can be selected which reflects the individual's concept of the optimum combination of frequency response, distortion, and signal-to-noise ratio within the capacities of a particular machine.

Conclusions

It has been pointed out that IM distortion can be a serious problem in tape recording, especially if one attempts to cut close to the line in maximizing signal-to-noise ratio; that adjustment of bias current can be quite critical if distortion is to be kept to a minimum at high recording levels; that departures from this critical bias point can exaggerate the consequences of excessive recording levels; and that, if the necessary test equipment is available, a definite procedure can be followed to determine first the bias current for minimum distortion and secondly the bias current which at speeds below 15 ips provides the most satisfactory compromise among the requirements of low distortion, wide frequency response, and high signal-to-noise ratio.

A number of judgments are required in determining maximum recording level and optimum bias current. How wide need frequency range be in order to give essentially satisfactory results? How much IM distortion is tolerable? How much for a split second? How much for a few seconds? How much for half an hour?

These of course are subjective judgments. Consequently the determination of maximum recording level and optimum bias current is not a hard and fast procedure.

The writers have heard a number of professional master tapes, one or two generations removed from the original, which, according to indications of a properly calibrated VU meter in playback, were recorded at excessively high levels; the VU pointer frequently kicked to full scale instead of staying below 0. Yet many of these seemingly over-recorded tapes nevertheless sounded clean to the ear. Although IM distortion was undoubtedly present in substantial degree, perhaps it was occurring in such short bursts as not to be disturbing; or perhaps the nature of the musical selection was such as to mask the effects of distortion. On the other hand, the writers have listened to master tapes seemingly recorded at conservative levels, yet less clean-sounding than desirable. Possibly other factors than recording level and bias setting intervened between the original source and good reproduction. At still other times the writers have listened to recordings velvety smooth except for a relatively high background of hum, noise, and tape hiss. They would gladly have accepted more distortion for less background distraction.

The above observations point up the fact that top quality tape recording is both a technique and a craft. It is advisable to have a technical grasp which enables one to adjust a tape recorder, if feasible, so as to make the most of its capabilities with respect to distortion, frequency range, and signal-to-noise ratio. At the same time, one must have the craftsman's touch, which is based on experience, qualitative judgment, and—the best instrument of all in audio work—an acute ear.

Equalization in Tape Recorders

HERMAN BURSTEIN

Complete understanding of the fundamentals of equalization as applied to a tape recorder should make it easy to understand the circuits encountered in commercial products. The author leads the way through these fundamentals.

To anyone who ever expects to become involved in the use, alignment, repair, or construction (of the electronics) of a tape recorder, an insight into the recorder's equalization—and the resulting effect upon frequency response, distortion, and signal-to-noise ratio—should be of value. The following will attempt to illuminate a somewhat elusive subject.

Unless stated otherwise, the discussion is predicated upon a tape speed of 7.5 ips, the most popular speed for high fidelity purposes so far as home use and commercial prerecorded tapes are concerned.

Figure 1 is an introduction to the problem of equalization. It shows the typical response of a tape recorder at 7.5 ips without equalization of any sort, based upon a high-quality record-playback head (or separate heads) and normal bias. The figure also presents suitable bass and treble equalization curves that would achieve virtually flat response between 20 and 15,000 cps.

The story told by *Fig.* 1 is far from complete. Moreover, while the scheme of equalization therein is practical—in fact, something like it is employed by many moderate-price tape recorders—it is not the one almost universally employed in professional American machines and generally used for production of commercial recorded tape. Instead, so-called NARTB equalization is employed. Actually, NARTB has officially promulgated an equalization standard only for recordings made at 15 ips. However, the 15 ips standard has been found practical at 7.5 ips, at least in high-quality machines, and therefore it has been widely adopted, particularly in professional and semi-professional use. So today NARTB equalization characteristics and principles constitute a *de facto* standard for 7.5 ips.

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The advantage of NARTB equalization lies in a higher signal-to-noise ratio, although in exchange it causes the recorder to skirt the borderline of distortion more closely and accentuates the problem of maintaining high-frequency response. The reasons for this will appear in the section on Turnover Frequency.

Using NARTB equalization for purposes of illustration, *Fig.* 2, in eight parts, traces the equalization process in a tape recorder from signal input to signal output.

(A) represents an input signal to the recorder, flat from 20 to 15,000 cps.

(B) shows the typical record equalization in a machine conforming to NARTB requirements. The slight bass boost is a specific characteristic, namely 3 db up at 50 cps and rising thereafter with declining frequency at a rate approaching 6 db per octave; this rise, however, is not maintained below the audio range. The treble-boost curve is not specific but is that required to make over-all (record-playback) response flat over the audio range, as will be shown in (G), a specific curve covering the bulk of the audio range is provided only for playback.

A record head is operated so that the current through the head, and thus the magnetic flux *applied* to the tape, is essentially proportional to the signal voltage. Since a flat signal, (A) is assumed, (B) therefore essentially represents the magnetic flux applied by the record head to the tape.

(C) shows the losses that typically take place in the record process at 7.5 ips. That is, the magnetic flux *recorded* on the tape is less than the flux *applied* by the head. These record losses are principally due to the erasing effect of bias current, which becomes more severe with increase in bias, and to the phenomenon of demagnetization. The latter may be explained as follows. Frequencies recorded on the tape form equivalent bar magnets. The higher the frequency, the shorter the magnet. The opposite poles of a magnet exert a mutual cancelling (demagnetizing) effect, which varies inversely with distance between poles. Thus demagnetization losses are greatest at high frequencies, where the opposite poles of the equivalent bar magnets are closest together.

(D) is the net result of applied flux and record losses, that is (B) and (C). Thus it is the magnetic flux actually recorded on the tape, as ***implicitly*** required by the NARTB standard. To measure recorded flux directly is a complex laboratory procedure. Therefore NARTB does not specify recorded flux as such. Instead it sets forth a playback characteristic, (F), which bears a complementary relationship to recorded flux, as will be explained. Thus, given a playback-equalization characteristic, recorded flux is implicitly defined, on the assumption that over-all response is to be essentially flat.

It should be noted that NARTB re-

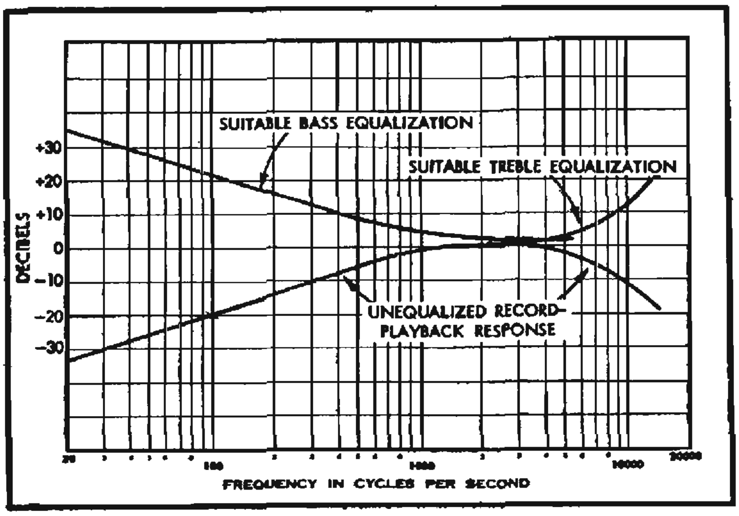

Fig. 1. Typical response of tape recorder at 7.5 ips without equalization.

corded flux has a turnover frequency at 3180 cps, at which point it drops 3 db. Thereafter it declines with increasing frequency at a rate approaching 6 db per octave. There is also a turnover frequency at 50 cps, where recorded flux is 3 db up; and the rise approaches 6 db per octave at lower frequencies.

(E) is the typical response of a high-quality playback (or record-playback) head at 7.5 ips, assuming that recorded *flux is constant at all frequencies.* The head is an inductive (velocity) device, so that its output is essentially proportional to frequency. In other words, its output tends to rise at the rate of 6 db per octave. However, high-frequency response of the head is limited by width of its gap. The particular response curvature in (E)—that is, departure from a 6-db-per-octave characteristic—is based upon a physical gap of .00025 in. and allows for *the fact that the effective*, or magnetic, gap tends to be somewhat larger than the physical one. Typically, the physical gap is about nine-tenths of the magnetic gap. Some heads have physical gaps as small as .00015 in., so that below 15,000 cps their response departs substantially less from *a 6-db-per-octave line than the head* represented in (E).

By adding the record losses of (C) *to the playback response of* (E), one obtains the unequalized record-playback characteristic of *Fig.* 1.

(F) is the net result of recorded flux and the response of the playback head —that is, (D) and (E). Thus it is the signal output of the playback head.

The solid line in (G) is the playback equalization specifically stipulated by NARTB. This characteristic has two turnover frequencies, which are the same *as for recorded flux:* (1) *At 3180* cps the rise in response with declining frequency reaches 3 db and thereafter *approaches a rate of 6 db per octave.* (2) At 50 cps bass boost is 3 db below the maximum it ever attains.

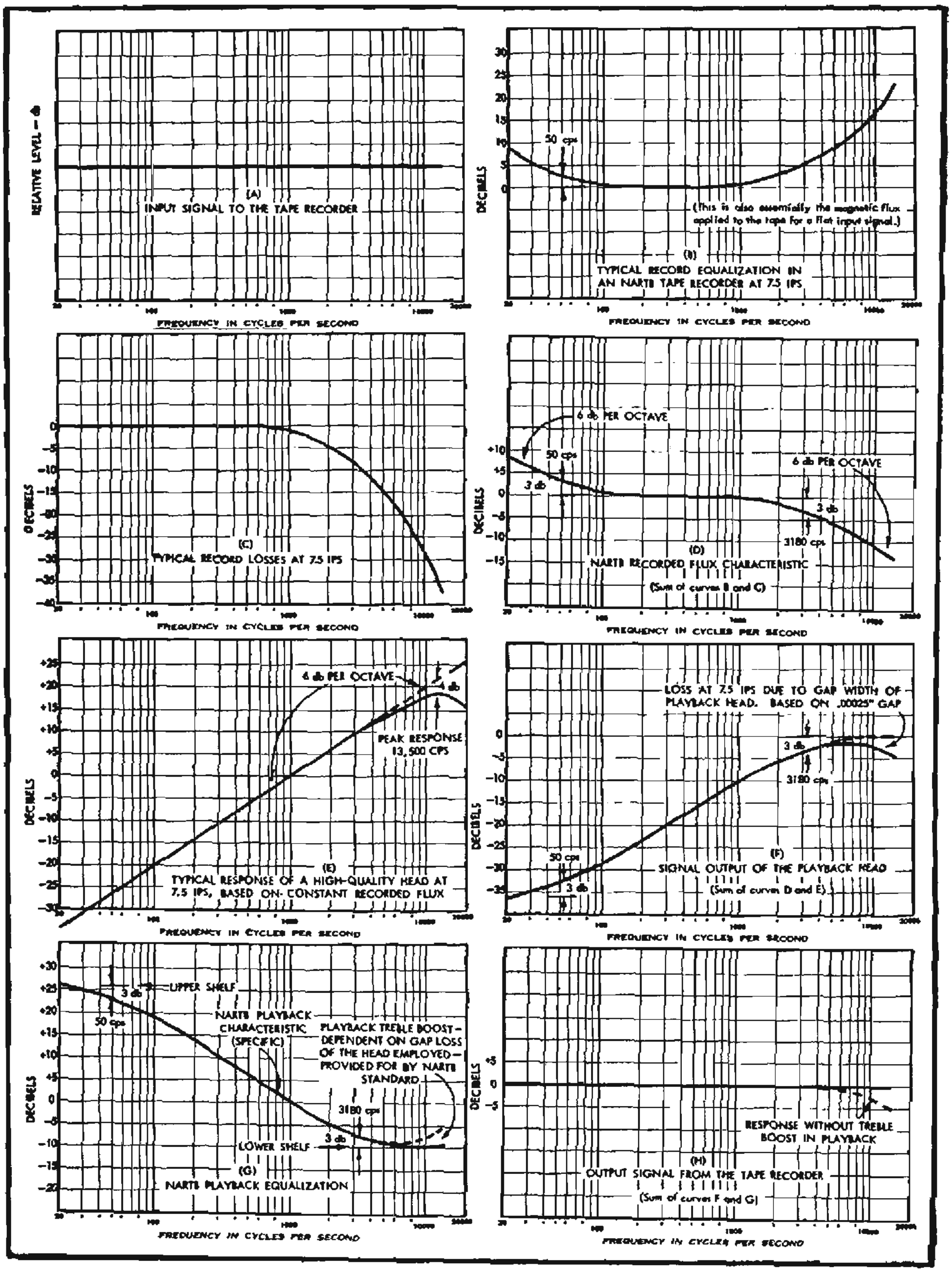

Fig. 2. Eight curves tracing the equalization process through a tape recorder and its associated amplifiers from signal input to playback-signal output.

The distance between the solid and dashed lines in (G) represents the playback treble boost required for flat response, as provided for by NARTB in general rather than specific terms. The *actual amount of treble boost depends* upon the losses in the playback head, chiefly due to gap width, as suggested in (E). Some heads, including the one upon which (E) is based, have gaps so narrow and losses so limited that adequate, though not perfectly flat, high-frequency response is obtained without playback treble boost.

The solid line in (H) is the net result of signal output of the playback head *and playback equalization—that is,* (F) and (G). Thus flat response is obtained, unless playback equalization does not incorporate treble boost, in which case there is a moderate decline at the high end, as *shown by the dashed line.* Flat, or nearly flat, response is not a coincidence but the result of maintaining the same turnover frequencies of 3180 and 50 cps in the recorded flux, in the signal *output of the playback head, and in the* playback equalization characteristic.

Turnover Frequency

In the case of NARTB playback *equalization, as just indicated, the first* turnover frequency is 3180 cps. On the other hand, as shown in *Fig.* 1, flat reponse can equally well be achieved with the aid of bass boost having a turnover frequency of approximately 1200 cps. One may well ask: How can different bass equalization characteristics achieve flat response in the same situation? And why is one turnover *frequency superior to another?*

To find the answers, let us commence with the fact that the response of a high-quality playback head at 7.5 ips essentially rises 6 db per octave with increas*ing frequency. For simplicity,* we shall ignore the relatively slight head losses of (E) in *Fig.* 2. Thus playback head response may be depicted in the simple fashion of (A) in *Fig.* 3.

Obviously, the rest of the tape recording system must then have a slope that *declines* 6 db per octave with increasing frequency, as shown at (B) in *Fig.* 3. From the discussion pertaining to *Fig.* 2 it can be realized that such a declining

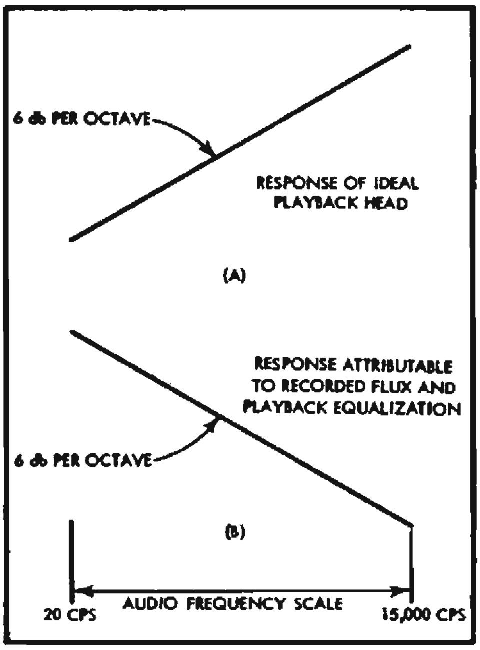

Fig. 3. Comparison of the playback head response (A) and the response of the balance of the recording system (B) in order to arrive at a flat over-all response curve.

slope is due to two factors: (1) recorded flux; (2) playback equalization. If the reader takes the trouble to add the curves (D) and (G) of *Fig.* 2, he will see that they produce a straight line declining 6 db per octave with increasing frequency.

Thus recorded flux and playback equalization bear a complementary relationship to each other, the sum of the two being a 6-db-per-octave declining slope. Given either characteristic, the other is implicitly defined. (Since we are here ignoring treble losses due to the playback head, it is assumed there is no treble boost in playback. If there were treble boost in playback, then the reference to playback equalization would be exclusive of any treble boost intended to compensate playback head losses. Thus in the case of (G) in *Fig.* 2 the reference would be only to the solid line.)

Recorded flux and playback equalization may be combined with each other in an infinite number of ways to produce the 6-db-per-octave slope. Five possibilities appear in *Fig.* 4. For ease of illustration, the characteristics are shown with sharp transition points—turnover frequencies—although in actuality they would have the gradual transition of *Fig.* 2. Furthermore, for simplicity, the 50-cps turnover frequency is ignored.

In *Fig.* 4, (A) represents a situation where recorded flux declines 6 db per octave throughout the audio range, so that zero playback equalization is required. At the other extreme, (E) in *Fig.* 4 indicates constant recorded flux throughout the audio range, in which case playback equalization must rise with declining frequency throughout the spectrum. (B), (C), and (D) of *Fig.* 4 are intermediate cases, where both the recorded flux and playback equalization have turnover frequencies. As recorded flux changes from flat to a 6-db-per-octave slope, playback equalization undergoes a complementary change from a 6-db-per-octave slope to flat.

At 7.5 ips, (A) in *Fig.* 4 is unrealistic because it is extremely wasteful of signal-to-noise ratio. Taking into account the nature of record losses at this speed and the permissible record treble boost, based largely upon the distribution of audio energy over the spectrum, more flux could be applied to the tape without overloading it; therefore a good deal more flux could also be recorded on the tape, resulting in an improved ratio in playback between audio signal and noise produced by the tape (hiss) and the playback amplifier.

On the other hand, the recorded flux of (E) in *Fig.* 4 is virtually impossible to achieve at 7.5 ips in the present state of the art (although feasible at 30 ips). Excessive record treble boost would be needed to overcome the record losses indicated at (C) in *Fig.* 2 at 7.5 ips. Such treble boost would produce considerable distortion by overloading the tape.

Between the extremes of (A) and (E) of *Fig.* 4 are combinations of recorded flux and playback equalization practical at 7.5 ips. (B) and (C) may be viewed as approximating equalization practices found in moderate-price recorders. It can be seen immediately that these practices involve fairly limited playback bass boost, thereby mitigating the problem of hum, which of course becomes more severe as such boost increases. Also, they indicate inferentially that relatively moderate amounts of treble boost are needed (because of substantial decline in recorded flux), which permits a saving in gain and tubes in the record amplifier.

(D) in *Fig.* 4 represents a tape recording system conforming to the NARTB standard. The recorded flux is just about as much as can be put on the tape at 7.5 ips without incurring appreciable distortion due to the large amount of treble boost required in the record amplifier, on the order of 23 db at 15,000 cps as indicated at (B) in *Fig.* 2. In complementary fashion, a great deal of playback bass boost is necessitated—36 db all told—so that very careful precautions against hum must be exercised in playback. In exchange for these disadvantages, namely a close risk of distortion and greater danger of hum, one obtains a high signal-to-noise ratio because of the relatively large amount of recorded flux.

In sum, the questions posed at the outset of this section have been answered as follows: (1) various playback equalization characteristics can be used to obtain flat response because each one is complemented by a different recorded flux characteristic, the sum of the two producing a 6-db-per-octave downward slope. (2) Playback equalization with NARTB turnover (3180 cps) is superior to equalization with a lower turnover frequency in that it reflects a scheme of equalization which permits a better signal-to-noise ratio because of greater recorded flux; at the same time, high fidelity requirements as to distortion and high-frequency response can be met.

Qualifications and Addenda

The first two sections of this article contain the basic story of tape recorder equalization as exemplifed by the NARTB standard. To facilitate explanation, certain factors have been omitted

(*Continued on page 85*)

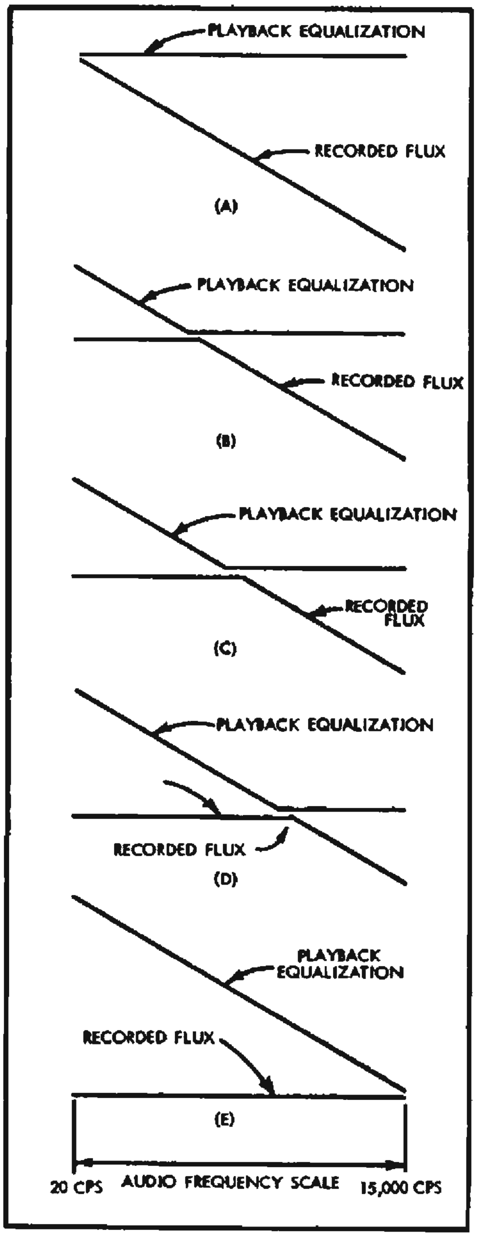

Fig. 4. Five possible combinations of recorded flux and playback equalization which combine to produce an over-all flat response curve.

TAPE RECORDER EQUALIZATION

(*from page 30*)

which deserve mention, although they do not change the essentials.

1. As indicated in *Fig.* 2, bass boost is provided predominantly in playback and treble boost predominantly in record by tape recorders that conform to the NARTB standard. This practice minimizes distortion by preventing tape overload when recording middle and low frequencies, where distortion is most apt to occur; and it minimizes noise by avoiding unnecessary treble boost in the playback amplifier, which is a high-gain affair inasmuch as playback head output is but a few millivolts. However, moderate-price machines often use "half-and-half" equalization; that is, they use equal amounts of bass boost in record

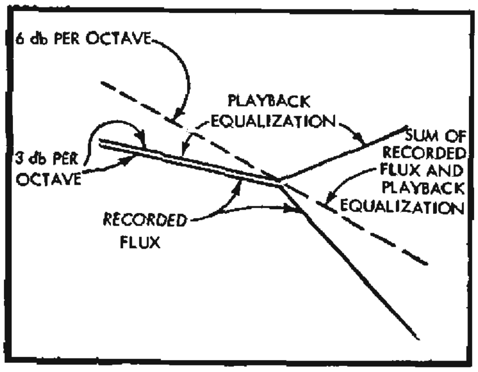

Fig. 5. Combination of "half-and-half" equalization often used in inexpensive tape recorders to provide over-all flat response without changing amplifier equalization between the record and playback modes.

and playback, and equal amounts of treble boost in both functions. As a result, playback bass boost essentially has a 3-db-per-octave instead of 6-db-per-octave slope, and recorded flux declines at a rate greater than 6 db per octave (greater to the extent of playback treble boost, excluding any such boost intended to overcome playback head losses). *Fig.* 5 illustrates half-and-half equalization. The net result is still a 6-db-per-octave declining slope, which is compensated by the 6-db-per-octave rising slope of the playback head. A major disadvantage of the pattern of equalization in *Fig.* 5, apart from the choice of turnover frequency, is that the signal-to-noise ratio deteriorates because of the decline in recorded flux at high frequencies and the compensating treble boost in the playback amplifier. Another disadvantage is that the increase in recorded flux at low frequencies heightens the danger of distortion at these frequencies due to tape overload when recording.

2. No mention has been made of record and playback head losses due to eddy currents and hysteresis—so-called "iron" losses. These losses increase with frequency. In a modern high-quality head they are kept to minimum, perhaps 1 db or less at 15,000 cps, so that essentially they can be ignored. However, if the losses are appreciable, the NARTB standard requires that in the case of the record head they be compensated by treble boost in the record amplifier, and that in the case of the playback head they be compensated by treble boost in the playback amplifier.

3. The various figures have assumed flat response from 20 to 15,000 cps. Actually, this is seldom achieved, particularly at 7.5 ips. At the very low end, the efficiency of the heads decreases. At the high end, to the extent that treble boost is used in the playback amplifier to compensate for playback-head losses (gap width, eddy currents, hysteresis), there is a corresponding increase in reproduced tape hiss and playback amplifier noise. Consequently, perfectly flat response to 15,000 cps is seldom sought at 7.5 ips; often not even at 15 ips. The practical possibilities are reflected in the NARTB standard, which permits response to be 4 db down at 50 and 15,000 cps.

4. Some of the curves in *Fig.* 2 are idealized. Thus the response of the playback head is not as smooth as shown at (E) in *Fig.* 2 and does not conform precisely to a 6-db-per-octave/throughout the middle and low ranges. The record equalization and recorded flux and other resulting curves are drawn to correspond exactly to the theoretically desired variations of response with frequency, although practical circuitry usually does not achieve this degree of precision. As a result, the record-playback response is not truly flat over the bulk of the audio range, allowing for the permissible decline to 4 db down at 50 and 15,000 cps. Instead, over-all response contains some bumps and hollows. However, in top-quality tape recorders these deviations are kept within ±1 db of a smooth line, while other, also excellent, machines maintain response within ±2 db. Even ±3 db variation is considered quite creditable. Æ

Tape Tension—The Neglected Dimension

Excessive tape tension increases wear on magnetic heads, breaks or permanently distorts tape physically, and increases the possibility of magnetic print-through. Optimum results are obtained when tension is no greater than that required for adequate head contact.

DR. ERWIN J. SAXL *

Tape tension must be kept to a minimum to avoid breakage during recorder operation. Moreover, excessive stress in the film will also set up changes in the character of the magnetically susceptible ferrous oxide layer that covers the supporting film and thus impair the quality of sound reproduction.

Tension also influences the internal pressure from layer to layer within the wound reel. Excessive pressures in the reel can be avoided and printing of the sound from one magnetically exposed location to adjoining layers above or below can be reduced by maintaining tension at the lowest practicable value

Tape tension is critical not only in recording and playback, but also in the manufacture of a high-quality tape. When operating with magnetic tapes for high-fidelity recording, the distance with which two signals are separated from each other is influenced by the stretch of the tape.

Stretching beyond the limits of elastic recovery of the film not only damages the tape mechanically but also engenders distortion in its ferromagnetic layers. Excessive pull produces an orientation of the ferrous oxide layer causing alignment of the magnetic domains and subsequent impairment of the electromagnetic quality. The erasing of the previous recording particularly becomes imperfect if same has been "set" by mechanical deformation, thus increasing the level of the background noise of subsequent recordings on the same tape.

Aside from influencing the structure of the tape proper and the ferrous layer covering it, there is the question of head wear. The heavier the film and its oxide layer (which is an abrasive equivalent to jeweler's rouge) presses on the head. the more the latter is worn. Add to this the braking action and there is considerable build-up of tension. Thus the minimum practicable pressure should be used which will maintain adequate, proper contact between tape and head.

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Fig. 1. Tensitron's new Tape Tension Meter.

Everything else being equal, the tape recorder which has the minimum of tape tension has the better quality.

By keeping the contact pressure (which is a function of tape tension) between tape and head low, fewer oxides are removed, and with less flaking-off the life of the tape is increased. Furthermore, with less damage to the ferrous layer, the background noise level is reduced and the over-all performance of the tape is increased.

Aside from the deleterious effects high constant tension, variations in tension will affect reproduction quality. To maintain reduced flutter and wow, it is essential to have the lowest possible tension.

Low tension also permits the use of comparatively thinner film materials, permitting a greater length to be wound on a reel of standard size. Last but not least, less expensive tape base materials can be used because the less the tension the less the stretch. Inexpensive long-playing film carriers are usable provided they are not stressed beyond the limit of elastic recovery. Thus, low tension also has the indirect advantage of reducing the cost in the film material.

Methods of Controlling Tension

Several methods are possible for the control of tape tension and for the maintenance of its uniformity—from electrical means for controlling tape transport to flywheels for minimizing short-time fluctuations of speed. However, before one can approach the correction of tension fluctuations, *a quantitative knowledge of the true tensions under performance conditions* is necessary.

Damage is caused not only by the smooth running tension, but also by the peaks—the sudden starting, stopping, and reversing peculiar to the operation of recorders. This differs vitally from the static conditions when a tape is pulled at a quasi-stationary speed. Thus, the use of ordinary spring scales to measure tape tension is inadequate because they do not show tensions under *performance* conditions. Accordingly, an instrument was designed for checking tensions of films and tapes *while in operation,* and this novel Tension Meter[1] is shown in *Fig.* 1. It consists essentially of a tension-sensitive roller between two guide rollers. To insert the running tape, the trigger is pulled back in the same manner on a gun. Lowering the two outer rollers and opening the space for easy insertion of the tape. Even while it is running tape can be inserted in the Tension Meter. To do this, the tape is placed on top of the two outer rollers and the trigger is then released gently.

The position of the two outer reference points that lift the tape into test position constitutes a constant mechanical shunt. The center roller is connected to a lever which is pivotally deflected with a minimum of friction in response to the tension applied to the tape.

To keep the conditions of checking uniformly constant, the center sensing roller is only lifted a small distance.

[1] U.S. Patent No. 2,591,724.

For all practical purposes this can be neglected, since the influence of the testing instrument upon the tape remains essentially constant under all comparative tension conditions. The small lever motion is amplified over a gauge movement. It shows with clarity, directly on a dial facing the observer, the tension of the tape while in motion. The rollers over which the tape advances run on anti-friction bearings so there is no significant braking of the tape.

In view of the large range of tensions encountered under the practical conditions of recording and playback, a rather large tension range has to be considered for visual indication. To provide an easily read scale, the meter was designed for a dual range, using an extended dial. The first revolution of the pointer goes from 0 to 200 grams, calibrated in black figures, and the second revolution goes up to 1000 grams and is calibrated in red figures. This combines adequate sensitivity for low tension measurements (where discrimination between individual tensions is particularly necessary) and covers the larger variations of occasional shock tensions.

Measurement of Variations

It is known that the tension of a reel changes in relation to the volume of tape on the reel. There is the long-time change as the reel gradually builds up from the layers close to its mandrel until it reaches the maximum outside diameter when it is full. In addition, superimposed over the long-time variation, there are short-time tension fluctuations. This causes tension changes as the reels unroll and build up.

These and other factors, singly and together, must be summarized to arrive at a realistic picture of the tape tension. They present an involved relation for the tensions existing during normal recording and playback, as well as during starting, stopping, and rewind and fast-forward operation.

As the full storage reel is unwound and the finished form builds up, we are dealing with a relation that may be indicated as in *Fig.* 2. The radius of the large storage reel is shown as R_2. The

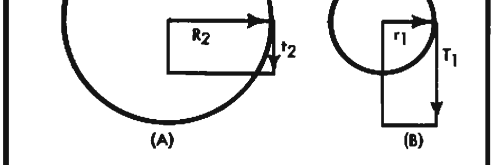

Fig. 2. Tape tension and the radius of reeled tape influence the work required to pull tape off a spool of large diameter (A) as compared to a near-empty reel (B).

Fig. 3. Checking operating tension in a typical tape recorder. (Courtesy Reeves Equipment Corp.)

tension, t_2, required to produce a given de-reeling action is then shown in the work necessary for unwinding the reel. In other words, for constant work in terms of a vector-diagram, the work is represented by the areas

$$r_1 \times T_1 = R_2 \times t_2$$

After the de-reeling from a storage package has progressed to a certain point, we are then dealing with a smaller diameter of the reel. A quicker rotation is now required to deliver the same amount of tape per time unit. Accordingly, more tension such as shown in T_1 is required now for the smaller radius r_1.

Thus between (A) and (B) we have a variety of conditions which indicate the gradual change from one to the other when a spool is de-reeled. As the storage reel gets smaller, tension has a tendency to build up unless compensation is made for this condition.

The opposite condition exists when we wind layers of tape upon an inner mandrel. Then we start with a small diameter such as shown in (B). It is gradually built up, finally resulting in a form which is shown in (A), and which is characterized by what may be compared to a larger lever.

Considering the entire system, we are confronted on one hand with the diminishing effective diameter represented by the de-reeling mechanism, and on the other hand with the increasing size of the wound reel which gradually is built up. This changes the relation between one reel which unwinds and delivers its tape to a mandrel that builds up by the same amount.

This condition of variable tensions is aggravated by the need for quick starting and stopping, particularly in computing, recording, and dictating work. Here we deal with the ambient tensions inherent in moving and stopping comparatively large masses of changing weights. This imposes difficult design requirements in order to arrive at a uniform rate of tape-transport tension. Since for adequate sound reproduction it is necessary to accelerate the tape from standstill to full running speed, the use of proper tension engineering becomes a necessity to avoid extremes of tension variations.

Braking

The de-reeling tension as applied to the braking mechanism that is linked to the storage reel is but one part of the summation of all the tensions. Another factor is the frictional retardation incurred by contact between recording-head and tape. Accordingly, these restraints have to be added to the de-reeling tension in the reel to arrive at the final tension.

The establishment and control of tension as measured under conditions of dynamic transport are mandatory to reduce film breakage. It is axiomatic that the complicated tension picture has to be known quantitatively, if we are to engineer proper compensations into the tape-moving chain. If we want to avoid tape breakage and achieve uniform recording without stretching or damage to the tape, we have to make sure that ambient tensions are well within the limits of elastic recovery of the base material. Speed control to compensate for different effective reel diameters, flywheels, slipping clutches, eddy current devices, variable resistors contacted by dancer arms and other methods are known to compensate in part at least for this variation.

By the use of the Tension Meter, precise tension measurements can now be made on which proper corrective action can be based. For instance, the characteristics of torque motors can be engineered so as to standardize the tension with which they move tape. This results in fewer stresses imposed upon the supporting film, increases in life, and a more uniform translatory motion during recording and playback.

Figure 3 shows how the Tension Meter is used with an operating tape recorder, in this case the Tandberg, which has a tape tension of only 10 grams.

(*Continued on page 64*)

TAPE TENSION

(*from page 36*)

Since the effective tension with which the tape operates represents the total history of all the tension events encountered during its passage from reel to reel, the tension measurement should be made just before the tape enters the final wind-up reel. Under operating conditions tension measurements have shown as much as 300 per cent variation between different makes of recorders. The lower and more uniform the tape tension is, the better the condition for the proper performance of the recorder.

From the viewpoint of computation and high fidelity recording the distance between two signals can be made precise at uniform tension since the tape is stretched the same amount at all times. This is an important factor in data handling, in precise computing mechanisms, magnetic tape memories, and similar applications where the distance from peak-to-peak of the magnetically recorded signal is critical.

Tension, a factor still frequently neglected in electronic engineering and its associated fields, should be given the serious consideration it deserves. Tension measurement and subsequent tension control will help the engineer to improve the mechanism for the adequate translatory motion of tape in recording devices of every type.

The VU Meter in Tape Recording

There are many advantages to the use of a standard VU meter as a level indicator and the author clarifies them, in addition to telling how the meter is connected and what it does and does not indicate. Every serious tape recordist will find this information valuable.

HERMAN BURSTEIN* and HENRY C. POLLAK

RECORDING A TAPE at too high or too low a level respectively entails excessive distortion or a poor signal-to-noise ratio. There is no great margin of safety between these dangers even in the best of tape machines. Consequently the record-level indicator plays a vital role in tape recording. How well it serves depends upon type of indicator, its stability, accuracy of calibration, manner of connection to the record-amplifier circuit, the prevention of false readings due to bias pickup, and the operator's skill in interpreting what he sees.

The VU meter is not inexpensive, and its use as a record-level indicator was largely confined to professional machines until recently. Other machines employed an electronic indicator, either an electron-ray (magic eye) tube or neon lamp. But with expansion of the home market for tape recorders suitable for high fidelity application, the VU meter has come into increasing use. Now it is found in a number of semi-professional tape recorders favored by audio fans and in several of the still lower-priced "home-type" machines. There is a continuing trend to ever-greater use of the VU meter or a similar meter by home units.

A full understanding of the role of the VU meter in tape recording should be of value to the technician and audiofan concerned with the repair, maintenance, modification, selection, or use of a tape recorder.

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Advantages of the VU Meter

The VU meter has a number of advantages over the electron-ray tube and neon lamp. Among them are:

1. It indicates the extent to which the record level varies from that producing maximum permissible distortion. The neon lamp can only indicate when level is too high or too low, but not by how much. The electron-ray tube does show a continuous variation, from open eye to closed eye, but its meaning is uncertain. The VU meter enables the recordist to make adjustments in record level easily and accurately on the basis of what he is recording and for what purpose.

2. It is a standard, relatively uniform product. If replaced, the new meter gives essentially the same readings as its predecessor. Electronic indicators have tolerances such that individual tubes or lamps of the same type may produce significantly different readings in a given circuit.

3. It is stable over time.

4. It is very sensitive and therefore has minimum driving requirements.

5. It permits the very important function of checking bias current accurately. Too much bias reduces both distortion and high-frequency response; too little has the opposite effects. The optimum amount of bias is fairly critical, particularly at 7.5 ips, if high fidelity results are sought. By means of a switching arrangement (*Figs.* 1, 2, and 6), the VU meter can measure bias current flowing through the record head. A calibrating resistor is employed so that optimum bias corresponds to a specific point on the meter, usually 0 VU.

6. It permits measuring playback level (*Figs.* 1, 2, and 6). This is important in professional applications, so that the amplitude of the playback signal can be adjusted by means of the level control to meet the requirements of following equipment in a recording or broad-

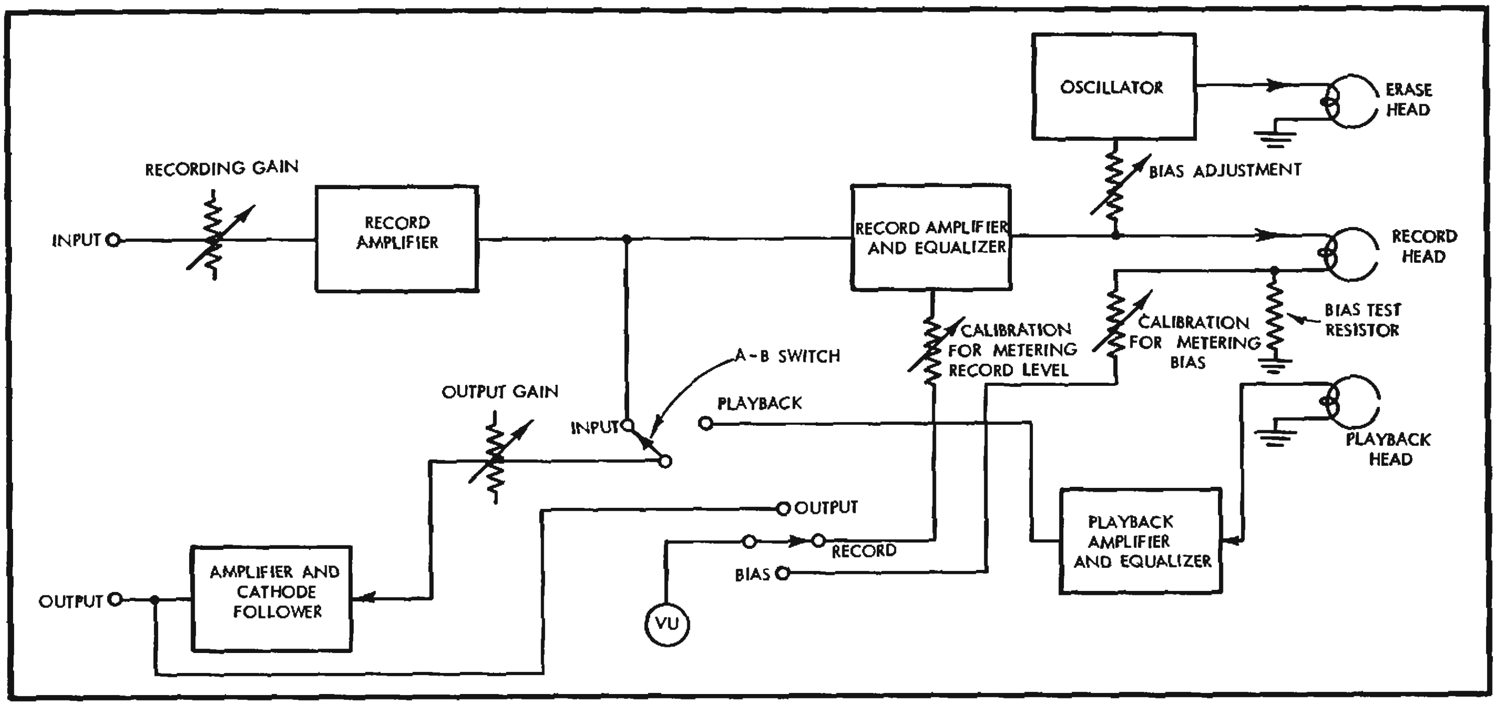

Fig. 1. Typical employment of a VU meter in a tape recorder with separate record and playback heads.

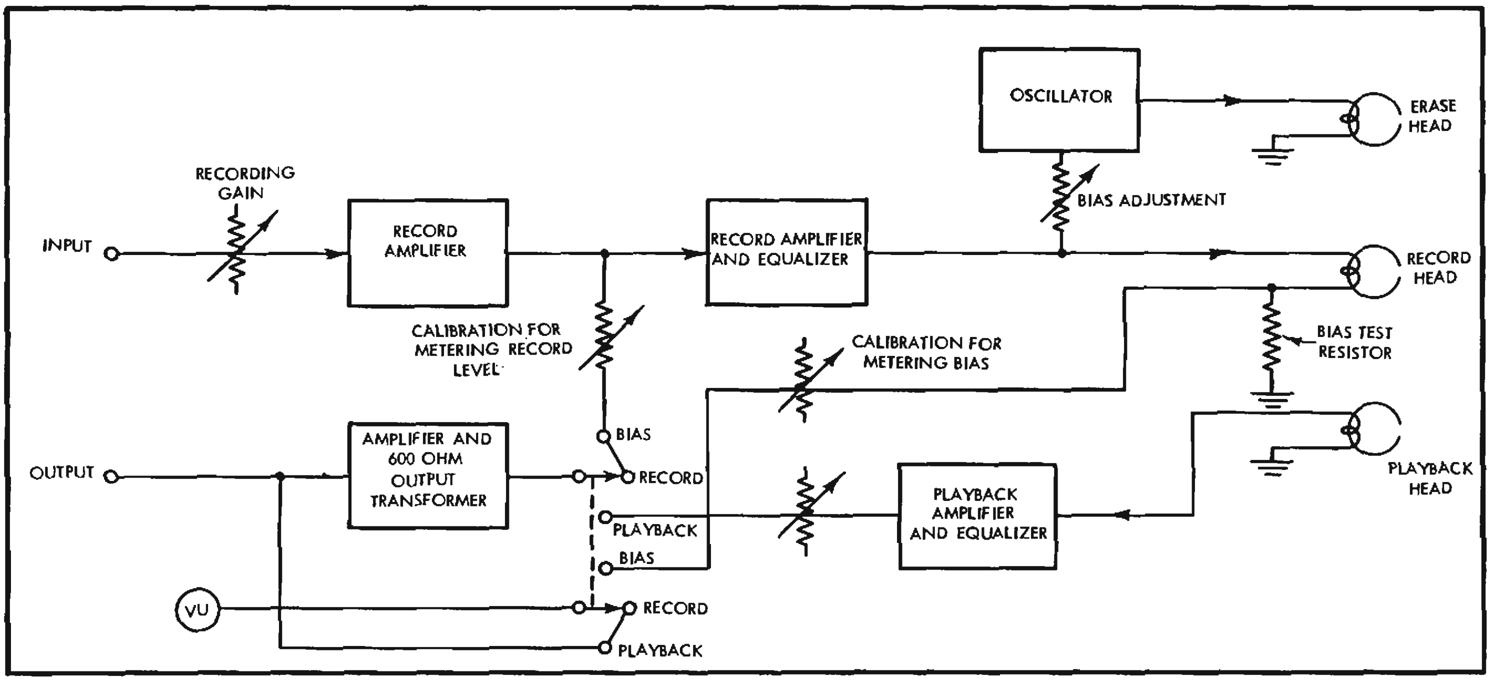

Fig. 2. Alternative employment of the VU meter in a tape recorder with separate record and playback heads.

cast studio. If the signal is too low, there may be interference from adjacent audio lines operating at a higher level, or the following equipment may produce insufficient amplification. On the other hand, a playback level that is too high may produce crosstalk on other audio lines, or cause distortion or unnecessary compression in associated line amplifiers without level controls.

Characteristics of the VU Meter

The VU meter (*Fig.* 3) contains a 50-microampere d.c. movement with a full-wave copper-oxide rectifier. The standard meter has a 4 in. dial with a double scale, one reading from −20 to +3 VU, and the other from 0 to 100 (percent). Usually the VU units are featured, with the VU scale in black, this being known as an "A" scale. In the "B" scale the percentages are featured and show as black figures. The secondary scale is in red, and the color of the dial background is standardized as an "easy-on-the-eyes" yellow.

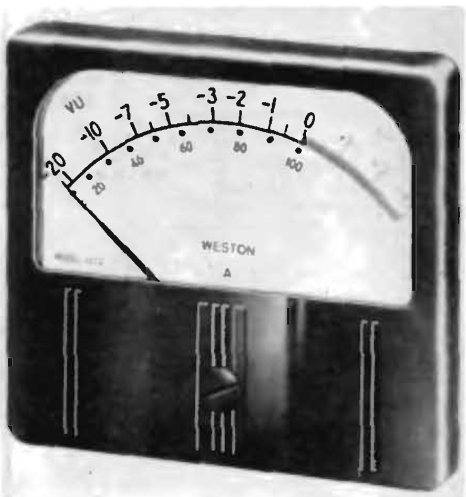

Fig. 3. Photo of a Weston VU meter with the "A" scale. (Courtesy Daystrom, Inc.)

VU units are simply decibels, with 0 being an arbitrary reference level: one milliwatt of power passing through a 600-ohm resistance.

The VU meter is designed to be placed in series with a 3,600 ohm external resistor, as shown in *Fig.* 4. Resistance of the meter movement plus that of the rectifier plus that of an enclosed resistor totals 3900 ohms. Therefore the total load of the *meter circuit* across the 600-ohm line is 7500 ohms. And now the meter reads 0 VU for 2.5 milliwatts of power in the 600-ohm line, which is actually +4 VU, since volume units are by definition referred to 1 mw in a 600-ohm circuit.

When employed in the standard manner (across a 600-ohm line and in series with a 3600-ohm resistor), the standard VU meter must exhibit certain characteristics, which enable the practiced operator to rely upon its readings and interpret them correctly. The overshoot must be between 1 and 1.5 per cent when a sine wave of 2.5 milliwatts power is suddenly introduced in the line; the pointer must reach 99 on the percentage scale within 0.3 seconds; frequency response must be within ±0.2 db between 35 and 16,000 cps; loading distortion must not exceed 0.2 per cent harmonic when the meter is placed across a 600-ohm line; the meter must withstand for half a second ten times the voltage which produces a 0–VU indication (1.228 volts), and to withstand continuously a five-fold voltage overload.

The very high sensitivity of the VU meter is due to a high-flux-density magnet of special alloy, and if the meter is mounted in a steel or iron panel some of the flux is shunted, thereby upsetting the calibration. VU meters intended for such mounting must be especially calibrated by the manufacturer upon the basis of panel thickness.

Drive Requirements and Circuitry

When the VU meter is used as a record-level indicator and as a means of measuring bias current, it is unimportant what the reading signifies in terms of power across a 600-ohm line. The important thing is that a given point on the scale, usually 0 VU, should correspond to the record level producing maximum permissible distortion on the tape, or to the correct bias current. On the other hand, when the VU meter is driven as it was designed to be (across a 600-ohm line and in series with a 3600 ohm resistor), its dynamic characteristics (overshoot and response time) will be preserved, which is important to the recordist.

Only 1.228 volts is required to drive the VU meter when it is connected in the standard manner. The necessary drive is easily available in the record amplifier. Also, it is consistent with the

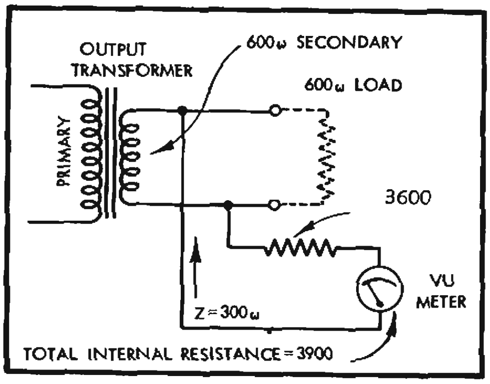

Fig. 4. Standard method of connecting a VU meter.

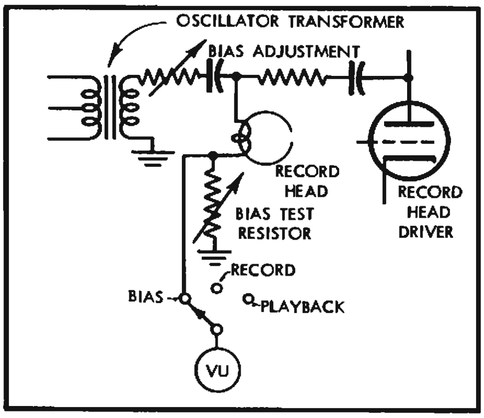

Fig. 5. Circuit for metering bias current and for calibrating the meter indication.

output that may be expected of the playback amplifier.

To measure bias current through the record head, a "test" resistor is employed between one lead of the head and ground; and the meter, via a switching arrangement, is placed across the test resistor. In order that a given point on the meter, usually 0 VU, should correspond to correct bias current, a variable calibrating resistor is used. Sometimes this calibrating resistor is the same as the test resistor, which is made variable, as in *Fig.* 5. Frequently, however, a separate calibrating resistor is employed, as shown in *Figs.* 1 and 2.

Figure 1 illustrates how the VU meter is driven in some tape recorders. The VU meter can be switched across a 600-ohm transformer, which is associated with a stage of amplification. This amplifier stage can be connected to the record section, via a calibrating resistor, so that it serves as a record-level indicator, or it can be connected to the playback section so that the meter serves as a playback-level indicator. In the third position the meter measures bias current.

A cathode follower typically has an output impedance of about 500 ohms, and can be satisfactorily used to drive the VU meter, as illustrated in *Fig.* 2. Here the meter is driven by the cathode follower only when measuring playback level. For measuring record level and bias current, it is connected to the appropriate points in the circuit through calibrating resistors. Note that by means of an A-B switch the meter can compare the incoming signal with the playback signal; this requires that the VU meter be switched to the "output" position.

In some circuits the VU meter is not placed directly across the output, as in *Figs.* 1 and 2, but is driven by its own cathode follower, as illustrated in *Fig.* 6. The advantage is that the VU meter does not load down the audio signal. Though specifications call for the VU meter to produce no more than 0.2 per cent harmonic distortion when connected in the standard manner, this may correspond to a greater, and significant, amount of intermodulation distortion.

Loading Distortion

If the VU meter, including its external 3600-ohm resistor, is placed across a signal circuit with an impedance much greater than 600 ohms, excessive loading distortion will result unless the external resistor is suitably increased. Impedance of the meter circuit must be at least ten times that of the signal circuit to avoid excessive loading. Heavy loading also attenuates the signal.

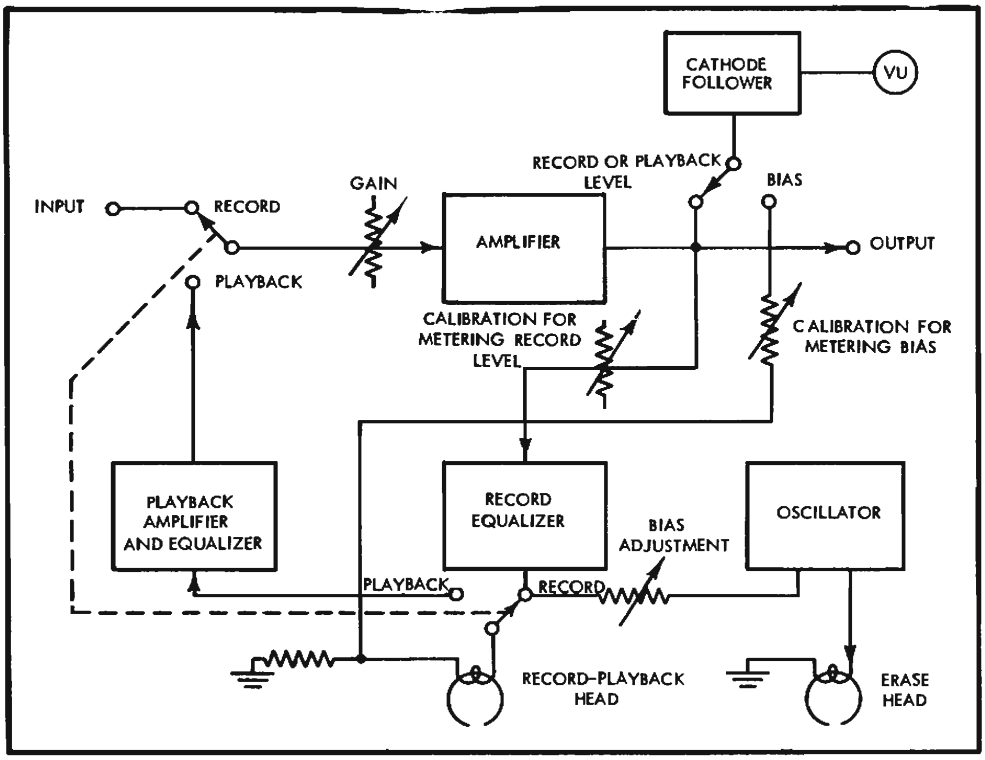

Fig. 6. Employment of the VU meter in a tape recorder with a combination record-playback head.

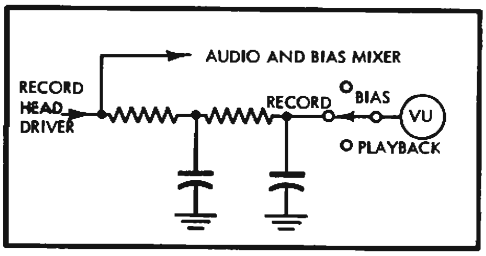

Fig. 7. Filter for preventing bias pickup by the VU meter.

The copper-oxide rectifier used in the VU meter (and other meters) has nonlinear characteristics. Its impedance varies with signal polarity and instantaneous voltage. Assume that the meter circuit, including the 3600 ohm external resistor, appears to the signal circuit as a 7500-ohm resistance fluctuating over a 10 per cent range. If a 10 per cent change in load resistance has a significant effect on signal voltage, which is the case if load impedance is appreciably less than ten times the source impedance, the nonconstant load resistance will cause significant distortion.

Record-Level Connection

In most instances the VU meter is connected to the record signal at a stage prior to equalization, as illustrated in *Figs.* 1 and 6, rather than after equalization as in *Fig.* 2. Record equalization in high-quality tape recorders generally conforms to NARTB standards, so that it consists of a little bass boost (about 3 db at 50 cps) and of considerable treble boost (20 db or more at 15,000 cps at 7.5 ips).

The post-equalization connection, *Fig.* 2, has the advantage of indicating the actual amount of signal applied to the tape at *all* frequencies, so that one may guard against tape overload at the high frequencies, which are so greatly boosted at the commonly used speed of 7.5 ips.

It may be questioned, then, whether the pre-equalization connection satisfactorily warns against distortion in the treble range. Essentially, yes. Record treble boost largely affects the audio spectrum above 3000 cps, and in this area the decline of audio energy with rising frequency tends to compensate the treble boost. Furthermore, for the same amount of distortion, somewhat more signal can be applied to the tape at high frequencies than at the mid-range and low ones.

The pre-equalization connection has the following possible advantages: (1) Taking the record signal at a point before the equalization stages will provide better isolation between the VU meter and the record head; this helps prevent bias current in the head from producing an unwanted and misleading indication on the meter. (2) After pronounced treble boost, high frequencies,

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VU METER

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particularly transients, may be of such amplitude as to damage the meter, especially if record level were accidentally set too high. (3) If the meter is used to compare playback level with the incoming signal, proper comparison would not be obtained if the incoming signal were metered after treble emphasis.

Record Level Calibration

As shown in *Figs.* 1, 2, and 6, a variable calibrating resistor enables the VU meter to read 0 VU at a recording level which produces maximum permissible distortion on the tape. Miscalibration defeats the basic purpose of the indicator.

Calibration is usually based on a record signal producing 2 or 3 per cent harmonic distortion, although 1 per cent is also used. While these amounts of harmonic distortion seem relatively innocuous for peak signals, it should be borne in mind that the corresponding IM distortion may be much greater. Thus 1 per cent harmonic may correspond to about 5 to 10 per cent IM, while 3 per cent harmonic may entail 20 to 30 per cent IM.

The principal disadvantage of the VU meter compared with electronic indicators is that it reads average rather than peak levels. Due to mechanical inertia, it cannot follow sharp transients, which may exceed the average level by 10 to 20 db. Such transients can cause very severe, though brief, distortion.

Therefore in calibrating a VU meter it is desirable to allow for the difference between the indication of average level and the actual peak level. Many, though not all, professional machines provide a margin of 6 to 10 db by causing the VU meter to read 0 VU for a sine wave signal (usually 400 cps) which is 6 to 10 db less than that which produces maximum permissible distortion (1, 2, or 3 per cent, depending upon the manufacturer's sights).

It is not strictly necessary to set the meter ahead in this manner, for the recordist could instead adjust record level so that the meter pointer always stays about 6 to 10 db below 0 VU. However, this crowds the working-range of the meter into a relatively small part of the scale.

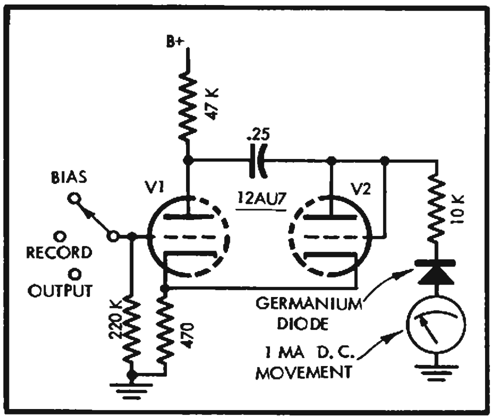

Fig. 9. A VTVM-type meter used in a tape recorder. (American Electronics.)

Interpreting the Meter Indication

Even though the VU meter allows for the difference between indicated and actual level, the recordist must still exercise judgment and bring experience to bear. Various types of sound have varying relationship between peak and average level, so that allowing the pointer to hit 0 VU may result in over-recording in one case and under-recording in another. Moreover, distortion is less objectionable in some circumstances than in others, and this too should be taken into consideration in setting record level on the basis of what the VU meter shows.

Eliminating Bias Pickup

Tape recorders must take precautions to prevent bias current from inadvertently reaching the VU meter and thereby causing it to indicate higher than it should. One measure has already been discussed, namely separating the meter circuit from the record head by connecting this circuit prior to the equalization stages.

Other precautionary devices consist of filters or traps. *Figure* 7 shows a two-stage low-pass filter, which permits the audio frequencies to reach the VU meter, but rapidly attenuates the bias current, which is much higher in frequency, 60 to 100 KC being typical in high-quality recorders.

Figure 8 shows a resonant trap having a very high impedance at the bias frequency and a relatively low impedance at audio frequencies. Audio current can flow from the record amplifier through the trap to the record head, but bias current cannot flow to a significant extent in the reverse direction.

A breakdown in circuits such as the above can impair the validity of the meter indication and thus affect the quality of the tape recording.

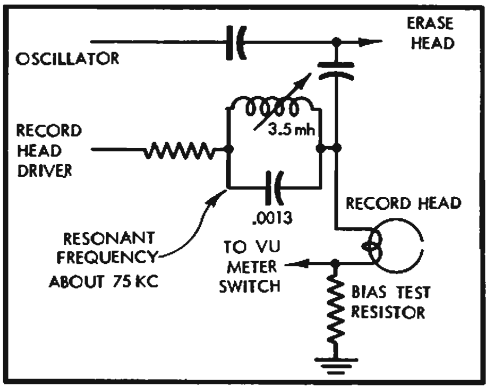

Fig. 8. Use of a resonant trap to prevent bias current from reaching the record head driver and earlier stages. (Presto SR-27.)

The VTVM Indicator

In closing it is appropriate to mention that not every meter with a VU scale is a standard VU meter. There are also some non-standard ones, which may or may not be equally satisfactory.

Since the standard VU meter is very sensitive and therefore costly, manufacturers of tape recorders sometimes employ a less sensitive movement, typically 1 ma, and drive it by means of a voltage amplifier. This is in effect a VTVM. An example of one appears in *Fig.* 9. Although 1-ma movements with the same characteristics as a VU meter are not available as a stock item, they can be obtained by a manufacturer on special order. In such a case there would be no disadvantage to the user. Æ

What Kind of Tape Machine for Your Audio System?

HERMAN BURSTEIN*

The first step in buying a tape recorder is to determine what your requirements are—some users want only to play recorded tapes, others to record their favorite music off-the-air, and still others want to engage in the most complicated of tape maneuvers.

If you purchase an FM tuner, it is quite well understood what you are getting from a functional point of view. Essentially you are acquiring a device that will convert a radio signal into an audio voltage. It remains for *other* audio components to amplify this voltage, adjust its tonal balance if necessary, and finally convert it into sound. Similarly, if you acquire a control amplifier (often called a preamplifier) or a power amplier or a speaker, its functions are quite well defined; you do not expect a power amplifier, for example, to do what a control amplifier does (selection of signal source, control of gain, adjustment of bass and treble, filtering of highs and lows, preamplification and equalization of low-level signals, and so on), or vice versa.

But the province of a tape machine is not nearly so well defined. (For the time being we shall use the term tape machine in lieu of tape recorder because the device does not necessarily have to record.) The scope of a tape machine can range from merely transporting the tape to providing a complete self-contained audio system for the recording and reproduction of sound on tape.

There are five types of purchase that one can make in order to bring tape reproduction into the home:

1. Transport only
2. Transport and separate playback electronics
3. Transport and separate record-playback electronics
4. Tape recorder proper—transport and integrated record-playback electronics.
5. Self-contained tape recorder—including power amplifier and speaker

The purchase suited to a given individual depends partly upon his wants,

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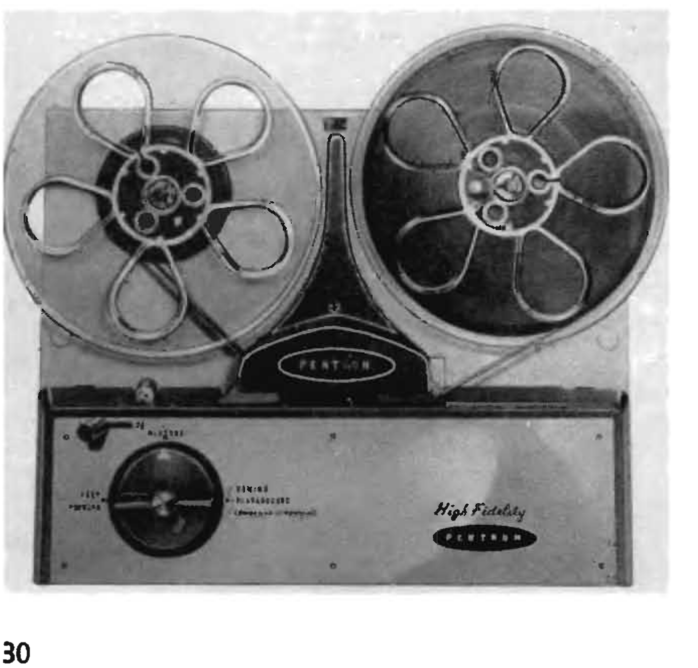
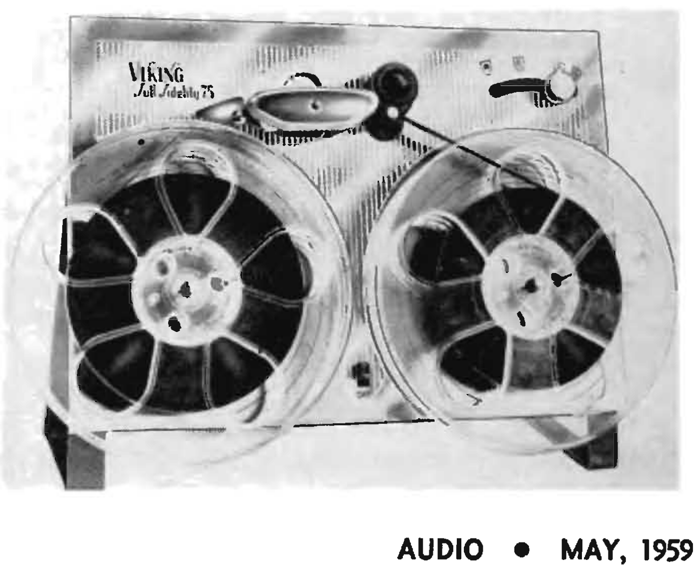


Fig. 1. (left) A typical transport mechanism—the Pentron TM Series.

Fig. 2. (below) Another type of tape mechanism—Viking FF75.



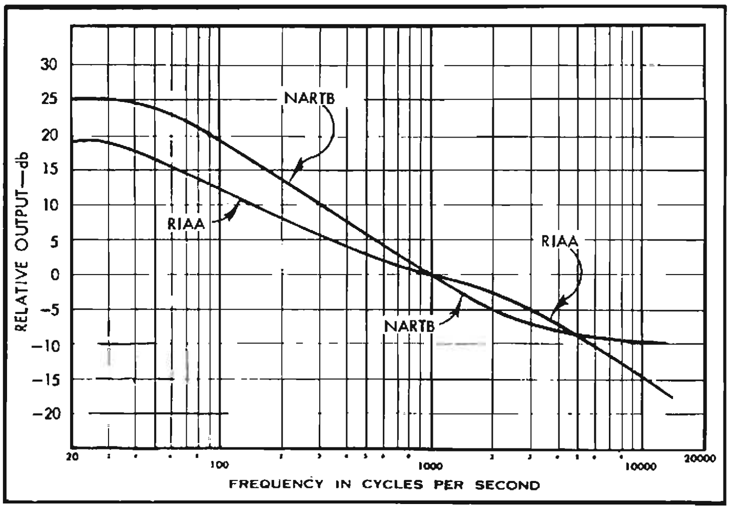


Fig. 3. NARTB tape playback equalization compared with RIAA phono equalization.

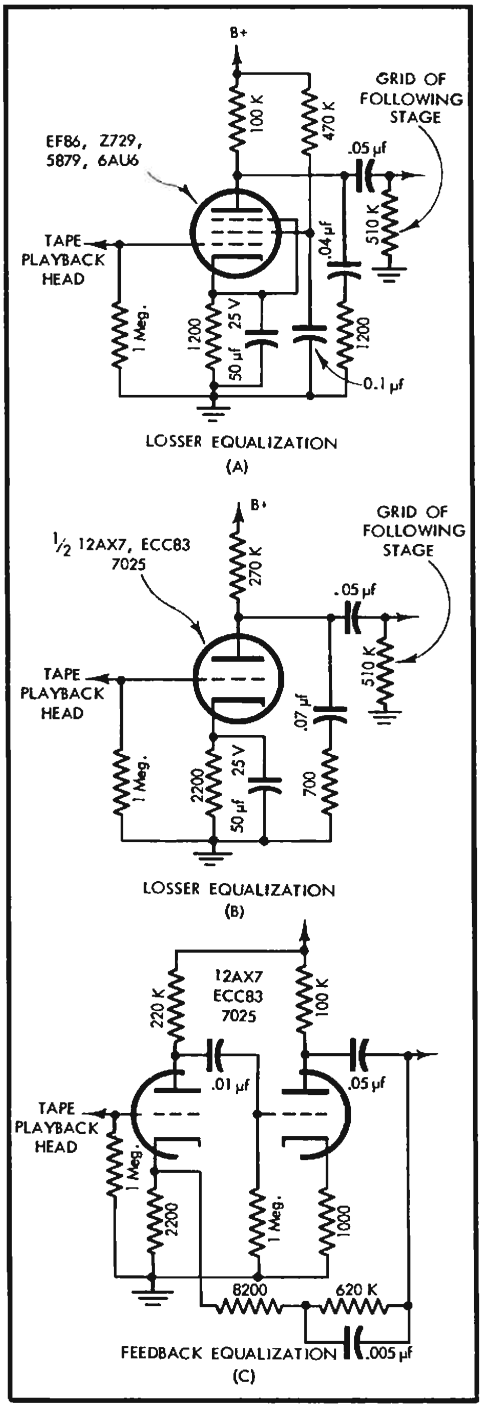


Fig. 4. NARTB playback equalization circuits.

partly upon his standards of audio reproduction, and partly upon the audio equipment he already owns. Through an understanding of each of these types of acquisition in terms of functions performed, quality of performance, limitations, and economy, the audiofan can decide which acquisition is best for him.

Transport Only

As many people know, it is feasible to purchase simply the transport (*Figs.* 1 and 2), the mechanical device that moves the tape from a supply reel, past the tape heads, and onto a takeup reel. Transports capable of good performance can be had for substantially less than $100, although it is also possible to pay several hundred dollars for units of semi-professional and professional quality. Ordinarily, the transport comes with one or more heads, and in some cases the unit may provide space and facilities for adding more heads. To illustrate, if the owner wishes only to play recorded tapes, he can purchase a transport containing a playback head. However, if he plans on recording, he will need to add at least an erase head; if he desires the best possible results, he may wish to add a separate recording head instead of using one head for both recording and playback. The transport may contain room for as many as five heads; considering the diversity of types of heads—full-track monophonic, half-track monophonic, two-track stereo, and four-track stereo—and taking into account special practices such as sound on sound recording, some audiofans might want as many as five heads.

The transport plays a role in the audio system analogous to the phonograph. It is a mechanical device (like the turntable) incorporating a transducer (like the phono cartridge) that delivers a small signal requiring amplification to bring it up to a usable level and equalization to achieve flat frequency response.

To be able to limit one's purchase to a tape transport without electronics, it is necessary that the control amplifier in one's audio system contain preamplification facilities (including equalization) specifically designed to accommodate the signal directly from a tape playback head. This corresponds to the requirements imposed by a magnetic phono cartridge, where the control amplifier must provide RIAA compensation (equalization for earlier recording characteristics, such as LP and AES, are also usually supplied). In the case of the tape head, and assuming operation at 7.5 ips, NARTB equalization is required instead; it is also becoming the practice to use NARTB equalization at 3.75 ips, although this is not yet as common as at 7.5 ips. *Figure* 3 shows the RIAA and NARTB curves.

Most control amplifiers made today contain equalization for the signal obtained directly from a tape playback head. That is, they have an input jack marked "tape head." However, they do not all provide NARTB equalization. The writer has measured the tape equalization curves of a number of control amplifiers, and while some follow the NARTB curve within a decibel or two,

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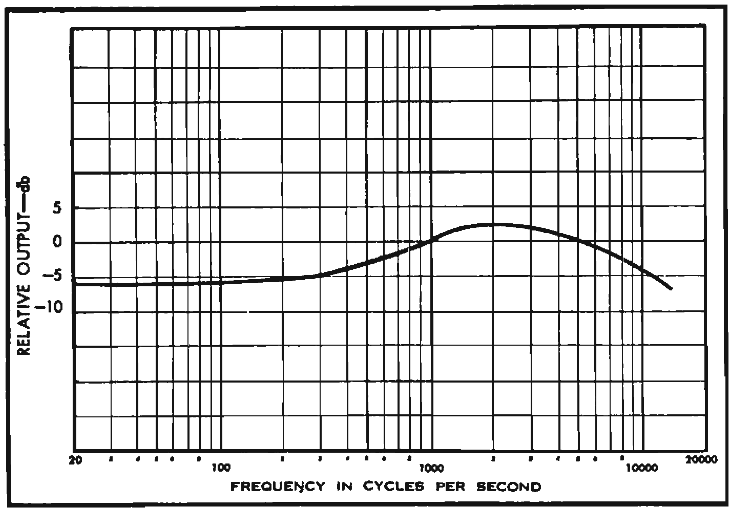

Fig. 5. Frequency response resulting from the use of RIAA compensation for a tape requiring NARTB equalization.

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others fall significantly wide of the mark, usually in the direction of inadequate bass boost, causing the sound to be definitely on the "thin" side.

Therefore if one intends to use his control amplifier for equalizing the signar from a tape head, it is a good idea to have the control amplifier checked out by a technician aganist the NARTB curve. This can be done by playing back a test tape and checking for flat response at the output of the control amplifier, or by feeding signals from an audio oscillator into the tape-head input and checking whether the response at the output of the control amplifier conforms to *Fig.* 3. If response does not deviate more than 3 db from the NARTB curve between 50 and 15.000 cps, it is acceptable.

For those having older control amplifiers without an input for tape head, it is fairly simple to convert one of the phono positions—such as LP or "European"—so that the equalization will instead be NARTB. Thus one can accommodate the signal from a tape head which is fed into the magnetic phono jack. But one must then of course remove the phono plug from the jack. On the other hand, some control amplifiers have two magnetic phono jacks that can be used simultaneously, and in this case one of the jacks can be converted to tape head use. Or, if there is an input jack marked "microphone," this can be converted. *Figure* 4 shows some typical circuits that will produce NARTB playback equalization; parts A and B show losser type equalization, while C shows feedback equalization. If none of these circuits is suitable for the particular control amplifier, information can generally be obtained from the manufacturer of the amplifier in question on modifying the phono equalization circuit to produce NARTB equalization instead.

If no other alternative is available, one can simply feed the signal from the tape head into the magnetic phono input jack. Amplification will usually be sufficient, but because of the difference between the RIAA and NARTB curves frequency response will be as shown in *Fig.* 5; bass will be insufficient, there will be a slight hump in the middle range, and the upper treble frequencies will suffer. Some degree of correction can be achieved by using the bass control for bass boost, but there is still apt to be a significant departure from flat response in this region because it is

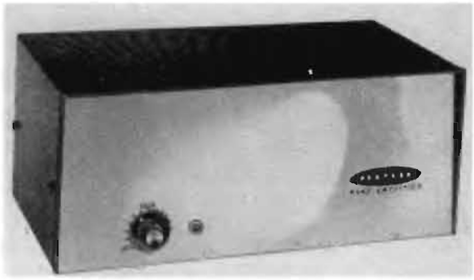

Fig. 6. Tape playback amplifier—Pentron CA-11.

unlikely that the characteristic of the bass control will exactly complement the bass deficiency shown in *Fig.* 5. It is not feasible to compensate the hump in the mid-range, because attenuation by means of the treble control would aggravate the treble deficiency at the high end.

Recording

Unlike a phonograph, the tape transport is also capable of recording, given suitable electronics. But recording electronics differ significantly from playback electronics in two vital respects:

1. An oscillator is required to furnish high-frequency bias current to the record head, in order to bring distortion

Fig. 7. Another type of tape playback amplifier—Viking PB60.

down to acceptable levels and to increase the amount of signal recorded on the tape. Bias current is also used to energize the erase head, so that the previous signal on the tape will be erased before it reaches the record head. (However, this is not a strictly necessary feature, although it is virtually universal; one could instead use a bulk eraser (a large a.c. electromagnet) to erase the tape, which is often done by those seeking maximum results.)

2. A record level indicator—meter, magic eye tube, or neon lamp—is required to indicate whether the signal being recorded is too great, breeding excessive distortion, or whether it is too small, resulting in a poor signal-to-noise ratio.

These additional requirements mean that it is far from simple to modify the electronics of a control amplifier so that they will be suitable for recording as well as playing a tape. To date, the writer has seen no control amplifier incorporating complete tape electronics, although there is always the possibility that some manufacturer will eventually bring out such a unit.

But as things stand, by purchasing only a transport the audiofan limits himself to playing tapes recorded by others—either commercial tapes (that are considerably more expensive than phono discs of equivalent playing time) or tapes recorded by friends. If one does not own an FM tuner, or if the content and quality of FM programs in one's locality offer no incentive to preserving them on tape, the absence of recording facilities may not be missed. On the other hand, many use their tape recorders extensively for copying discs they have purchased. With proper care of the machine and the tape, the latter can be played thousands of times without suffering blemish (such as scratches, ticks, and pops in the case of discs) and without significantly undergoing change with respect to frequency response and distortion. Hence the disc can be played once in order to record a tape and then be put aside, and the tape can be played as often as wished instead. In the event of misadventure to the tape—loss, accidental erasure, damage to several feet because of a tangle, or the like—the disc can be brought out again for making a new tape.

It is very important that the cable connecting the playback head on the transport to the control amplifier be as short as possible. The head is customarily a high impedance affair, and 200 μμf or less capacitance across it produces a sharp drop in response close to or within the audio range. The greater the capacitance, the lower is the frequency at which treble droop begins. A major factor in this shunt capacitance is that of the cable. Therefore one should use cable of minimum capacitance per foot (about 25 μμf per foot is available) and of minimum length.

Because the signal from the tape head is very small, just a few millivolts, it is subjected to tremendous amplification. Moreover, as shown in *Fig.* 3, a great deal of bass boost is applied to it. Hence any hum picked up by the cable will be amplified to the extent where it is apt to be audible. Accordingly, one must be careful to route the cable from the head to the control amplifier so that it does not encounter magnetic fields produced by motors or transformers.

Transport and Electronics

Those desiring only the playback

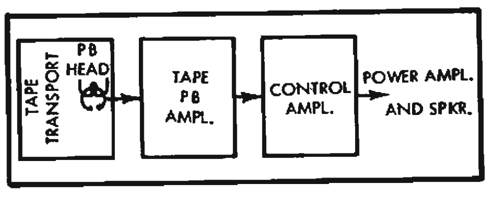

Fig. 8. Use of a separate tape playback amplifier in an audio system.

Fig. 9. Record and playback tape amplifier — Webster Electric.

function, but not owning a control amplifier with proper equalization for a tape head, can purchase a tape playback amplifier separately (*Figs.* 6 and 7) and use it with the transport of one's choice. Thus the signal is fed from the playback head to the tape playback amplifier, and from the latter to the control amplifier, as illustrated in *Fig.* 8. Again, if such a course is followed, one must guard against excessive cable length between the head and the playback amplifier, and one must route the cable to avoid hum pickup.

If one plans to record as well as play tapes, it is possible to purchase a complete tape amplifier separate from the transport, that is, a unit containing both playback and record electronics. (*Figs.* 9 and 10). Or one might build a complete tape amplifier on the basis of an article appearing now and then in the literature. In either case, one must proceed with greater care and caution than when purchasing tape electronics that are integrated with the tape transport. By an integrated unit we do not necessarily mean that the electronics is physically part of the transport; instead we refer to a tape amplifier *specifically designed* for use with a given transport, whether or not they are on one chassis.

An integrated transport and tape amplifier are apt to have the following advantages:

1. Length and routing of the cable will minimize high-frequency losses and hum pickup.
2. The oscillator circuit will be designed to supply the correct amount of bias current required by the particular record and erase heads used on the transport. Requirements for optimum performance differ by at least slight amounts and sometimes by major amounts among heads of different manufacture.
3. The record-level indicator will supply a proper reading. The correct amount of audio recording signal depends not upon the signal (electrical) delivered to the head but upon the signal (magnetic) delivered to the tape. For the same amount of signal fed to two different heads, different amounts of magnetic flux may be applied to the tape, resulting in different levels of recorded signal and therefore different amounts of recorded distortion.
4. Provision for switching from the play mode to the record mode will be at the transport rather than at the tape amplifier, accompanied by an interlock feature to prevent accidentally putting the machine in the record mode and erasing a valued tape. Usually the interlock consists of an auxiliary button or lever that one must actuate in order to bring the record electronics into use.

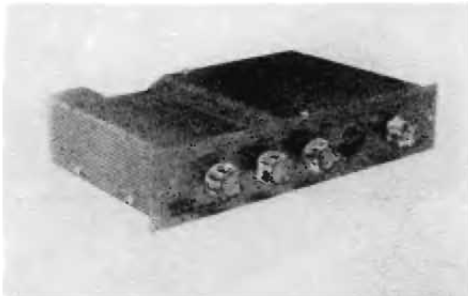

Fig. 10. Record and playback tape amplifier—Viking RP61.

5. Equalization will be more specific. Depending upon construction of the head, particularly gap width, high-frequency losses will differ somewhat from one make of head to another. Also, the frequency response at the very low end may vary among heads. In an integrated tape machine, the equalization will probably take into account the deviations of a particular make of head from ideal response, so that over-all frequency response is relatively flat.

The term tape recorder should properly be applied to a unit (for example, that in *Fig.* 11) comprising a transport accompanied by record and playback electronics, with the playback electronics limited to delivering a signal voltage capable of driving a control amplifier or power amplifier—about 1 volt on peaks. Many tape machines, however, particularly the so-called "home" units, which are mostly relatively low in price, also include a power amplifier and a speaker, as in *Fig.* 12, so that one does not have to rely on external facilities for playback. Such a machine may be referred to as a self-contained tape recorder. The advantages of a self-contained unit are obvious enough to require no more than a few words of comment. The machine can be taken anywhere—school, church, friends' homes, etc.—and one need not wait until returning to one's high fidelity system to hear the results of a recording session. One can check on the spot whether the recording is satisfactory by playing back the recording. If it is not satisfactory, circumstances frequently permit one to re-record.

Fig. 12. Complete tape recorder, including power amplifier and speaker.

However, there *may* be some disadvantages. To offer more functions at the same price or even lower price, there must be a sacrifice in quality somewhere —in the transport mechanism, in the electronics, or both. There is a substantial likelihood that the oscillator will be a single-ended rather than push-pull affair. Typically in a tape machine including a power amplifier, the output stage is a single tube such as a 6V6 or 6AQ5. In the record mode, this tube is switched to serve as an oscillator. However, a single-ended oscillator has more distortion than a push-pull one that uses a tube such as the 12AU7. Minimum waveform distortion of the bias frequency is important inasmuch as this distortion produces noise in recording. It should be added that even in tape machines incorporating a push-pull audio output stage, such as two 6AQ5's, it is general practice to use only one of these tubes as the oscillator when recording. Æ

Fig. 11. A tape recorder proper—Magnecord S-36B.

How Many Heads for the Tape Recorder?

HERMAN BURSTEIN*

The great variety of head arrangements available on different tape recorders makes it necessary to evaluate your needs before selecting a particular machine. Here is all the information you will need to find the answer.

IN PURCHASING A TAPE RECORDER, one has a choice between acquiring a machine with two heads or with three heads (usually in the more expensive units of semi-professional or professional caliber). One of these heads is for playback and the other for erasing the tape. In the case of a two-head machine, the playback head also serves for recording purposes through a switching arrangement, as shown in *Fig.* 1. In the case of a three-head unit, the third head serves for recording purposes and the playback head is restricted to the playback function.

It is also possible to acquire a machine with merely one head, for playback only. Ordinarily this would occur when purchasing a transport intended just for playing back prerecorded tapes (through the control amplifier or through a separate tape amplifier, as discussed in the preceding article.)

It is further possible to acquire a tape machine with as many as four or five heads. There are several possible reasons for such additional heads, as we will come to later. The principal part of this discussion, however, will deal with the advantages and disadvantages of two-head versus three-head machines.

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Two-Head Machines

An obvious advantage of a two-head machine is that it is cheaper to incorporate one head that serves two functions (record and playback) than separate heads for each function. This economy extends to the tape amplifier. Virtually all tape recorders with separate record and playback heads also have separate record and playback amplifiers (except for the power supply), as shown in *Fig.* 2, permitting one to play back the tape as it is being recorded. In the case of two-head machines, however, to a substantial extent the same electronics are employed in both the record and playback modes, with a switching arrangement to change from one mode to the other, as indicated in *Fig.* 3. Switching includes the change from record equalization to playback equalization and turning the oscillator on in the record mode and off in the playback mode. Thus there is a saving in electronics as well as in the cost of an extra head.

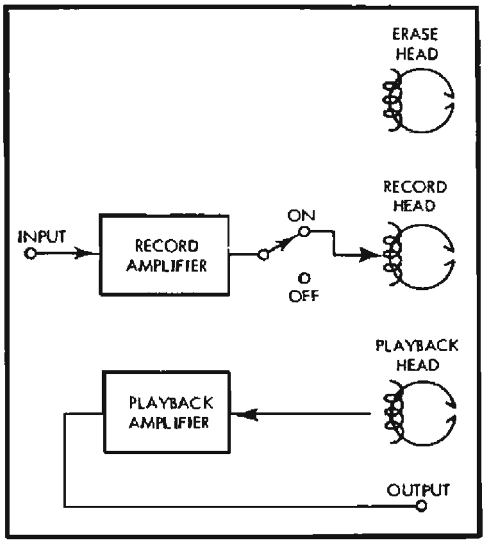

Fig. 2. Tape machine with separate record and playback heads.

A distinct advantage of using the same head for record as well as playback is that the problem of azimuth alignment becomes non-critical, provided that the only tapes to be played are those recorded with the same head. Azimuth refers to the angle formed by the gap of the head with respect to the long dimension of the tape, as shown in *Fig.* 4. Standard alignment requires that this angle be exactly (or as close as possible to) 90 deg. A minute azimuth error can result in very substantial high frequency losses. Thus if the head that recorded the tape has an azimuth angle of 90 deg. while the azimuth of the playback head deviates a few seconds of arc from 90 deg., there will be considerable treble losses. The slower the tape speed, the greater are the losses at any given frequency for a given difference between the azimuth angles of the record and playback heads. On the other hand, if one uses the same head for recording and playback, then a moderate error has no significance, because the error in

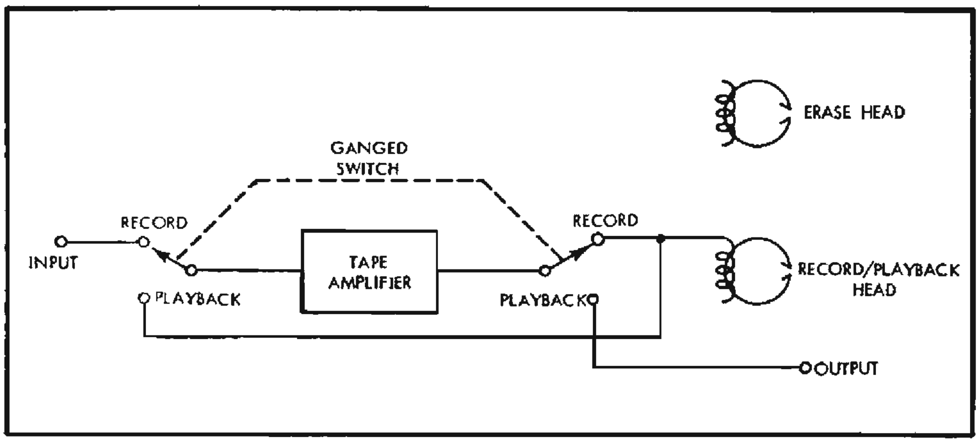

Fig. 1. Switching arrangement in a tape machine using the same head for record and playback.

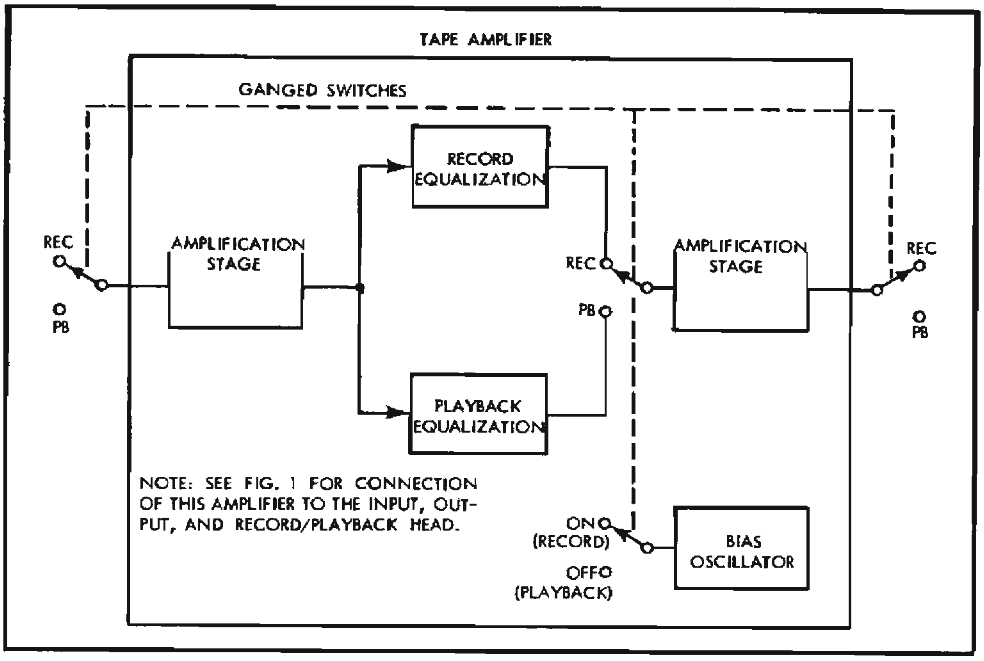

Fig. 3. Elements of a tape amplifier used with a single record-playback head.

recording is compensated by an equal error in playback.

But if it is one's intention to play recorded tapes—commercial tapes or those recorded on friends' machines—then correct azimuth alignment of 90 deg. becomes imperative, and there is no advantage in this respect over a three-head machine, except for the fact that there is less work in aligning one head than two.

When a single head is used for both record and playback, the playback requirements take over. That is, the requirements are more exacting for playback than for recording. A head suitable for playback can generally be used for recording, but not vice versa.

A prime requirement of the playback head is a narrow gap. The narrower the gap, the better is the high-frequency response, all other things remaining the same (same tape speed, head quality, equalization, and so on). The slower the tape speed, the proportionately finer must be the gap in order to maintain the same frequency response. Thus if a gap of .0002 in. is adequate at 7.5 ips to maintain response out to 15,000 cps, then a gap of .0001 in. is required at 3.75 ips.

A narrow gap is not a prime requisite of a record head. To the contrary, a relatively wide gap—in the range of .0005 in.—tends to be optimum. The narrower the gap in recording, the more difficult it is for the magnetic flux produced by the head to permeate the tape. The tape acts as a bridge from one edge of the gap to the other, so that the magnetic flux courses through the tape. But if the gap and therefore the bridge (the tape across the gap) are too narrow, not enough flux travels through the tape, so that the record head has to work all the harder to impress the desired amount of signal on the tape; the resulting increase in signal requirements raises the possibility of distortion. On the other hand, it is claimed that the increased efficiency of modern heads makes it possible to record satisfactorily with the narrowest of present-day gaps (.00009 in.).

Playback heads produce a very small signal voltage, on the order of a few millivolts at the most, and generally much less. Hence there is a very difficult problem of keeping noise and hum in

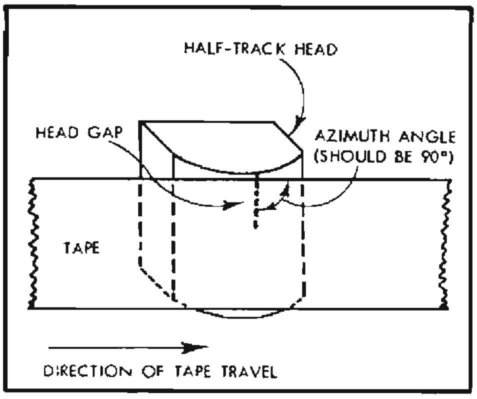

Fig. 4. Azimuth angle.

the tape amplifier sufficiently low to permit a satisfactory signal-to-noise ratio; the problem is substantially more difficult than in the similar case of the magnetic phono cartridge. To maximize the signal-to-noise ratio, a high-impedance playback head is desirable, that is, one having a large number of turns for high voltage output. (If a transistorized tape amplifier were used instead of one with vacuum tubes, then current rather than voltage would be the prime requirement; as yet, however, virtually all tape amplifiers employ vacuum tubes, which have a high input impedance, so that voltage rather than current is the significant electrical quantity.)

On the other hand, for recording purposes a head with relatively low impedance is desirable. A certain amount of audio voltage is required to drive audio current through the record head, thereby generating magnetic flux and impressing a signal on the tape. The higher the impedance of the head, the greater is the required driving voltage. But it is more difficult to develop a low-distortion signal at high voltage than at low voltage. If the recording has low impedance, the required driving voltage is reduced, and the difficulty is avoided.

By the use of separate heads for recording and playback, the contradictory requirements of each type of head with respect to gap width and impedance can be met. But if the same head is to be used for both purposes, then compromises are evidently in order. Thus record-playback heads typically have an inductance of about 0.5 henry, whereas a head designed expressly for playback may have an inductance of as much as 2.5 henries, and therefore greater output. On the other hand, a head designed specifically for recording may have an inductance of 50 millihenries or less.

At the same time, it should be recognized that the compromises entailed in record-playback heads are not so severe as to prevent quite good results from being obtained with them. But the perfectionist, desiring the best possible results in the present state of the art, can obtain even better results with separate heads designed for each function.

Three-Head Machines

Machines with separate record and playback heads generally are in the semi-professional and professional class, where maximum assurance is required that (1) the unit is in proper condition before the recording session commences, and (2) that the recording is proceeding satisfactorily during the session.

The three-head machine, with separate amplifiers for recording and playback, permits one to monitor the tape as it is being recorded, as illustrated in *Fig.* 5. That is, one can listen either to the incoming signal (which is about to be recorded) or one can switch to the playback signal (which has just been recorded). The time difference between

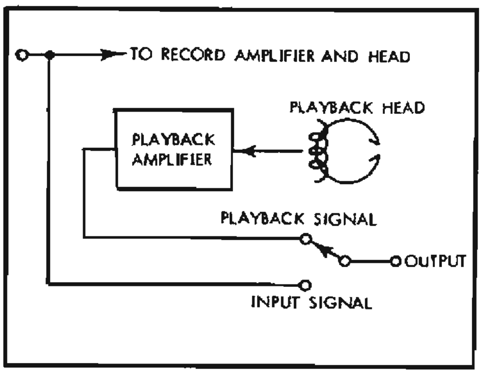

Fig. 5. Monitoring provision in a tape machine with separate record and playback heads.

the two signals is so small—roughly about 1/6 second at 7.5 ips—that alternate switching between them affords a very accurate indication whether the recorded signal is similar to the original signal. If there is a noticeable increase in distortion in the recorded signal, one can reduce the recording level. Conversely, one can increase the level if the monitoring arrangement indicates no perceptible increase in distortion. Similarly, if frequency response of the taped signal sounds reasonably close to that of the original signal, the recording may proceed; if not, then the frequency response of the tape recorder bears looking into (which is the subject of a later article). Ordinarily, one would not wait until a recording session to check frequency response by the monitoring method. Instead, one would make a test run, using a phonograph record or FM program as a signal source.

There are three basic tests required to ascertain whether the tape recorder is operating at its maximum potential for recording and reproduction of sound. These concern the frequency response of the machine, distortion, and signal-to-noise ratio. A high-quality tape recorder, into which category most three-head machines are apt to fall, contains adjustments for frequency response and permits one to adjust bias current, which in turn affects both frequency response and distortion. The higher the bias current (within the normal range of adjustment), the lower is the distortion, but also the worse is the treble response.

While one can check frequency response, distortion, and signal-to-noise ratio on a two-head machine by first recording a tape with the proper signals, rewinding the tape, and playing it back, this is a cumbersome, time-taking, and somewhat inaccurate process. Each time an adjustment is made, it is necessary to repeat the procedure to learn the result of the adjustment. With a three-head machine, where recording and playback are virtually simultaneous, one can check performance as swiftly as in the case of control amplifiers and power amplifiers. (The techniques of testing will be discussed in a later article.)

In the case of frequency response, there are two tests which should be made: (1) to ascertain whether playback equalization corresponds to the standard curve, which is NARTB at 7.5 ips; (2) to ascertain whether flat response is achieved when playing a tape recorded on the same machine. While the first test can be made as easily on a two-head machine as on a three-head one by using a standard test tape and checking for flat playback response, the second test is greatly facilitated by three heads.

A few of the better (and higher-priced) two-head machines can match the performance of the three-head ones, at least to the extent where the differences are inaudible, or negligible. The advantage of the three-head machine, then, lies not in actual performance but in greater *assurance* of proper performance. Such assurance is obtained through greater ease of testing and through greater ease of adjustment when testing reveals that something is wrong.

The importance of periodic testing, and adjustment if necessary, should not be underestimated if high fidelity standards are in force. If the operator of a tape recorder is interested merely in operation—whether mediocre, fair, or good—that is one thing. But if he considers mediocre or fair operation as equivalent to failure, then regular testing and adjustment become an essential part of the operating procedure. The professional and semi-professional recordists check their equipment before trouble develops, not only after it happens. Similarly, the audiofan desiring high fidelity performance from his tape recorder will try to head off trouble before it spoils a recording session, often irretrievably.

On the other hand, the importance of ease of testing and adjustment depends upon the amount of use that the tape recorder sees. A professional or semi-professional machine may be used many

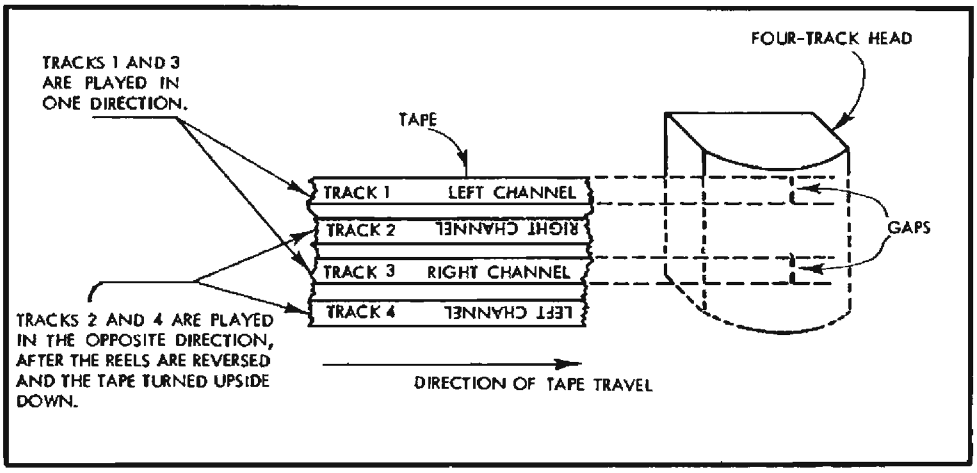

Fig. 6. Four-track stereo tape.

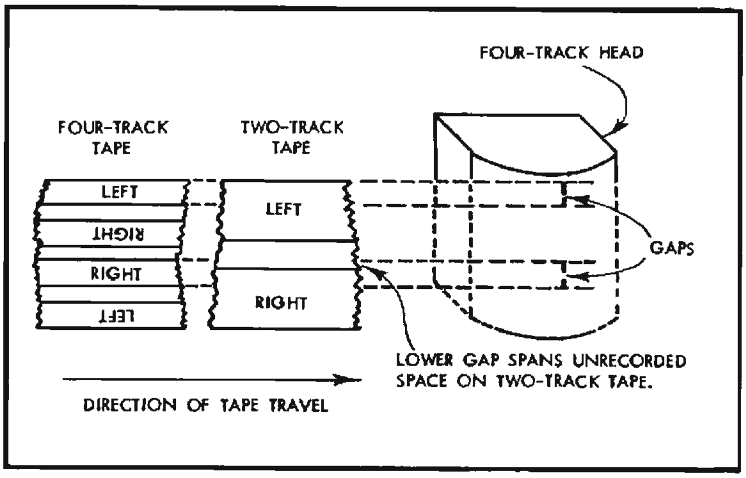

Fig. 7. Result of using a four-track head to play a two-track stereo tape.

hours a week, so that quick, easy checking and adjustment are imperative. But in the home, a machine may receive only a few hours of use each week or each month, so that facility of checking and adjustment becomes relatively less important. For one thing, the smaller amount of use means that trouble is less apt to occur. For another, when trouble does develop, there is apt to be less urgency about curing it.

A disadvantage of a three-head machine, compared with a two-head one, concerns azimuth alignment. Not only is it more time-consuming to align two heads instead of one, but more equipment is necessary. Where a single head is used for record and playback, an azimuth alignment tape is all that is needed. But where separate heads are employed, one needs in addition an audio oscillator or other source of single frequencies (such as a test record) in order to align the record head.

A *possible* disadvantage is that due to the oscillator. The frequency of the oscillator is usually between 30,000 and 100,000 cps, and the signal it generates has the characteristics of a radio wave. There is a problem of preventing this wave from being picked up and amplified by the playback amplifier, which is a very high gain affair. If picked up, the bias frequency signal can be amplified to the extent where, though not audible in itself, it may cause malfunction of the playback amplifier by driving it to excessive distortion, or perhaps by blocking the audio playback signal. While commercial equipment employs proper shielding, layout, and other measures so that the problem ordinarily does not arise, the amateur builder who ventures upon construction of a tape amplifier containing separate record and playback sections for a three-head machine may run into considerable difficulty on this score. The author speaks here from experience with one such tape amplifier that he sought to build; after his first try, he had to tear down the unit and redesign it with respect to layout and shielding.

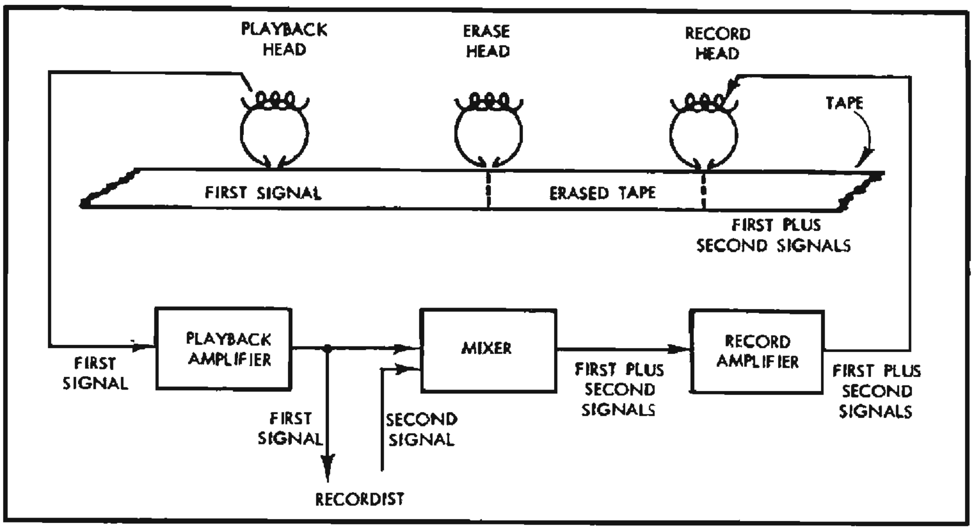

Fig. 6. Four-track stereo tape.

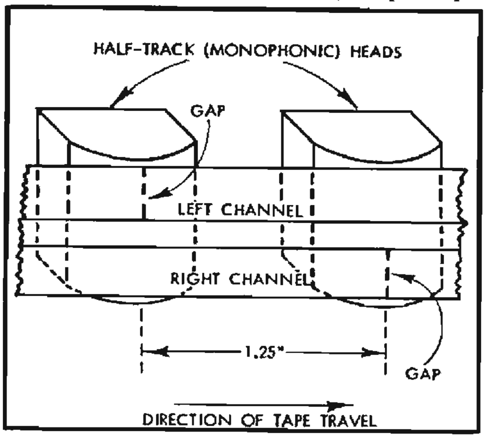

Fig. 8. Stereo tape using staggered heads.

Tape Recorders with Extra Heads

The introduction of four-track stereo tape has brought new problems. As shown in *Fig*. 6, two tracks are recorded in each direction, so that a stereo tape may be reversed just like a monophonic half-track tape. Obviously, each of the four tracks is considerably narrower than those on a two-track stereo tape, and the signal output of each section of the four-track playback head is proportionately reduced, resulting in a lower signal-to-noise ratio. If one plays a two-track tape with a four-track head, there is an unnecessary sacrifice in signal-to-noise ratio, because part of the recorded signal is not picked up. Therefore the individual may wish to add an extra two-track head so that he can obtain better results when playing two-track stereo tapes (or half-track monophonic ones).

If a four-track head is used to play a two-track stereo tape, one section of the head will partially span an unrecorded space on the tape, as shown in *Fig*. 7. Therefore this section of the head will produce less signal output than the other section, resulting not only in deterioration of signal-to-noise ratio, but also in lack of balance between the two signals. This problem can be met by a mechanical device that moves the four-track head up or down—up for playing four-track stereo tapes and down for playing two-track stereo tapes. However, there is some danger of impairing azimuth alignment as the head is moved up and down. Thus the audiofan may prefer to install an extra two-track head, so that it and the four-track head can each remain in fixed position.

Four-track stereo tape is associated in the main with the 3.75 ips speed. As previously pointed out, this speed requires the playback to have an extremely fine gap if frequency response is to be maintained out to 15,000 cps or thereabouts. But, as also pointed out, the gap *might* be too fine to be suitable for recording. Consequently, it may be desirable to have an extra head for recording purposes.

Initially, the majority of stereo tape machines—at least those for the home—employed staggered heads for playback, spaced about 1.25 in. apart, and positioned so that one operated on the

(*Continued on page 45*)

Comparison between stereo record-playback heads. Left, two-track (Shure TR-40); right, four-track (Shure TR-48).

upper half of the tape and the other on the lower half, as illustrated in *Fig.* 8. A substantial number of recorded tapes were sold for the staggered-head arrangement. The individual who invested in staggered tapes may be loath to write off this investment, preferring instead to add another playback head in order to be able to reproduce his staggered tapes correctly. Thus he would have a stacked head for playing conventional stereo tapes, and he would use one section of this head together with the extra playback head to reproduce his staggered tapes. If he were to

Half-track monophonic head (Shure TR-35).

play staggered tapes with a stacked head, there would be a relatively enormous lack of synchronization between the two channels. The discrepancy would be 1/6 second at 7.5 ips, whereas a number of experts consider that the two channels must maintain synchronization within .0001 second for proper stereo effect.

The recordist wishing to achieve special effects may have to add one or more heads to those already on his tape machine. Two of the best known of these special effects are sound-on-sound and echo effect.

In the case of sound on sound, it is necessary to place the playback head before the erase head, as shown in *Fig.* 9. The sequence is as follows. The first recording is made. The tape is rewound. The recorded tape is reproduced by the playback head, and this signal is monitored by the performer by means of earphones. At the same time the performer makes a second recording. The first signal (from the playback head) and the second recording signal are combined in a mixer and fed to the record head. Meanwhile, the tape has been erased by the erase head. The clean tape that reaches the record head receives the combination of the first and second signals. This process can be repeated as many times as desired; however, noise on the tape will increase each time, so there is a limit to the procedure. While it is not strictly necessary to add an extra playback head for sound-on-sound recording, it is quite inconvenient to have to transfer the playback head from its normal position (following the erase and record heads) to a position before the erase head each time that one wishes to create this special effect. Accordingly, the individual who plans to do an appreciable amount of sound-on-sound recording customarily will add an extra playback head prior to the erase head.

Figure 10 illustrates how the tape recorder may be used to achieve an echo effect. Part of the signal picked up by the playback head is fed back to the record head, and is therefore repeated in the manner of an echo. One can obtain superior results, more akin to the natural echo, by adding additional playback heads, each one feeding part of the signal back to the record head. The amount of signal fed back is decreased at each successful head. Æ

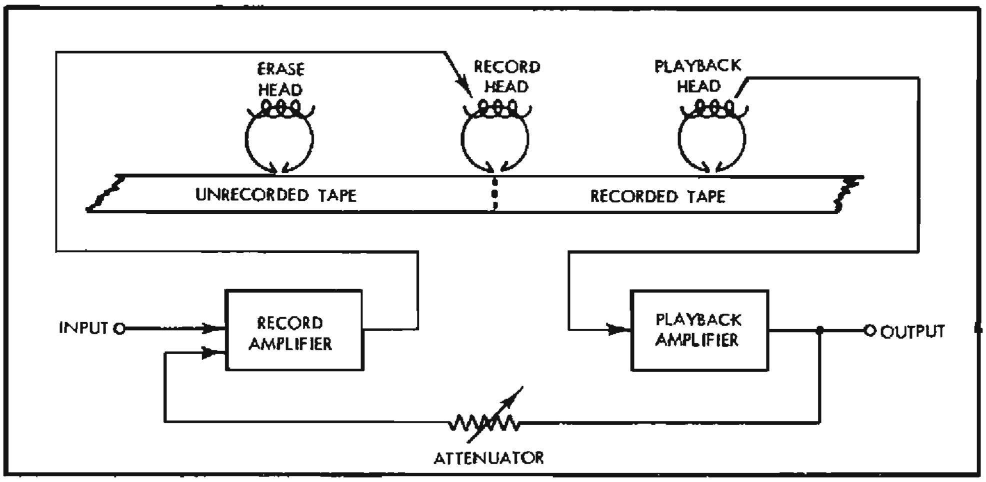

Fig. 10. Method of producing an echo effect in recording.

What Kind of Record-Level Indicator?

HERMAN BURSTEIN*

While tape recorders are inherently capable of a high signal-to-noise ratio and low distortion, it is imperative that the recording level be maintained within close limits in order to achieve best results. Therefore some means of measuring level must be used—the question is, "What kind"?

ONE OF THE FACTORS that will govern the thoughtful audiofan's choice of tape recorder, and perhaps help him decide whether to spend somewhat more than he had originally planned, is the record-level indicator incorporated in the unit. The type of indicator, its calibration (correct indication of maximum allowable recording level in terms of distortion), and its skillful use are all-important factors in the attainment of high-quality recordings. Although a tape machine may be potentially capable of making excellent recordings, it does not necessarily follow that the operator will obtain such results. Of all the components in a high fidelity system, the tape recorder requires the most skill on the user's part, and it is here that the record-level indicator plays a vital role.

The tape recordist must thread his way between two major pitfalls of high fidelity: excessive distortion on the one hand; a poor signal-to-noise ratio on the other hand. If he records at a very high level, he can substantially elevate the signal-to-noise ratio. As much as 10 db improvement can be obtained compared with normal recording levels compatible with distortion low enough to deserve the description high fidelity. Considering that it is quite an achievement to attain a signal to noise ratio of 55 db, and fairly creditable to attain one of 50 db, there is a very strong temptation to over-record.

Unfortunately, it is characteristic of tape recording that beyond the point where intermodulation distortion has reached about 3 to 5 per cent, such distortion rises very sharply, as shown in *Fig.* 1. Thus a try for a few more db

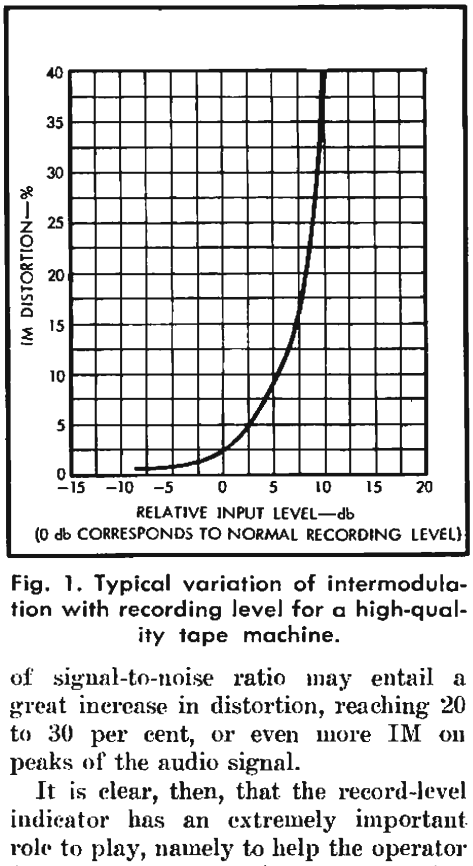

Fig. 1. Typical variation of intermodulation with recording level for a high-quality tape machine.

of signal-to-noise ratio may entail a great increase in distortion, reaching 20 to 30 per cent, or even more IM on peaks of the audio signal.

It is clear, then, that the record-level indicator has an extremely important role to play, namely to help the operator find an optimum point between high signal-to-noise ratio and low distortion. The more desirous one is of high fidelity results, the more important becomes this role. At the other extreme, if the tape recorder is employed in a manner where high fidelity standards are not applicable—for example, as a dictating machine—the importance of the record-level indicator become secondary.

Where high fidelity standards are applicable, the operator will want a record-level indicator which is easy to read and which shows *accurately* whether the recording level is too high, too low, or about right. Accordingly, it is the purpose of this article to discuss the types of level indicators commonly employed in tape machines and to indicate their relative merits. Through an understanding of how each type of indicator operates and its relative advantages and disadvantages, the audiofan can arrive at a decision as to which type is best, or at least suitable, for his purposes.

Electronic Indicators

The so-called "home machine," which tends to be relatively on the inexpensive side and to have a single record-playback head instead of separate heads for both functions, generally employes an electronic indicator, which is basically one of two kinds: the magic eye (electron ray) tube or the neon lamp. Contrasted with the meter type of indicator, commonly found in semi-professional and professional machines, the electronic indicators are much less expensive, have no moving parts, and respond accurately to signal peaks. Apart from these traits in common, the two kinds of electronic indicator are substantially different in how they operate and the information they furnish.

Before going on to discuss each type of electronic indicator, it is important to dwell on the basic advantage that they both possess over the meter indicator in reading true peak level of the audio signal. The meter, because of its mechanical nature, cannot follow very rapid signal changes. The electronic indicator, on the other hand, responds immediately to the sharp transients found in music and speech, thereby indicating the actual level of the material being recorded. In the case of the meter, an

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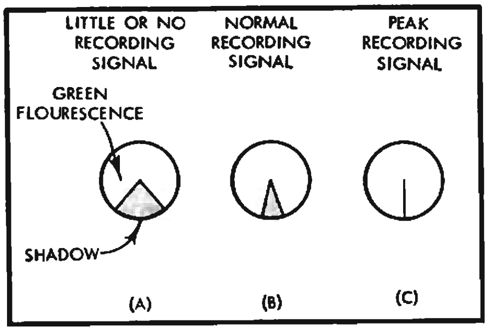

Fig. 2. Magic eye indicator.

allowance has to be made for the difference between the meter indication and the actual (higher) level of the transients.

The Magic Eye Tube

While there are several versions of the magic eye tube, the most common type employed in tape recorders is that represented in *Fig.* 2. With no signal applied to the tube (grid), a green fluorescent glow suffuses about three quarters of the face of the tube, as at (A). The remaining quarter is in shadow; in other words, the eye is open. As audio signal is applied to the tube, the shadow narrows, as indicated at (B). If sufficient signal is applied, the eye closes completely, as at (C), or even overlaps.

The problem—of the manufacturer or service technician—is to correlate the amount of signal fed to the magic eye tube with the amount of signal impressed on the tape so that when the eye barely closes this corresponds to maximum permissible recording level. On home recorders, the maximum level is usually between 3 and 5 per cent harmonic distortion, corresponding roughly to 30 per cent or more of intermodulation distortion. This is a tremendous amount of IM distortion, but occurs only—or is *supposed* to occur only—on peaks. At normal levels, which are typically 10 to 20 db below peaks on audio material, intermodulation distortion will drop to levels consistent with fidelity standards.

Figure 3 shows how the magic eye indication is tied in with the signal going to the record head and thence onto the tape. At a suitable point in the record amplifier, the audio signal is tapped off and fed to the magic eye tube. This signal goes through a voltage divider, which supplies the proper proportion of the signal required to drive the indicator. This proportion is experimentally determined by the manufacturer of the tape machine. It is more the exception than the rule to find a variable voltage divider in home machines so that one can adjust the amount of signal fed to the record-level indicator. However, a few home machines do contain a control—usually accessible internally—which permits the service technician or any other person equipped with the necessary instruments and knowledge to adjust the signal going to the magic eye tube so that its indication will correspond with maximum permissible distortion on the tape.

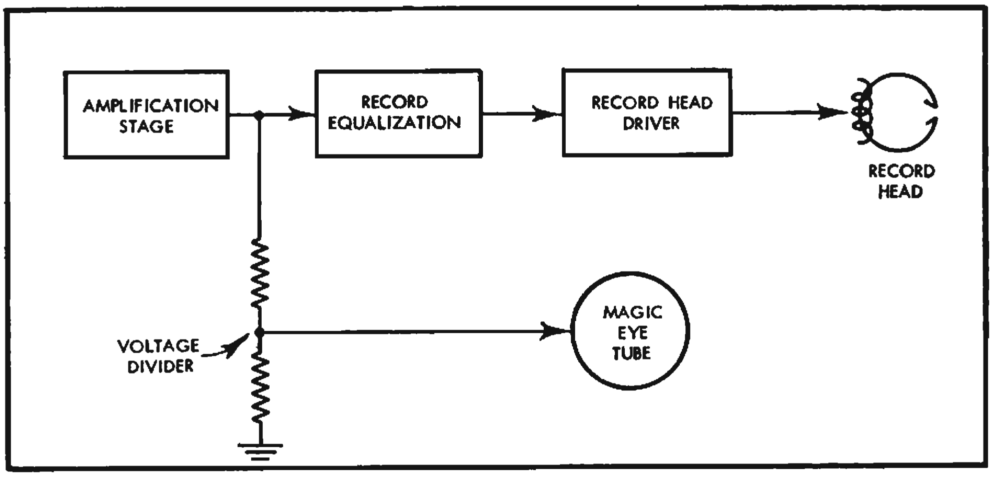

Fig. 3. Feeding the audio signal to the magic eye indicator.

Assuming that the magic eye tube is properly calibrated, optimum results are obtained if the operator allows the eye to close just barely on audio peaks. If the eye overlaps once in a great while, there is probably no significant harm. But if the eye completely closes or overlaps many times a minute, then the recorded signal is apt to take on the roughness and mushiness symptomatic of excessive distortion. On the other hand, if the eye seldom or never quite closes, the chances are that the recording level is too low, which means an unnecessary sacrifice in signal-to-noise ratio; and there is none of this to spare in any machine, much less one of the home type.

To some extent, the permissible recording level will vary with the nature of the program material. Ordinarily, more distortion is tolerable on speech than on music. And on certain kinds of music a given amount of distortion is less offensive than on others. This is where experience and skill in recording enter into the picture.

As mentioned before, a prime advantage of the electronic indicator is that it responds instantaneously to transients (usually responsible for the peak audio levels), so that one obtains a correct indication of how much signal is going onto the tape. On the other hand this immediate response is not only an advantage but can also be a source of difficulty. When the transients are strong and frequent, the magic eye will fluctuate so rapidly that the operator finds it difficult to discern its meaning and to set recording level properly. The extent of this difficulty, depends of course upon the nature of the program material being recorded. A fiery composition is apt to offer much more of a problem than quiet, relaxed music.

To minimize the problem, a number of tape recorders, as indicated in *Fig.* 4, incorporate a "floating action" circuit which maintains the eye for a brief period at the maximum degree of closure. This circuit may be described as a "one way street." It permits the eye to respond (to close) very quickly when a transient comes along. But it does not permit the eye to open with the same degree of rapidity. Thus the high reading is maintained for a short while. A typical floating action circuit may allow the eye to close in about one-thousandth of a second, but may not allow it to open for about one-twentieth of a second. Although one-twentieth of a second may seem extremely brief, yet it is long enough to increase substantially the facility with which the magic eye can be read. If the persistence time were increased greatly, it would become difficult to judge the frequency of transients and make a corresponding adjustment of recording level, bearing in mind that this adjustment depends not only upon how great the signal peaks are but also how

(*Continued on page 35*)

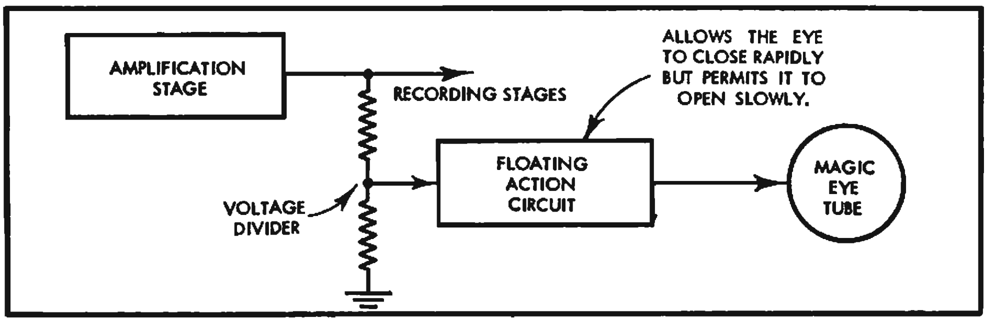

Fig. 4. Use of a floating action circuit to facilitate reading the magic eye indicator.

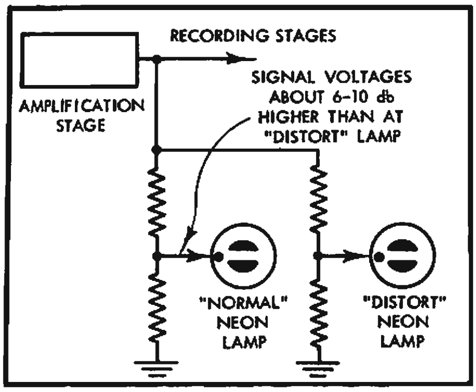

Fig 5. Use of two neon lamps as record-level indicators.

(*Continued from page 32*)

often they occur. In other words, increased persistence time would cause the transients, as they appear on the magic eye tube, to become a blur.

The Neon Lamp

Least expensive of all record-level indicators is the neon lamp. This is an on-off device, with no intermediate indication. It produces an indication only when the recording level is at or above a certain point. If the recording level is too low (with signal-to-noise ratio unnecessarily reduced), the lamp does not indicate how much too low. Even the magic eye tube, with its varying shadow, provides some indication of the extent to which the recording level is below normal. If the recording level is too high, the neon lamp fails to indicate by how much. The magic eye tube provides a little information in this respect, as indicated by eye overlap. The only indication of over-recording in the case of the neon lamp is how frequently it ignites.

The greatest flaw is the failure to provide an indication when recording level is too low. However, this is corrected in some machines by using two neon lamps, as in *Fig.* 5. One, called the "distort" lamp (or similar term), ignites at the maximum permissible recording level. The other, called the "normal" lamp (or similar term), ignites at a lower level—usually about 6 to 10 db below the acceptable distortion point. The objective for the operator is to try to adjust the recording level so that the normal lamp is ignited most of the time but the distort lamp is ignited as seldom as possible. Again, the nature of the program material must affect the operator's decision concerning recording level.

The neon lamp has inherent floating action. That is, the signal required to fire the lamp is appreciably greater than the voltage at which the lamp goes out. Accordingly, the glow produced by a transient lasts longer than the transient.

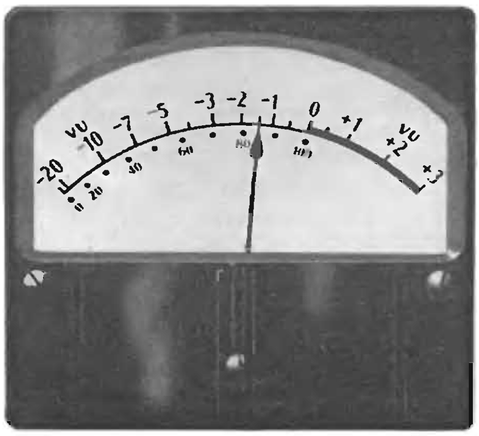

Fig. 6. The VU meter, Type A scale.

The VU Meter

The VU meter—or a similar type of meter—is most commonly found in semi-professional and professional tape recorders, although on occasion it also appears in the so-called home machine. As a matter of fact, there has been an increasing trend toward use of meters in home-grade machines, and some familiarity with the operation and characteristics of meters as record-level indicators is therefore all the more likely to be useful to the home recordist.

Characteristics of the VU Meter

VU means volume units. These units are simply decibels. The zero point on the scale (see *Fig.* 6) is an arbitrary reference level; when the VU meter is connected in the standard manner—with a 3600-ohm series resistor across a 600-ohm line (See *Fig.* 7)—a 0 reading denotes 1.23 volts of signal at the source (or 2.5 milliwatts of power in the line). All other readings on the VU scale are simply in terms of decibels above or below the 0 reference level. For recording purposes, the reference level is significant only in *relative* terms, denoting that maximum permissible recording level has been reached, after which point tape distortion becomes excessive. The *absolute* meaning of the reference level—1.23 volts—is of no consequences for recording purposes.

As may be seen in *Fig.* 6, the VU meter also contains a 0 to 100 (per cent of maximum permissible voltage) scale, which some users may find more convenient for recording purposes. In fact, in some VU meters the positions of the two scales are reversed, as shown in *Fig.* 8.[1]

The VU meter has been designed not only to have a certain sensitivity but, much more important, to have certain characteristics that facilitate the operator's understanding of the nature of the audio signal. For one thing, the standard VU meter must have a frequency response within ±0.2 db between

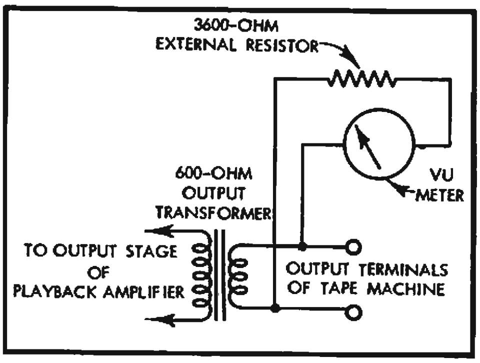

Fig. 7. Standard method of connecting a VU meter.

35 and 16,000 cps. It must respond quickly to audio signals; the standard requirement is that when a sine wave of 2.5 milliwatts power is suddenly introduced in the line, the pointer should reach 99 on the percentage scale within 0.3 seconds. On the other hand, sudden application of power should not cause the meter to overshoot and give a false indication; here the requirement is that sudden application of the same signal should cause overshot of no more than 1.5 per cent. The meter, furthermore, must be a hardy device. It should be able to withstand continuously five times the voltage that produces a 0 VU indication, and it should be able to withstand for one-half second a ten-fold voltage overload.

Advantages of the VU Meter

Compared with electronic record-level indicators, the VU meter has the following advantages for tape recording purposes.

1. It provides a quantitative indication of the extent to which the recording level is above or below that corresponding to maximum permissible distortion. Thus the recordist can vary the recording level by a desired amount. To illustrate, he may know from experience that he should record a certain type of program material about 4 db below the usual level in order to keep distortion satisfactorily low. With the aid of a meter indicator, he can achieve a db

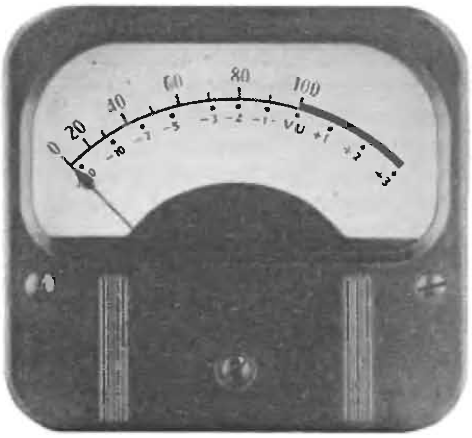

Fig. 8. Type B VU meter scale, which features units showing percentage of maximum permissible voltage.

[1] Both are "Standard;" Type A is usually employed where levels are being read in db. Most recording and radio studios choose Type B, which then shows "percentage utilization" of the channel. Ed.

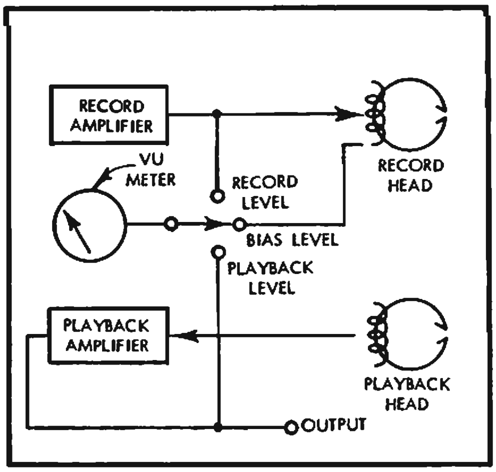

Fig. 9. Use of the VU meter to measure bias current and playback level, in addition to normal use in measuring recording level.

reduction quite closely. With an electronic indicator, however, he could not be sure whether the reduction is of the correct amount.

2. The VU meter, if made by a reputable manufacturer, is a standard and relatively uniform product, so that one meter provides essentially the same indications as another. If the meter must be replaced (much more likely due to accident than normal usage), the new one will provide very nearly the same indications as its predecessor. This is not nearly as true of electronic indicators, where the tolerances are such that significantly different readings may be obtained between two magic eye tubes or two neon lamps for the same signal. Thus one neon lamp may fire at a voltage 3 db higher or lower than another neon lamp of the same kind.

3. The characteristics of the VU meter remain stable with use and the passage of time.

4. If one insists upon top quality recording in terms of low distortion and wide frequency response, accompanied by a high signal-to-noise ratio, and if at the same time one wishes to record at speeds below 15 ips, the value of bias current supplied to the record head is quite critical. It is very important then to adjust bias current to the correct value as indicated by the tape recorder manufacturer or as determined by the recordist equipped with the instruments for checking frequency response and distortion. Assuming that the correct bias current is known, it is highly desirable to be able to check quickly and easily whether the actual value corresponds to the desired value. In many high-quality machines containing a meter, a switching arrangement is incorporated that permits one to use the meter to measure bias, as illustrated in *Fig.* 9. The machine will also have a control (usually on the rear panel or internal) that permits bias current to be adjusted if its value proves to be incorrect. (However, enough warmup time—15 minutes or more—should be allowed for bias current to stabilize.)

The electron-ray tube and the neon lamp are not sufficiently accurate in their characteristics to enable them to be used for measuring bias current with the necessary precision. If bias current is too great, high-frequency response will suffer, although distortion usually will decrease at the same time. If bias current is too low, there will be considerably better high-frequency response, but at the expense of more distortion. Thus the correct value of bias represents a fairly critical compromise point.

5. In some situations, as in a recording or broadcast studio or in other instances where professional equipment is employed, it is necessary to know the level of the tape playback signal to insure that the following equipment is being neither overloaded nor supplied too little signal for proper operation. Therefore in semi-professional and professional tape machines it is the practice to have a switching arrangement that enables the meter to measure the playback level, also shown in *Fig.* 9.

On the other side of the coin, there are also disadvantages to the use of a meter as a record-level indicator. One is that the meter is relatively expensive compared with electronic indicators. Another is that the meter requires special circuitry to drive it properly and to isolate it from the recording signal (a low impedance source is required), which further raises the cost. A third, as already mentioned is that the meter does not follow transients but lags behind them; this important problem is discussed at greater length in the next section.

Calibration of the VU Meter

Due to mechanical inertia, the meter pointer cannot follow very sudden and strong impulses. Thus one may obtain a meter indication as much as 10 db, and on some occasions as much as 20 db, below the true signal level, as illustrated in *Fig.* 10. Therefore in calibrating the VU meter so that its indication corresponds to maximum permissible recording level, a different procedure may be in order than for electronic indicators. In the case of the meter, it is desirable to make an allowance for the difference between the pointer indication and the actual level.

Accordingly, a number of manufacturers of tape recorders adjust the calibration so that the meter will read 0 VU when a sine wave (usually 400 cycles) is being recorded at a level substantially less than that which causes maximum permissible distortion on the tape. The safety margin is usually between 6 db and 10 db, depending upon the tape recorder in question. In other words, the meter is "set ahead," so that on a steady signal it indicates distortion "too soon." But on program material, where the meter fails to keep up with transients, the amount by which the meter is set ahead more or less compensates for the amount that it lags behind.

It is not absolutely necessary that the meter be set ahead in the manner described. In fact, some tape recorders fail to do so at all. It then becomes necessary for the recordist to make full allowance for the lagging nature of the meter. This means that he should not allow the pointer to exceed approximately the –10 VU mark. However, this crowds the operating range of the meter into a relatively small part of the scale.

It should be clear from the above discussion that if one acquires a tape recorder with a meter type of indicator, it is important to the user to find out if the meter is set ahead and by how much. To illustrate the point, a recent review of a tape machine employing a meter emphasized that clean recordings could be obtained with this unit only if the pointer were kept below the –10 VU mark.

Whether or not the meter is set ahead, the adjustment of recording level is far from a mechanical operation. Instead, experience and judgment must be brought to bear. The relationship between average levels, as indicated by the meter, and peak levels, which the meter cannot follow, will change in accordance with the material being recorded. As previously indicated, peaks may be as much as 20 db higher than the program average. Or they may be only 6 db higher. Thus in some cases the recordist may allow the VU meter to hit 0 VU or even higher without incurring audible distortion. For other program material, he may find it necessary to hold the pointer well below 0 VU in order to keep distortion inaudible.

Other Meters

Not all meters found in tape recorders are VU meters. In some instances, particularly machines of the home variety, the manufacturer has incorporated an inexpensive meter that simulates professional appearance without professional performance. Its main virtue is that it wriggles. The writer has

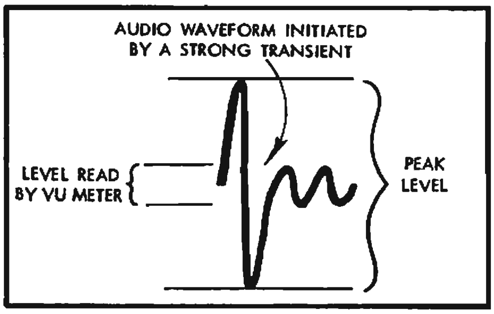

Fig. 10. Difference between actual signal level and VU meter indication.

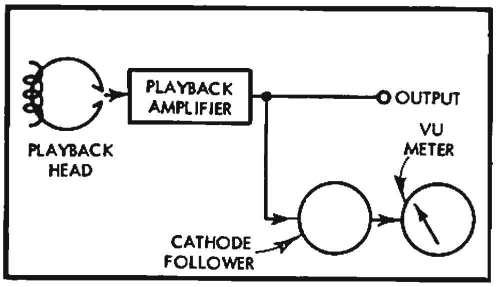

Fig. 11. Isolating the VU meter from the playback circuit to minimize distortion.

come across instances of this kind where not only was the meaning of the meter reading open to serious question, but the meter was connected to the circuit in a manner seriously deleterious to operation of the record amplifier. The user would have been far better served by a magic eye tube or neon lamp indicator.

On the other hand, a meter does not have to be a VU meter in order to render equivalent service. In some tape recorders, a meter of moderate sensitivity is driven by a vacuum tube amplifier in order to achieve the same sensitivity as a true VU meter. And the meter movement, obtained on special order, is designed to have the same characteristics as a VU meter with respect to frequency response, speed of response, overshoot, and so on.

Loading Distortion

A meter is a non-linear device. That is, depending upon the voltage of the audio signal at a given instant and whether the voltage is positive or negative at this instant, the effective resistance of the meter changes. When placed across the audio signal, the VU meter presents a changing load, which results in distortion. That is why it is necessary to have a 3600-ohm resistor in series with the VU meter, as was shown in *Fig.* 7; this resistor helps keep the loading distortion suitably low. One of the standard requirements of a VU meter is that when connected in the manner of *Fig.* 5, it shall cause not in excess of 0.2 per cent harmonic distortion.

In some tape recorders, as an extra precaution, the VU meter is isolated from the audio signal by an extra tube stage, usually a cathode follower, as shown in *Fig.* 11.

Recording on the Basis of Playback Level

In some tape recorders, either through the operator's choice or through the design of the machine, the recording level is determined on the basis of the signal coming off the tape rather than on the basis of the signal going to the record head. As illustrated in *Fig.* 12, the VU meter is connected to the playback amplifier; a suitable proportion of the playback signal, obtained through a voltage divider, is fed to the meter so that the latter gives the proper reading. At the same time, the gain control of the playback amplifier is placed in a predetermined position so that the meter reading may correctly indicate recording level.

Through the above technique, one is judging recording level on the basis of the signal that actually gets onto the tape. When using different brands of tape or different lots of the same brand, there may be differences of a few db in tape efficiency; that is, for the same signal *presented* to the tape there may be different amounts of signal *recorded* on the tape. But the amount of distortion tends to vary with the signal recorded on the tape. Therefore it may be more desirable to set recording level in terms of the amount of signal on the tape rather than in terms of the signal presented to the tape. Furthermore, in a recording or broadcast studio there may be operational advantages in leaving the meter always connected to the playback amplifier (except when checking bias current).

On the other hand, there is at least one disadvantage to the above procedure. The level of the program material cannot be checked unless the tape is in motion and being recorded; otherwise there is no playback signal. Many recordists, however, will wish to evaluate first the level of the program material, adjust the recording gain control accordingly, and then put the tape into motion. Æ

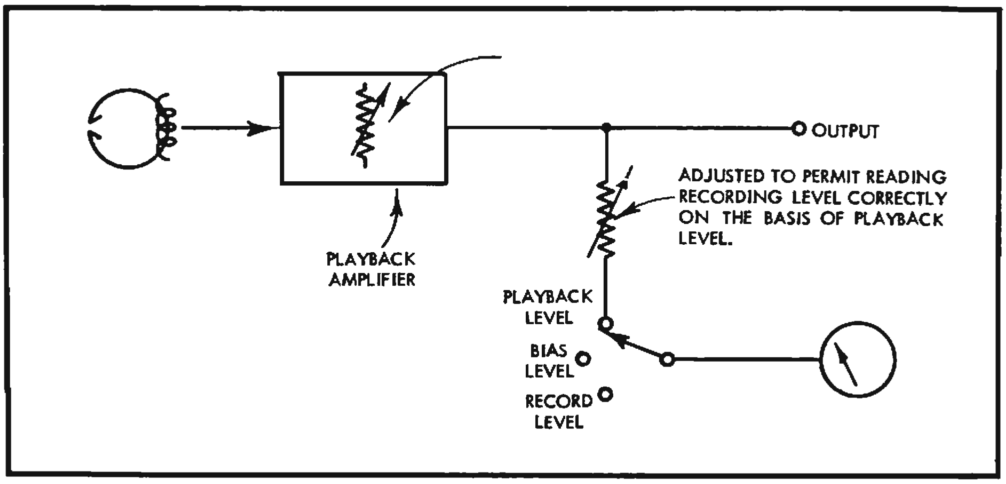

Fig. 12. Reading recording level on the basis of playback level.

What to Look for in a Tape Recorder

HERMAN BURSTEIN*

THE PRECEDING THREE ARTICLES have dealt with some basic matters that the audiofan will wish to consider in purchasing a tape machine: whether to buy a transport only or a transport and electronics; how many heads the machine should have; what kind of record level indicator is suitable for his purposes. The present article deals with additional factors, of varying degrees of importance, that play a part in a purchase decision. Some of these concern electrical performance, some mechanical performance, and others operating convenience.

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Fig. 1. A tape machine with provision for stereo playback and monophonic recording. (Norelco)

Fig. 2. A stereo tape machine utilizing a plug-in attachment for recording second channel. (Tandberg) Unit at the left is the auxiliary recording amplifier for the second channel; without it, the machine plays stereophonically and records monophonically.

Probably no one machine contains all the features that may be desired by all audiofans. On the other hand, needs vary from one tape recordist to another. Through a preliminary familiarity with the features available in one machine or another, the audiofan is in a position to choose that tape recorder which is most likely to satisfy both his wants and his budget.

Stereo Versus Mono

In view of the pace of stereo, the individual who purchases a tape machine for serious music listening is well advised to consider one equipped for stereo at least in the playback mode. Of course it is possible to modify a monophonic machine by replacing the mono playback head with a stereo head, but this calls for an additional playback amplifier, raising two problems: (1) that of closely matched equalization and amplification facilities in the playback amplifier for each channel; (2) that of a cable run to the additional amplifier, with high frequency losses taking place if the cable is too long, and with the possibility of hum pickup if the cable is improperly routed.

A number of tape machines now provide for stereo playback but only mono record, as illustrated in *Fig.* 1. If the audiofan has any thoughts of eventually wanting to record stereophonically, he should inquire whether such a machine has facilities for *properly* adding a record amplifier. The second channel record amplifier should have the same equalization and gain as for the other channel, and—very important—there should be means for synchronizing the bias oscillators in the two amplifiers, if separate oscillators are used, so that they will operate at the same frequency. Bias current in each channel will to some degree leak through to the other channel. If the two currents are of different frequency, they will beat together, and the resulting beat frequencies will appear on the tape, causing birdies and other objectionable sounds. *Figure* 2 shows a tape recorder designed to permit addition of a second record amplifier; in this case the oscillator of the first channel also serves the second channel.

Tape Speeds

For home purposes, in the past few years the virtually standard speed compatible with high fidelity has been 7.5 ips, which permits frequency response to about 15,000 cps and at the same time allows satisfactorily low distortion and a satisfactorily high signal-to-noise ratio. The 3.75-ips speed has also been widely used, although not considered compatible with high fidelity. The principle difficulty at the lower speed lay in high-frequency response. All other things remaining equal, the frequency response of a tape machine varies directly with tape speed. Thus a machine capable of maintaining flat response to, say, 12,000 cps at 7.5 ips (response may be 3 to 6 db down at 15,000 cps) will be able to maintain flat response only to 6,000 cps at 3.75 ips.

The problem at 3.75 ips occurs in large part in playback, being due to the fact that treble response varies inversely with width of the playback head gap. The recent introduction of heads with extremely narrow gaps—as fine as .00009" —has made it possible to extend frequency response to about 15,000 cps at 3.75 ips *so far as playback is concerned.* But there are also very serious *recording* losses at high frequencies due to bias current and to the phenomenon known as self-demagnetization (recorded frequencies on the tape are equivalent to small bar magnets; the higher the frequency, the smaller is the equivalent magnet and the greater is the tendency of the opposite poles of each magnet to cancel each other).

By using somewhat less bias current than at 7.5 ips (which reduces treble losses), by recording at somewhat lower levels (which compensates for the greater distortion because of reduced bias), and by using somewhat more treble boost in recording, it has been found possible to put on the tape at 3.75 ips a signal with frequency response corresponding at least to minimum high fidelity standards and having acceptably low distortion and acceptably high signal to noise ratio. This does not mean that 3.75-ips tapes are yet as good as 7.5-ips ones of recent vintage. However, they are already as good as the 7.5 ips tapes of several years ago, and it can be expected that technological progress will bring further improvement.

Accordingly, the serious audiofan may wish to include the 3.75-ips speed and to make sure that the machine he purchases does all that is possible in the present state of the art to achieve maximum performance at this speed. Specifically, he will want a machine with a playback head that has a gap of .0001" or less, and having the recording equalization and bias current that allow frequency response to extend to 10,000 cps and beyond. On the other hand, since equalization and bias current requirements will be different at the 7.5-ips speed, he will want to make sure that performance at this higher speed is not compromised by failure of the machine to change the bias and equalization when the tape speed is switched to 7.5 ips.

The audiofan will probably find that that 3.75-ips speed is quite suitable for various types of program material not of the highest fidelity, such as old records one wishes to copy (in fact, the loss of the higher frequencies can be a distinct blessing in this situation since these frequencies will consist more of noise than music), AM station programs, etc. Or there may be situations where one is willing to exchange some sacrifice in quality for the privilege of doubling the recording time on a reel tape. Thus in taping an opera or other lengthy work, one can get from two to four hours of time on a 7-inch reel, using half-track monophonic recording or quarter-track stereo recording. Regular tape will yield two hours, long-playing tape three hours, and extra-long-playing tape four hours.

For non-high-fidelity applications, such as recording speech, dance music for parties, and so forth, the 1.875-ips speed is coming into increasing use. In fact, this speed now enjoys about the same status as 3.75 ips formerly occupied. A number of tape machines now offer this speed along with a surprisingly satisfactory quality for non-critical uses where length of playing time is important. Thus on a 7-inch reel of tape one can record from four to eight hours of material at 1.875-ips, depending upon whether one is using regular, long-playing, or extra-long-playing tape.

Frequency Response

At 7.5 ips, a modern tape recorder should be able to cover the range of 40 to 15,000 cps, being no more than 3 or 4 db down at either extreme and achieving quite flat response—within ±1 db or ±2 db between 50 and 10,000 cps. Response within ±3 db may be considered satisfactory, but not of top quality. At 3.75

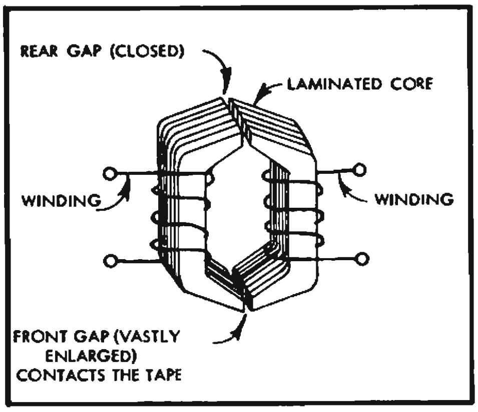


Fig. 3. Tape head of laminated construction.

ips, response should extend at least to 10,000 cps, remaining reasonably flat between 50 and 8000 cps. At 1.875 ips, response to about 5000 cps may be expected.

Distortion and Signal to Noise Ratio

The playback amplifier is generally the dominant source of noise in a tape reproducing system. The amount of signal produced by the tape playback head is at the most a few millivolts in the audio mid-range and is a fraction of a millivolt at low frequencies. When this weak signal undergoes the necessary amplification and equalization (bass boost), the noise and hum of the first stage in the playback amplifier and the hum picked up by the head, are also greatly magnified. The more signal on the tape—that is, the higher the recording level—the greater is the magnitude of the audio signal relative to playback noise and hum. In other words, the signal to noise ratio is greater. Unfortunately, as the recording level is increased, there is also an increase in distortion due to the characteristics of the tape. In sum, then, distortion and signal-to-noise ratio go hand in hand; the more distortion one is willing to tolerate in a tape system, the higher is the feasible signal-to-noise ratio, assuming that all else remains the same.

It follows that one must define how much distortion is acceptable. However, this is not a straightforward problem. To begin with, tape distortion is almost invariably stated in terms of harmonic rather than intermodulation distortion, because the amount appears respectably low in terms of harmonic distortion but tends to assume outlandish proportions when stated as intermodulation distortion. Whether maximum harmonic distortion should be 1, 2, 3 per cent, or possibly more is a viewpoint that varies considerably among tape machine manufacturers.

Signal-to-noise ratio of the top quality machines tends to be rated by their manufacturers on the basis of 1 or 2 per cent maximum harmonic distortion. This may correspond roughly to about 5 to 10 per cent IM. Many machines, however, state performance in terms of 3 per cent harmonic distortion, and some even in terms of 5 per cent; these amounts may correspond to 30 per cent and more IM. Considering that the difference between recording at a level that results in 1 per cent harmonic distortion and recording at a level productive of 5 per cent harmonic distortion represents an increase of about 8 db in recording level, it can be understood why some manufacturers rate their machines on the basis of 5 per cent. They are adding 8 db to the signal-to-noise ratio they can claim for their units.

The audiofan desiring truly clean, silky recordings—assuming such program material is available to him—will probably not want to operate his machine at levels that take him into 5 per cent harmonic distortion. More likely, he will want to stop at about the 1 or 2 per cent level. Therefore, a tape recorder should be rated for signal-to-noise ratio in terms of a signal, in the range of 250 to 500 cps, recorded at a level producing no more than 1 or 2 per cent harmonic distortion. If the ratio is based on a higher distortion figure, one can make a rough adjustment by subtracting 2 db for each 1 per cent of distortion above 1 per cent level.

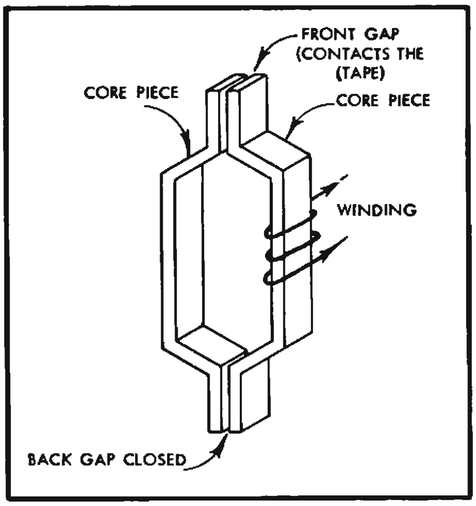


Fig. 4. Tape head of non-laminated construction.

Based on 2 per cent harmonic distortion, which is the NARTB (now NAB) standard, a tape recorder may be considered excellent if it achieves a signal-to-noise ratio of about 55 db, and very good if the ratio is closer to 50 db. Below 50 db begins to get out of the category of high fidelity. Less than 45 db tends to be unsatisfactory. With a machine having a signal-to-noise ratio that approaches 55 db, one can make a clean recording and play it back at life-like levels, yet have virtually no discernible background noise during quiet passages. Such machines, unfortunately, are still much more the exception than the rule so far as home tape recorders are concerned. On the other hand, there are a few, some at relatively moderate prices, that are the equivalent of professional machines in this respect.

Quality of Heads

Audiofans are wont to be very discriminating about the phono cartridges they choose for their audio systems. In similar fashion, there are quality differences among tape heads that deserve attention. Some of the factors involved in head quality are as follows:

1. *Gap Width.* As pointed out before, the narrower the gap, the better the high-frequency response in playback. Most playback heads encountered today have gaps sufficiently narrow to permit relatively flat response throughout the audio range at 7.5 ips and a close approximation to such response at 3.75 ips. The gap should be .00025″ or smaller for speeds of 7.5 ips or higher. It should be .0001″ or smaller for 3.75 ips.

2. *Gap Linearity.* Recording of the tape takes place at the trailing edge of the record head gap (the last edge contacted by the moving tape). To achieve a well-defined signal on the tape, it is necessary that this gap edge be equally well defined. It must be as perfectly sharp and straight as possible. Sharpness of the gap edge—in this case both edges—is also vital in playback. If the edges are rounded, then the gap *magnetically* behaves as though its *physical* width were increased. Thus a head with a .00025″ gap and very linear edges may provide better high-frequency response than a .00015″ head with a less well-defined gap. In a high-quality playback head, the magnetic gap is about 10 per cent wider than the physical gap.

3. *Head Construction.* Heads are basically of two types, laminated and non-laminated, as illustrated in *Figs.* 3 and 4. The laminated head tends to have greater output because of its greater volume of magnetic material. Moreover, the laminations serve to reduce eddy current losses (by interrupting eddy current flow), which increase with frequency.

Figure 5 shows another type of non-laminated head, whose gap has considerably less depth than in *Figs.* 3 and 4. This means that the gap wears more

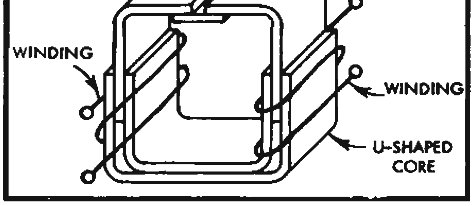


Fig. 5. Another type of non-laminated tape head.

quickly, with attendant loss of high-frequency response.

4. *Hum-Bucking Windings.* Whereas the heads in *Figs.* 3 and 5 each have two sets of windings, the head in *Fig.* 4 has only one winding. Two windings are desirable because this permits connecting them in series so as to balance out hum and at the same time increase voltage output. The manner of connection is illustrated in *Fig.* 6. Hum polarity tends to be the same at each output terminal, so that there is little or no hum potential between the terminals. On the other hand, the signal polarity at one terminal is positive when the other is negative.

5. *Saturation.* The core and design of the record head must be such as to permit sufficient magnetic flux to be developed to magnetize the tape, but without saturating the head and thereby causing distortion. While tape heads are generally satisfactory in this respect, the writer has come across instances where the record head saturated before the tape did.

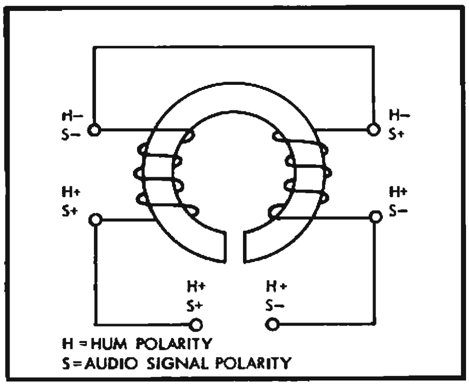


Fig. 6. Connecting the dual windings of a tape head in series for maximum voltage output and for hum cancellation.

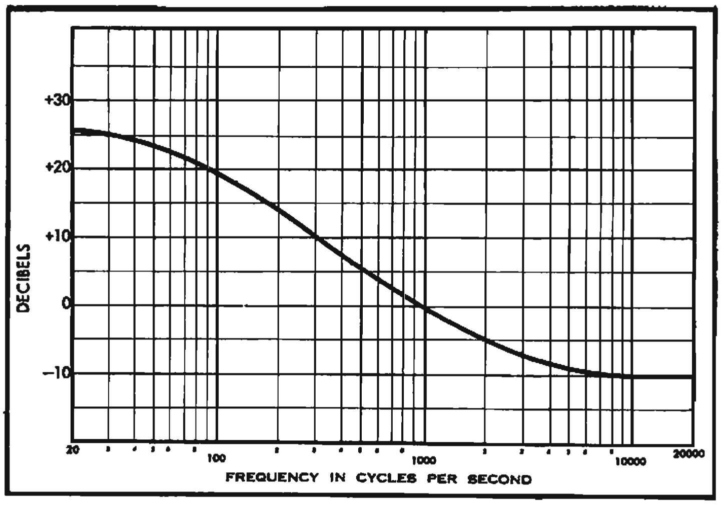
Wow and Flutter

Wow refers to slow variations in speed, below ten times per second, heard as a quavering or "sourness" in the frequency being reproduced. Flutter refers to rapid variations in speed, up to thousands of times per second, which tend to be heard as extraneous sounds in the nature of noise. That is, one hears a frequency corresponding to the rate of fluctuation.

Professional performance calls for wow and flutter not to exceed 0.2 per cent, and preferably to be less than 0.1 per cent. This is not very easy to achieve, particularly when tape speed is below 15 ips. One may say, then, that for home purposes, about 0.25 per cent is the maximum amount consistent with high-fidelity performance.

The ear is a good instrument for checking wow and flutter. By playing a test tape having a recorded frequency of about 3000 cycles (or by making such a tape with the aid of an audio oscillator), one can readily determine whether wow and flutter are unduly offensive. Wow will be apparent as an unsteadiness in the sound. Flutter will be noticeable as imparting a grainy or noisy quality to the note.

Fig. 7. NARTB playback Equalization.



Speed Accuracy

Professional requirements are that tape speed be correct within ± 0.3 per cent. Since there are 1800 seconds in one half-hour (approximately the playing time for one track on a 7-inch reel of standard tape), a 0.3 per cent error translates into 5.4 seconds slow or fast per half hour. Professional machines, and sometimes the semi-professional ones as well, generally achieve an accuracy of 0.2 per cent, which is 3.6 seconds slow or fast per half-hour.

So long as a tape machine is employed to play only tapes recorded on the same machine, the speed error is of no consequence, assuming the error remains the same over time. However, if commercially recorded tapes or other tapes made on different machines are to be played, speed errors appreciably greater than 0.3 per cent, particularly those over 1 per cent, are apt to be noticed as significant deviations from correct pitch.

The better tape machines employ synchronous motors, whose speed is essentially determined by the line frequency, namely 60 cps. But use of a synchronous motor for driving the capstan does not in itself guarantee accurate speed. The diameter of the capstan may be out of tolerance, resulting in excessive speed error. Or there may be other misadjustments.

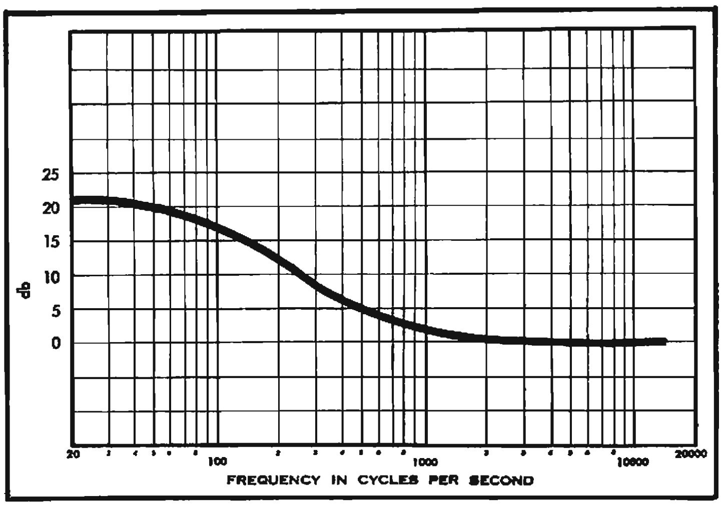
The individual who pays the extra cost of a tape machine containing a synchronous motor is entitled to a speed accuracy within 0.3 per cent. Should he find, through a test timing tape or use of a tape stroboscope, that the error exceeds 0.3 per cent, he is entitled to have this excessive deviation corrected by whatever means are appropriate, including replacement of the machine.

On the other hand, if the machine does not have a synchronous motor, speed errors up to 1 per cent should be expected and tolerated. Over 1 per cent may be considered excessive for a high-quality home machine with a non-synchronous motor.

In measuring speed accuracy, this should be done at several portions of the reel, because the error will tend to vary from beginning to end of a reel of tape.

While it is desirable for the individual to measure speed accuracy (by means of a stroboscope or test tape), in the great majority of instances there is nothing he personally can do to correct the situation. Whereas a fair number of phono turntables and even record changers provide the operator with means for readily adjusting speed, it is a rare tape recorder that makes such provision. Before

Fig. 8. Playback equalization employed at 3.75 ips.



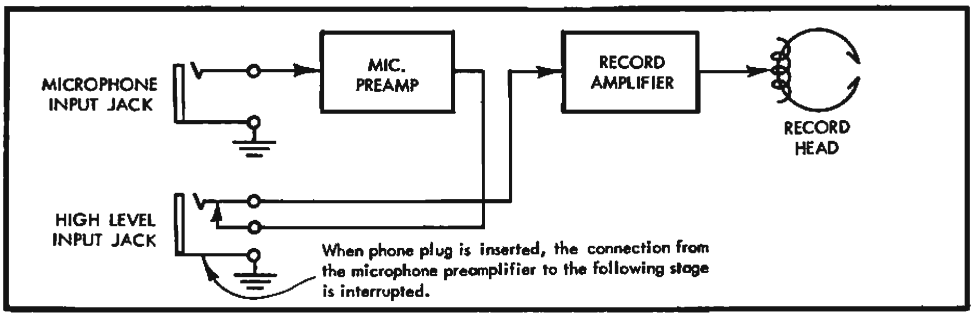


Fig. 9. Tape machine permitting one to record from a microphone or a high-level source, but not both at once.

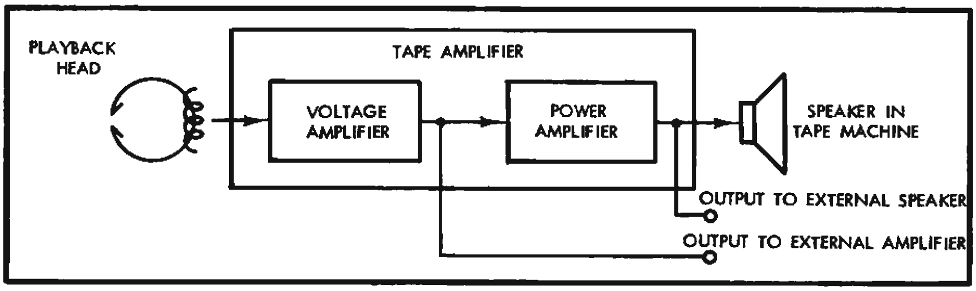


Fig. 10. Tape machine with separate outputs for external speaker and external amplifier.

the user seeks to tinker with the transport mechanism in order to speed it up or slow it down, he should take into consideration that this attempt is likely to backfire. He may improve speed accuracy, but at the same time he may cause an increase in wow and flutter, which are generally more deleterious to satisfactory musical reproduction than are moderate speed errors. The user's best recourse is to take a seriously inaccurate machine back to the point of purchase.

Equalization

NARTB equalization (*Fig.* 7), or a close approximation thereto, is considered virtually standard today for tape recorders operating at 7.5 ips. Accordingly, the tape recorder should provide NARTB playback equalization within ± 2 db at 7.5 ips. Otherwise, when playing commercial recorded tapes, frequency response may depart significantly from flat. Inasmuch as most machines that depart from the NARTB playback characteristic provide inadequate bass boost, the resultant response when playing a recorded tape will be a thin bass sound. These machines sometimes apply considerable treble boost in playback, whereas none is called for by NARTB (except to compensate for head deficiencies), so that shrillness is introduced when playing a recorded tape.

With respect to the 3.75-ips speed, the equalization question is not settled at the present writing. For a time, equalization such as in *Fig.* 8 was employed. Recently, however, there has been a trend toward employing NARTB equalization (*Fig.* 7) for 3.75 ips as well as for 7.5 and 15 ips.

Assuming that NARTB playback equalization is employed at both 7.5 ips and 3.75 ips, nevertheless, different record equalization will be required at each speed because the recording losses vary with tape speed. Therefore the tape recorder should contain switching facilities to vary the record equalization with speed. Some machines, however, employ the same record equalization at both speeds. The result is that frequency response—in terms of smoothness as well as range—is not as good at either speed as it might be, because compromise equalization is used; or else the result is that, if good frequency response is maintained at 7.5 ips, then response is considerably short of as good as it might be at 3.75 ips because the other speed has been favored.

Inputs

Tape recorders customarily have two inputs. One is for high-level sources, such as the signal from a tape output jack of a control amplifier, or the signal obtained directly from a tuner, TV, or the like. The other input is for microphone. In many cases these are alternative inputs, as illustrated in *Fig.* 9, so that one can record from one input or from the other, but not from both at once. Insertion of a phone plug into the high-level input jack disconnects the microphone signal. In other instances, it is possible to record from both sources simultaneously. Too often, however, only the high-level input has a gain control, so that it is difficult to achieve satisfactory mixing. In the better machines, there are individual gain controls for each input.

The microphone input in the lower-price machines is customarily intended for a piezoelectric (ceramic or crystal) microphone or for a high-impedance magnetic microphone. If one intends to use a crystal or ceramic microphone, it is necessary to ascertain that the input impedance of the tape recorder is sufficiently high to permit full bass response. Typically, an input impedance of 5 megohms or more is required; this depends upon the particular microphone used. Information on the necessary input impedance should be obtained from the microphone manufacturer. If the input impedance of one's tape recorder is less, the necessary modification should be made by a service technician.

A high-quality tape recorder (usually the semi-professional and professional ones) will provide an input for a low-impedance microphone, which permits a long run of cable to the tape recorder without loss of high frequencies and which is less sensitive to hum pickup than the high impedance type.

Outputs

Although a tape machine may contain its own power amplifier and speaker, it should still provide an output jack for

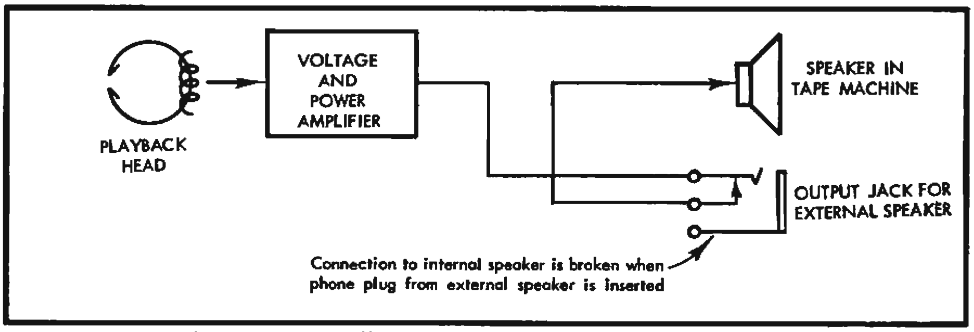


Fig. 11. Means for automatically disconnecting a tape machine's internal speaker when an external speaker is plugged in.

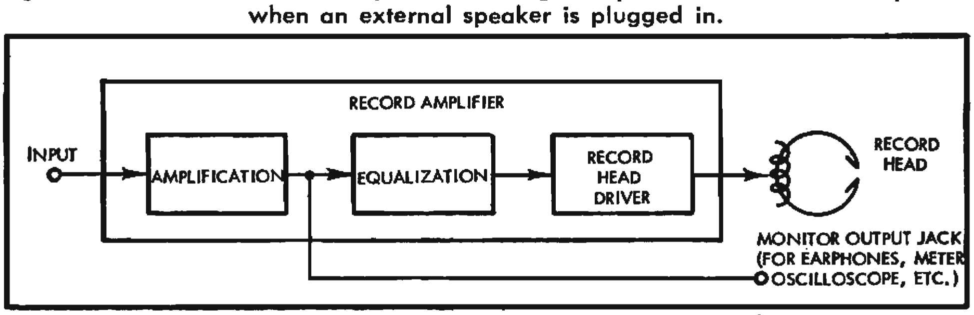


Fig. 12. Tape machine with a monitor output jack.

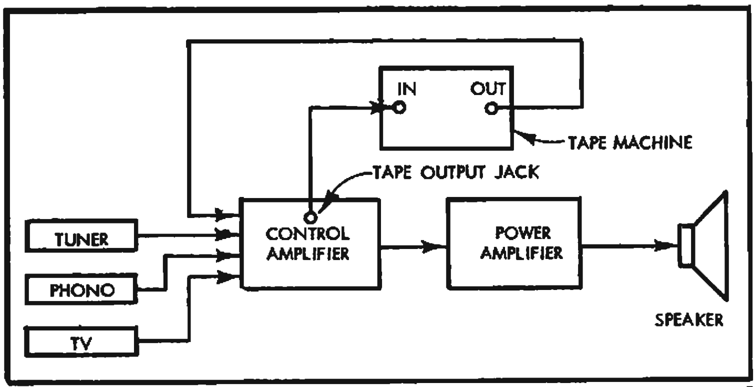


Fig. 13. Most frequently used method of feeding signals from a high fidelity system to a tape machine.

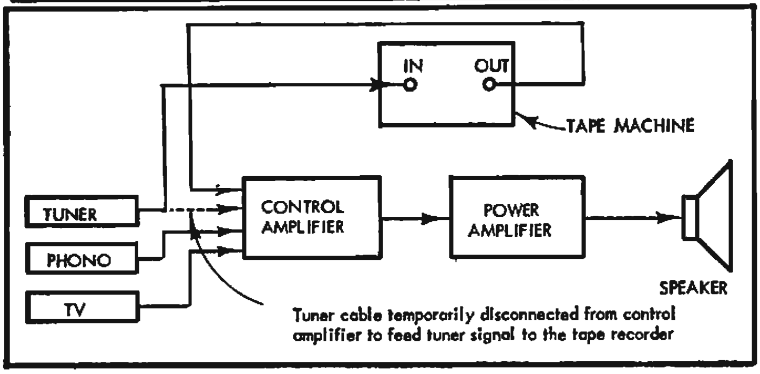


Fig. 14. Seldom-used method of feeding a signal from a high-fidelity system to a tape machine.

feeding an external audio system. Preferably, for minimum distortion, the signal should be taken from a point *prior* to the power amplifier of the machine. In some units, however, there is an output jack designated for feeding either an external speaker or an external amplifier; in this case the signal is taken *after* the internal power amplifier. The better situation is where there are two output jacks, one for feeding an external amplifier and the other for feeding an external speaker, as illustrated in *Fig.* 10.

To obtain as flat response as possible, it is ordinarily desirable that the signal at the output jack be taken from a point prior to the machine's tone controls, if any. In some machines, however, the tone controls are employed as part of the equalization circuit, and in this case one would want the playback signal *after* the tone controls. At the same time, it is necessary to ascertain the position of these controls that achieves flat response.

If the tape machine contains its own power amplifier and speaker, means should be provided for cutting off the internal speaker when the signal is fed to an external sound system. In some cases this is done automatically when a plug is inserted into the output ja^k, as shown in *Fig.* 11.

Some tape recorders contain a monitor jack, so that when recording one can listen to the incoming signal with earphones, as shown in *Fig.* 12. While this gives some evidence that the recording signal is getting through, it is not positive proof that the signal is being satisfactorily recorded. Such proof is obtained only by using a machine with separate record and playback heads, which permits the signal being recorded to be played back immediately and checked. However, a monitoring jack does have worthy uses. Thus if one is recording directly from a tuner or phonograph, one can at least check the quality of the incoming signal. Or one could attach an oscilloscope or meter (high impedance, to avoid loading effects) to evaluate the nature of the signal with respect to amplitude, transients, frequency response, and so on.

To permit a long cable run from the tape machine to the following equipment without high-frequency loss, a low output impedance is desirable. It is for this reason that the output jack in some machines is connected after rather than before the power amplifier stage (we are speaking, of course, of those units having their own power amplifier and speaker). A preferable course is for the machine to incorporate a cathode follower or other low-impedance circuit (such as a plate follower) in the output stage.

Input Sensitivity

When recording from a source other than a microphone—FM tuner, AM tuner, TV sound, phono pickup—most audiofans will obtain the signal from a control amplifier, as illustrated in *Fig.* 13, rather than by feeding the source directly into the tape recorder, as shown in *Fig.* 14 (where the source is a tuner). In a number of control amplifiers, the incoming signal is routed directly to the tape output jack (for feeding a tape recorder), without amplification or attenuation of the signal, as illustrated in *Fig.* 15. Since high-level sources generally produce at least 0.5 volts on peaks, it would appear that a tape recorder sensitivity—input signal required to drive the machine to full permissible recording level—of 0.5 volts is sufficient for high-level signals. However, it is advisable to allow for two contingencies: (1) the occasional high-level source that produces less than 0.5 volts; (2) the desirability of going above normal recording level, as for example on speech, where distortion is less apparent than on music. Accordingly, the high-level input sensitivity of a tape recorder should be about 0.1 to 0.2 volts.

In some control amplifiers, as illustrated in *Fig.* 16, the incoming signal first goes through an input level-set control. Then it goes through a stage of gain before reaching the tape output jack, at which point it *may* be restored to approximately its original level. However, there is no assurance that such

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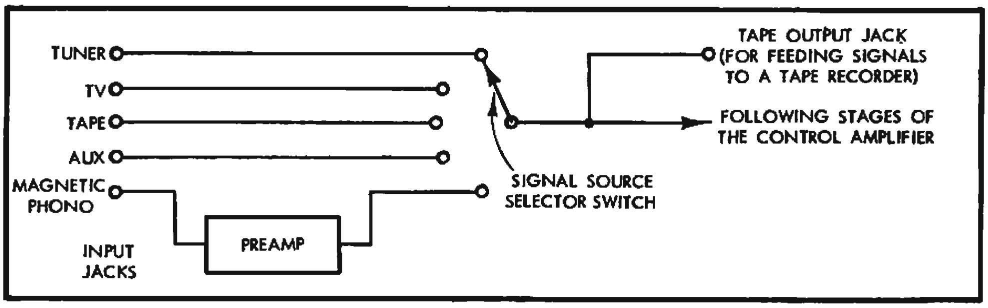

Fig. 15. Method employed in some control amplifiers for feeding incoming signals directly to a tape recorder.

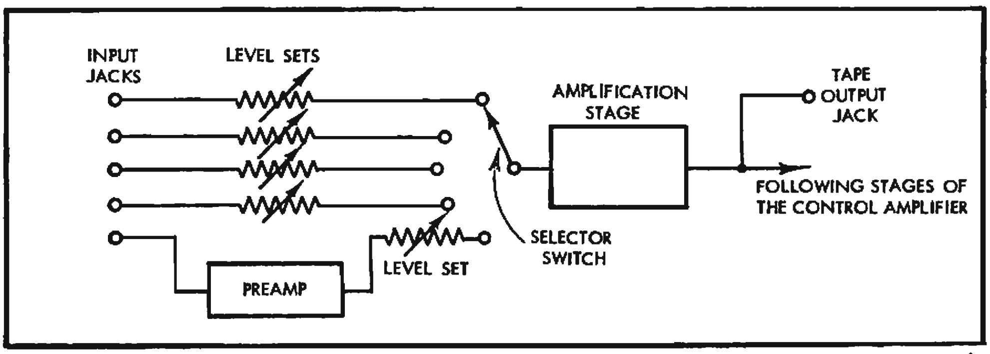

Fig. 16. Method employed in some control amplifiers for feeding incoming signals to a tape recorder.

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(*from page 40*)

restoration will be complete. Therefore it is all the more desirable in this situation that the tape recorder be capable of being driven by a signal well below 0.5 volts.

The sensitivity required for microphones depends a great deal on the type of microphone used. Of the two types most apt to be used by the home recordist—piezoelectric and magnetic microphones—the latter generally produce considerably less signal. At ordinary speaking levels a few feet distant, the magnetic microphone may produce about 2 or 3 millivolts at average levels and perhaps 10 to 15 millivolts on peaks. Allowing for a reasonable reserve of gain, an input sensitivity of about 2 millivolts is desirable.

Output Level

If the tape machine (that is, the signal from the tape amplifier) is played back through a high-fidelity system, most often it will be fed to the control amplifier, which generally can be driven to the desired level (enough to drive the power amplifier and speaker in turn) by signals of about 0.1 to 0.5 volts, depending upon the particular control amplifier. Allowing for a reasonable reserve, it may be said that an output of 1 volt from the tape machine should be sufficient in virtually all situations. One volt should also be enough to drive a power amplifier directly, as is sometimes done, because most power amplifiers can be driven to full or very high output by 1 volt or less.

(If the playback signal is taken directly from the tape head, then one must look to the control amplifier rather than to the tape machine to assure there is sufficient gain. On signal peaks at 1000 cps, the tape head will produce roughly 5 millivolts or less on a half-track tape; and correspondingly less on a quarter-track tape. Thus at 1000 cycles the control amplifier should be capable of being driven to about 1 volt for a signal of about 2 millivolts from a tape head.)

Adjustment Facilities

Every tape recorder should have means for *readily* adjusting the azimuth of the heads, so that the gaps are perpendicular to the length of the tape. This is commonly achieved by locating a spring under one or more of the head mounting screws, so that tightening or loosening the screws slightly will tilt the head about its vertical axis.

Adjustments for equalization, bias current, and calibration of the record-level meter are seldom found in machines of the home type, but are generally incorporated in the better grade machines associated with semi-professional and professional use. If a tape recorder is to be capable of consistently providing high-grade performance, then the following adjustments should be available.

1. *Playback Equalization.* At 15 and 7.5 ips (and perhaps at 3.75 ips as well), equalization of the playback amplifier should conform to the standard curve, which is NARTB at present. Equalization can be checked and adjusted on the basis of a series of frequencies fed into the amplifier from an audio oscillator or by playing a test tape. Some tape machines include compensation for treble losses due to the playback head. Many or most playback heads used today have gaps sufficiently fine so that treble losses at 7.5 ips and higher speeds are negligible. On the other hand, as a playback head wears, its gap tends to widen and treble response to deteriorate. Thus an adjustment for frequency response at the very high end—above 10,000 cps or so—can prove useful. However, such equalization has limits. For one thing, after the gap widens to a certain extent, the drop in high-frequency response becomes too sharp to compensate satisfactorily. For another, excessive high-frequency boost in playback accentuates noise of the playback amplifier.

2. *Record Equalization.* Once playback equalization is adjusted to conform to the standard curve, then equalization of the record amplifier—in particular, treble boost—should be capable of adjustment to yield relatively flat response on a record-playback basis. Some machines incorporate two adjustments; one determines the maximum amount of treble boost, and the other the point at which treble boost commences. This permits very accurate shaping of the recording characteristic.

3. *Bias Current.* As pointed out earlier, the amount of bias current fed to the record head governs the amount of recorded distortion. Up to a point, an increase in bias reduces distortion. Before this point is reached, however, bias current causes severe high-frequency losses in recording. At 15 ips, one can usually adjust bias for minimum distortion without seriously affecting treble response, because the high-frequency losses become very severe *above* the audio range. But at 7.5 ips and lower speeds, these losses are severe *within* the audio range. Therefore at 7.5 ips and lower speeds, the bias setting is critical, being less than that which produces minimum distortion. One must make sacrifices both in distortion and in frequency response, and the problem is to find the optimum amount of bias that does not unduly sacrifice one performance characteristic for the sake of the other. Therefore the ability to adjust bias to the optimum level is important for the person desiring the best possible results. A previous article pointed out that tape machines which use a meter as a record-level indicator generally employ a switching arrangement so that the meter can be used to check whether bias is correct. Inasmuch as the proper amount of bias current will vary with tape speed, and even with brand or type of tape, it is desirable that the bias control be fairly accessible.

4. *Record-Level Indicator Calibration.* A high-quality machine will permit one to adjust the amount of signal fed to the record-level indicator so that it accurately indicates when the amount of signal fed to the tape produces a given amount of distortion—2% or 3% harmonic distortion usually being considered the maximum permissible amount.

Bias Frequency

In order to avoid discernible beats between the bias frequency and harmonics of the audio frequencies, the bias frequency should be about four to five times the highest audio frequency, namely between 60,000 and 75,000 cps. While 75,000 cps or higher is even more desirable, a limit is set by the fact that capacitive losses in the record and erase heads increase with frequency. Accordingly, the bias oscillator has to work proportionately harder as bias frequency increases, which raises the problem of distortion in the bias waveform and attendant noise. Hence 75,000 cps or so is a practical maximum for the bias frequency. A frequency much below 60,000 cps is open to serious question as to its compatibility with high-fidelity performance.

A-B Switching

In a machine having separate record and playback heads, it is highly desirable that there be an A-B switching facility, as illustrated in *Fig.* 17, to permit comparison between the incoming signal and the signal recorded on the tape. Specifically, the output jack of the tape machine and the monitor jack should be switched between the incoming signal and the playback signal. Comparison between the two signals can then be made by earphones connected to the monitor jack or by means of a sound system fed from the output jack.

Record Interlock

One of the catastrophes that occasionally befalls the tape recordist is that of inadvertently erasing part or all of a valued tape because the machine is accidentally set in the record instead of playback mode. To minimize this danger, most tape machines provide a safety interlock that prevents putting the machine into record position unless one simultaneously actuates a special record button or lever. This button or lever should automatically disengage when the machine is put into any other mode of operation. To further minimize the danger of accidental erasure, some tape recorders have a warning light that goes on when the machine is in the record mode.

Automatic Equalization Change

It is desirable that the record equalization, and if necessary the playback equalization, be automatically changed when going from one tape speed to another.

Number of Motors

The transport has three basic mechanical functions so far as the record and playback modes are concerned: (1) To turn the takeup reel in a given direction; (2) to cause the supply reel to *tend* to turn in the opposite direction so as to provide back tension on the tape; (3) to drive the capstan, which in conjunction with the pressure roller grips the tape and pays it out at a prescribed rate of speed. As a general rule, the best performance—most accurate speed and least wow and flutter—is achieved by transports having a separate motor for each function. On the other hand, there are a few single-motor transports that through excellent design and construction achieve results about as good.

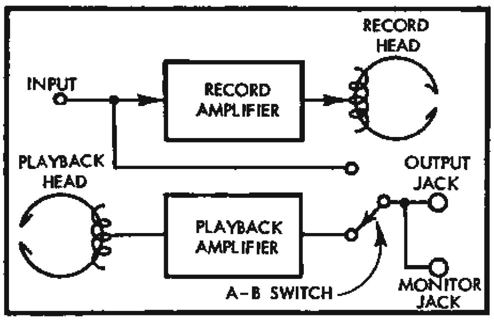

Fig. 17. A-B switching arrangement.

Tape Handling

The speed, ease, and smoothness of tape handling are among the factors to be considered in acquiring a tape recorder. Starts and stops should be fast, but accomplished with sufficient smoothness to avoid breaking or stretching the tape. While some professional machines can come up to operating speed or to a full stop in as little as 0.1 second, as much as 1 second is usually adequate for home purposes.

A transport should be able to wind a 1200-foot reel of tape in either direction in about 60 to 90 seconds; semi-professional and professional units require as little as 30 seconds. Smoothness of winding is of greater importance than speed, however, so one should not be overly distressed about a slow-winding machine provided that it winds uniformly. The slower the winding speed, the less the tendency to stretch the tape or generate stresses that can result in distortion. Some tape recordists, where utmost quality is sought, have been known to rewind a tape at operating speed by reversing the reels and putting the machine in the playback mode. If the winding speed varies according to the tape speed setting (7.5 ips, 3.75 ips, etc.), it may be advisable to rewind the tape in the slowest available speed.

Tape Lift

To minimize head wear, which is due to abrasive action of the tape against the heads, it is desirable that the tape be automatically spaced away from the heads during rapid wind or rewind. In some machines, this spacing is deliberately kept small to permit a slight amount of signal pickup (chiefly low frequencies) to facilitate locating a desired passage on the tape.

Loading

Loading of the tape machine should be a simple, rapid procedure, without the tape having to be threaded through a complex system of guides, rollers, and so on. Most transports today feature "in-line" loading, where the tape is merely dropped in a straight or slightly curved slot and thereby properly engages the driving mechanism, heads, and so on. There are times when the operator will have to reload as rapidly as possible—for example, when taping a program off the air—and facile loading can then be a most important asset of a tape machine.

The purchaser will want to check that the tape path is such as not to skew the tape, but allow it to wind from one reel onto the other without scraping the top or bottom of either reel (assuming the reels are not warped). If the path causes the tape to skew, not only is there a disturbing noise as the tape scrapes the reel, but the azimuth relationship between the tape and heads may be affected.

Tape Index

To facilitate location of a passage on a tape, a tape index of some sort is desirable. Most transports at least provide markings under the reel to indicate elapsed time or remaining time at a given speed. In addition, a number incorporate a mechanical counter of one type or another. Some employ a clock-type dial and revolving hand. Others use rotating numbered discs.

Editing Facilities

The serious recordist may be concerned with editing problems, where it is necessary to make splices at a given note or syllable. In this case, it is important to the recordist that the machine provide easy access to the tape when it is in front of the heads, so that he can mark the exact point on the tape where a splice is to be made.

Controls

Push-button controls—either mechanical or employing solenoids—are found in many tape recorders, both of the home type and of professional grade. While they offer operating convenience compared with transports manually actuated by levers, on the other hand the more complex the mechanism the greater is the possibility of malfunction. Moreover, there is less "feel" to a machine with push-button control, and tape breakage or spillage may occur if the push-button mechanism functions improperly. Æ

Superimposed Tape Recording

WILLIAM D. BELL*

One way to save tape at the possible expense of quality provides an interesting experiment in geometry. At least it is not too much trouble to try out—with a little ingenuity.

WHAT WOULD YOU SAY to making 10 different recordings, one on top of the other, on a ¼-inch magnetic tape with a conventional single-track head? And then playing *any* of the recordings at will? Sounds impossible, doesn't it? Yet, at least two computing-machine laboratories have independently discovered this recording trick—a method you can try for yourself on your own tape machine.

Recording multiple tracks on the same section of tape is easy, of course, if separate heads are employed and the tracks are kept separate. This was the method used in an elaborate system, built by Ampex for Les Paul and Mary Ford, which recorded eight tracks on one-inch tape. With separate record, erase, and playback controls for each track, this Ampex-built machine was ideal for the multiple recording techniques for which Les Paul is famous.

However, recording separate and distinct tracks on different parts of the tape is not the method I am describing here. Rather, one track is recorded right on top of another, as stated before.

The guided-missile people also record several separate channels of data on one track. Each measuring transducer modulates an audio tone of restricted band width. A number of transducer signals are then mixed and the composite signal radioed to a ground receiving station. The tape-recorded signal can then be played back through appropriate filters and each separate signal isolated. Thus, this telemetry method works because there is no overlap of the frequencies of the various channels. Again, this is not the method described here.

What, then, is the method that permits multiple recordings to be made on top of each other? The answer, quite simply, is to change the angular orientation of the head gap with respect to the tape for each recording. To play back, a recording is selected by choosing the same angular position for the head as was used for the original recording.

Alignment

In normal operation, the head gap is accurately aligned at right angles to the tape. Ampex claims in its literature that this azimuth angle is held to plus or minus a single minute of arc.

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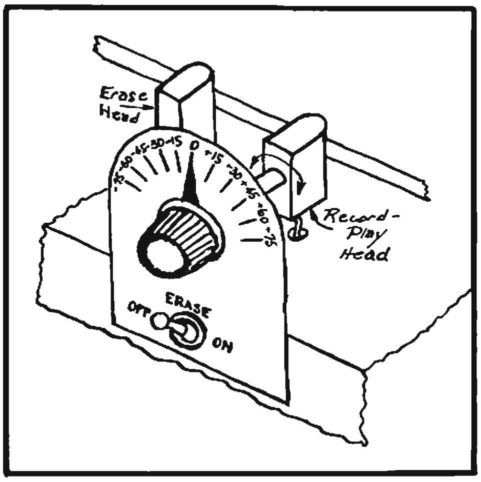

Fig. 1. Diagram of the basic arrangement of the tiltable head. It would be preferable if the head were mounted closer to the indicating scale, however.

Why is such an accurate manufacturing specification imposed? What if the head is not accurately perpendicular to the tape?

In reply, if a tape recorded on an accurate machine is played back on one with a bad azimuth angle adjustment, the signal strength will be reduced. Worse yet, there will be distortion. In fact, if the azimuth angle is deliberately increased, the output will become increasingly garbled until it finally can no longer be heard.

These are the reasons a recorded tape can sound horrible when played on a home machine. While you can indignantly return the tape to the dealer, it may be that the playback machine is the villain. If the azimuth angle of the playback head is out of adjustment, then it can't possibly sound right. By the same token, if the head on the machine that made the original recording is out of adjustment, then the tape will sound poor on any well-aligned playback equipment.

Recognizing this alignment problem, a logical question is: what happens if a tape is both recorded and played back on the *same* machine, when the azimuth is not critically adjusted? As the great number of inexpensive tape machines clearly shows, reproduction is surprisingly good.

Just how far the azimuth angle can be modified is demonstrated by a special modification supplied by Rangertone, Inc. An extra head is mounted on stock machines with the gap *parallel* to the tape! The head is used to record 60 cps along with either single-channel or two-channel material on ¼-inch tape. Although the 60 cps track is in the middle of the tape, the standard heads, perpendicular to the tape, don't even "see" the special track. The purpose of this third track is to synchronize tape playback exactly for movie making.

How to do it—

The above facts furnish a background for understanding the phenomenon of superimposed tape recording. If you'd like to experiment yourself, do this:

1. Mount the record-playback head on a pivot along with an angular pointer or indicator. *Figure* 1 is a suggested plan, although the head should be closer to its mount.
2. Fix the erase circuit so it can be disabled with a switch. Obviously, re-recordings can't be made if the previous information is erased.

That's all there is to it! Recording procedure is as follows:

1. Erase the tape.
2. Disable the erase circuit.
3. In separate steps, record different material, always rewinding and starting over at the beginning of the tape. In doing this, remember to set the azimuth angle to a *new* position for *each* recording.

What angles should be used? That depends upon your heads, tape speed, and other factors. A good trial value is increments of 15 deg. Thus, if we call the perpendicular position 0°, then we can record at −75°, −60°, −45°, −30°, −15°, 0°, +15°, +30°, +45°, +60°, and +75°.

After a tape has been multiple-recorded, how is playback accomplished?

1. Set the indicator on the head to the angular position that will select one of recordings.
2. Play back and listen!

"Tune in" the Program You Want

You'll soon discover that on playback you can "tune" the head to an accurate

(*Continued on page 113*)

SUPERIMPOSED TAPE

(*from page 62*)

setting. The effect is very much like tuning a station on a radio. As the angular position of the head gap approaches one of the settings used for recording, sound will begin to be heard. As the position gets closer, the sound becomes clearer and clearer.

How is the fidelity? Surprisingly good in view of the mad things we have done to the magnetic oxide on the tape. But, the results are definitely *not* high fidelity. However, the results are good enough that it's reasonable to believe that technique and equipment could be improved to a point of satisfying high-fidelity standards. If that could be done, then very interesting possibilities exist.

A seven-inch reel of long-play tape runs for a full hour at 3¾ ips. With ten recordings on the tape, it would play for a full ten hours! This is not only competitive with LP records—it is superior on a straight dollar-and-cents basis. The cost of reproduction should not increase, since multiple heads could be placed "in-line" and all recordings made in a single pass of the tape.

Intriguing? Æ

Tape Recorder Accessories

HERMAN BURSTEIN*

While you can dig a post hole with a tablespoon, you can do it much easier with the proper tool; similarly, these accessories become the desirable "tools" to the recordist, making his work much easier.

TAPE RECORDER ACCESSORIES are considerable in number. As to their usefulness, an anology with automobile accessories may be in order. Some are a virtual necessity (like a heater if you live in the northern part of the country). Some are highly desirable (like a radio or power steering). And some are a matter of convenience (like the gadget that squirts water on your windshield): nice to have but not indispensable.

It is impractical to attempt to sort the various accessories into these three categories of usefulness. What is merely a convenience to one tape recordist may be a necessity to another because of the manner or extent of use of his tape recorder. Therefore it should be clearly understood that the order in which accessories are discussed below constitutes only an extremely loose approximation to their relative importance. And this order of importance is purely the subjective view of one person—the writer.

Tape Splicer

One of the accessories most likely to fall into the category of necessities is the tape splicer. Whether or not you plan to go in for an appreciable amount of tape editing, which involves the excision of undesired portions and the joining of desired portions, you will still find a tape splicer handy for the accidental break that will inevitably occur. True, you can use a scissors for the job, but it is rather difficult to do a precise, neat job, which means bringing the two sections of tape together in exact alignment so as to avoid a thump when the spliced portion passes the playback head.

Tape splicers come in a great variety of types and prices, ranging from as little as approximately $1.50 to as high as $50 or more. *Figures* 1, 2, and 3 show three models, which are respectively low-, medium-, and high-priced units. The expensive ones are more rugged, have a greater degree of automaticity, and operate more quickly. Yet one can do a perfectly adequate job of tape splicing with the least expensive units.

If your only need for a tape splicer arises from an occasional accidental break, then a low-priced unit may be your best purchase. But if you plan to do a fair or substantial amount of editing, it is wise to investigate the more sophisticated models as most suited to your needs. One of the factors to be taken into consideration is the fact that some splicers are sufficiently compact so that with the aid of an adhesive they can be mounted directly on the tape machine, where they are always conveniently at hand.

It is vital that one use a splicing tape specifically made for this purpose rather than conventional cellophane tape. If the latter is used, the adhesive is apt to ooze out under the pressure exerted by the capstan and pressure roller, and will be deposited upon these and other parts, causing wow and flutter and otherwise impairing operation of the machine.

As a further precaution against fouling of the mechanism by the splicing material, it is desirable to cut the tape so that it narrows slightly in the area where the splice occurs. A number of splicers make such a cut, as do those in *Figs.* 2 and 3, which produce what their manufacturer calls a "Gibson Girl" cut.

In the event you rely upon a scissors rather than a tape splicer, be sure to make the cut at an angle of about 30 to 45 deg. rather than 90 deg. A cut of right angles to the length of the tape will result in a thump as the tape passes the playback head.

* 280 Twin Lane E., Wantagh, N.Y.

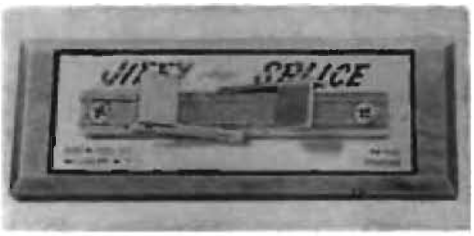

Fig. 1. A low-priced tape splicer—"Jiffy-Splice," by Rason Mfg. Co.

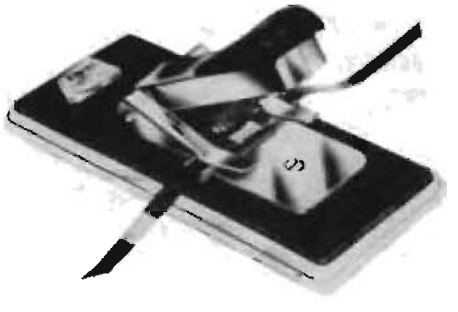

Fig. 2. Robins Industries Corp's. medium-priced splicer—Model TS4A-STD.

Fig. 3. A high-priced splicer,—Robins' Model TS-250-1000.

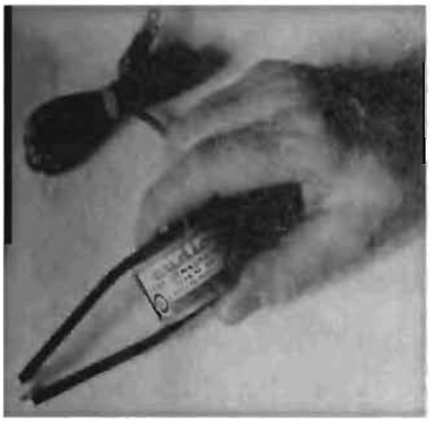

Fig. 4. Recording head demagnetizer—Audio Devices, Inc., Type 400.

Head Demagnetizer

The record and playback heads tend to become magnetized gradually as the result of the asymmetrical wave-forms contained in speech and music; this asymmetry is in effect a d.c. component producing magnetization. The heads may also become magnetized due to surges of current as the machine is turned on or off. A magnetized head is a source of noise; magnetized record and playback heads inscribe noise on the tape, while a magnetized playback head also produces noise directly. Moreover, such heads may erase the higher frequencies.

Therefore the individual wishing to preserve his valued tapes in the best possible condition should demagnetize the heads on his machine at frequent, regular intervals. About every 8 to 10 hours of use is a suitable period.

Figures 4 and 5 show several makes of head demagnetizers. They operate from house current and apply an alternating magnetic field of decreasing intensity to the head, accomplished by bringing the pole pieces of the demagnetizer in contact with the head and then slowly removing them. The unit in *Fig.* 5 comes with three sets of removable pole pieces, each shaped differently to permit access to heads in various types of housing.

If you are slightly handy and happen to have an old, inexpensive, two-pole motor, such as is commonly found in a cheap phonograph of the $20 variety, you can construct your own head demagnetizer, as shown in *Fig.* 6. First, remove the armature, so that only the field of the motor is left. Next, remove portions A and B, which are respectively a metal bar serving as a magnetic link for the motor and a heavy copper wire used as a short-circuiting ring. Attach a strip of iron—a silicon steel strip from the core of a junked transformer is fine—as shown, and you will find there is sufficient magnetic field at the end of this bar to demagnetize the heads. The bar is attached by one of the screws holding the field laminations together.

Cleaner for the Heads and Other Parts

As important as the demagnetizer is head cleaner, frequently suitable for cleaning other parts as well. Due to friction, a layer of tape oxide builds up on the heads, capstan, pressure roller, and guides. These and other components may also become contaminated with oil. As a general rule, a cotton swab dipped in alcohol will serve effectively as a head cleaner. Whether it serves effectively for other parts depends upon the nature of the contamination.

One can purchase fluids specifically intended for removal of tape oxide from heads and other parts, such as HC-2, made by Robins Industries. Or one can obtain an all-purpose cleaner, such as Long Life, made by Electrical Chemical Specialty Company. The latter consists of four different kinds of solvents, one for oxide, the second for grease, the third a general cleaner, and the fourth a diluting agency to limit the potency of the active solvents.

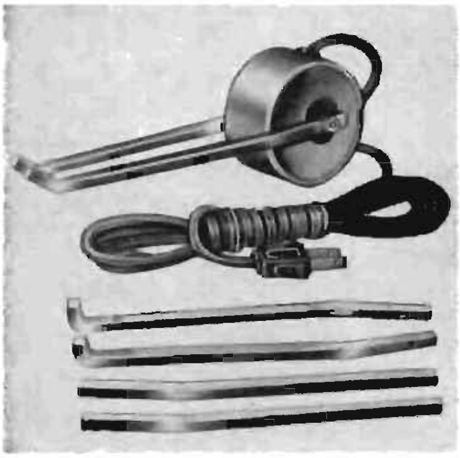

Fig. 5. Head demagnetizer with interchangeable pole pieces—Lafayette Radio's Model PK-238.

The accumulation of tape oxide on the heads prevents intimate contact between them and the tape, resulting in high-frequency losses. Extremely minute accumulations can produce significant losses. Accumulation of oil and other materials on the guides, capstan, pressure roller, and soon tends to produce speed irregularities, manifest as wow and flutter. Also, they can cause deviation from correct speed.

Fig. 7. Walsco "Kleen-Tape," designed for cleaning heads.

Another form of head cleaner is that shown in *Fig.* 7, namely a tape which bears not the usual magnetic coating, but instead a special material that cleans and polishes the tape. This comes in a 100-foot length on a conventional reel and is operated past the heads in the same manner as a regular tape.

Tape Conditioner

Some tapes, particularly after long use, develop increasing friction as they pass the heads, resulting in audible squeal and distortion. Several compa-

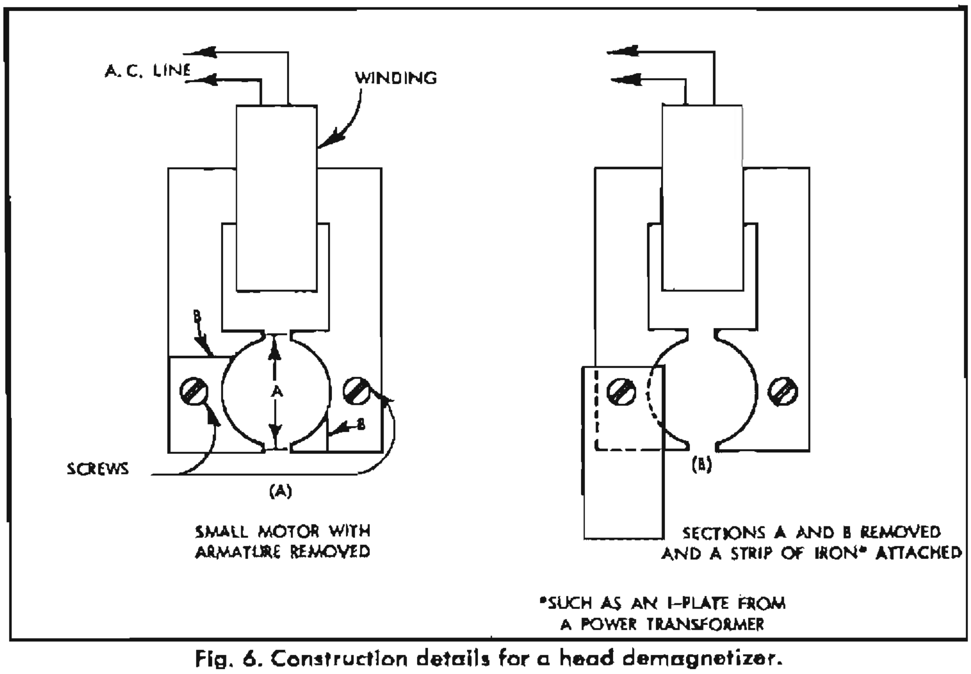

Fig. 6. Construction details for a head demagnetizer.

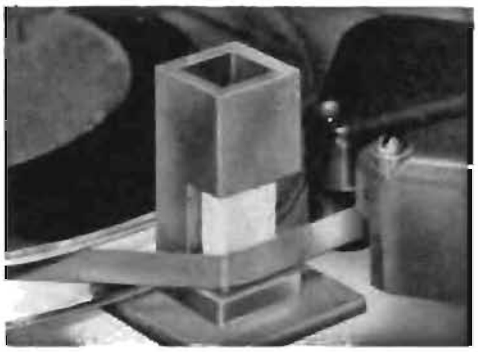

Fig. 8. Applicator used to apply Long Life Conditioner to the tape.

nies have therefore brought out products to lubricate and clean the tape. One is the Long Life Tape Conditioner, made by the same company that manufactures the Long Life Cleaner previously described. A special applicator, as shown in *Fig.* 8, is used to make contact with the tape as it is run from reel to reel. Robins Industries produces what is calls a Jockey Cloth, which is held in contact with the moving tape and serves to clean as well as lubricate the tape.

Head and Guide Lubricator

For minimum wow and flutter, minimum distortion due to erratic tape movement, and elimination of tape squeal, it is desirable to lubricate the heads and guides. One of the products designed for this purpose is Long Life Lubricant (the third of a trio of companion products), which contains silicone. An important caution is in order here. Be sure that the lubricant does not get on the capstan or pressure roller, as this will cause tape slippage.

Bulk Eraser

A bulk eraser is a powerful electromagnet that can completely erase a reel of tape in a matter of seconds and a good deal more effectively than the erase head does. The entire reel of tape is brought into contact with the bulk eraser and then slowly removed, meanwhile describing a circular motion so that the magnetic field cuts all parts of the tape. In some cases, the bulk eraser has a handle, so that it rather than the reel of tape may be moved.

The bulk eraser is useful when dealing with a tape that has been heavily over-recorded, because it is then difficult or impossible for the erase head to completely remove the signal, at least on the first pass. The noise level on the tape may be brought to a minimum by subjecting the tape to a bulk eraser prior to recording. The bulk eraser can prove a necessity rather than a mere convenience in the event the tape recorder contains no erase head, which occurs in some instances where the erase head has been removed to make room for another head, such as a separate record head, a four-track stereo head, etc.

A major disadvantage of the bulk eraser is that it erases the entire tape, not just the one or two tracks that one wishes to eliminate. Another disadvantage is that one cannot confine erasure to a given length of the tape with the same precision as when using an erase head.

(As an incidental note, a bulk eraser can be useful in other ways than in connection with tape recording. Thus it can be employed to magnetize or demagnetize tools, to demagnetize tubes, to demagnetize tube shields, and so on.)

One should not attempt to bulk erase a reel of tape as it lies on the tape machine, particularly if the machine employs a meter as a record level indicator. Bringing the bulk eraser into the vicinity of the meter is apt to upset its calibration or perform even greater injury.

Bulk erasers, several of which are shown in *Fig.* 9, are a relatively expensive accessory, generally costing between \$20 and \$40. However, it is not difficult for the recordist to make his own at a very little cost. This assumes he can obtain an old power transformer, such as is found in a power amplifier, TV set, or transformer-operated radio; these are often to be had in a surplus radio parts store for a few dollars. The procedure then is as follows. Disassemble the transformer by removing the nuts and screws that hold the casing and laminations together. Remove the E-shaped and I-shaped plates from the transformer core. Reinsert the E-plates so they all face in the same direction; put the I-plates aside (you can use one of these as the metal strip for a head demagnetizer, as previously described). Bolt the E-plates together, using the screws and nuts previously removed; do not replace the casing. Attach lamp cord and a plug that mates with the house socket to the leads of the primary winding. Be sure that you know which are the leads of the primary winding before you start on this job. Snip all other windings and tape them so they will not make contact with each other. Wind the bulk eraser with rubber tape to protect the core and windings. Insert the plug into the house socket, and you have a powerful electromagnet that can erase a tape in seconds.

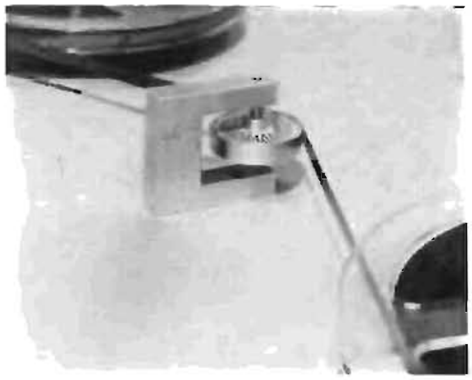

Fig. 10. The Tape Strobe, a product of Scott Instrument Labs.

Do not leave this bulk eraser connected to the house line for more than about one minute at a time, because it heats up fairly quickly. However, in one minute you can erase several reels of tape. If your needs call for the bulk eraser to be operated an appreciable length of time, you should consider the purchase of a commercial unit.

Test Tapes

Test tapes will be discussed in detail in a later article concerned with the testing, alignment, and adjustment of tape machines. It may be briefly stated here that it is desirable to have one or more

Fig. 9. Various types of bulk erasers—left to right, Lafayette Radio ML-120, Rason Mfg. Co. "Jiffy-Rase," and Robins Industries' Model ME-99.

Fig. 11. Orr Industries' Stroboscopic Tape Disc.

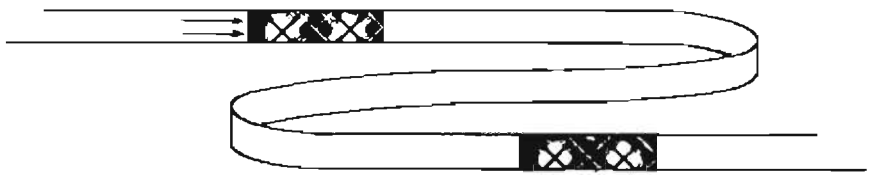

Fig. 13. Typical leader tape, Minnesota Mining and Manufacturing Co.

test tapes that provide test tones to check the following:

1. Azimuth (a tone of about 10,000 cps is desirable).
2. Frequency response (tones covering the range of 50 to 15,000 cps).
3. Wow and flutter (a tone of about 3000 cps, at which frequency wow and flutter tend to be most noticeable).
4. Correspondence of the record level indicator with the maximum permissible recording level. (The test tape should have a test tone at a frequency between 250 and 500 cps, recorded at a level resulting in 1, 2, or 3 per cent harmonic distortion.)

Tape Stroboscope

It is desirable to have a means of checking whether the tape is running at reasonably accurate speed. Of course, what is "reasonably accurate" depends upon one's acuity in detecting departures from true pitch. In the case of semi-professional and professional users, accuracy in timing programs also becomes important. To most human ears, a speed error not greater than 1 per cent will be unnoticeable; in fact, many persons can tolerate errors as great as 5 and even 10 per cent. Thus the 1-per cent criterion will be acceptable in the majority of instances. On the other hand, professional practice calls for speed error to be kept within limits of ±0.3 per cent. Thus the deviation in a half-hour program would be no more than 3.6 seconds.

At least two stroboscopic devices are now on the market. One is the TapeStrobe made by Scott Instrument Labs, shown in *Fig.* 10. The device is placed on a convenient part of the tape deck and pressed slightly against the moving tape, causing the strobe to turn. The TapeStrobe has sufficient weight so that it will remain in the position where it is placed. The strobe markings are viewed under the light of a bulb operating on 60-cycle current. This may be an ordinary light bulb; to obtain a sharply defined strobe image, however, a neon or fluorescent lamp is preferable. If the pattern appears stationary, speed is correct. If it appears to be moving forward, that is, in the same direction as the tape, speed is fast. If the pattern appears to be moving backward, the speed is slow. To judge the degree of speed inaccuracy, count the number of bars that appear to be passing a fixed point during one minute (hold a pencil point over a fixed spot on the strobe). If the number of bars is 72, speed error is 1 per cent. A greater or smaller number of bars denotes a proportional error. To illustrate, 36 bars would signify a speed error of 0.5 per cent; 144 bars would signify an error of 2 per cent, and so on. This holds true for all speeds.

Another stroboscopic device is the Stroboscopic Tape Disc made by Orradio Industries, Inc., shown in *Fig.* 11. This operates in exactly the same fashion, except that it is held by hand against the moving tape instead of being positioned on the tape deck.

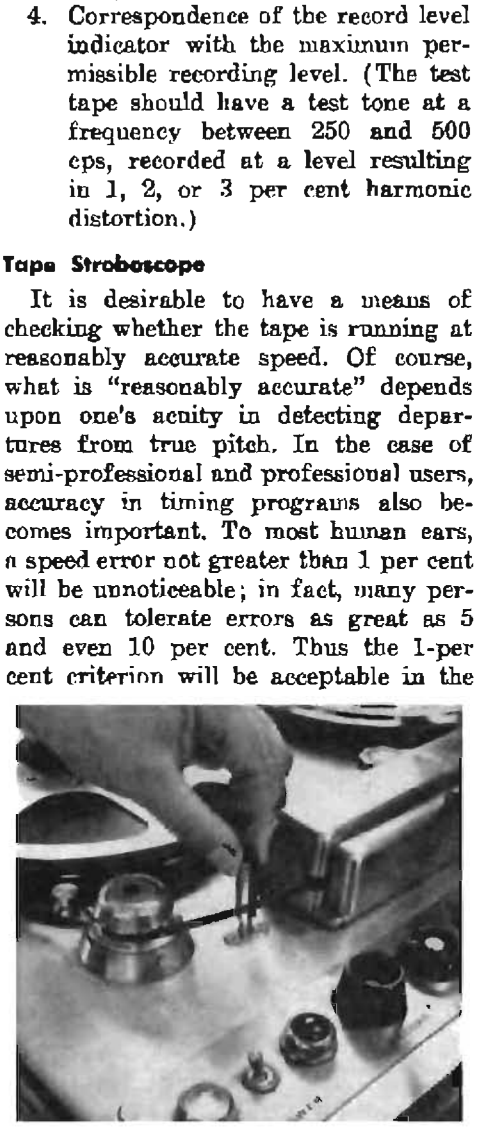

Fig. 12. Audio Devices' "Echoraser."

Print-Through Remover

When a tape has been heavily recorded and/or stored for a considerable length of time, the phenomenon known as print-through tends to occur, namely the transfer of the signal on one layer of tape to the adjacent layers. Hence one may hear what is known as "pre-echo" and "post-echo." This effect is more severe for the long-playing and extra-long-playing tapes than for the standard tape, because the former are thinner and therefore present less of a barrier to signal transfer.

Of quite recent vintage is a device known as the Echoraser, developed by Audio Devices, Inc., that can remove print-through from a tape without substantially affecting the original audio signal. As shown in *Fig.* 12, it is designed to be slipped onto pins that are permanently attached to the tape deck. The tape contacts the Echoraser, and as the tape passes through the magnetic field of this device the print-through is reduced by various amounts, depending upon the age of the recording, the magnetic properties of the tape, conditions under which the tape has been stored, and the signal frequencies.

Echorasers are supplied in two strengths. One strength, according to the manufacturer, achieves up to 9 db removal of print-through without significantly affecting high frequency response. The other strength, intended for the most serious cases of print-through, achieves up to 18 db removal and involves losses of about 2 to 3 db at high frequencies.

Other Accessories

There are a number of other tape recorder accessories which require a minimum of comment. These include the following.

1. Write-on labels for identifying the contents of a recorded reel of tape. These come on a dispenser in the same fashion as cellophane tape.

2. Leader tape (*Fig.* 13), which may be attached to the beginning, end, or intermediate portions of the reel of tape. This can serve at least three important purposes: (1) it can protect the ends of the tape from wear and tear. (2) It permits one to write in the contents of a reel of tape. Although the reel may bear a label, it is quite possible that one may neglect to rewind the tape back onto this reel after it has been played, particularly in the case of a one-way stereo tape. Such neglect may lead to accidental erasure of a valued tape, unless one takes the precaution identifying the contents on a strip of leader tape. (3) The leader can be used to separate various portions of a tape, permitting quick and accurate location of desired sections.

3. Tape clips (*Fig.* 14), which prevent the end of the tape from unraveling off the reel.

(*Continued on page 107*)

Fig. 14. Robins Industries' Tape Clip.

TAPE GUIDE

(*from page 38*)

4. Metal storage cans and storage chests are also available. Some cans are made of special material to shield the tape from magnetic fields produced by motors, transformers, and so on.

5. The Tape-Time Ruler is manufactured by Ferrodynamics Corp., Lodi, New Jersey and is available—at least at the time of writing—free of charge. Not all tape machines contain footage counters or clear and accurate markings under the reel to indicate elapsed time and thereby help the operator locate a desired passage on the tape. The Tape-Time Ruler is placed on the supply reel spindle, and a reading is taken at the outermost layer of tape to indicate the number of feet and playing time left.

While the foregoing has sought to be a comprehensive account of the accessories available to the operator of a tape machine, it is quite possible that some have been inadvertently omitted. If so, the writer asks the indulgence of the manufacturers of such items and will attempt to correct the omission in the near future. Æ

Kinds of Tape

HERMAN BURSTEIN*

A compendium of information about all the kinds of tape that are available for the home and professional recordist with respect to the playing time, print-through, output, and backing material.

MAGNETIC TAPE is in essence a coating of iron oxide on a plastic base. The base may range from about 0.5 to 1.5 mils (thousandths of an inch) in thickness, while the coating may range from about 0.35 to 0.65 mil. By varying the chemical formulation of the coating, the type of base material, and the thickness of the coating and base, and by suitable attention to such factors as uniformity of coating and base, fine dispersion of the magnetic particles that constitute the coating, lubrication, and so on, the tape manufacturer controls the physical and magnetic properties of his product.

The magnetic properties of the tape principally concern the following:

1. Frequency response
2. Output (for a given level of distortion)
3. Noise
4. Print-through

The physical properties of the tape principally concern the following:

1. Playing time
2. Strength
3. Smoothness
4. Limpness

For selling purposes, manufacturers classify their tapes on the basis of the following four characteristics: playing time, strength, output, and print-through. For a given combination of the above characteristics, the purchaser may then decide what brand to buy, or which one of several tapes within the same brand to buy, on the basis of the following characteristics: frequency response, noise, smoothness, limpness, and other attributes to be discussed later. His choice may rest upon personal experience with various tapes, upon the recommendations of others (including advertising), or possibly upon price alone.

The extent to which differences may be observed among brands or within brands depends in part upon the tape equipment employed. To illustrate, the fact that one tape is less noisy than another may not be apparent in a low cost tape recorder which generates a substantial amount of noise, thereby over-riding tape noise. On the other hand, a high-grade tape machine may permit differences in tape noise to be obvious.

The likelihood of perceiving differences among tapes also depends upon how the tape equipment is used. For example, print-through may never bother some recordists because they record only at moderate levels or because they do not store recorded tapes for long periods before playing them (print-through increases with storage).

Before proceeding with the discussion of tape characteristics, it should be pointed out that it is not possible to produce a tape which produces maximum performance in all desired respects. Frequently, it is necessary to sacrifice performance somewhat in one respect in order to improve performance in another. On the other hand, it sometimes happens that an improvement in one direction also brings an improvement in a second direction.

The next four sections will deal with the characteristics generally employed to classify tapes. The fifth section will classify several leading brands of tape according to these characteristics. The last section will discuss other important characteristics.

Playing Time

A very simple and obvious, as well as important, distinction among tapes concerns playing time. We are speaking of different lengths of tape, depending upon their thickness, that may be accommodated on the same reel size. For convenience, we shall refer to the 7-in. reel, which is virtually standard for home use. Tape may be sorted into three categories:

1. Standard-play
2. Long-play
3. Double-play

Standard-play tape is about 2 mils (.002 in.) thick; the plastic base is about 1.5 mils and the coating about 0.5 mil. A 7-in. reel can accommodate 1200 feet of such tape; this translates into 32 minutes playing time if the tape is operated in one direction at 7.5 ips, which is the speed commonly used in the home for high fidelity reproduction. If the reel is reversed and the tape also recorded in the opposite direction—in the case of mono half-track or stereo four-track recording—the playing time is 64 minutes.

Long-play tape has a base of about 1 mil and a coating of about ⅓ mil, for a total thickness approximately two-thirds that of standard-play tape. Since tape length on a given reel size is inversely proportional to tape thickness, tape length can be increased by a factor of 3/2, which means 50 per cent more playing time on a reel. Hence one can obtain 48 minutes of playing time from a 7-in. reel of tape operated in one direction at 7.5 ips: or 96 minutes in both directions.

Double-play tape uses a base in the region of 0.5 mil; including the coating, total thickness is about 1 mil, which is one-half that of standard tape. Thus one

* 280 Twin Lane E., Wantagh, N. Y.

TABLE I
TAPE PLAYING TIME
7-in. reel operated in one direction

TAPE SPEED ips	KIND OF TAPE		
	STANDARD-PLAY	LONG-PLAY	DOUBLE-PLAY
1.875	2 hrs., 8 min.	3 hrs., 12 min.	4 hrs., 16 min.
3.75	1 hr., 4 min.	1 hr., 36 min.	2 hrs., 8 min.
7.5	32 min.	48 min.	1 hr., 4 min.
15	16 min.	24 min.	32 min.

NOTES: A — For a 10-in. reel, double all playing times
B — For a 5-in. reel, divide all playing times by two
C — For a tape operated in two directions, double all playing times

can double the amount of tape on a reel and thereby double the playing time.

Table 1 shows the playing time obtainable from a 7-in. reel of each kind of tape operated in one direction at the various speeds encountered in home use —15, 7.5, 3.75, and 1.875 ips. For standard-play tape, the time ranges from a minimum of 16 minutes at 15 ips to a maximum of 2 hours and 8 minutes at 1.875 ips. For long-play tape the range corresponding is from 24 minutes to 3 hours and 12 minutes. For double-play tape the range is from 32 minutes to 4 hours and 16 minutes. If the tape is operated in two directions, all the above figures are doubled; then the minimum playing time would be 32 minutes at 15 ips for standard-play tape, and the maximum would be 8 hours and 32 minutes at 1.875 ips for double-play tape.

It may be noted at this point that the three kinds of tape are often identified in terms of their (approximate) base thickness. Hence the expression "1½ mil tape" signifies standard-play tape; "1 mil tape," long-play; "½ mil tape," double-play.

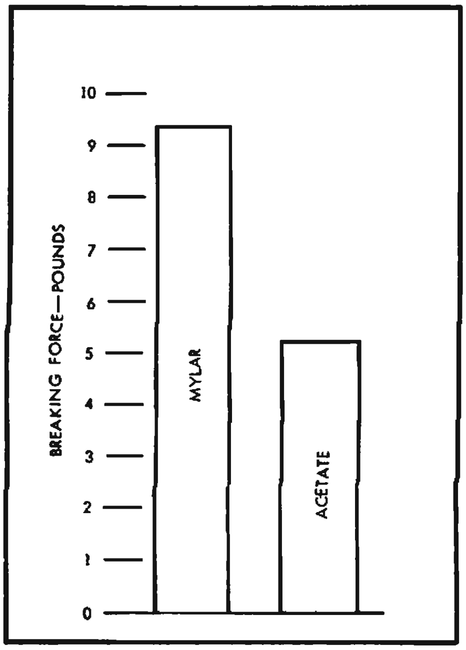

Fig. 1. Comparison of breaking forces for acetate and Mylar (DuPont brand of polyester film). (Orr Industries.)

In going to a thinner base in order to permit more playing time per reel, there is of course a sacrifice in tape strength, of which more will be said later. Hence the long-play and double-play tapes may be more easily subject to breakage or to elongation, which entails distortion. On the other hand, improvements in base materials, particularly the use of polyester film—Dupont's Mylar—instead of cellulose acetate as a base has made it possible for the long-play and double-play tapes to hold up satisfactorily. In addition, improvements in tape machines with respect to their handling of tape has made it possible to get away with weaker tapes.

Another disadvantage of the thinner tapes is increased print-through, because the thinner base offers less of a barrier to this magnetic phenomenon. To indicate how much the print-through increases as the base thickness is decreased, one may refer to the specifications for Reeves Soundcraft tape. Soundcraft long-play tape is stated to have 3 db more print-through than its standard-play tape, while its double-play tape is stated to have 6 db more print-through.

On the other hand, the thinner tapes have advantages as well as disadvantages. One benefit is that a thin tape tends to contour better to the shape of the head, which means intimate contact with the gap and therefore better high-frequency response. Moreover, the thinner coating tends to emphasize the high frequencies. The reason is that the lower frequencies penetrate the tape more deeply than the high ones do, so that reducing the thickness of the coating reduces the low frequencies more than the high ones; in relative terms, the high frequencies are emphasized. Altogether, the thinner tapes may be superior in frequency response at the very high end to the extent of 2 or 3 db or more.

Another advantage of the thinner tapes is that improved contact between the tape and the head tends to reduce dropouts, namely sudden, brief drops in sound level.

Tape Strength

Although at one time tapes with a paper base were quite common, these have virtually disappeared, and today the backing is either cellulose acetate or Mylar. In distinguishing between tapes as to strength, one is essentially distinguishing between those having an acetate base and those having the stronger Mylar as a base.

Tape undergoes various stresses as it is shuttled back and forth by the transport. During normal operation the tape is under tension due to the opposing pressures of the supply and takeup reels. There are other stresses during rapid wind and rewind, particularly when shifting quickly between these two modes of operation. And there are the stresses of quick starts and stops. Tape must be able to endure all these without breaking or stretching.

In addition to considerations of breaking and stretching, the term strength has to take into account the extent to which the tape is impervious to humidity, temperature, and age.

The differences between Mylar and acetate are indicated by data supplied by Orr Industries for its 1.5 mil tapes with each type of base. The force required to break the tape (*Fig.* 1) is 9.4 pounds for Mylar compared with 5.3 pounds for acetate. Mylar can be stretched 150 per cent before breaking, whereas acetate can be stretched only 25 per cent before breaking (*Fig.* 2). A force of 5.6 pounds is required to stretch the tape 10 per cent in the case of Mylar, and of 4.5 pounds in the case of acetate (*Fig.* 3). Thus, although Mylar may seem more susceptible to stretching (it can be stretched 150 per cent before breaking), actually a greater force is required to produce a given amount of

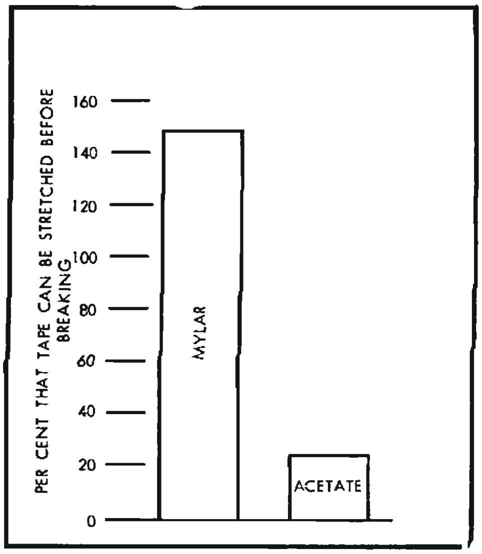

Fig. 2. Ability of 1.5-mil tape to stretch without breaking. (Orr Industries.)

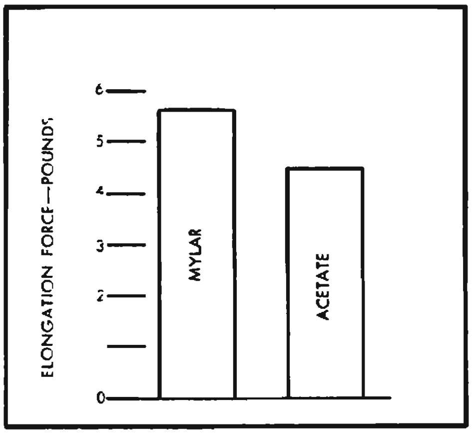

Fig. 3. Force required to elongate 1.5-mil tape ten per cent. (Orr Industries.)

elongation than in the case of acetate. Moreover, Mylar recovers to a greater degree from stretching, as indicated by data on "residual elongation." This refers to the elongation after the tape has been stretched for a given period of time and then allowed to recover for another given period of time. Orr Industries indicates that residual elongation is but 0.75 per cent for the 1.5 mil Mylar compared with 2.75 per cent for the 1.5 mil acetate (*Fig.* 5).

Mylar is less subject to expansion due to heat and humidity. The Orr data indicate that Mylar expands to the extent of 2 parts in 100,000 per degree Fahrenheit, compared with 3 parts in 100,000 for acetate. In brief, acetate expands 50 per cent more under the same conditions of change in temperature. With respect to effects of humidity, Mylar has a very distinct edge over acetate. It expands to the extent of 1.1 parts in 100,000, compared with 15 parts in 100,000 for acetate.

The greater resistance of Mylar to the effects of humidity and temperature changes means it is more proof than acetate against the ravages of age. Hence Mylar does not tend to chip, crack, become brittle, dry out, stretch or shrink. In sum, the individual desiring a tape that can best withstand the stresses of normal tape recorder operation and the effects of age should consider investing in a tape with a Mylar base. On the other hand, if cost is an important consideration, the savings obtained through the purchase of acetate tape may outweigh other considerations. Where an individual uses the same reel of tape over and over, a premium for Mylar tape may be of no consequence. But if he uses many reels of tape (for example in building up a library of recordings), the savings afforded by acetate tape can be considerable. Still, if this library is to be preserved for a substantial period of time, a minimum of deterioration requires a base of polyester film.

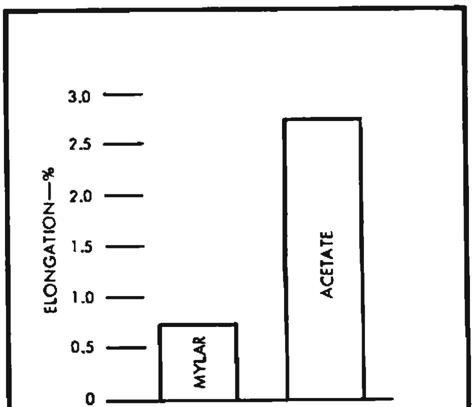

Fig. 4. Residual elongation of 1.5-mil tape. (Orr Industries.)

High-Output Tape

At the time of writing, to the best of the author's knowledge, only one company—Minnestota Mining and Manufacturing Company (3M)—offered the so-called high-output tape, which enables one to record a substantially higher signal level at a given amount of distortion than in the case of conventional tape. The increase in recorded signal level is claimed to be about 7.5 db (133 per cent).

This is achieved in part through the type of oxide used as a coating and in part through a coating of extra thickness. Whereas the 3M Company employs a 0.55-mil coating for its regular tape, the coating is 0.65 mil for the high-output tape. An increase in output level takes place in two ways: (1) For the same amount of signal (magnetic field) applied to each kind of tape, a higher recorded level is obtained on the high-output tape. (2) For the same amount of distortion, the high-output tape can accept a greater applied signal than regular tape.

In exchange for higher output, there is some sacrifice in *relative* high-frequency response. That is, high-frequency output relative to low-frequency output is about 3.5 db lower for high-output tape than for regular tape. It also appears that high-output tape is more subject to print-through because of the greater intensity of the magnetic field recorded on it.

It may well be that the individual possessing a tape recorder of mediocre quality in terms of hum and noise will find that the increase of about 7.5 db in signal-to-noise ratio possible through the use of high-output tape is a considerable blessing, well worth the slight drop in high-frequency response and the possible increase in audible print-through. But the individual owning a fine tape machine that generates virtually no audible hum and noise may find that the increase in output signal does not outweigh the other consequences of using high-output tape.

Low Print-Through Tape

Print-through (the transfer of the signal on a layer of tape to the adjacent layers, resulting in "pre-echo" and "post-echo") increases with recording level and with age of the recorded tape. To the extent that the recordist seeks to impress as much signal as possible on the tape without incurring appreciable distortion, thereby maximizing the signal-to-noise ratio, the print-through problem increases. Even though print-through may not be initially apparent—that is, when playing back the tape immediately or a few hours or days after the tape has been recorded—it may become apparent after a substantial period of storage. *Figure* 5 shows how print-through increases with time.

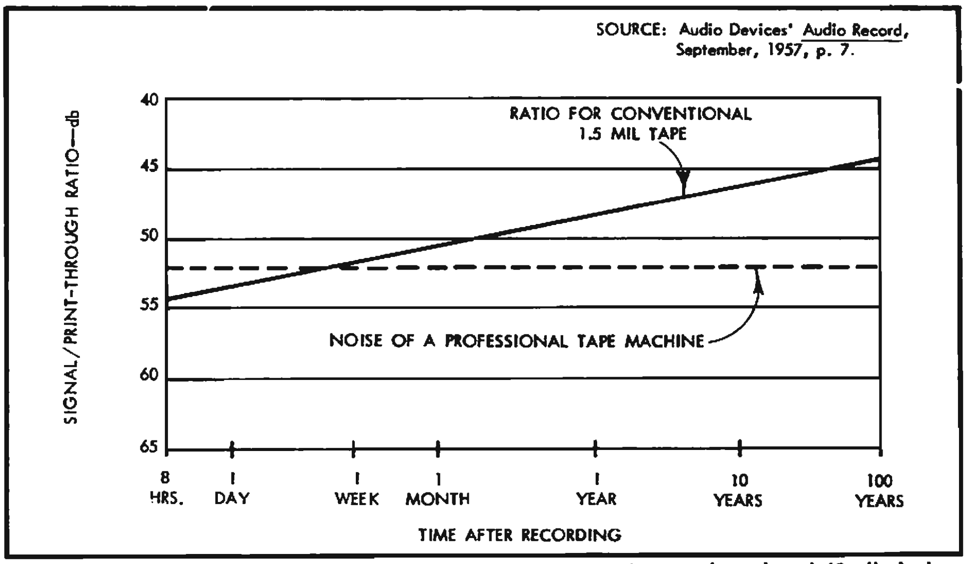

Fig. 5. Ratio of audio signal to print-through at normal recording level (8 db below 3 per cent harmonic distortion level).

The data in *Fig.* 5 are for a 1000-cps signal recorded on conventional tape at a normal level (8 db below the level producing 3 per cent harmonic distortion). The figure also shows the noise level of a tape recorder of professional quality. It may be seen that about one week after recording, print-through has reached the point where it is at the same level as tape-machine noise and therefore is

TABLE II
TYPES OF TAPE SOLD BY FOUR COMPANIES

Based on data supplied to the author by the manufacturing companies

BRAND OF TAPE ⟶	AUDIOTAPE	IRISH	SOUNDCRAFT	SCOTCH
STANDARD-PLAY				
Acetate:				
Conventional	X	X X X	X X	X
High Output				X
Low Print				X
Mylar:				
Conventional	X	X	X	X
High Output				X
Low Print	X			X
LONG-PLAY				
Acetate	X	X		X
Mylar	X	X	X	X
DOUBLE-PLAY				
Mylar	X	X X	X X	X

no longer obscured by such noise. Print-through continues to increase so that it becomes more and more audible.

On the other hand, if tape machine noise were greater than portrayed in *Fig.* 5, it would take longer for print-through to become noticeable. Therefore it is to be realized that the print-through problem concerns users of high-quality tape machines more than users of medium- and low-quality ones.

To reduce or eliminate the print-through problem, two tape manufacturers—Audio Devices and the 3M Company—have introduced tapes with especially low susceptibility to print-through, about 8 db less than for regular tape. *Figure* 6 shows the extent to which Audio Devices' "Master Audiotape" reduces print-through. According to the company's estimate, it would require over 100 years before print-through approached the level of tape machine noise, based on a normal recording level (8 db below 3 per cent harmonic distortion).

Low print-through is achieved by a combination of a special oxide, a relatively thick base, and a relatively thin magnetic coating. The thin coating limits the extent of the recorded magnetic field, while the thick base acts as a barrier between this field and the adjacent layers of tape. The output level of this kind of tape compares with that of regular tape, while high-frequency response is somewhat better (because of the thin coating, for reasons previously explained).

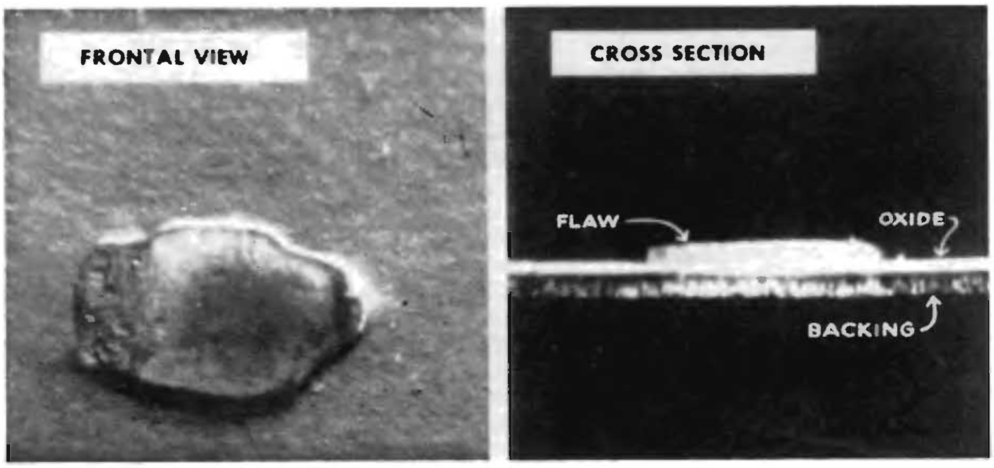

Fig. 7. Photomicrograph of an oxide flake imbedded in the coating of magnetic tape.

Comparison of Some Well-Known Brands

Without getting into the question of the merits of one brand of tape compared with another, Table 2 simply seeks to classify the varieties of tape offered by four prominent manufacturers. Their tapes are classified first by playing time, next by base material (cellulose acetate or Mylar), and then according to high-output and low print-through characteristics. An "X" indicates that the manufacturer produces a particular class of tape. Two or more "X's" indicate that the manufacturer produces a corresponding number of tapes within a particular category; there are quality differences among these tapes of the kind to be described in the next section.

Table 2 shows that no two manufacturers make exactly the same assortment of tapes. They all make tape of standard playing time in both the acetate and Mylar versions. Also, they all make the long-play and double-play tapes with polyester base. Three of the four make the long-play tape available in an acetate base for extra economy. None makes double-play tape with an acetate base because this would be too fragile.

Only one company produces a high-output tape. This is available only in standard-play tape, with either an acetate or Mylar base.

Two produce a low print-through tape. In each case this is available only in standard-play tape; in one instance the base is acetate or Mylar while in the other it is Mylar only.

In several instances a company has more than one tape in a given class. For example, Orr Industries offers three kinds of standard-play acetate tape, at varying prices and of varying quality. Two of the companies offer two kinds of double-play tape; in each instance the principal difference lies in the use of a superior form of Mylar with extra strength. Again, one pays more for the extra strength.

(Continued on page 86)

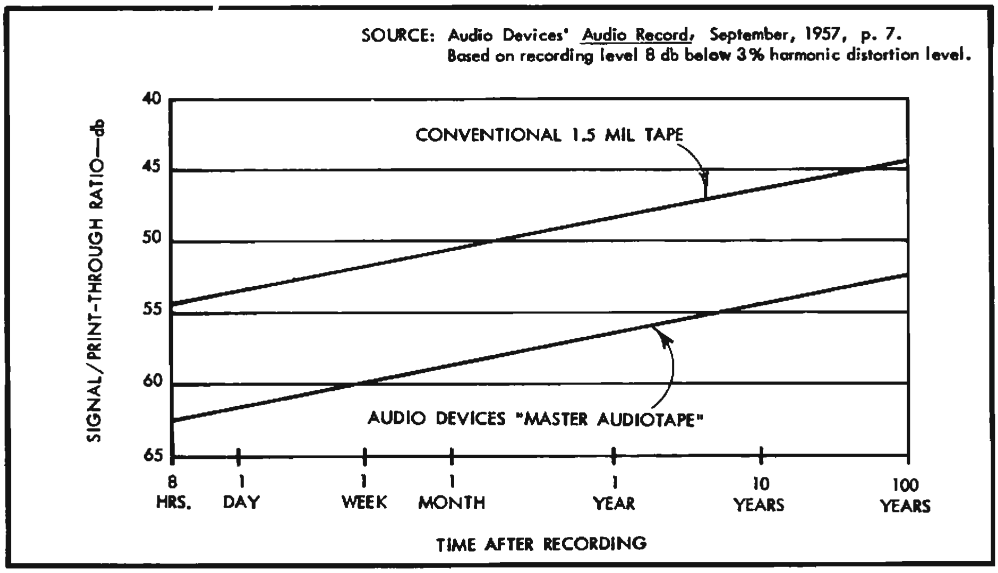

Fig. 6. Reduction in print-through obtained through use of a low-print tape.

(*from page 38*)

Other Qualities

Playing time, type of base material, output level, and print-through characteristics are far from a complete description of tape properties. Additional properties of importance to the user are:

1. *Uniformity of Magnetic Coating.* An uneven magnetic coating—in terms of thickness or distribution of magnetic particles—produces what is called modulation noise. When an audio signal is recorded on the tape, the amplitude of the signal varies in accordance with the irregularity of the coating. This variation is in effect a noise signal added to the

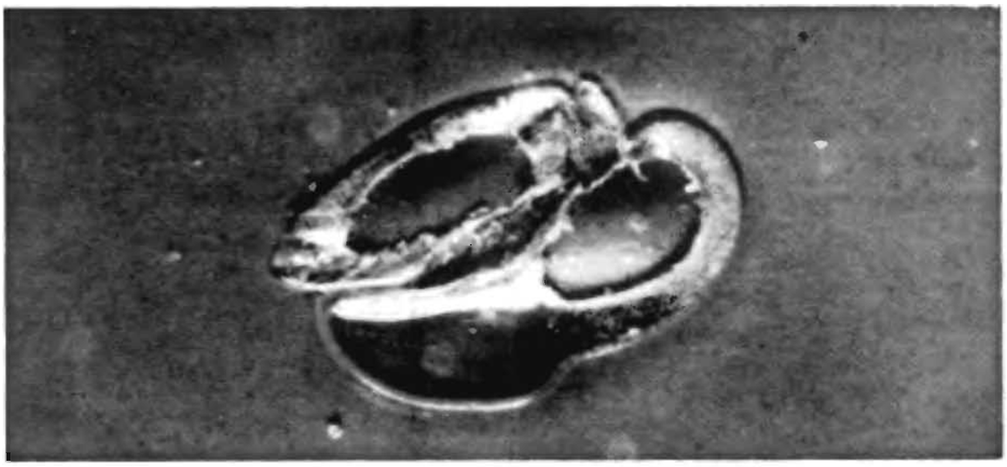

Fig. 8. Photomicrograph of an irregularity in the backing of magnetic tape.
Minnesota Mining & Mfg. Co.

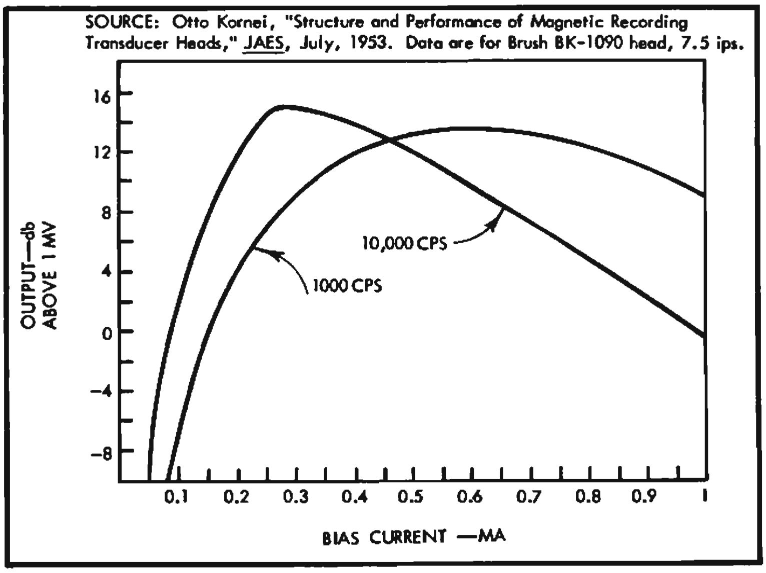

Fig. 9. Variation of output with bias current.

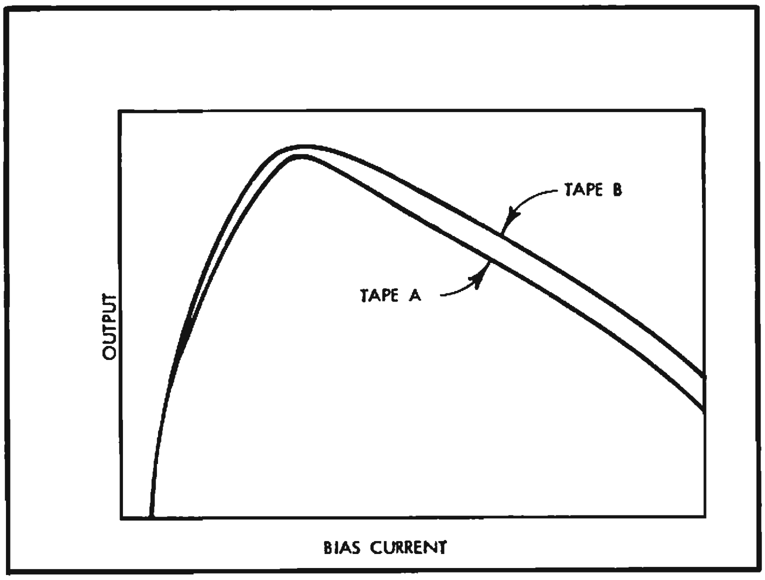

Fig. 10. Output vs. bias current for different tapes; 10,000 cps recorded at 7.5 cps.

audio signal. The louder the audio signal, the greater the modulation noise.

2. *Uniformity of Base Thickness.* If the base varies in thickness, the oxide deposited on the base tends to vary correspondingly, resulting in modulation noise.

3. *Absence of Foreign Materials.* Foreign matter impregnated in the tape coating causes the magnetic coating (tape oxide) to be separated from the head, resulting in signal attenuation (dropouts). *Figure* 7 portrays the nature of such tape flaws. The higher the frequency, the more severe is the dropout. *Figure* 8 shows an irregularity in the tape backing, which has similar consequences.

4. *Smoothness of the Magnetic Coating.* Tapes of good quality are polished and impregnated with a lubricant in order to minimize friction between the tape and the heads, guides, and so on. Excessive friction between the tape and the heads can produce audible squeal. Even though squeal is not heard, friction can cause excessively rapid head wear and a high order of flutter and frequency-modulation noise, evident as a grainy quality in the reproduced sound.

5. *Limpness.* The nature of the materials used in the coating and the base must be such as to impart a limp quality to the tape, permitting it to conform intimately to the contour of the heads and thereby maximize high-frequency response; as previously stated, treble response is severely reduced by extremely slight separation of the tape from the head.

6. *Good Frequency-Response Characteristics.* In part this refers to the output that can be obtained at high frequencies relative to low frequencies. It also refers to the variation of frequency response with bias current. At any given frequency, response rises at first with an increase in bias, but eventually falls as bias is further increased; this is shown in *Fig.* 9. The higher the frequency, the earlier and the sharper the drop in response with increasing bias, as may be seen in *Fig.* 9, which presents data for 1000 and 10,000 cps recorded at 7.5 ips. However, for some tapes the shape of the curve is not as sharp as for other tapes, as may be seen in *Fig.* 10, which compares the output *vs.* bias curve at 10,000 cps for two brands of tape. Curve B in *Fig.* 10 has a broader plateau; the significance of this is that the value of bias current becomes less critical. A slight departure from correct bias current—because of changes in line voltage, insufficient warmup time, aging of components, and so on—will have less effect upon output and frequency response in the case of curve B, which represents the tape with the broader plateau.

7. *Stability.* It is desirable that tape characteristics remain, as nearly as possible, the same within the reel and from reel to reel. Thus it is often important, as in professional recording for the purpose of making commercial tapes or phonograph records, that for a given signal level *applied* to the tape the amount of signal *recorded* on the tape remain virtually constant throughout the reel and from one reel to the next one. Similarly, it is desirable that the tape's frequency response, distortion, and noise characteristics remain as stable as possible.

The degree to which the above tape characteristics concern the recordist—whether it is worthwhile paying something extra for the very best in tapes—depends upon his standards of performance and upon the particular tape machine he is using. The higher his standards and/or the quality of the machine, the more apt one is to find a difference among brands or among grades of tape within the same brand.

The recordist whose requirements are critical should experiment with various brands of tape. Depending upon what factors are of particular importance to him, he may find that one brand suits his needs better than another. The brand best suited for one recordist may not necessarily be so for another. In the case of the less critical recordist, the differences among brands may be minor or completely non-apparent. Æ

Improving the Signal-to-Noise Ratio

HERMAN BURSTEIN*

If the performance you are getting from your tape recorder does not come up to the standards you would like, one or more of the suggestions offered may improve your lot. Most of the ideas are fairly simple, but collectively they could make even a poor machine satisfactory.

In Two Parts—Part I

THE CRITERIA of a tape recorder's quality may be placed on two broad categories: (1) mechanical motion (wow, flutter, timing accuracy); and (2) electrical performance. The latter comprises three basic criteria: frequency response, distortion, and signal-to-noise ratio. Of these three, signal-to-noise ratio is at least as important as the other two and is probably the characteristic most noticeable to the average user. It takes no particular skill and very little time to ascertain whether a given tape machine is excellent, average, or mediocre with respect to keeping noise and hum at a suitably low level with respect to the audio signal.

The problem of obtaining a high signal-to-noise ratio is generally greatest in playback, because the playback head delivers a very tiny audio signal. Hence it is difficult to keep noise and hum produced by the playback amplifier sufficiently below the signal as to be inaudible. In recording, one generally deals with a higher order of audio signal, unless using a microphone of quite low sensitivity, so that maintaining a high signal-to-noise ratio is less of a problem.

Obviously, one should pay careful attention to what a tape machine's specifications have to say about signal-to-noise ratio. Based on a recording level that produces 3 per cent harmonic distortion, a high-quality machine will provide a signal-to-noise ratio of at least 55 db in playback. If the rating is on the basis of 2 per cent harmonic distortion, one can subtract about 3 db, so that a ratio of at least 52 db is called for. If the reference level is 1 per cent harmonic distortion, one can subtract another 3 db.

Since specifications and actual performance can differ, one should, if possible, check the performance of the machine one intends to buy. A quick way to check is to play a commercially recorded tape recorded at a level low enough to result in clear sound. The noise and hum produced by the tape machine should be of about the same order, relative to the audio signal, as encountered in the case of an FM tuner or when playing a phonograph record; at least this is so for a high-quality machine. At the worst, the noise from the tape machine should be only slightly greater relative to the audio signal than in the case of a tuner or phonograph.

Another way to check the playback noise of a tape recorder is to play a blank tape. If the dominant noise is tape hiss—as can be determined by listening to the tape amplifier with the tape still and with the tape in motion—it is safe to say that the tape machine has a very good signal-to-noise ratio in playback.

It is also advisable to check the noise in recording. One should record from a high-level source, as from an FM tuner, and, more important, from a microphone. Of course, the greater the sensitivity of the microphone (the higher its output for a given sound level), the higher will be the ratio of audio signal to noise produced by the record amplifier. One should attempt to use a microphone of average sensitivity, about −55 db/microbar, at fairly close range. In playback, the signal-to-noise ratio should be roughly comparable with that from a tuner or phonograph.

A tape machine that originally had a very good signal-to-noise ratio may fail in this respect with age. Or the ratio may not have been as good as desired to begin with. In either case, there are various factors to be considered and measures that may be taken by the person wishing to improve the signal-to-noise ratio of his tape machine. Some of the measures to be discussed require a slight amount of technical ability, about the same order of ability acquired by those who have assembled an amplifier kit, which many thousands with no knowledge of electronics have done. Hence it does not seem amiss in these pages, intended especially for the layman, to deal with a few simple procedures requiring one to go inside the tape amplifier. Moreover, many readers probably can count among their friends at least one with sufficient knowledge of electronics to assist them.

Tube Selection

Tubes of a given type tend to vary from one to another because of manufacturing tolerances. Two of the respects in which they vary is the amount of noise and hum produced. In these respects, the most important tube is that in the first stage, because its noise and hum are amplified in all successive stages. Hence, to assure minimum noise and hum one should obtain three or four of the tube type employed in the first stage and try these successively. Since

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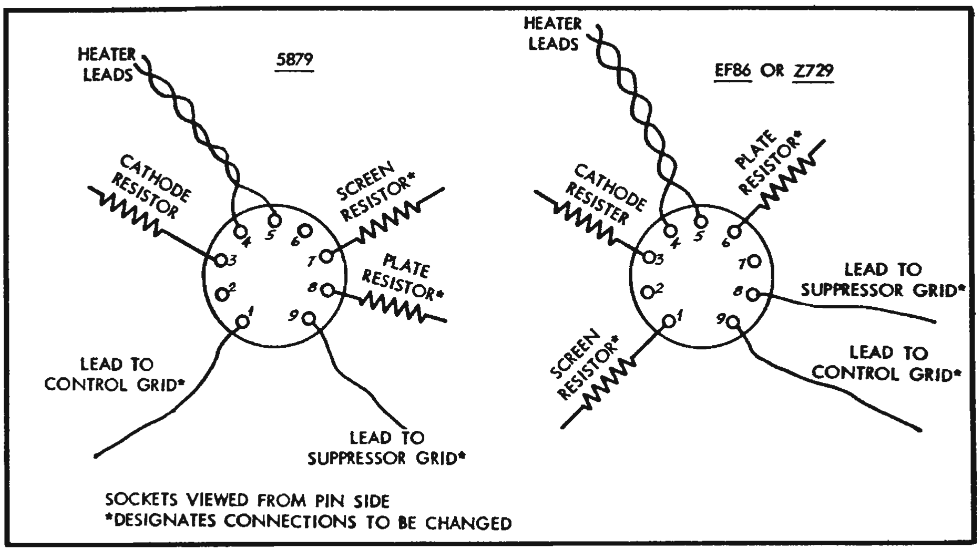


Fig. 1. Rewiring a tube socket to accommodate an EF86 or Z729.

tape amplifiers frequently employ the same tube for more than one stage, the extra tubes generally are not wasted.

The differences among tubes of a given type can be fairly profound—as much as 10 db in the case of hum if the heater is operated on a.c. The ear will generally serve as an adequate instrument for rejecting a tube that is poor so far as hum is concerned. Similarly, the ear can enable one to detect tubes that are outright hissy. However, if one is trying to select among tubes that differ by only a few decibels in their hum and noise characteristics, it is probably necessary to measure the output of the tape amplifier with a sensitive vacuum tube voltmeter to be sure which is the best tube. Moreover, when one does find the tube that is best, there is a possible pitfall: be sure that this tube provides as much amplification as the others, because sometimes a tube apparently has less noise and hum only because it provides less amplification. The amplification of a tube can be determined on a suitable tube tester; or, using a test tape with a constant signal, one can measure the output of the tape amplifier when various tubes are used in the first stage.

Tube Substitution

A significant reduction of noise and/or hum often can be achieved by substituting a simialr type of tube for the one originally used in the first stage. Perhaps the best example is the use of an ECC83 in place of a 12AX7, which is frequently the first stage tube. The ECC83, basically a European tube (although some are made in America), is electrically identical to the 12AX7 but on the average has superior hum and noise characteristics. Recently, American manufacturers have brought out the 7025, also intended to be a superior replacement for the 12AX7. Other high-quality replacements include the American 12AY7 and the Telefunken 12AX7.

Sometimes a pentode is employed in the first stage. Here the European EF86 has won a position of very high esteem

If the tape amplifier presently has a 5879 (American) or Z729 (European) in the first stage, these can be replaced by the EF86. No rewiring of the tube socket is required, if the original tube was a Z729, whereas rewiring is necessary if it was a 5879. *Figure* 1 shows the pin locations of the 5879 and EF86. If the latter is substituted, it is necessary to rewire the resistors, capacitors and other elements of the first stage so that they are connected to the proper tube pins; this task requires probably no more than 10 to 15 minutes.

Reversing the Power Plug

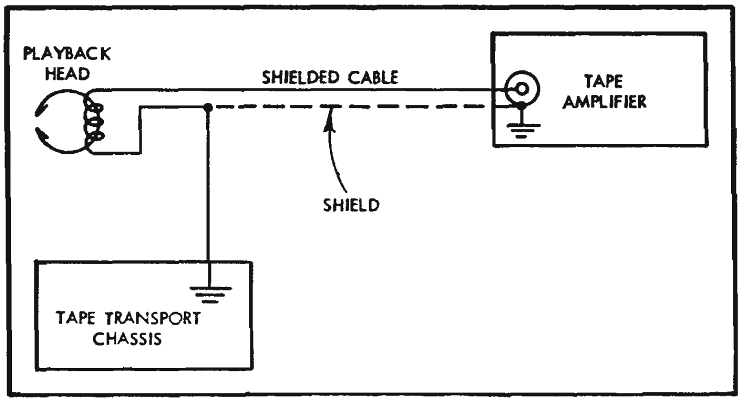
As simple a measure as reversing the power plug in the house outlet can sometimes reduce hum by a significant amount. If the tape machine has a potentiometer for canceling hum, this should be readjusted for each position of the power plug.

Tube Shields

The first stage tube is, or at least should be, shielded against hum pickup. The shield should make firm contact with ground through the socket mounting or through other means. If the shield has worked loose, it may be ineffective; in fact, the loose shield may serve to increase hum above the level produced by an unshielded tube. For maximum protection against hum pickup, shields are available that are made of special material, such as the "Co-Netic" shields made by Perfection Mica Co. If one has access to Mumetal, it is possible to fashion a tube shield out of this material.

Demagnetization of Tubes and Shields

A magnetized tube, particularly in the first stage of the tape amplifier, can be a source of hum. While tube selection can eliminate this difficulty, another technique is to demagnetize the offending tube by means of a bulk eraser, being careful not to bring the tube too close to the eraser lest the powerful pull of



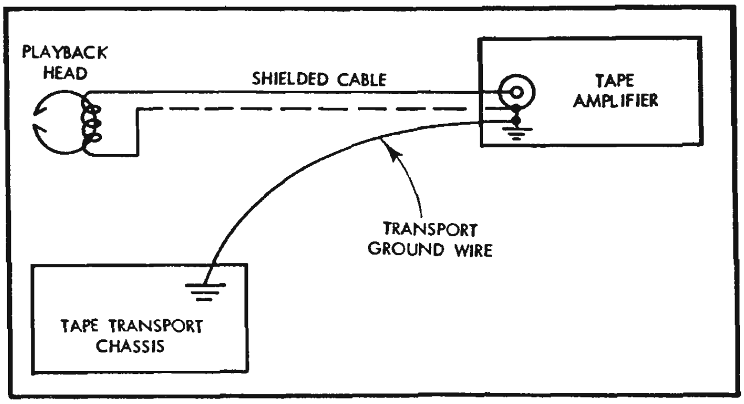


Fig. 2 (left). A method of grounding the tape transport that may produce hum. Fig. 3 (right). Method of grounding the tape transport that generally results in the least hum.

the latter dislodge any of the tube elements.

Similarly, a magnetized tube shield can be an unsuspected source of hum. Here tube substitution would probably do no good. However, the shield can be demagnetized by a bulk eraser.

Routing of the Cable from the Playback Head

If one employs only a tape transport, using the control amplifier instead of a tape amplifier to provide the necessary amplification and equalization of the signal produced by the playback head, careful attention must be paid to the routing of the cable between the head and the control amplifier. If the cable passes close to leads carrying a.c., or close to a power transformer, motor rectifier tube, or other component where a.c. is present. it may pick up enough hum to become audible; keep in mind that the signal presented by the cable to the amplifier undergoes tremendous amplification, which is greatest in the vicinity of the hum frequencies, namely 60 and 120 cps.

Sometimes one may run into a cross-talk problem if the cable from the playback head runs close to another cable carrying a high-level signal, for example from a tuner. Should the tuner be on, then one might hear a radio program along with the signal from the tape machine.

Grounding of the Tape Transport

Whether an integral tape amplifier or one's control amplifier is used in conjunction with the tape transport mechanism, there should be a good ground connection between the transport chassis and the amplifier. An increased hum level may result if one depends upon the cable shield for a ground between the transport and the amplifier, as in *Fig.* 2. Instead, a separate, heavy ground connection is generally preferable, as in *Fig.* 3. In this case the playback head should not be grounded to the transport chassis, for this will provide a second path to the amplifier ground point; the dual paths, as shown in *Fig.* 4, constitute a ground loop, which picks up hum.

Lead Dress

Sometimes excessive hum is due to improper routing of the hot lead from the playback head to the grid of the first tube in the playback amplifier; the reference here is not to the routing of the cable between the head and the amplifier but to the path followed by the hot lead of the cable *inside* the amplifier. If the lead comes too close to a.c. heater wiring, other wires containing a.c., a power transformer, or the like, it may pick up enough hum to be audible. Moving the lead by a slight amount, perhaps a small fraction of an inch, can sometimes result in appreciable improvement. Of course this is a trial-and-error procedure, but the results on occasion justify the effort expended. It may be preferable to leave the lead from the playback head in its original position and try moving other leads, such as a.c. heater wiring.

Installing a Hum-Bucking Pot

To cancel hum due to operation of the tube filaments on a.c. instead of d.c., a number of inexpensive tape machines connect a tap at the center of the heater winding on the power transformer to ground, as in *Fig.* 5. While this does greatly reduce hum, seldom does it minimize hum as fully as possible. A superior procedure, illustrated in *Fig.* 6, is to place a potentiometer across the heater winding, with the arm of the pot connected to ground, and adjust the arm for minimum hum. Usually it is not difficult to find room to install a pot. Drilling a hole takes a few moments, and the necessary connections are few and very simple. Note in *Fig.* 6 that the center lead of the heater winding is removed from ground.

Construction of a D.C. Heater Supply

The more ambitious home constructor may wish to install a source of d.c. for the heaters of the tubes in his tape amplifier, replacing the original a.c. supply. *Figure* 7 shows a suitable d.c. supply, which can be used for the first stage or first two stages of the tape amplifier. It will do no harm to continue operating the remaining stages on a.c.

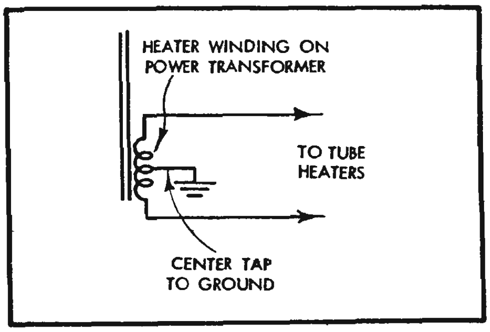


Fig. 5. Method of hum cancellation often used in moderate-priced tape recorders.

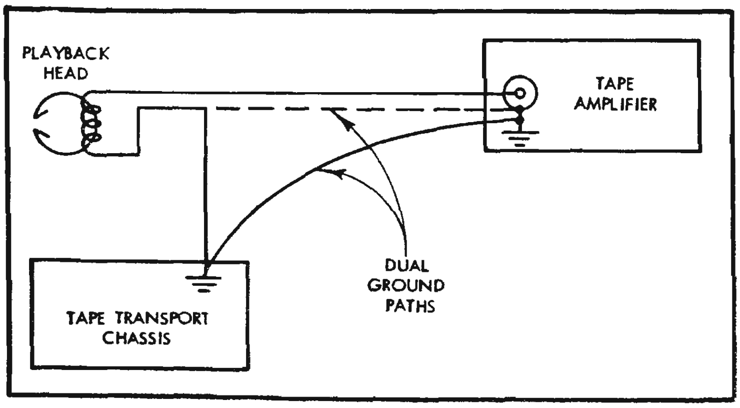


Fig. 4. Formation of a ground loop.

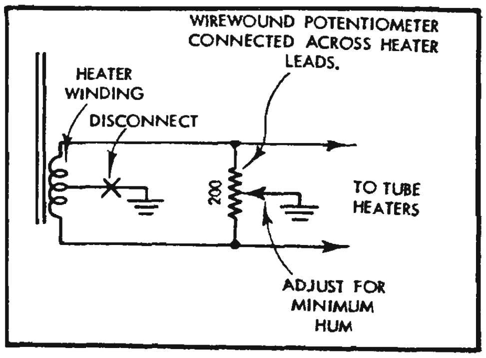


Fig. 6. Installation of a hum-bucking potentiometer.

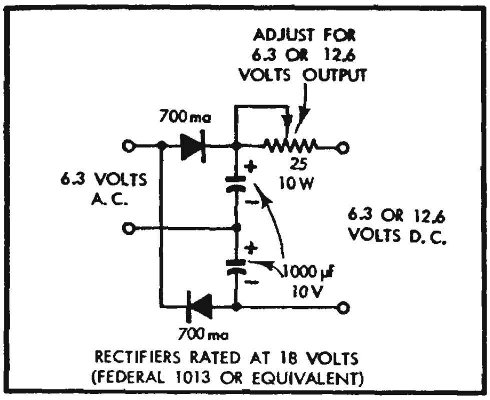


Fig. 7. Construction of a d.c. heater supply.

Re-Orienting the Motor

The transport motor is a major potential source of hum picked up by the playback head. Sometimes hum can be significantly reduced by re-orienting the motor, namely rotating it a number of degrees about its axis. If the motor is suspended by three bolts, as is frequently the case, one can then turn the motor 120 deg. in either direction.

Relocation of the On-Off Switch

It is common practice for the on-off switch to be located on the gain control of the tape amplifier. However, the 60-cps current at the switch creates a minute hum field which may be picked up in significant quantity by the gain control, particularly if the control is situated at an early stage of the amplifier so that there is a great deal of subsequent amplification. Installation of a separate toggle switch, as in *Fig.* 8, for turning the tape machine on and off can sometimes effect a worthwhile reduction in hum.

Adding Filter Capacitance

If hum is of the 120-cps variety—that is, pitched an octave higher than the 60-cps hum encountered when bringing a screwdriver or similar metal object close to the playback head—it may be possible to reduce it by adding filter capacitance, as in *Fig.* 9. Additional capacitance of about 30 to 60 µf will usually prove effective, particularly when added to the filtering stage closest to the rectifier stage. Of course, it is possible that a filter capacitor has opened or greatly decreased in value, so that one is really replacing rather than adding capacitance.

Shielding the Playback Head

Normally the playback head is partly encased in Mumetal or other special material designed to prevent hum pickup. In addition, the better tape machines surround the heads with a h[illegible] shield with an aperture just wide enough to permit the tape to pass through. In other machines, a piece of Mumetal or other shielding material is sometimes mounted on the pressure pad holder so that when the pad is brought against the playback head the shield guards the face of the head against hum. As illustrated in *Fig.* 10, it may be possible for the handy audiofan to rig up, something of this kind himself. In addition to Mumetal, one can use Co-Netic for shielding purposes, or even a piece of silicon steel, fashioned from a transformer lamination. Co-Netic is very easily cut and bent to the desired shape.

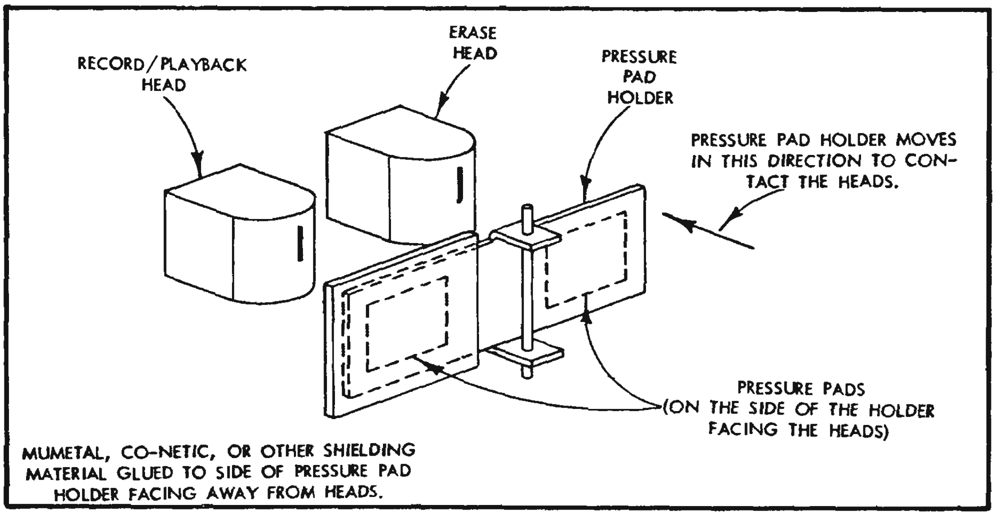


Fig. 10. Example of how a playback head can be shielded against hum pickup.

Shielding Other Components

If hum is picked up by the playback head from the motor or power trans-

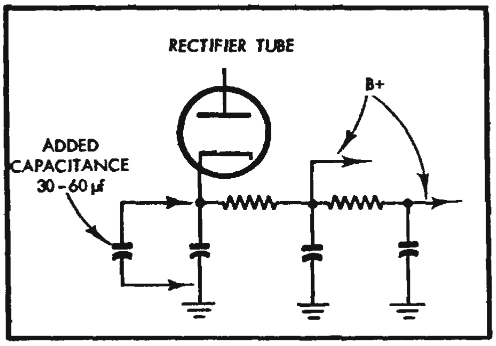


Fig. 9. Adding filter capacitance to reduce hum.

former, it may be possible to remedy this situation by placing a shield around the offending component. The shield may consist of Mumetal, Co-Netic, silicon steel laminations, or copper.

Defective Playback Head

Most playback heads have two windings which, when properly connected, serve to cancel hum picked up by the head. However, if for any reason the windings are electrically unequal, cancellation will be imperfect, resulting in an increase in hum. Shorted turns in one of the windings could produce such a situation. Replacement of the head is then called for.

Substitution of Resistors

One of the principal sources of noise—high-pitched rushing or hissy sounds as contrasted with hum—is the garden-variety resistor. Imperfect contact of the particles within the ordinary molded carbon resistor results in internal arcing and consequent noise voltages. It is generally possible to obtain a substantial reduction in noise by substituting resistors with low-noise properties in the first stage or perhaps in the first two stages of the tape amplifier. Such resistors are called for in the plate, cathode, and grid circuits.

Wire-wound resistors have the best noise characteristics, but they are also quite expensive, as much as $2 or $3 apiece. Moreover, since they consist of a number of turns of wire, they may be-

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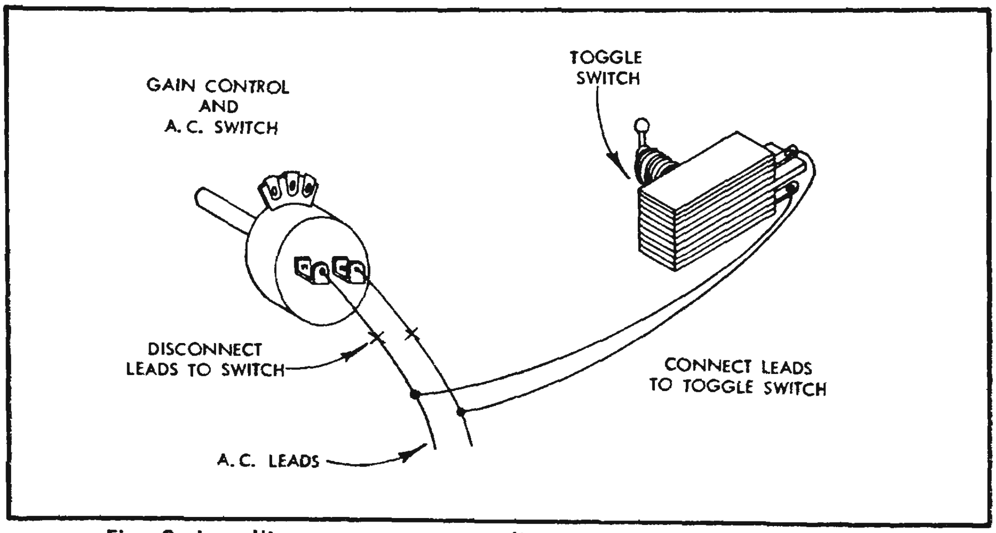

Fig. 8. Installing a separate on-off switch to reduce hum pickup.

TAPE GUIDE

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have as an inductance and pick up a slight amount of hum, which is then subject to great amplification in the tape amplifier.

The audiofan seeking to improve the signal-to-noise ratio of his tape recorder will find that some of the deposited-metal-film resistors are virtually as good as the wirewound and cost only about half as much or less. The Davohm 850 series is recommended.

Deposited-carbon resistors have frequently been acclaimed for their low-noise properties, but one must be very careful here. It is true that certain deposited-carbon resistors, particularly of foreign manufacture, produce extremely little noise and are suitable for high-grade applications. Unfortunately, a number of other deposited-carbon resistors are hardly, if at all, better than the conventional molded ones that sell for a few cents apiece.

Accordingly, the audiofan desiring a high degree of assurance of good results will invest in wirewound or deposited metal resistors. If he uses the latter, he should bear in mind that occasionally a poor one will get into the lot, and that failure to get improved results might be due to such an occasional mischance.

If a cost of $1 or more for each resistor seems too high for a not necessarily successful attempt to reduce noise, the audiofan may try another expedient, which is to use conventional resistors of relatively large wattage rating. Whereas ½-watt resistors are usually employed in the plate, cathode, and grid circuits, he may substitute 2-watt resistors and thereby possibly achieve an appreciable noise reduction.

Demagnetization of Heads

Noise is often due to heads that have become magnetized. Every time that a recorded tape is played back, noise is added to it by the magnetized heads. In recording, a magnetized head presents a d.c. component that serves to bring up modulation noise on the tape, which is due to unevenness in the coating and/or base.

The tape heads can be quickly and easily demagnetized by exposing them for a few seconds to a head demagnetizer. One should follow a regular course of preventive maintenance in this regard, namely demagnetizing the heads after about every 8 hours of use. Once noise has been added to a tape by a magnetized head, the noise cannot be removed without also removing the audio signal.

(TO BE CONTINUED)

Improving the Signal-to-Noise Ratio

HERMAN BURSTEIN*

If the performance you are getting from your tape recorder does not come up to the standards you would like, one or more of the suggestions offered may improve your lot. Most of the ideas are fairly simple, but collectively they could make even a poor machine satisfactory.

In Two Parts—Part II

Noise can be defined as any *undesired* audible signal. Accordingly, noise may be said to include signal remaining on the tape from a previous recording, left there because of imperfect operation of the erase head. The erase head can be ineffective for mechanical or electrical reasons.

Figure 11 shows how erasure may be incomplete for mechanical reasons. The erase head gap is not positioned vertically in proper relationship to the gap of the record head, so that not all of the recorded track is subject to erasure.

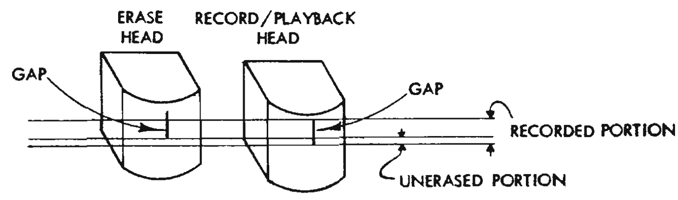

Fig. 11. Incomplete erasure due to improper vertical positioning of the heads.

Electrical reasons for imperfect erasure include the following: (1) poor design; (2) shorted turns; (3) a weak oscillator or other defective components so that insufficient current reaches the head; (4) an oscillator frequency that is too high; the higher the frequency, the less efficient the head tends to be.

Finding and curing electrical malfunction is largely a task for the electronic technician rather than for the audiofan. However, if the cause is a weak oscillator tube, the audiofan can at least ascertain this and cure the difficulty by substituting another tube. Frequently it is also within his power to adjust the oscillator frequency by turning the screw protruding from the container that holds the oscillator transformer. Turning the screw so that it recedes into the container will lower the frequency and possibly increase the effectiveness of the erase head by a significant amount. However, when the bias frequency changes, this may also result in a change in the amount of bias current that reaches the record head, thereby affecting frequency response and distortion. Unless one possesses the necessary equipment for measuring bias current through the record head, it is best not to tamper with the oscillator adjustment.

If one cannot obtain satisfactory results from the erase head, one may have recourse to a bulk eraser. The drawback, however, is that one cannot erase just one track in mono half-track recording or just two tracks in four-track stereo recording; the entire width of the tape is necessarily erased at once.

In connection with imperfect erasure it is necessary to bear in mind that the erase head and oscillator may be functioning properly but that the fault lies in excessive recording level, one that produces a high degree of distortion.

Print-Through

Print-through is a form of noise. The problem may be mitigated or eliminated through one or more of the following approaches: (1) Use of low-print tape, made by at least two manufacturers. (2) Use of a print-through eraser, described in the article on tape accessories.[1] (3) Reduction of the recording level. (4) Storing recorded tape properly in a cool place and away from the magnetic fields produced by motors, transformers, etc.

Tape Hiss

In the case of a very good tape recorder, where the tape amplifier produces a minimum of noise, it may well be that tape hiss is the dominant noise factor distinguished by the ear. (Although hum may seem dominant on an oscilloscope, high-pitched noise may be the only kind apparent to the ear.) Several courses of action are possible. First, subjecting the tape to a bulk eraser may reduce tape hiss by a significant amount. Second, changing to a different brand of tape or to a higher quality within the same brand might result in noticeable improvement. Third, it is sometimes advisable to look beyond the tape machine,

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[1] November, 1959.

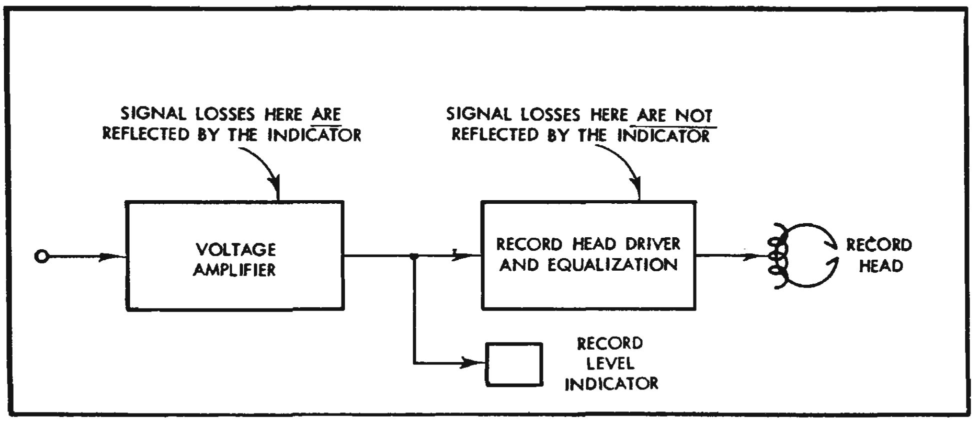

Fig. 12. How the record-level indicator may fail to reflect signal losses.

namely to other components in the audio system; an appreciable departure from smooth frequency response in the control amplifier, power amplifier, or—most likely—the speaker system, will tend to accentuate tape hiss, particularly if there is peakiness in the range of about 3000 to 5000 cps. The course then is to adjust tone controls, filters, or speaker-level controls as best as one can to remove the objectionable peakiness. The audiofan might even want to consider replacing his speakers, or perhaps moving them to a different location in the room if a change in acoustic environment reduces the treble peak.

Departure from NAB (formerly NARTB) equalization may account for inordinate tape hiss. At speeds of 7.5 and 15 ips, a treble droop of 10 db between 1000 and 15,000 cps is called for in playback. In some tape amplifiers, however, substantially less treble droop or quite possibly an appreciable amount of treble boost is encountered, resulting in emphasis of tape hiss. Treble boost in playback is found in tape recorders employing so-called half-and-half equalization, where the boost required to compensate for severe treble recording losses is provided partly in recording and partly in playback, instead of entirely in recording as stipulated by NAB standards.

Tape Squeal

If tape has dried out and lost its lubricant, it may produce an unpleasant squeal as it passes the heads. This squeal can be recorded on the tape, so that even if measures have been taken in playback to avoid squeal, nevertheless the unpleasant sound will be repeated.

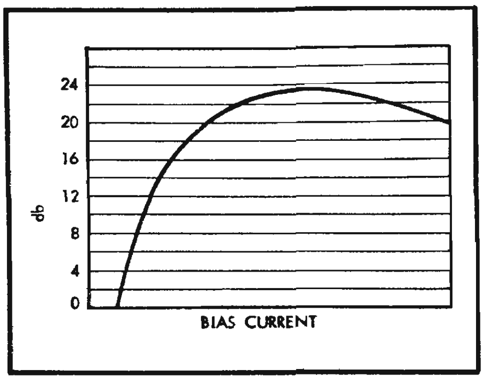

Fig. 13. Variation of output with bias current at 1000 cps at 7.5 ips.

Tape squeal can be avoided by the following measures: (1) purchasing tape of high quality; (2) lubricating the heads, guides, pressure pads, and so on, with substances described in the article on accessories; (3) lubricating the tape with materials described in articles on accessories; (4) replacing worn pressure pads.

Accuracy of the Record-Level Indicator

Thus far in discussing improvement of the signal-to-noise ratio we have dealt with measures to reduce noise. But this is only one side of the coin, and it is also necessary to consider the problem of getting as much audio signal on the tape as possible without running into excessive distortion.

In this connection, accuracy of the record-level indicator is of vital importance. Should the indicator show a high recording level when actually the magnitude of the signal recorded on the tape is small, obviously the result will be a deterioration in the signal-to-noise ratio. Generally, the technically untrained or unequipped individual cannot correct the calibration of the record-level indicator. Usually this is the province of the technician.

However, what the home recordist can do is to make a more or less rough check whether the record-level indicator is operating properly. Thus he can record a tape at what is presumably maximum recording level, listen to its quality in playback, then try successively higher recording levels. If he can make clean recordings with the gain control advanced well beyond the point where the record-level indicator tells him he should have stopped, this *suggests* the possibility that the indicator is miscalibrated. It is necessary to bear in mind that the point at which some persons find distortion to be offensive may be considerably different than the point at which others find distortion intolerable.

Another procedure is to compare the playback level of a commercially recorded tape that sounds clean with the level of a tape the audiofan has recorded himself. If one's own tape appears to have a distinctly lower level in playback, this points to miscalibration of the record-level indicator.

If one is recording at too low a level, however, the fault is not necessarily in the indicator. It may be, as illustrated in *Fig.* 12, that the recording signal undergoes losses at a point in the tape amplifier following the record-level indicator. Thus a weak tube in the stage that drives the record head, or possibly a defective head, may produce such losses. Hence if it appears necessary to adjust the record-level calibration in the upward direction, thus permitting a higher input level, it is advisable to have a check made on the recording stages following the take-off point for the indicator.

Use of High Output Tape

As indicated in the article on Kinds of Tape,[2] one can obtain an increase of nearly 8 db in recorded signal level by using high output tape, although this tends to involve a moderate loss in high-frequency response and may involve greater print-through. However, if one is struggling with a tape machine that produces an undue amount of noise and hum, use of high output tape can produce a very worthwhile improvement in over-all performance.

Bias Current

The amount of bias current fed to the record head is vitally linked to the signal-to-noise ratio that can be attained. The greater the bias current (up to a point), the lower the distortion, so that it becomes possible to put more signal on the tape for a given degree of distortion, resulting in a higher signal-to-noise ratio. But there is a drawback in that increased bias current causes a loss in high-frequency response. Hence, particularly at speeds of 7.5 ips and less, it is not feasible to increase bias current as much as one would like to do from the viewpoint of maximizing signal-to-noise ratio.

On the other hand, if bias current is reduced appreciably below its normal value, this will directly reduce the signal level recorded on the tape, apart from considerations of distortion. *Figure* 13 shows how recorded level varies with bias current at a frequency of 1000 cps at 7.5 ips. Furthermore, distortion will increase with reduced bias, so that it is necessary to reduce the signal level further to avoid an increase in distortion. Hence it is important to avoid letting bias current fall much below its normal value, although sometimes this is an expedient employed to maintain frequency

[2] December, 1959.

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TAPE GUIDE

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response at the extreme end of the treble range.

Some tape recorders provide a ready means of adjusting bias current. In others, adjustment involves replacement of components, which is a matter for the service technician. In either case, adjustment requires certain equipment, which, together with the procedure, will be discussed in a later article.

Not only is the magnitude of the bias current important to maintaining signal-to-noise ratio, but so is the purity of the waveform. Ideally it should be a sine wave. Departure from a symmetrical waveform results in noise. Checking the waveform requires an oscilloscope. Some tape recorders incorporate an adjustment for balancing the bias current oscillator to minimize asymmetry and noise. Usually in such a machine there are separate record and playback heads, so that one can make the adjustment while listening to playback of a tape which is put through the recording process but without putting an audio signal on tape. Æ

Maintaining Frequency Response in Recorders

If you are not satisfied with the frequency response you are getting from your tape recorder, this article may tell you why and it may also tell you what you can do to correct it.

HERMAN BURSTEIN*

IN TWO PARTS—PART I

As is usual in discussions of frequency response, we are concerned with response that is smooth, full at the low end, and maintained substantially to the upper limit of the audio range—at least to 12,000 cps and preferably to 15,000. In the case of tape recording and playback, the greatest problem is to preserve the high frequencies, and so it is this particular aspect of the subject of frequency response that will figure largest in the following discussion. Preservation of the upper audio range is relatively more difficult in tape recorders than with other elements of an audio system. This is particularly true when tape speed is 7.5 ips or less.

While a tape machine may perform well initially, its frequency response may deteriorate with age, use, or mishap. Maintaining frequency response as it should and can be requires care on the owner's part. On the other hand, frequency response of a tape machine may not initially be all that it should be. Alert to this possibility, the purchaser is in a position to reject a particular model or a particular unit of a given model that cannot deliver a desired standard of performance. Through comprehension of the various factors that enter into a tape machine's frequency response, the prospective purchaser or the present owner maximizes his chances of obtaining suitable frequency response.

At the same time, one obtains very little for nothing in the electronics realm. When maximizing frequency response in the sense of extending the range to 15,000 cps or so, sacrifices may be required with respect to distortion, signal-to-noise ratio, or both. Accordingly, the problem may be of finding a suitable compromise among conflicting considerations, namely treble response, distortion, and noise.

Tape Speed

Frequency response is closely associated with tape speed. In recording, certain losses occur that increase with frequency, and the slower the tape speed the greater the loss at any given frequency. In playback, there are losses associated with the playback head that similarly increase with frequency and

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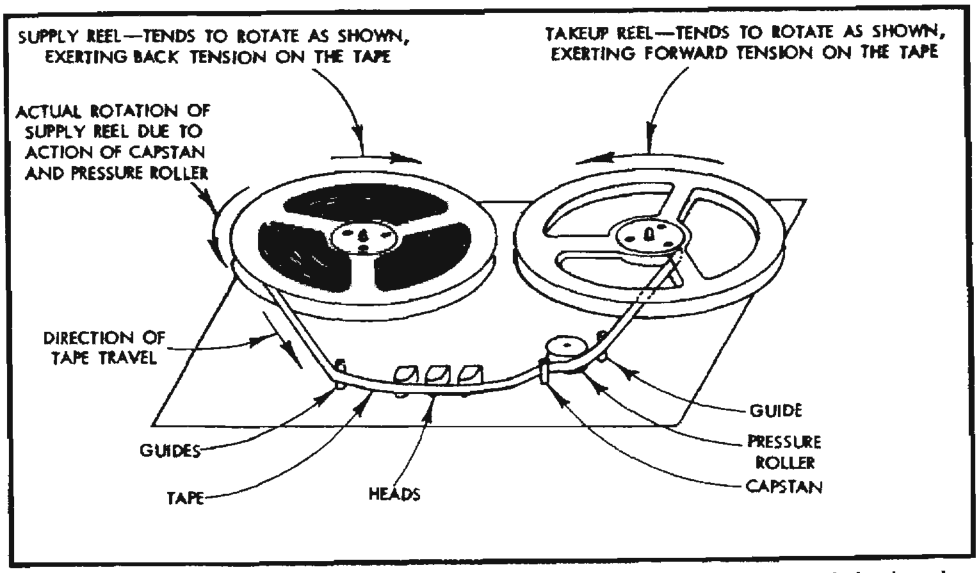

Fig. 2. Use of tape tension to achieve firm contact between the tape and the heads.

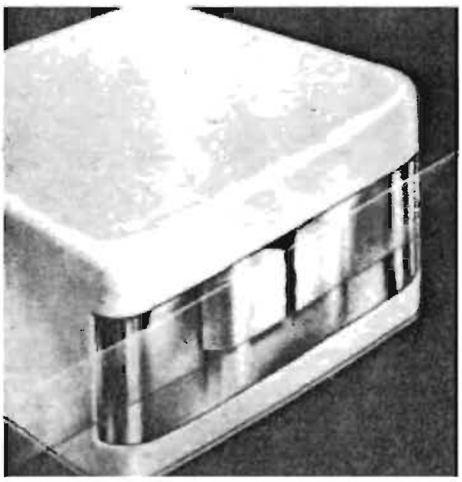

Fig. 1. Erosion of the gap of a tape head due to abrasive action of the tape.

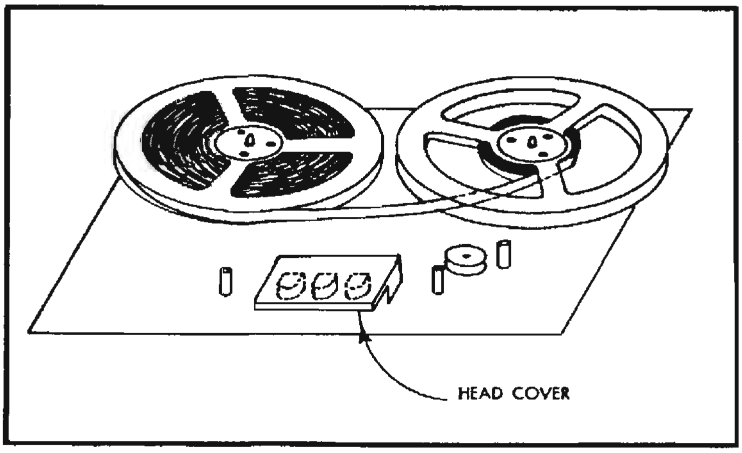

Fig. 3. Winding the tape directly from reel to reel can reduce head wear during rewinding.

become more acute as speed is reduced.

For some time it has been possible at 7.5 ips to achieve results consistent with the concept of high fidelity—namely, response extending to 15,000 cps or at least to 12,000 cps. Quite recently, it has appeared feasible to reach out to 12,000 cps or better at 3.75 ips as well, and even 1.875 speed has been gaining a place in home use. While it may be adequate for moderate quality reproduction of the voice and some forms of background music, as yet this last speed is incapable of high fidelity performance. Nevertheless, considering the constant progress that takes place in the tape art, it is conceivable that not too many years from now it will be possible to have high fidelity at 1.875 ips. At such a time the 15/16 ips might then play the role of a secondary speed where results of only moderate quality are required. Returning to the present, it may be said that nothing less than 3.75 ips is compatible with a first-rate home music system, and that to be really sure of good results it is still necessary to operate at 7.5 ips.

Head Losses

Head losses are of two kinds: (1) frequency-dependent and (2) speed-dependent. Frequency-dependent losses have nothing to do with tape speed and are electrical in nature. Specifically, they are eddy current and hysteresis losses, which have to do with the construction and material of the head, and they increase with frequency. In modern heads, these losses are very small within the audio range and may be left out of the following discussion.

The principal head loss is due to gap width of the playback head and varies inversely with tape speed. The narrower the gap, the higher is the maximum frequency that the head is capable of reproducing. As a rough approximation, one can use the following formula to estimate the upper response limit of a playback head:

$$f = \frac{S}{2G},$$

where f is the approximate upper frequency limit in cps, S is tape speed in inches per second, and G is the physical gap of the head, in inches, as specified by the manufacturer.

To illustrate, assume that tape speed is 7.5 ips and the gap of the playback head is .00025 in. according to its manufacturer. Then $f = 7.5/(2 \times .00025) = 15,000$ cps. At a tape speed of 3.75 ips, however, the upper response limit for this head would be only about 7500 cps. It is therefore apparent why gaps considerably narrower than .00025 in.—heretofore widely used in machines of

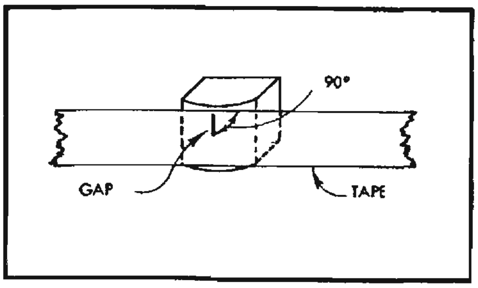

Fig. 4. Meaning of azimuth alignment.

good quality—are required if extended response is to be achieved at 3.75 ips. The newer heads have gaps in the vicinity of .0001 in. Inserting this value into the above formula, with speed at 3.75 ips, the upper response limit appears to be 18,750 cps. This is the feasible response *in playback*. In recording there are very serious losses that make it difficult to maintain this kind of treble response at 3.75 ips.

It should be noted that the physical gap is not the same thing as the magnetic gap. The above formula takes into account that in a well-made head the magnetic gap tends to be about 10 per cent wider than the physical gap. However, in a poorly constructed head, where the gap is not extremely straight and sharply defined, the magnetic gap may be considerably more than 10 per cent in excess of the physical gap, so that the upper response limit is correspondingly lower than indicated by the formula. As a result, it is quite possible that a head with an advertised gap of .0001 in. may afford better treble response than another head with an advertised gap of .00009 in., or 90 microinches.

While a head may initially have a gap sufficiently narrow and linear for good treble response at the speed in use, the gap may widen due to head wear and thereby cause a noticeable fall-off in reproduction of high frequencies. The rapidity and extent of head wear depend upon the following factors:

1. *Head Construction.* Laminated heads generally wear better than nonlaminated ones.

2. *Smoothness of the Tape.* Depending upon the brand and quality of the tape and therefore upon the extent to which the tape has been lubricated and polished, head wear will vary. *Figure* 1 *suggests* the nature of head wear due to abrasive action of the tape; for visual clarity, the effect of abrasive action has been exaggerated in the drawing.

3. *Pressure of the Tape Against the Head.* For good treble response it is important that the tape and the heads maintain intimate contact. However, the pressure required for close contact results in friction as the tape moves past the heads. Thus the pressure should be just enough to maintain good contact and no more. The reels *tend* to pull the tape in opposite directions, so that the tape is held more or less taut against the heads, as illustrated in *Fig.* 2. This is the scheme generally employed in semi-professional and professional tape recorders to achieve close contact between the tape and the heads. Excessive pressure can result from excessive back tension exerted on the tape by the supply and takeup reels. There is usually provision for adjusting back tension.

Most home machines rely on pressure pads to obtain firm contact between the tape and the heads, because the path followed by the tape does not assure such contact. If the pressure pad holder is improperly adjusted, head wear may take place at an excessive rate.

On the other hand, it sometimes happens that a brand new head will offer improved treble response after a moderate period of wear. What happens is that the head wears down to the point where the gap is narrowest. But eventually the gap will begin to widen with increased wear and high-frequency response will deteriorate.

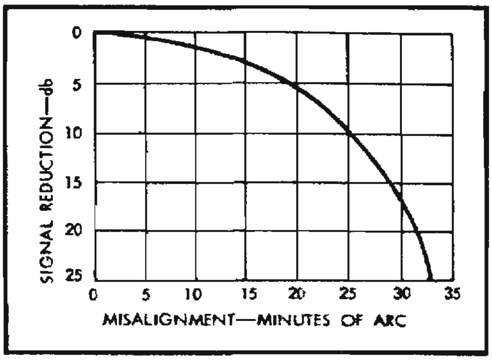

Fig. 5. Effect of azimuth misalignment upon response at 7500 and 750 cps at 7.5 ips.

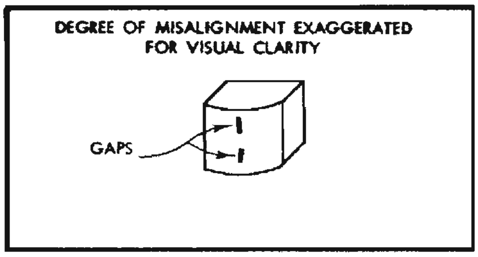

Fig. 6. Relative misalignment of the gaps of a stereo head.

4. *Manner in Which the Tape is Wound and Rewound.* When the tape is wound rapidly in the forward or reverse direction, some machines "lift" the tape slightly away from the heads. Many, possibly most, home machines fail to space the tape away from the heads during rapid wind and rewind, thereby causing appreciable head wear, perhaps more wear than occurs during normal record and playback. To avoid this unnecessary head wear, it is generally possible to wind the tape directly from one reel to the other without going past the heads, as illustrated in *Fig.* 3. It is merely necessary to lift the tape out of its normal path past the heads—usually a guide slot—and allow it to take the shortest path between reels. The possible disadvantage of this procedure is that the tape may not be wound as smoothly as if it were following its normal path.

5. *Care of the Heads.* Head wear can be minimized through suitable care, which includes regular cleaning of the heads to remove accumulated tape oxide, and the application of lubricants to minimize friction between the heads and the tape. Once the gap of the playback head has widened appreciably, nothing can be done except to replace the head, which is a good deal more costly than preventive maintenance. The gap does not have to widen very much before the head becomes unable to reproduce high frequencies. To illustrate, a gap of .0001 in. permits response to 18,750 cps at the 3.75 ips speed. If the gap widens by just one ten-thousandth of an inch, the upper response limit is reduced to 9375 cps at 3.75 ips, which is too low for high-fidelity purposes.

Azimuth Alignment

Improper azimuth alignment is one of the most common reasons for inadequate treble response. The gap of a correctly aligned head forms an angle of exactly 90 deg. with respect to the length of the tape, as shown in *Fig.* 4. If the angle differs from 90 deg., however slightly, there are losses that increase with frequency. For a given degree of misalignment, the loss at any given frequency goes up as tape speed is reduced. On the other hand, the narrower the track—for example a half-track recording compared with a full-track one, or a four-track stereo tape compared with a two-track one—the proportionately smaller are the azimuth losses.

The foregoing assumes that different heads are used for recording and playback. If the same head is employed for both modes of operation, the azimuth error cancels. However, when a misaligned record-playback head is used to play a commercial recorded tape, then treble response of course suffers.

Figure 5 shows how severe the losses due to azimuth misalignment can be. The curve shows the drop in response at 7500 cps for various degrees of misalignment when a half-track head is employed at 7.5 ips. It may be seen that misalignment of one half of 1 degree reduces output by more than 17 db

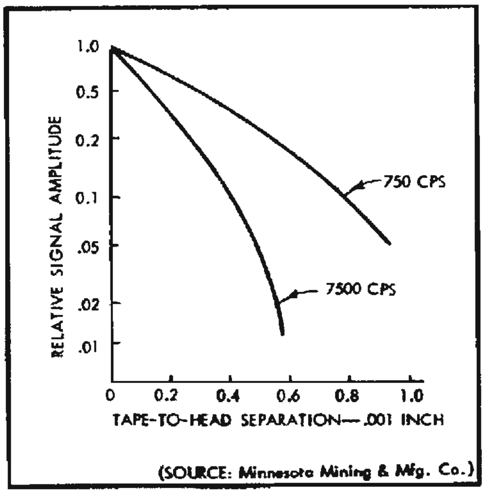

Fig. 7. Separation losses when recording frequencies of 7500 and 750 cps at 7.5 ips.

under the stated conditions. If the frequency were greater than 7500 cps, if the tape speed were reduced, or if the track width were increased, the losses would be greater.

In the case of stereo heads, it is important to realize that it is possible for the two gaps to be out of alignment with each other, as shown in *Fig.* 6. Ideally, the two gaps should be in a perfectly straight line. If they are not, as sometimes happens, then it is not possible to obtain correct azimuth alignment on both tracks. Aligning one gap automatically throws the other gap out of alignment. Else one has to find a compromise position where both tracks are equally affected. The best solution is to replace the head, unless the degree of misalignment is so slight as not to cut response more than two or three db at the upper end of the audio range.

As stated before, the effect of azimuth misalignment decreases as track width is reduced. Hence four-track stereo heads have an advantage over two-track heads, because for a given degree of misalignment between gaps the effect upon treble response will be less with four-track heads.

Tape-to-Head Contact

Intimate contact between the tape and the heads is vital to preservation of high-frequency response. Failure of the tape to hug the heads may be due to various factors; inadequate pressure when pressure pads are used; inadequate tension when pads are not used; accumulation of tape oxide on the heads.

Losses due to separation of the tape and the heads can occur in recording as well as playback, although they are generally more severe in playback. *Figure* 7 shows the losses at 7500 cps and 750 cps for various amounts of separation in recording at 7.5 ips. The curve for 7500 cps shows that separation of about 0.4 thousandths of an inch, which can occur due to accumulation of tape oxide on the record head, will reduce the recorded level to about one-tenth of the level in the case of perfect tape-to-head contact—a loss of 20 db. The curve for 750 cps exhibits considerably smaller, but nevertheless substantial, losses. At frequencies above 7500 cps, the losses would be much greater than indicated in *Fig.* 7.

Figure 8 shows the separation losses for a playback head. Here it may be seen that at 7500 cps at a speed of 7.5 ips a separation of 0.1 thousandth of an inch reduces the signal to about one-fourth of its potential level, a loss of 12 db, compared with a loss of about 6 db in recording for the same amount of separation. If the same head is used in recording and playback, and if separation is .0001 in. at both modes, then a loss of 18 db altogether can occur at 7500 cps.

The moral is clear. Heads must be regularly cleaned every few hours to remove tape oxide. To be on the safe side, cleaning should take place before every recording or playback session. In addition, at suitable intervals, the machine should be checked to ascertain that tape tension is correct or that pressure pad holders are properly adjusted.

TO BE CONTINUED

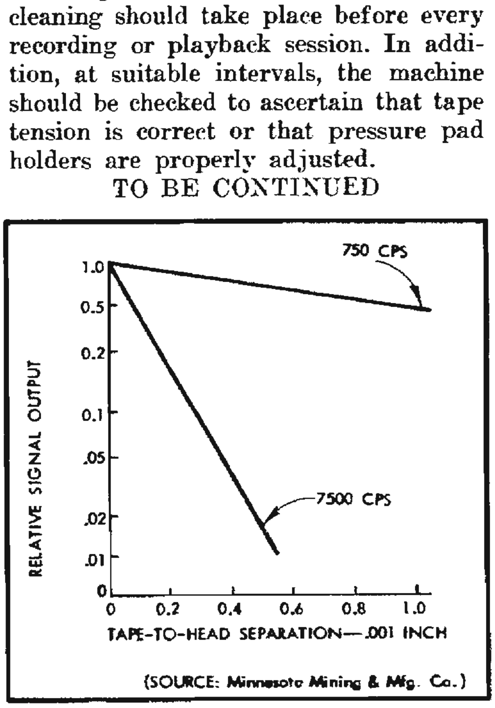

Fig. 8. Separation losses when reproducing frequencies of 7500 and 750 cps at 7.5 ips.

Maintaining Frequency Response in Recorders

If you are not satisfied with the frequency response you are getting from your tape recorder, this article may tell you why and it may also tell you what you can do to correct it.

HERMAN BURSTEIN*

IN TWO PARTS—PART II

The high-frequency bias current fed to the record head, if applied in sufficient quantity, causes the head to behave in the manner of an erase head. This is shown in *Fig.* 9, which represents the variation in output of a record-playback head operated at 7.5 ips at 1000 and at 10,000 cps as bias current is varied in magnitude. At first the recorded level goes up with an increase in bias, but eventually the level goes down as bias is increased further. By comparing the curves for 1000 and 10,000 cps, it may be seen that the level goes down faster at higher frequencies.

The effect of bias current upon frequency response can be observed more directly in *Fig.* 10, which shows the unequalized response of a record-playback head at two values of bias current.

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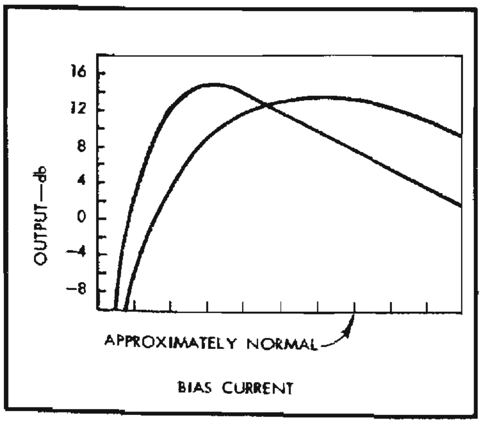


Fig. 9. Variation of output with bias current at 10,000 and 1000 cps for a record-playback head operating at 7.5 ips.

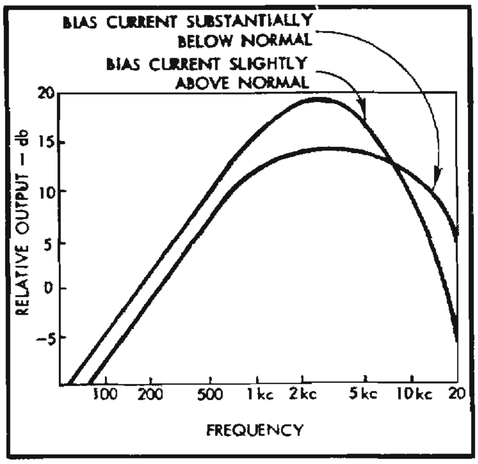


Fig. 10. Unequalized response of a record-playback head at 7.5 ips for two different values of bias current.

At the larger bias current, the drop in treble response is considerably greater.

The reason that the higher frequencies are more susceptible to an increase in bias current is that such frequencies, when recorded on the tape, do not penetrate the tape as deeply as do the lower frequencies. Therefore the upper frequencies are more easily erased by the alternating magnetic field due to the bias current in the record head.

One of the simplest measures that can be taken to improve high-frequency response is to decrease bias current or, from a different point of view, to prevent bias current from exceeding the value specified by the manufacturer of the tape machine or of the record head. In fact, in the attempt to maintain full-range response at 3.75 ips, appreciably smaller amounts of bias current are often used than at 7.5 ips.

The better tape machines often contain a control (variable resistor or variable capacitor) that permits one to adjust readily the amount of bias current fed to the record head. In other machines, however, it is necessary to change the value of a component—resistor or capacitor—in the circuit supplying the bias. In either case, measuring bias current and adjusting it is a procedure requiring technical competence and suitable instruments. While the layman is ordinarily not equipped to do this, it is well for him to be aware that deficient treble may be simply due to excessive bias rather than to something which is much more expensive to remedy, such as a playback head with a gap that is too wide. It can happen that one goes to the effort and expense of replacing a

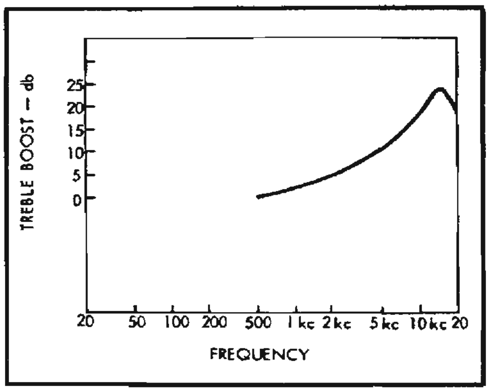


Fig. 11. Typical treble boost employed in recording on a tape machine at 7.5 ips with NAB playback equalization.

playback head, only to find that the fault lay in too much bias.

One must guard against excessive reduction of bias in order to achieve the desired treble response. The penalty for too little bias is excessive distortion. And the increase in distortion is quite sharp as one reduces bias. It can easily happen that in the effort to extend treble response by a relatively moderate amount, say from 12,000 to 15,000 cps, one decreases bias to an extent which results in a severe increase in distortion. More about this in the next article.

One can partly or completely avoid an increase in distortion by reducing the recording level, but then one has less recorded signal on the tape, and this means a reduced signal-to-noise ratio in playback. In sum, the effort to extend frequency response by decreasing bias involves an increase in distortion or a reduction in signal-to-noise ratio or a combination of the two. On the other hand, it is quite possible that for accidental reasons bias current is above the level consistent with reasonably low distortion and a satisfactory signal to noise ratio. In such cases, particularly at speeds of 7.5 ips and less, it is important that bias be reduced to its proper value.

Record Equalization

In recording there are two kinds of magnetic losses which become increasingly severe as frequency rises. One of these has already been described—the loss due to bias current in the record head. The other, known as demagnetization loss, refers to the fact that as frequency goes up the equivalent bar magnets recorded on the tape grow shorter, with the consequence that the opposite poles of each magnet are closer and therefore tend to cancel each other to a greater extent.

Altogether, the recording losses require a great deal of treble boost in order to make it possible to achieve response out of 12,000 cps or beyond. *Figure* 11 indicates how much treble boost is necessary by showing the typical treble equalization for a machine operating at 7.5 ips and designed to yield a relatively flat response when playback equalization conforms to the NAB (formerly NARTB) curve. With such a large amount of boost required, it may be realized that anything which prevents correct operation of the treble boost circuit—a faulty resistor, capacitor, inductor, or other component—can deal a severe blow to treble response.

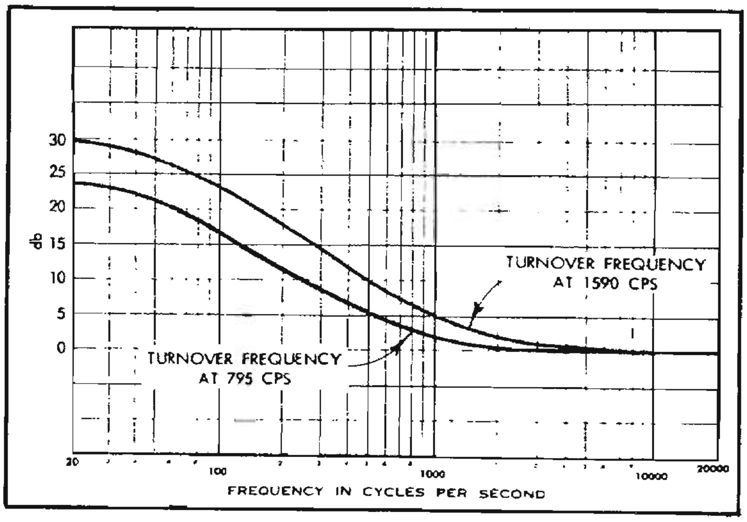


Fig. 13. Playback curves that have been used at 3.75 ips.

In some tape machines, especially the ones of semi-professional and professional quality, the amount of record treble boost can be controlled, within limits. Accordingly, as one adjusts bias current or as one changes to a different kind of tape with different high-frequency characteristics, one can make a compensating change in treble boost. In most home machines, however, there is no such adjustment. Unless precision components have been employed in the treble boost circuit, it may be necessary to replace a component in order to achieve treble response as flat as possible. It sometimes happens that there is a peak in treble response—usually in the region of about 6000 to 10,000 cps—which may be great enough to warrant removal through a change in the equalization circuit. Conversely, treble response may be deficient due to a component that is too far from design value. All in all, variable treble equalization in recording is a desirable feature in a machine to be operated by the audiofan who is meticulous about flat frequency response.

As tape speed is changed, the required amount of treble boost and the frequency (turnover) at which boost commences also change. In many tape machines, treble boost is automatically changed as speed is changed. In others, however, particularly those with external tape amplifiers, the equalization change must be made manually. It is quite easy for the recordist to forget to make this change when shifting speeds. If he has been recording at 7.5 ips and then goes to 3.75 ips without changing equalization, the result is deficient treble in the signal recorded on the tape. If he goes from 3.75 ips to 7.5 ips without changing equalization, then excessive treble is the result.

Some tape machines contain no provision for changing record equalization when shifting from one speed to another. Accordingly, inadequate frequency response is achieved at one speed or the other, unless a compromise equalization is used, which produces less than the best results at both speeds.

Playback Equalization

Figure 12 shows the NAB equalization that is standard for playback at 15 ips and virtually, though unofficially, standard for 7.5 ips as well. *Figure* 13 shows the two playback equalization curves that have been most commonly employed at 3.75 ips, although some tape machines have used the NAB curve at this speed. At the time of writing it appeared that curve (A) in *Fig.* 13 would become standard for 3.75 ips, although there is no assurance about this. In view of the uncertainties about playback equalization at 3.75 ips, the following discussion will be conducted in terms of the 7.5 ips speed, for which there seems to be little dispute, if any, about playback equalization.

When playing most 7.5-ips commercial recorded tapes, relatively flat response will be obtained if the tape machine provides NAB equalization. But to this day a substantial number of home machines provide different equali-

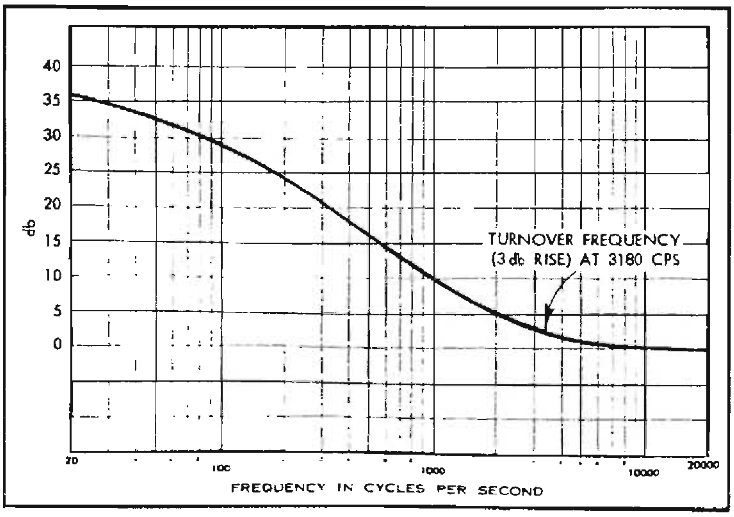


Fig. 12. NAB (formerly NARTB) playback equalization.

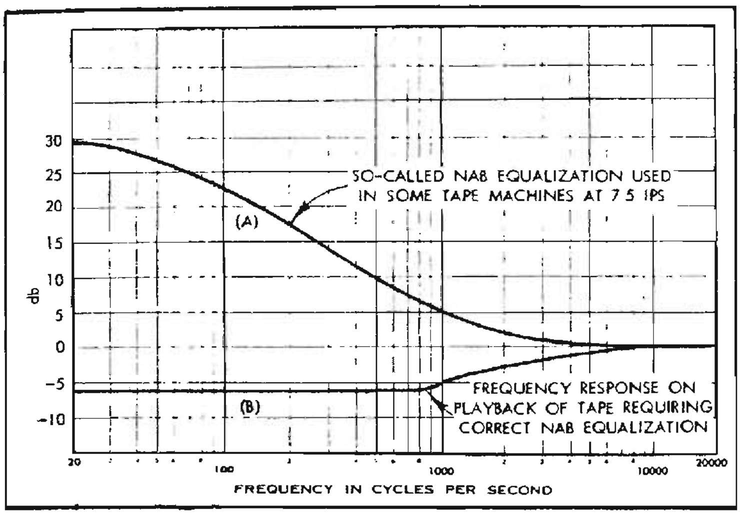


Fig. 14. Effect of improper playback equalization upon frequency response.

zation, typically that if curve (A) in *Fig.* 14, which the manufacturers often call NAB equalization even though it is not. If playback equalization is that of curve (A), the result will be too much treble and not enough bass, as shown by curve (B). This frequency imbalance can be more or less compensated through bass boost and treble cut elsewhere in the audio system.

A certain number of home machines employ so-called half-and-half equalization, as in *Fig.* 15, whereby half the required bass boost is supplied in recording and the other half in playback; similarly, treble boost is equally divided between recording and playback. When playing a commercial recorded tape, there will again be a thinness in the bass region, probably to a greater extent than with machines using the playback curve of *Fig.* 14. Moreover, there will be excessive treble, because half-and-half equalization provides treble boost in playback, whereas such boost (except to compensate playback head deficiencies) is not called for under NAB equalization.

All in all, the individual who wishes to play commercial recorded tapes is well-advised to ascertain that the machine he owns or plans to purchase provides accurate NAB equalization. As previously indicated, a number of tape machines or tape amplifiers that advertise NAB equalization fall short of the mark. Some say simply nothing on the subject. If one already has a machine that deviates appreciably from the NAB curve, it is quite simple for a qualified technician to make the necessary circuit change, often requiring replacement of only one component. However, in doing so, one upsets the record-playback frequency response of the machine in question, meaning that the record equalization must also be altered to achieve flat response when playing recordings made on the machine. Since the record equalization curve is a more complex affair than the playback one, it may be time-consuming and expensive to have record equalization adjusted accurately. What one can do instead is to have a switch installed, permitting one to use either the machine's original playback equalization or NAB equalization.

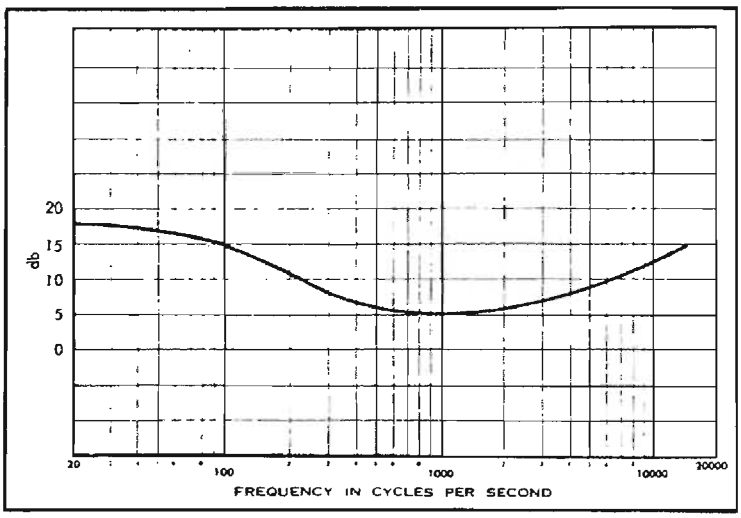


Fig. 15. Example of "half-and-half" equalization employed in some tape recorders for both record and playback.

As with record equalization, the tape machine may or may not provide for automatic change in playback equalization when changing from one speed to another. The comments in the preceding section on this problem apply here as well. However, the problem is less serious in playback. If a tape is played with the wrong equalization, one can correct the error; a new start can be made if desired. But if a tape has been recorded with wrong equalization, circumstances often do not permit the error to be retrieved, as when taping a program off the air.

In a number of tape machines and tape amplifiers, the treble drop above 1000 cps is less than shown in *Figs.* 12 and 13. The reason is to compensate for the falling response of the playback head in the upper range due to gap width. Modern heads, however, with extremely fine gaps on the order of .0001 in., generally do not require such compensation. By and large, the best procedure is to obtain a tape machine or tape amplifier with correct playback equalization and then compensate such deficiencies as may occur by means of the tone controls in the audio system.

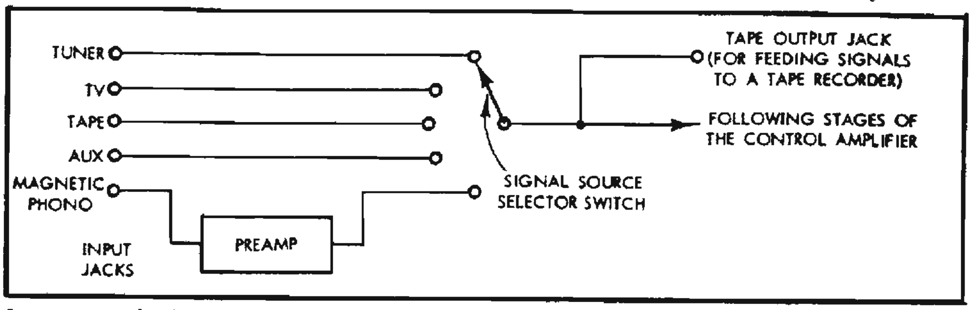


Fig. 16 Method employed in some control amplifiers for feeding incoming signals directly to a tape recorder.

Cables

High-frequency losses can be due to the cables between the tape machine and the rest of the audio system. Let us first consider the cable that carries the signal from the rest of the audio system to the tape machine for recording purposes.

In a number of control amplifiers or integrated amplifiers, the incoming signal is fed directly to the tape recorder, as illustrated in *Fig.* 16. If the signal source has a low impedance—for example, when a tuner has a cathode-follower output—a substantial run of cable between the control amplifier or integrated amplifier and the tape recorder will have no consequential effects upon treble response. On the other hand, if the signal source has a high impedance, then under the arrangement of *Fig.* 16 high-frequency response can be seriously affected by more than two or three feet of cable between the amplifier and the tape recorder.

To prevent the capacitance of the cable between the amplifier and the tape recorder's input jack from having a deleterious effect upon frequency response, some amplifiers feed the tape

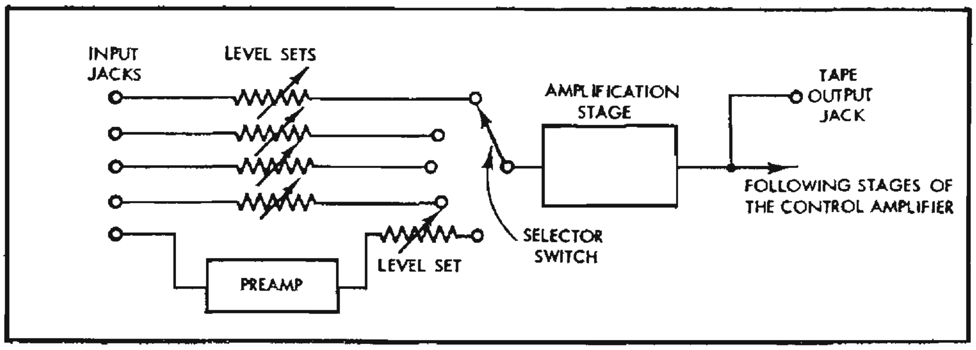


Fig. 17. A second method used in some control amplifiers to feed a tape recorder.

recorder from a cathode follower (or other low-impedance circuit), as in *Fig. 17*. Under these conditions a long cable has no appreciable effect.

The audiofan should ascertain whether the signal presented to the tape recorder comes from a low-impedance source—supplied by either the signal source (tuner, TV, etc.) or the amplifier—or whether it comes from a high-impedance source. In the latter case, he should make special effort to keep the cable to the recorder as short as possible, no more than three feet. and he should use low-capacitance cable.

The same kind of problem exists with respect to the cable leading from the output of the tape machine to the input of the control amplifier or integrated amplifier. Some tape machines have a low-impedance output, so that cable length, within reason, does not matter. But a number of home machines have high-impedance outputs, and here it is necessary to be careful about cable length.

The most serious problem, perhaps, occurs when a cable is run directly from the *tape head* to the control amplifier or integrated amplifier, there to undergo preamplification and equalization. This is the case when the audiofan is interested only in playing tapes, not in recording them, and therefore purchases a tape transport without electronics. In this situation, more than a foot or two of cable can significantly affect treble response.

Kind of Tape Used

Treble response tends to vary somewhat with the brand of tape used and with the kind of tape within the brand. Depending upon the formulation of the magnetic coating on the tape and the thickness of the coating, treble response may vary a few db at the upper end of the audio range.

The thinner tapes tend to have an advantage over standard tapes with respect to treble response. At the middle and low frequencies, the amplitude of the recorded signal increases somewhat with coating thickness. But the higher frequencies, which are recorded closer to the surface of the tape, are less affected by thickness of the coating. Therefore when the tape has a thin coating, as in the case of long-play and double-play tapes, response at low and middle frequencies is reduced in comparison with standard tape. In other words, high-frequency response, relatively speaking, is increased.

The thinner tapes tend to be more limp and therefore conform more easily to the contour of the playback head, assuring close contact between the tape and the head and thereby maximizing treble response. At 15,000 cps at the 3.75 ips speed, differences of as much as 5 db in response have been noted as the result of using thin tapes.

It has been pointed out that an increase in bias current results in a reduction in treble response. However, the extent to which treble response is affected by a slight increase in bias tends to vary somewhat among tapes. In other words, some tapes are less critical than others in terms of setting bias current to the correct operating value.

Location of the Tape Output Jack

In many or most monophonic control amplifiers, the tape-output jack is located ahead of the tone controls so that the setting of the latter has no effect upon the frequency balance of the tape recording. In other ampliers, however, particularly stereophonic ones, the tape-output jack is located *after* the tone contols, and frequency balance of the tape recordings depends upon how one sets the bass and treble controls (and possibly upon the setting of treble and bass filters as well). Quite possibly, excessive or deficient bass or treble in a recorded tape can be traced to the fact that recording did not take place with the tone controls in flat position. Moreover, the error tends to be compounded in playback. To illustrate, assume one normally turns up the bass control to a substantial degree to compensate for speaker and/or room acoustics. Consequently, a good deal of bass boost gets onto the tape. But if the bass control is left untouched in playback, then the repetition of bass boost can become objectionable.

On the other hand, location of the tone controls (and filters) ahead of the tape-output jack has a decided advantage. It permits one to restore frequency balance prior to recording program material on tape. Thus a substantial number of phonograph records have excessive treble in order to impart a false illusion of high fidelity, so-called. By turning down the treble when recording, one comes closer to a tape with natural balance. In addition, reduction of excessive treble tends to reduce distortion in recording. To take another illustration, AM reception is usually deficient in the high frequencies. Accordingly, one can boost the treble to achieve or approach natural frequency balance when recording a tape.

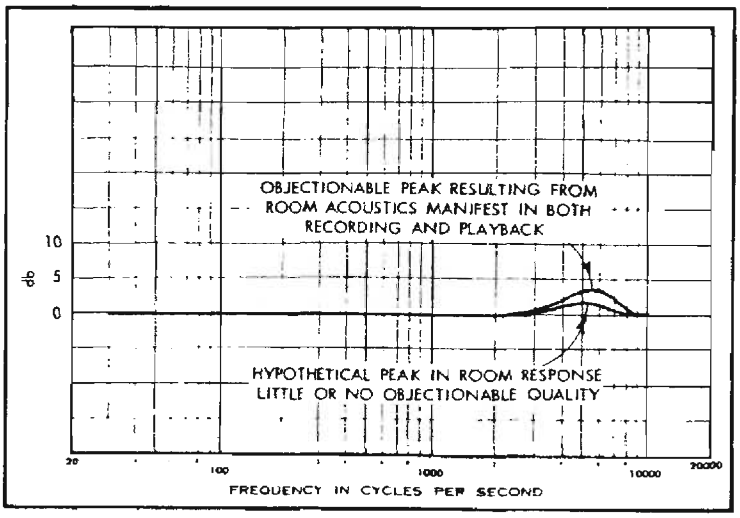
On the whole, location of the tape-output jack after the tone controls seem to be the more advantageous position. But this is an advantage only so long as the operator remembers to adjust these controls from the viewpoint of recording a tape rather than from the viewpoint of what sounds good at the moment over the loudspeaker. To make this point clear, assume that one wishes to record a tape while listening at low level to an FM program. Pleasant listening may require a substantial amount of bass boost to compensate for the apparent loss of bass at low volume. At the same time, it may be undesirable to have bass boost appear in the signal recorded on the tape.

Microphone and Room Characteristics

When recording through a micro-

(*Continued on page 61*)

Fig. 18. Effect of a peak in room acoustics upon record - playback response.



TAPE GUIDE

(*from page 34*)

phone, one is, of course, greatly dependent upon the range and smoothness of the particular microphone used. In addition, room characteristics are an important factor in determining frequency balance. It is possible to blame a microphone for faulty response when it is as much the room that is to blame.

To illustrate, assume that room acoustics produce a treble peak at 5000 cps when using a microphone that, for the sake of illustration, is perfectly flat through the audio range. This peak is mild enough, let us say, so that it is not disturbing when listening to music taped from a tuner, phonograph, or TV set. But when recording from a microphone, the peak manifests itself twice: once in recording and again in playback. Then, as illustrated in *Fig.* 18, the peak may become sufficiently severe to become objectionable.

In similar fashion, if the room causes a fall-off in treble response because of its sound-absorbing characteristics, the drop may become objectionable only when it occurs both in recording and in playback.

If the microphone has an undesirable peak or droop in response, it is possible that room characteristics may either compensate or not make any difference one way or other. However, should both the microphone and the room tend to peak or droop in the same frequency area, then the results can sometimes become intolerable. In other words, some microphones may sound bad in certain rooms and not in others. Æ

A 1 7/8-ips Magnetic Recording System for Stereophonic Music

P. C. GOLDMARK,* C. D. MEE,* J. D. GOODELL,* W. P. GUCKENBURG*

Rumors about the new tape system have been rampant for several months, but little actual information was available. Here for the first time is a complete description of the tape, cartridges, and handling mechanisms from the best authorities—those who developed it.

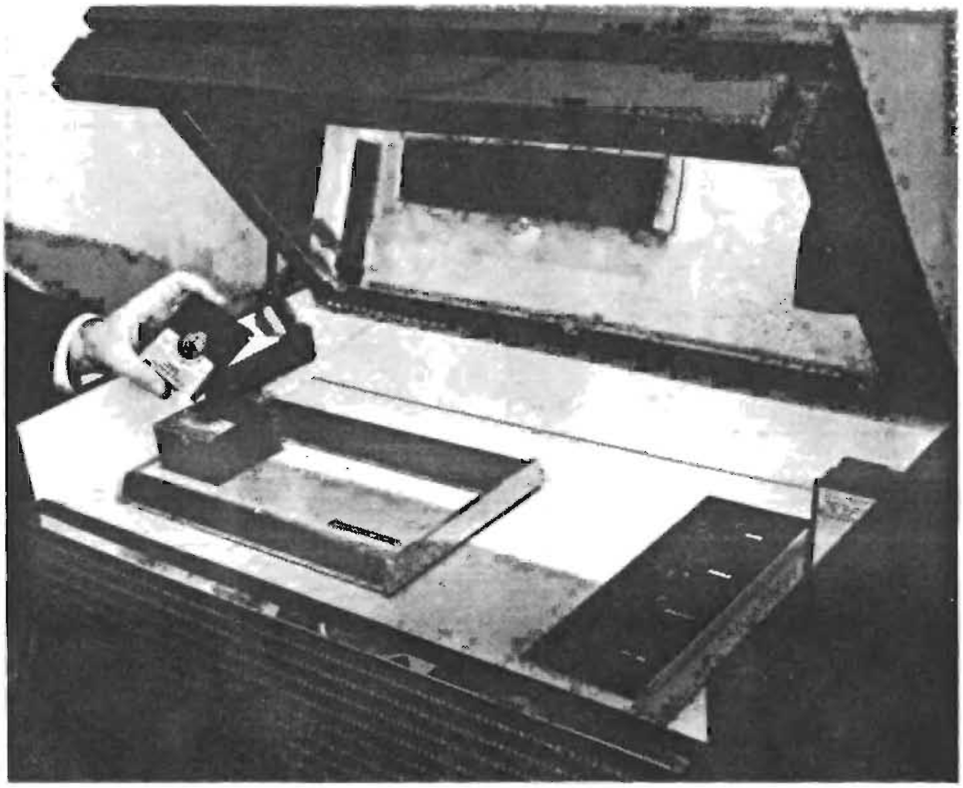
As part of a long range development program in the field of magnetic recording which CBS Laboratories undertook on behalf of Minnesota Mining and Manufacturing Company, recorded tape systems for the home have been under study over a period of several years.

In order that recorded tape can take an important place in the field of home entertainment, one must take into account a great many requirements, some of which are not easily met. For instance:

1. The tape must be contained in a compact cartridge in such a way that no part of the tape is exposed.
2. The amount of tape must be small and the cost of the cartridge low in order that the price of the final product can approach that of the disc record.
3. The sound should be stereophonic with provision for three tracks for maximum flexibility.
4. A complete musical composition should be played without interruptions; that is without reversing the cartridge or tape.
5. The quality of sound should be at least as good as the best of existing recorded media.
6. The durability of the tape and cartridge must be high enough so that after several hundred plays, the sound remains unchanged.
7. It should be possible to place a number of cartridges on a tape machine equipped with a changer-type mechanism so that one can provide music for several hours.

Here we will report on the outcome of these studies and subsequent developments which we believe will satisfy the preceding conditions and requirements.

It was clear from the outset that one was dealing with a system rather than just a few components. Thus intensive development work over a period of several years progressed simultaneously in such areas as methods of signal recording, magnetic transducers and playback heads, design of cartridges and tape transport mechanisms. The Laboratories' system work, in close cooperation with 3M, also included the development of a new tape with characteristics that provided optimum matching into the overall performance.



Prototype of Columbia 1⅞-in. tape unit in cabinet.

Late last fall the new recorded system was in a sufficiently advanced stage to demonstrate it to many members of this industry.

3M had, at that time, stated that the Zenith Radio Corporation had joined this effort and entered the design of commercial equipment based on these developments.

Some of the important features and parameters of the new tape cartridge system are as follows:

1. Tape speed is 1⅞ ips. The width of the tape is 150 mils; the thickness 1 mil, and there is provision for three tracks. Each track is 40 mils wide.
2. The cartridge is approximately 3½ in. square and 5/16 in. thick. The cartridge contains sufficient tape to play continuously for 64 minutes, and thus will carry more than 98 per cent of the music compositions available without interruptions. The space occupied by the cartridge in its container is approximately 4 cu. in. as compared with an LP record in its envelope with approximately 20 cu. in.
3. The tape machine can take five cartridges and play them automatically one after the other. A cartridge can be rejected during any part of its play similar to a record changer. The production versions of this machine now under development by Zenith will have fast forward and reverse speeds. The same instruments will also serve as a home recorder using the new cartridges with blank tape.

The Third Track

Earlier reference was made to a third track which is located in the center of the 150-mil tape.

Extended studies have been undertaken in the Laboratories to determine the optimum acoustic conditions desired by the listener in the average home while playing recorded music. Conventional stereophonic music, as now recorded, provides only a portion of the

* *CBS Laboratories, Stamford, Conn.*

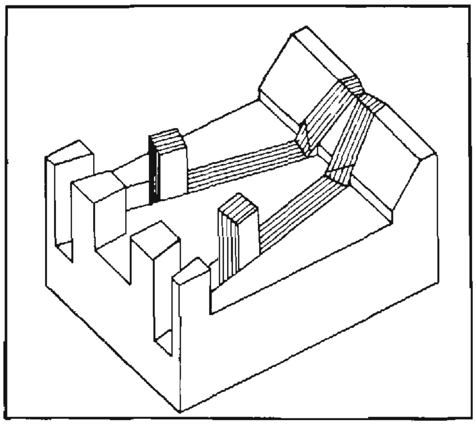


Fig. 1. Two-track playback head sub-assembly.

sounds that are perceived by the listener sitting in a concert hall. A large percentage of the total acoustic energy which reaches the listener's ears is reverberated and delayed sound which is considerably depleted of its original stereophonic character. Experiments in the Laboratories have shown that in a space simulating the average living room, a much more exciting and realistic sound can be produced giving an illusion of "being there." Thus, it is intended to record on the third track as an optional feature on the new recorded tape system, the stereophonic sum signal delayed and reverberated to an optimum degree.

The new medium will provide maximum flexibility and a new dimension in sound. The reproducing instruments can be manufactured for two or for three tracks.

Later some of the electrical and magnetic characteristics of the new system will be discussed. The data and curves shown are already based on the newly developed tape and represent the overall behavior of the entire system, that is, recording, tape, and playback. The new tape is now in pilot production at 3M, but the cartridges played in current demonstrations still use the older tape on which these programs were recorded last fall.

Following the section dealing with the magnetic aspects of the new system, some of the mechanical problems and their solutions as encountered will be described.

A comparison of the new tape system with the original 15-ips tape master from which both the stereo records as well as the new tape cartridges have been derived, has been demonstrated with success. For this purpose, some sections of music were alternately transcribed from the original master and the 1⅞-ips narrow track version onto a 15-ips half-track tape.

Magnetic and Electrical Characteristics

In order to achieve an adequate signal-to-noise ratio, frequency response, and dynamic range at a tape speed of 1⅞ ips, significant developments of most components used in magnetic recording are required. For instance, due to the shorter wavelengths encountered, developments have been aimed at reducing wavelength-dependent losses.

Among the losses in reproduction which have been minimized in the system at hand are those attending (1) separation of head and tape surface, (2) azimuth alignment of head and tape, and (3) playback-head efficiency. Losses minimized in the recording process are (1) tape-thickness loss, (2) recording-field configuration loss, and (3) loss caused by non uniformity of tape particles.

(A) *Losses in Reproduction*

1. There is an exponential reduction of the playback-head flux with decreasing recorded wavelength due to the finite separation between the surface of the tape and the playback-head pole pieces.

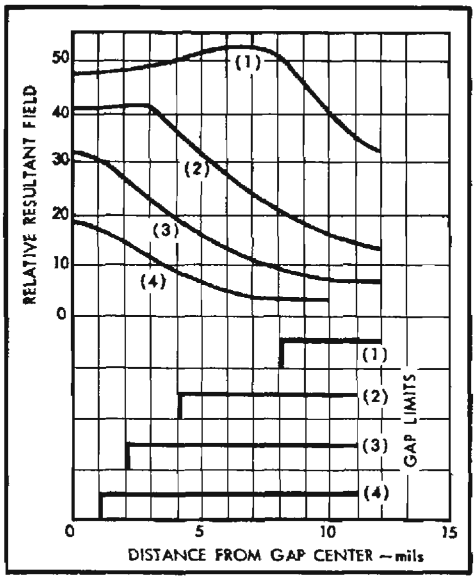


Fig. 2. Rate of recording-field extinction as a function of gap length.

At 15,000 cps and 1⅞ ips, this loss is almost 0,5 db per micro-inch separation.

2. Another important loss is associated with the azimuth alignment between the playback-head gap and the line of constant recorded magnetization across the track width. For a conventional 90-mil track a loss of 6 db occurs at 15,000 cps and 1⅞ ips. for a misalignment angle of 3 minutes.

3. The proportion of playback-head flux shunted by the gap will increase when using the narrow gaps necessary to resolve the shortest wavelengths recorded at a tape speed of 1⅞ ips. In order to maintain a high efficiency it is necessary to compensate for a reduction in gap length by a corresponding reduction in gap depth.

(B) *Losses in Recording*

1. A separation loss of the type described for reproduction occurs during recording due to the finite coating thickness. Those particles remote from the tape surface will thereby give an attenuated contribution to the tape-surface flux and so will contribute less to the playback-head flux.

2. The magnetization of a recorded tape will not be uniform throughout the coating thickness since it depends on the rate of extinction and the direction of the recording field when the critical value for recording is reached after the tape has passed the recording gap. In addition to this, a further loss can occur due to change in phase of the recorded signal through the coating thickness caused by the vertical curvature of the effective recording plane of the recording-head field.

3. For high resolution of the effective recording plane a sharp cutoff of the recording field must be accompanied by a high uniformity in the magnetization characteristics of the individual particles of the tape. Elimination of particles with low critical fields for switching will also reduce self-demagnetization effects. The separation loss has one advantage in slow speed tapes for audio, since, due to the shorter wavelengths involved, print through is correspondingly reduced allowing new thin tape backing materials to be used with safety.

New Developments in Magnetic Recording Components

Although the major loss component, called separation loss, is inherent in presently known magnetic recording systems, it has been possible by improvements of tape and heads to achieve performance characteristics approaching those presently obtained from 7.5-ips machines. Such performance is achieved with a track width of 40 mils. Having a narrow track reduces the alignment problem.

It has been found that a conventional laminated ring-type playback head can be constructed to be responsive up to 15,000 cps with a 1.5 mv output from a tape having ⅓ mil. coating thickness. A sub-assembly of the two-track version of such a head is shown in *Fig.* 1. The playback-head coils fit over the projecting laminations. Since the recorded wavelength at 15,000 cps is only ⅛ mil, it is necessary to form an effective magnetic gap of 1/16 mil (or 1.5 microns.). It has been found that a 1-micron spacer gives satisfactory head resolution in pro-

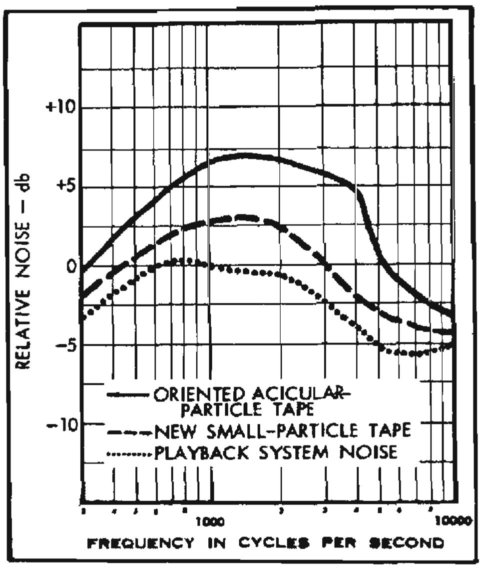


Fig. 3. Zero modulation noise curves with correct playback equalization plus the 40-phon ear characteristic.

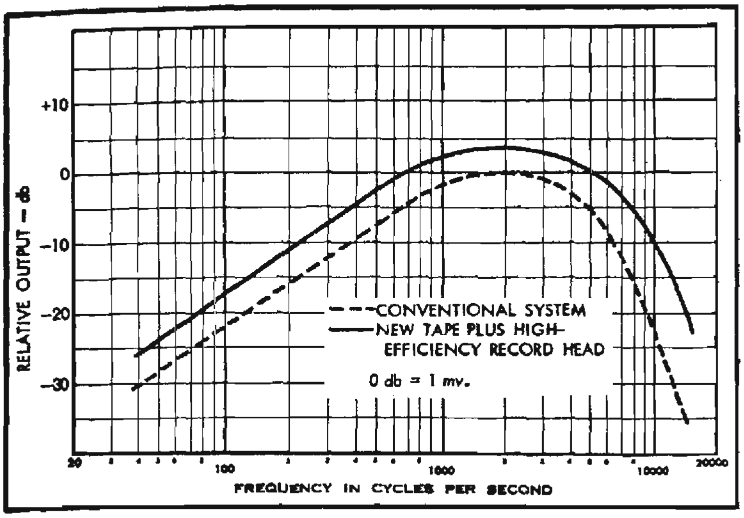


Fig. 4. Maximum output curves for 1⅞-ips tape.

longed use. By manufacturing the multitrack head in two halves, automatic colinearity of the gaps is assured and in practice the 10,000-cps sensitivity of the tracks differ by less than 4 db.

Similar mechanical refinement is necessary of course in the recording head. *Fig.* 2 shows a plot of the field distributions at 0.1-mil spacing for various gaps. It is seen that the field decrement increases somewhat with gaps which are large compared to the spacing. Thus a long gap might be thought advantageous especially since the vertical field decrement is also reduced. In practice, however, the expected improvement does not occur, probably due to the relatively greater vertical component of the effective recording field. Considerable development has been carried out to improve the recording-field configuration for the very short wavelengths involved in this system. This work will be reported at a later date.

Significant advances have been made by 3M in the recording media, leading to considerable reduction of the separation loss effects. Firstly, a tape lacquer formulation has been developed which is relatively soft, giving good head-to-tape contact. Particle rub-off on guides and heads has virtually been eliminated and the consequent amplitude variations considerably reduced at the shortest wavelengths. In addition, the Laboratories developed a higher-output and lower-noise tape as a result of changes in the magnetic material itself. Previous work has concluded that a reduction of effective particle size results in lower tape noise. The improvement achieved is shown in *Fig.* 3, where the weighted noise response for existing tape is compared with the new tape using optimum bias for each. A 4-db lower noise level is obtained in the mid-frequency range. Higher over-all output is also obtained from the new material. It is found that the short-wavelength efficiency is particularly improved. One reason for this is that a deliberate attempt was made to reduce the speed of critical fields required for magnetization change in the individual particles. For acicular particles better control of the size and shape is required and for effectively spherical or cubic particles it is necessary that the acicularity be kept low enough to make the crystal anisotropy dominant in all particles. *Figure* 4 shows the improvement resulting from recording with the new tape using one of the high-efficiency recording heads compared to that obtained with conventional 1⅞-ips recording.

Equalization Techniques and Performance of the System

The recording equalization adopted for the new 1⅞-ips record-replay system is shown in *Fig.* 5. This curve was derived by performing many listening tests on a variety of program material. It is the characteristic which meets the requirement to load the tape optimally at all frequencies without overload danger. Using this in conjunction with the playback equalization (also shown in *Fig.* 5) a flat response is obtained from 30 to 15,000 cps at −18 db relative to a level giving 3 per cent distortion at 1000 cps. Under these conditions the ratio of the maximum signal level at 1000 cps to the zero-modulation system noise is 54 db. The 10,000-cps signal response at this maximum signal level is −12 db relative to that at low frequencies. Typical equivalent signal-to-noise ratio for professional 7½-ips half-track systems is 54 db with a corresponding 10,000-cps signal response at −6 db. Thus the new system with its own recording and playback characteristics approaches the 7.5-ips performance available today and has been found to be entirely adequate for all types of musical programs.

Mechanical Design Problems and Solutions

One of the central problems in recorded tape systems is the design of the tape packaging. Obviously, it is necessary to satisfy requirements of convenience as well as to provide adequate protection for the tape. Naturally high-quality performance with respect to music reproduction is a prerequisite.

In order to popularize recorded tape it is essential to eliminate the process of manual threading between the reels. This requirement is dictated by the need for avoiding manual threading and also by the requirement to make the cartridge compatible with a practical automatic changer mechanism.

On first examination the notion of threading the tape permanently between two side-by-side reels contained in the cartridge is attractive. However, every practical design incorporating both the supply and take-up reels in the cartridge requires that sections of the tape be exposed through openings in the cartridge walls with consequent dangers of damage. Even in a single cartridge player there are many difficulties involved in coupling the tape of a dual-reel cartridge to the drive system and the heads, but when the design of an automated changer is considered, these problems increase rapidly in number and magnitude.

A basic consideration in any type of cartridge is the need for relatively high speed transport in so-called "search" operations. If flanges are used on the reels inside the cartridge, the bulk is considerably increased and many problems of stability are encountered. Thus,

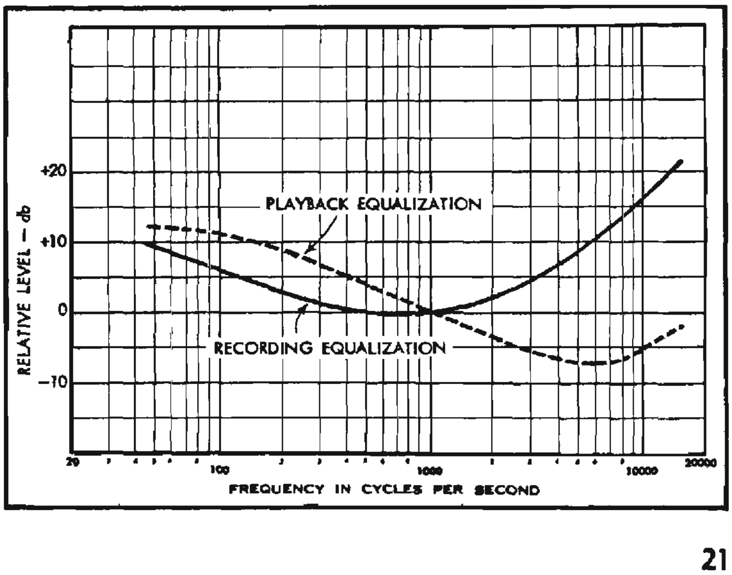


Fig. 5. Recording and playback equalization curves.

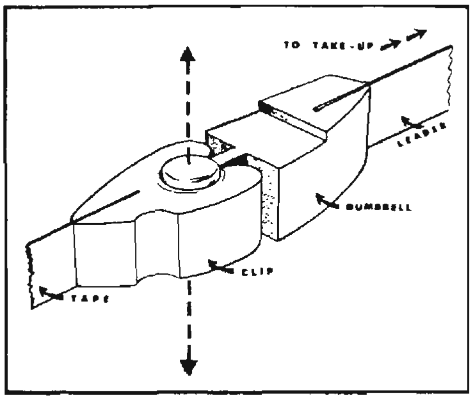


Fig. 6. Cartridge coupling members.

high-speed winding without flanges requires some method of maintaining a separation between the tape and the cartridge walls.

The three dimensional geometry of the reeled tape, the driving spindle in the transport mechanism, the walls of the cartridge and other components call for strictly orthogonal relationships or some automatic dynamic adjustment and an accurate system of tape guidance. Otherwise, the cumulative errors in repetitive reeling of the tape, even on the same machine, will lead to telescoping or angular displacement of the tape reel with respect to the cartridge walls. In brief laboratory experiments these problems may not be evident but in long-term field use the increasing friction produces instabilities in the tape speed and eventually may completely block the reel from rotating.

The problem of smooth reeling without any flanges was solved by introducing a novel guiding member in the cartridge with adequate compliance to insure a smooth rewind cycle. This arrangement allows a tape with an hour of playing time to be rewound in twenty seconds. (A five-second rewind time has been achieved in the laboratory.)

Threading of the tape is accomplished by means of a leader permanently attached to the takeup reel in the mechanism. The end of the rewind cycle leaves the permanent leader in the threading path of the machine.

A very simple and economic solution was used for the design of the coupling between the reeled tape and the permanent leader. This consists of a "U" shaped device attached to the end of the tape in the cartridge and so shaped that it seals off the only opening in the cartridge when the tape is fully rewound. The permanent leader terminates in a dumbbell-shaped element that readily mates with the "U" shaped clip. The dumbbell attached to the permanent leader can slip through the "U"-shaped clip in a vertical direction with only a light detenting restraint but provides an absolute coupling in terms of horizontal pull when the two members are engaged. (*Fig.* 6).

In order to eliminate variations in back tension with dynamic changes in effective reel diameter, a felt pad is

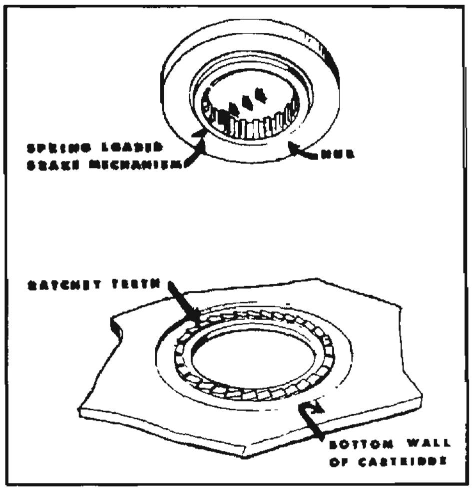


Fig. 8. Cartridge brake mechanism.

spring-loaded against the surface of the tape as it leaves the cartridge and the supply reel is operated in free-running bearings. This provides excellent tensioning characteristics and at the same time maintains the cartridge complexity cost at a minimum. *Figure* 7 shows the tape deck and the felt pad.

Some kind of braking mechanism is essential in order to avoid partial unreeling and fouling of the tape within the cartridge under normal conditions of handling. The brake must be positive,

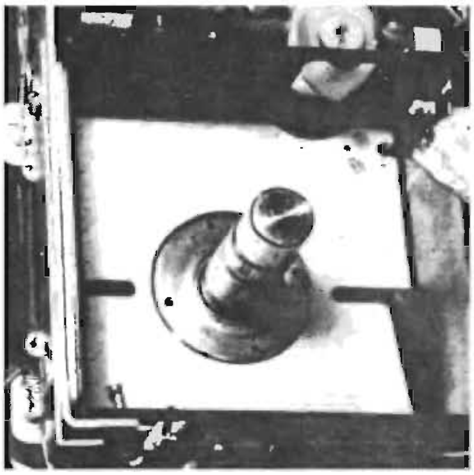


Fig. 9. Close-up of cartridge spindle and well.

reliable, and simple to assemble. The device selected consists of a linkage mounted in the cartridge hub and spring-loaded in a ratcheting relationship with teeth molded in the cartridge wall. When the cartridge is placed on the machine, the spindle releases the brake automatically. The brake is shown in *Fig.* 8.

The facility for driving the cartridge hub during the rewind cycle must be designed so as to permit random rotary orientations of the spindle with respect to the cartridge hub in the loading process. This is accomplished by means of radial slots around the inner periphery of the hub and a spring-loaded two-toothed drive in the spindle. See *Fig.* 9.

The cartridges are designed with mating surfaces that couple them (see *Fig.* 10) together in a stable vertical stack. This feature contributes considerably to the ease with which they may be handled and loaded in a changer mechanism. The patterns are unsymmetrical so that the cartridges must be correctly oriented or they cannot be fitted together. Other details of the mechanism make it impossible to load the cartridges in any way that results in improper operation.

The resulting cartridge design is compact, inexpensive and dependable. Actually, of course, the cartridge design was carried on in conjunction with the development of mechanisms capable of handling it in a fully automated changer so as to eliminate any mutually exclusive features. The actual changing mechanism consists simply of a spring-loaded platform in a well (*Fig.* 9) with which the supply spindle is coaxial, and an appropriate escapement. The latter is an essentially conventional device.

There are two escapement levers that operate in tandem on opposite sides of the cartridge well. One of the escapement levers is placed close to the corner from which the tape is fed in order to maintain accurate positioning between the clip terminal and the threading path.

The path for the tape is a straight

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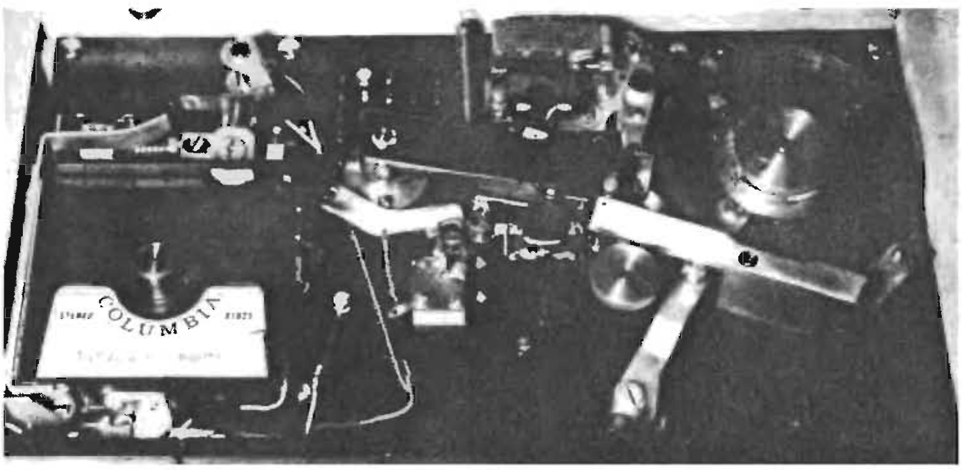

Fig. 7. Tape deck, showing felt pad.

1⅞-IPS MAGNETIC RECORDING SYSTEM

(*from page 22*)

line from the cartridge to the supply reel during the threading operation. When the tape has been pulled from the cartridge and starts to wind on the supply reel, the pressure pad that supplies the back tension and the pressure roller are automatically brought into position. (*Fig.* 13)

The takeup reel is operated with a conventional slipping clutch drive.

The successive cycles of operation are programmed by means of a multiposition rotary switch and several mechanical interlocks. The slipping clutches,

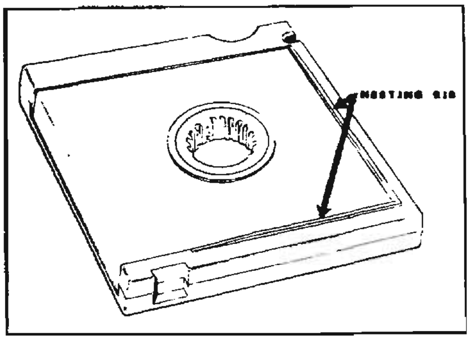

Fig. 10. Diagram showing cartridge nesting ribs.

brakes, speed-changing idlers, and the like are operated from the three-dimensional surfaces of a single complex cam which programs all pressure-pad, pressure-roller, and escapement operations. It is necessary to provide a number of mechanical and a few electrical interlocks to prevent improper manual interference with machine operations, but these are relatively simple and straightforward in design.

The straight-line character of the tape path does not require intermediate idlers and consequently the guidance problems are minimized. However, as in all such drives, it is important to maintain the pressure roller axis parallel to the axis of the capstan. This is accomplished by introducing sufficient compliance in the mounting of the pressure roller so that it is self-adjusting within small limits. The spring loading provides a simple adjustment for correcting major pressure differentials across the idler surface. (*Fig.* 11).

Obviously, there must be some means for sensing the end of the tape and various other portions of the operating cycle. In this machine these results are obtained by means of a simple analog computing linkage that cannot be disclosed in detail at this time. However, the method is independent of the length of the tape in a given cartridge and has displayed a very high degree of reliability.

The authors wish to express their appreciation for the advice and assistance during the course of this work by B. B. Bauer, A. A. Goldberg, J. C. Jeschke, H. R. Sherman, E. L. Torick, and J. C. Wistrand of CBS Laboratories and Barbara Ivins, formerly with CBS Laboratories. We also wish to acknowledge whole-hearted cooperation of Dr. W. W. Wetzel and his associates of 3M's Magnetic Products Division. Æ

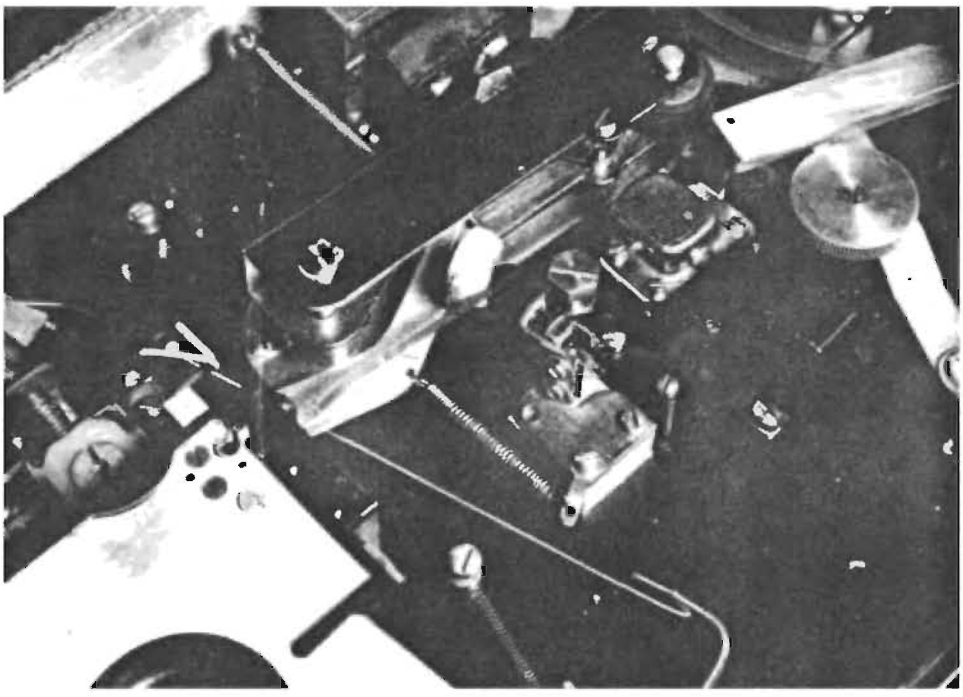

Fig. 11. Straight-line path for tape.

Distortion in Tape Recording

HERMAN BURSTEIN*

Types and causes of distortion should be understood by the recordist if he is to obtain the best results. Various compromises are shown to be effective under different conditions.

IN TWO PARTS—PART ONE

FROM TIME TO TIME the writer visits some friends who have in their living room a pre-war radio console, which cost over $500 and was considered one of the finest units of its day. It receives good care and enough service to maintain it in "as good as new" condition. To one whose ears have become attuned to modern high fidelity equipment, this console falls noticeably short of the mark in terms of frequency response and noise characteristics. But its most obvious deficiency, the greatest deterrent to pleasurable listening, concerns distortion. The instrument just does not have the smoothness and ease of reproduction afforded by modern equipment.

The foregoing illustrates the point that one of the most noteworthy developments in the audio art in the high fidelity era, at least to the ears of this writer, has been the reduction of distortion. While the power amplifier has come in for a great share of attention, it is also true that designers of control amplifiers, tuners, cartridges, speakers, and other components have concentrated on reducing distortion to imperceptible amounts.

In these days when most audio equipment is built to exacting standards with respect to clean reproduction—i.e. low distortion—one looks for comparable refinement in the tape recorder. The meticulous recordist will wish to preserve the original quality of the sound so far as possible. While satisfactorily low distortion can be achieved in tape machines, this is far from a simple matter. Overcoming distortion remains considerably more of a challenge in tape machines than, say, amplifiers. When used with today's better amplifiers, tuners, and speakers, a tape recorder must indeed be of high quality, and must be properly used, in order not to add noticeable distortion.

In tape recording, distortion is inextricably linked with several other aspects of the process—signal-to-noise ratio, frequency response, equalization, bias current, and tape speed. Therefore in the following discussion we shall discuss distortion in terms of its relationship to these factors. First, however, it would appear profitable to devote some space to a review of what is meant by distortion. Such an understanding can prove useful in various ways; for example, it enables one to appreciate why a given recording level results in no noticeable distortion for some kinds of sound and quite perceptible distortion for other kinds.

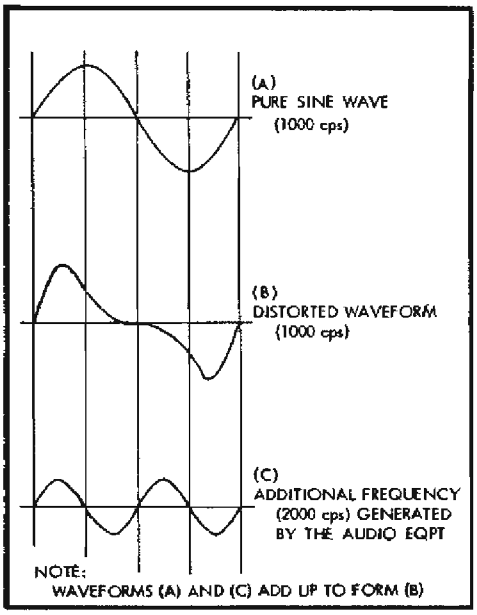

Fig. 1. Example of harmonic distortion.

Meaning of Distortion

Reproduced sound is never totally devoid of distortion. But in the present state of the art it can be kept so small in most audio components as to be unnoticeable, permitting the reproduced sound to retain the ease and naturalness of the original. In somewhat larger quantity, it may still not be immediately discernible but instead may produce a consciousness of aural fatigue after one has been listening for a moderate period of time. In successively larger quantities, distortion causes the sound to become grainy, gritty, coarse, and finally so broken up as to be partially or completely unintelligible.

Distortion consists of a change in the original waveform, due to improper functioning of one or more audio components. Such improper functioning is called non-linearity; that is, the waveform turned out by the component is not an exact replica of the incoming signal.

It can be demonstrated, mathematically and by suitable test equipment, that the change in the waveform actually consists of the addition of new frequencies to those that were originally produced by the sound source. This is illustrated in *Fig.* 1. At (A) we see the original waveform, a pure sine wave; (B) shows a distorted version of the original. The distortion consists of the waveform

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shown at (C). If the distortion frequency in (C) is added to the original frequency (A), the result is the distorted waveform of (B).

The new and undesired frequencies, which are termed distortion products, are produced by the audio equipment. Unlike noise and hum, which are also undesired frequencies produced by (some) audio components, distortion products appear only in the presence of an audio signal.

The principal kinds of distortion, those most offensive to the ear, are harmonic and intermodulation distortion. Harmonic distortion denotes the generation of frequencies that are multiples of the original frequency. To illustrate, in the course of reproducing a 1000 cps tone the audio equipment may, as the result of its non-linear behavior, also generate frequencies of 2000, 3000, 4000, etc. cps.

To an extent, the ear is not unduly offended by extraneous frequencies if they are harmonically related to—exact multiples of—the original note. As a rough rule, harmonic distortion products are compatible with pleasant listening when, in total, they constitute no more than about 1 to 2 per cent of the total sound; generally, 3 per cent is considered too great.

At the same time, the amount of harmonic distortion which is tolerable depends upon whether the distortion products are even or odd multiples of the original frequency. Even multiples tend to be less offensive. Furthermore, the "order" of the harmonic products is a determining factor. High-order products are many times the original frequency; low-order products are a few times the original frequency. High-order products tend to be more offensive. Thus if the original frequency is 1000 cps, distortion products of 8000 and 9000 cps would be more disagreeable than 2000 and 3000 cps. (It is appropriate to intersperse here that a tape recorder which cuts off sharply above 9000 or 10,000 cps may offer cleaner sound than one which goes out to 15,000 cps because the former eliminates high-order distortion products to a greater extent.)

Intermodulation distortion—IM for short—occurs only when two or more frequencies are simultaneously reproduced by the audio equipment. Deformation in the waveform of one frequency results in deformation of a second frequency, although it could well be that the second frequency, if reproduced alone, would not have been distorted by the equipment in question. Thus IM distortion refers to interaction among frequencies, with new frequencies being born out of this interaction. When a substantial number of frequencies are reproduced at once, as is often the case with music, the interaction, namely IM distortion, becomes very complex.

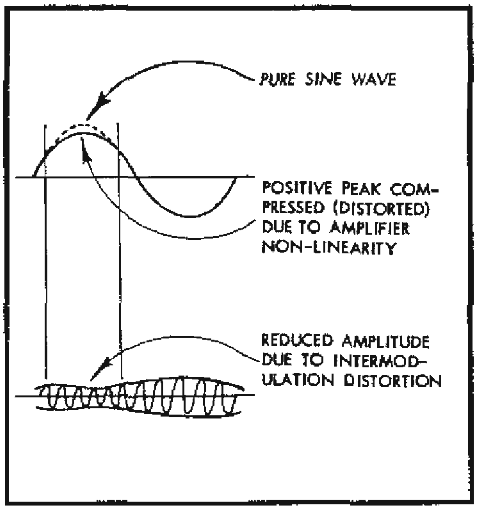

Fig. 2. Example of intermodulation distortion.

Figure 2 illustrates the process of intermodulation distortion. For simplicity, it is assumed that only two tones, 100 and 1000 cps, are present and that they are fed through an amplifier. Let us assume that the 100-cps signal is of substantially greater magnitude than the other, so that it causes the amplifier to operate in non-linear fashion at the positive peaks of the waveform. During these moments of non-linear operation of the amplifier the 1000 cps signal is also being treated in non-linear manner, despite the fact that this signal in itself is of too small magnitude to cause the amplifier to behave in non-linear fashion. At (A) we see the effect of amplifier non-linearity upon the 100 cps waveform. (B) shows the resulting effect upon the 1000-cps waveform due to the fact that the amplifier is periodically operating in *non-linear manner*. The 1000-cps waveform is compressed 100 times per second by the 100 cps signal. In other words, the 100-cps frequency is now present in the 1000-cps one.

Unfortunately the new frequencies created by IM distortion are not multiples of the original frequencies. The distortion products consist of various multiples of one frequency plus or minus multiples of the other frequency. For example, 100 and 1000 cps will form IM products of 1100 cps (sum of the original signals) and 900 cps (difference between the original signals). They will form 1200 cps (twice 100 cps plus 1000 cps) and 2100 cps (twice 1000 cps plus 100 cps). They will form 1900 cps (twice 1000 cps minus 100 cps) and 800 cps (1000 cps minus twice 100 cps). And so on and so forth. If there were more than two original frequencies involved, the distortion products would be still more complex.

IM distortion not exceeding 1 to 2 per cent is often considered compatible with high fidelity. On the other hand, it has been found that the ability to reduce IM to as low as 0.1 per cent in voltage amplifiers and power amplifiers has produced noticeable improvement.

Distortion Ratings for Tape Recorders

Extremely seldom does one find the specifications for a tape machine having anything to say about IM distortion. The reason will appear later, when we compare harmonic and IM distortion produced by tape recorders. The nigh-universal practice instead is to rate tape machines in terms of harmonic distortion at a stated signal-to-noise ratio, for example 50 db in moderate-quality machines or 55 db in high-quality machines. The record-level indicator is adjusted to provide an indication of maximum permissible recording level when the level is such as to produce anywhere from as low as 1 per cent to as high as 5 per cent harmonic distortion (at a frequency of 400 cps or so). The low-priced machines typically use 5 per cent harmonic distortion as maximum permissible recording level, while the top quality ones use 1 or 2 per cent. Many machines, of varying quality, use 3 per cent harmonic distortion as the reference. The official standard, applicable to 15 ips recording, considers 2 per cent harmonic distortion to be the maximum permissible quantity.

Distortion and Signal-to-Noise Ratio

In the process of recording and playing back a tape, there are two principal sources of noise to contend with: tape noise and amplifier noise. Tape noise is of two kinds. One, known as tape hiss, is due to incomplete cancellation of magnetic fields when the tape is erased. These magnetic fields are of random character and therefore produce random frequencies with a characteristic "hissy" quality. The other kind of tape noise is known as modulation noise, which appears only in the presence of an audio signal on the tape. Modulation noise is due to imperfections in the base and/or magnetic coating of the tape. When an audio signal is recorded, corresponding imperfections appear in the recorded signal and are manifest as noise. As the result of the improvements that have taken place in tape manufacture, modulation noise is less serious a problem than tape hiss.

Tape-amplifier noise occurs both in recording and playback. However, the signal fed to the tape amplifier is generally of much smaller magnitude in playback—the tape delivers but a fraction of a millivolt at many frequencies—so that it is principally noise of the tape playback amplifier which presents a problem.

In sum, the principal obstacles to a good signal-to-noise ratio are tape hiss and the noise (including hum) produced by the tape playback amplifier.

To achieve an adequate signal-to-noise ratio it therefore becomes vital to record as much signal as *practical* upon the tape. But the practical amount of signal that can be impressed on the tape is determined by the distortion characteristics of the tape, the tape head, and the

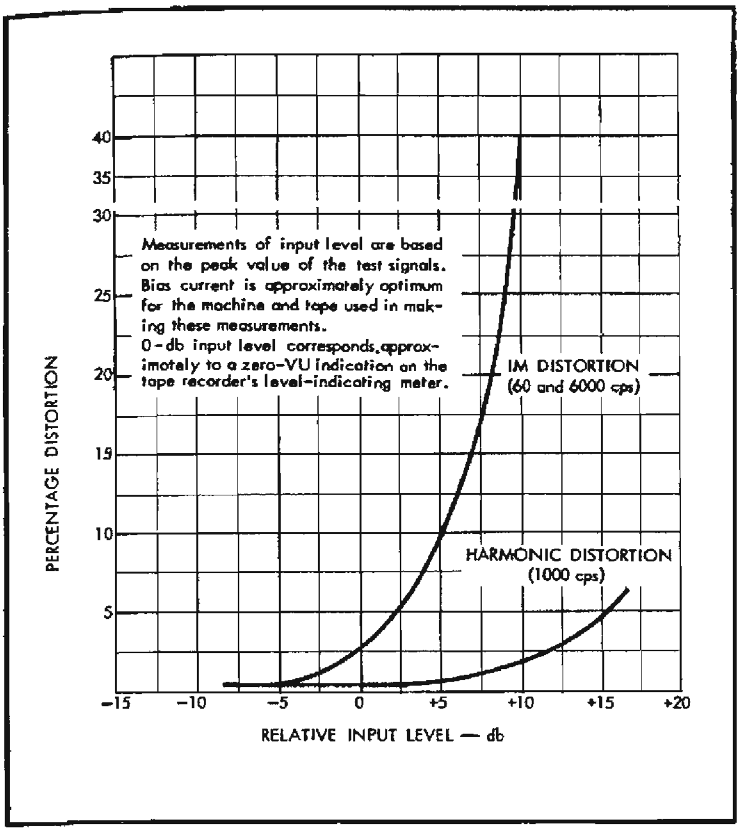

Fig. 3. Variation of tape distortion with changes in input level.

record amplifier. Ordinarily, the tape sets the bounds to how much signal can be recorded. That is, the tape overloads, or should do so, before the tape head and the tape amplifier go into serious distortion.

However, there have been instances where a poorly designed head has produced significant distortion in recording particularly at low frequencies, although the signal level was not such as to produce appreciable distortion on the tape. Laminated heads, which contain a greater amount of magnetic material, are generally apt to have superior distortion characteristics compared with those of nonlaminar construction.

There have also been instances where an improperly designed recording amplifier has gone into serious distortion at too low a recording level. For example, one instance of this kind involved a machine of professional calibre. Although the amplifier did not produce appreciable distortion when conventional tape was employed, it went into excessive distortion when the recording level was increased to a point consistent with the use of high-output tape, which can accept several db more signal for the same amount of distortion.

For the most part, however, we can assume that it is the tape which sets the limit to the recording level by overloading before any of the other components do.

Figure 3 indicates the variation of harmonic distortion and of IM distortion with changes in input signal. The measurements were taken on a professional-quality tape machine operating at 15 ips. While the results doubtless would be different with other machines, tapes, and speeds, nevertheless these curves can be viewed as representative.

It may be seen in *Fig.* 3 that distortion, either harmonic or IM, increases quite slowly for a while as signal level is increased, but that the rise in distortion becomes precipitous after a point. Severe IM distortion occurs much earlier than harmonic distortion. Hence at recording levels which breed innocuous amounts of harmonic distortion the IM distortion will have risen to unacceptable levels. It is understandable, therefore, why a recording may sound grating if made under conditions where the record level indicator permits 5 per cent maximum harmonic distortion.

On the other hand, a recording that permits IM distortion to reach 20 per cent or more is not always unacceptable. Sounds recorded at such distortion levels are tolerable if their duration is sufficiently brief. Characteristically, many sounds have peak levels 10 db, 20 db, or even more above their average level. While the peaks may be severely distorted, the major part of the sound may be at a level that escapes significant distortion. Whether the distortion in the peaks is tolerable depends upon their duration and how frequently they come along. If the peaks are occasional and very brief, large amounts of IM distortion in the reproduction of these peaks may escape attention.

The extent to which distortion is acceptable also depends upon the nature of the sound being recorded. Certain kinds of music must be recorded at lower levels than other kinds in order to maintain clean reproduction. Generally, higher levels of distortion are acceptable in reproducing speech than music. In recording a solo voice or a solo instrument, IM distortion is less apt to be serious than when recording a group of voices or instruments, because there will be fewer intermodulation products when there are fewer frequencies reproduced at one time.

In deciding how high a recording level one may employ for different source material, there is no substitute for experience. The neophyte recordist does well to invest a certain amount of time in experimenting with various recording levels for various kinds of material. In any event, he should remember that the desire for a slight improvement in signal-to-noise ratio—i.e., by raising the recording level just a few db—may bring with it a great increase in distortion if one happens to be at the point where distortion rises rapidly with a slight increase in recording level.

All in all, the recordist has three choices. First, he may be willing to accept occasional noticeable distortion, principally on signal peaks, for the sake of a relatively high recording level and therefore a superior signal-to-noise ratio. Second, he may be unwilling to accept any noticeable distortion whatsoever, but at the cost of a significant reduction in recording level and therefore in signal-to-noise ratio. Third, it is possible in a sense to eat one's cake and have it too by "riding gain." That is, one can record at a moderately high level, well below the point of noticeable distortion, during normal and quiet passages, then reduce the recording level just before loud passages come along. The last alternative requires one to be prepared with a score or other means of knowing when loud passages are about to occur. Also it implies that one is willing to compress the dynamic range (difference between the softest and loudest passages) in exchange for an improvement with respect to distortion.

It must be taken into account that the need to exchange signal-to-noise ratio, or possibly dynamic range, for a reduction in distortion depends upon the tape machine one is using. If the playback amplifier has superior characteristics in terms of low noise and hum, and if the head is specifically designed for playback and therefore has higher output than one intended for both recording and playback, the recordist's task of achieving a satisfactory compromise between the conflicting considerations of noise and distortion is lightened. On the other hand, if amplifier noise is relatively high and head output low, the recordist might conceivably decide he is willing to accept a fair amount of distortion in order to keep noise down relative to the audio signal.

To Be Continued

Distortion in Tape Recording

HERMAN BURSTEIN*

Types and causes of distortion should be understood by the recordist if he is to obtain the best results. Various compromises are shown to be effective under different conditions.

IN TWO PARTS—PART TWO

Tapes differ somewhat in their distortion characteristics. This is illustrated by a test that was made of four brands of conventional tape. At a relatively high recording level, the input signal was adjusted in each case so as to produce the same output level in playback; after all, it is the playback level in which we are ultimately interested. At the same time, bias current was adjusted so as to produce minimum IM distortion. The results appear in Table 1. While the differences in distortion are not profound, still the difference between minimum and maximum IM, 3.4 per cent, is not insignificant. On the other hand, Tape A would not necessarily be one's choice, assuming that one goes by laboratory tests. It would further be necessary to investigate the tape's characteristics with respect to frequency, noise, and other factors.

A high-output tape was tested in the *same manner as the four conventional* tapes just discussed. In this instance, minimum IM distortion was only 3.5 per cent, a substantial improvement.

The ability of the tape to accept a high recording level without serious distortion varies somewhat with frequency. At a tape speed of 7.5 ips, it appears that there is a rise in the amount of signal which can safely be presented to the tape. The rise starts at about 1000 cps and attains a maximum of some 4 or 5 db. The nature of this rise may vary among brands and kinds of tape. Conversely, it is indicated that the acceptable signal decreases at the low end of the audio range. *Figure* 4 suggests in approximate and relative terms the permissible recording signal that may be presented to the tape at various frequencies for the same amount of distortion. In view of what happens at the low end, it may be advisable to record at a somewhat lower level than usual when dealing with a sound source dominated by low notes.

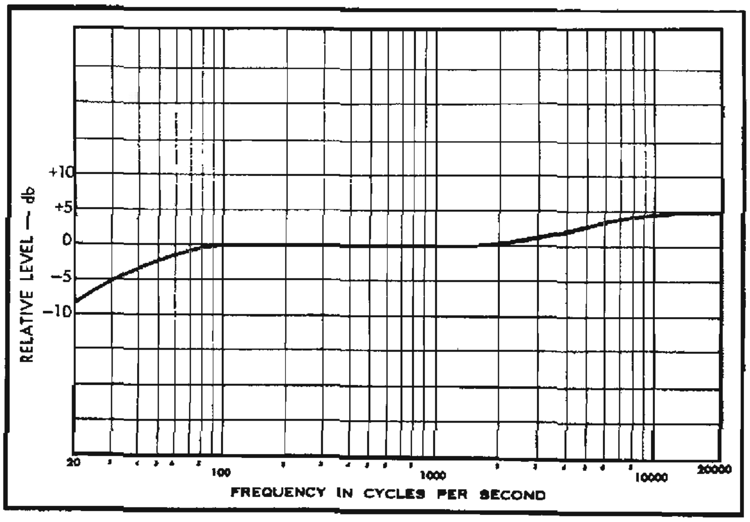

Fig. 4. Relative permissible recording level (approximate) at various frequencies at 7.5 ips.

Distortion and the Record-Level Indicator

To record at a level high enough for a good signal-to-noise ratio yet low enough *for tolerable distortion* depends a great deal upon the record-level indicator. This may be either of the electronic-eye type, which indicates peak recording level, or of the meter type, which tends to indicate average level. In either case, it is of paramount importance that the meter be properly calibrated in the sense of indicating accurately when maximum recording level is reached. Thus an electronic eye that is supposed to close at a level producing 3 per cent harmonic distortion but actually does so at 6 per cent can account for an unsatisfactory recording in terms of clean reproduction. On the other hand, an eye that closes at a level resulting in only 0.5 per cent harmonic distortion would lead to very clean

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TABLE I

Minimum Level of IM Distortion Obtainable With Four Brands of Conventional Tape at a High Recording Level

Tape	Minimum IM Distortion %
A	7.6%
B	9.0
C	11.0
D	10.0

recordings but probably with an unnecessarily low signal-to-noise ratio.

Accordingly, the individual who is meticulous about the maintenance of his tape recorder will see to it that on occasion the calibration of the record level indicator is checked and adjusted if necessary.

The VU meter presents a special problem. The electronic-eye indicator has an advantage in that it indicates the peak level of the signal (although the meter has other advantages discussed in an earlier article.) In the case of the VU meter, which tends to indicate average level, it is necessary to allow for the inability of the pointer, a mechanical rather than electronic device, to follow rapid signal changes. Hence the VU meter understates maximum signal level. Accordingly, it is important that an offsetting adjustment be made in the calibration of the VU meter. This means that the meter should be adjusted to indicate maximum permissible recording level when the average signal (or a steady sine wave) is actually about 6 db to 10 db below the level that would cause maximum permissible distortion. Thus the meter "reads ahead," providing a safety margin to compensate for the fact that signal peaks tend to be much higher than the average signal level. Even with this safety margin, the recordist must employ experience and judgment in setting his recording level.

Distortion and Frequency Response

With rising frequency there are increasingly severe losses that take place in the recording process. These losses have to be made up by treble boost in the record amplifier. In many tape machines this treble boost goes beyond 20 db by the time the upper end of the audio range is reached. Such amounts of boost carry with them the danger of overloading the tape.

To a substantial extent the danger is mitigated by the fact that in most musical material the amplitude of the high frequencies is considerably less than that of the middle frequencies. *Figure* 5 shows for a typical orchestral selection the relative peak amplitude of frequencies over the audio range; while it should be kept in mind that this figure applies only to one particular orchestral selection, nevertheless it is typical. To the extent that the high-frequency peaks are lower than the peaks of the other frequencies, there is an offset to treble boost used in recording.

However, in many musical sources the relative amplitude of the high frequencies may be considerably greater than shown in *Fig.* 5, so that excessive distortion may occur in recording unless the recording level is appropriately reduced.

The problem of excessive treble signal is often raised by the fact that in recording a phono disc (ultimately reaching the audiofan via a broadcast and then transferred by him to tape) the engineers may deliberately emphasize the treble range or a portion of it in order to impart a false brilliance that is frequently mistaken for high fidelity. It may be possible for the tape recordist to reduce this false treble boost, or some part of it, by means of the treble control in his control amplifier before the signal reaches the tape recorder. This can be done in those control amplifiers where the tape output jack is located after rather than prior to the tone controls.

A substantial part of the treble losses in recording are due to bias current. To reduce these losses and cut down the need for treble boost, it is expedient to reduce bias current. Unfortunately, reduced bias causes an increase in distortion. Were it not for the necessity of preserving treble response well out to the upper limits of the audio range, it would be feasible to increase bias and minimize distortion.

Distortion may also be traced occasionally to the desire to preserve response at the bass end. To maintain flat response to 50 cps and below, a slight amount of bass boost is often employed in recording. This boost reaches 3 db at 50 cps and increases as frequency declines. But, as mentioned previously, the tape is more susceptible to overloading at low frequencies than in the mid-range. If the sound source contains an abundance of very low frequencies at high amplitude, distortion may be appreciable unless, of course, care is taken to reduce the recording level.

Distortion and Equalization

Tape recorders require bass boost and treble boost, as indicated by *Fig.* 6, which shows the record-playback response of a tape machine at 7.5 ips in the total absence of equalization. The manner in which equalization is supplied affects distortion. For minimum distortion, bass boost should take place entirely or mainly in playback. Bass boost in recording imposes an excessive magnetic field on the tape. However, a number of tape machines employ half-and-half equalization, which consists of equal and ample amounts of bass boost in record and playback; and similarly for treble boost. The NAB standard, which applies to 15 ips recording, stipulates that bass boost shall take place essentially in playback.

For minimum distortion, it would be desirable to provide all or most of the necessary treble boost in playback. But this conflicts with signal-to-noise considerations. Playback treble boost emphasizes the noise of the playback amplifier, reducing the signal to noise ratio. Accordingly, it is the practice of quality tape recorders, in conformity with the NAB standard, to supply treble boost essentially in the recording process. Correspondingly, it becomes desirable to employ a pattern of equalization which minimizes the treble boost required in recording, thereby minimizing distortion.

The pattern of equalization revolves about the choice of a turnover frequency.

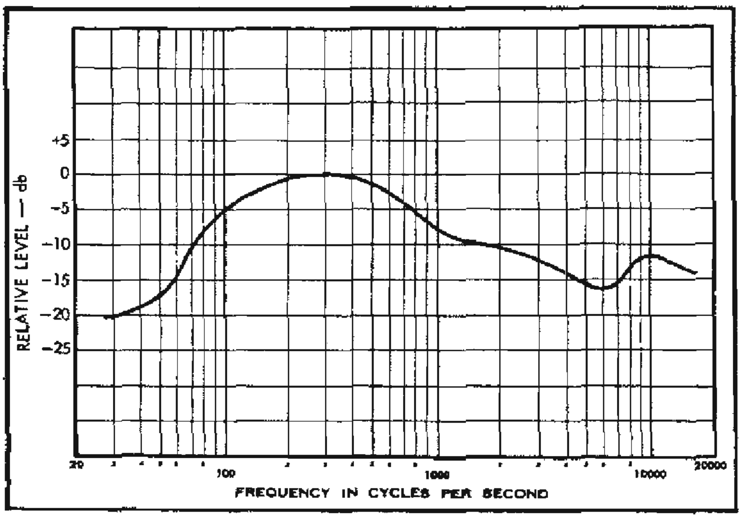

Fig. 5. Relative peak amplitude of various frequencies for a typical orchestral selection.

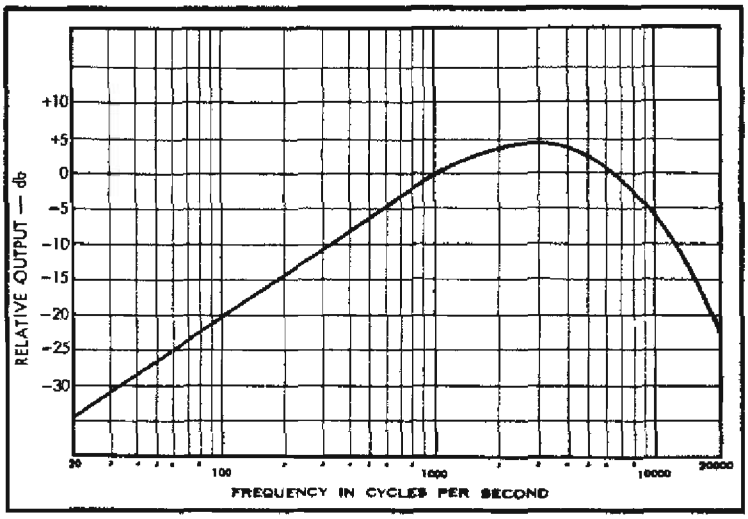

Fig. 6. Typical unequalized record-playback response of a tape machine operating at 7.5 ips.

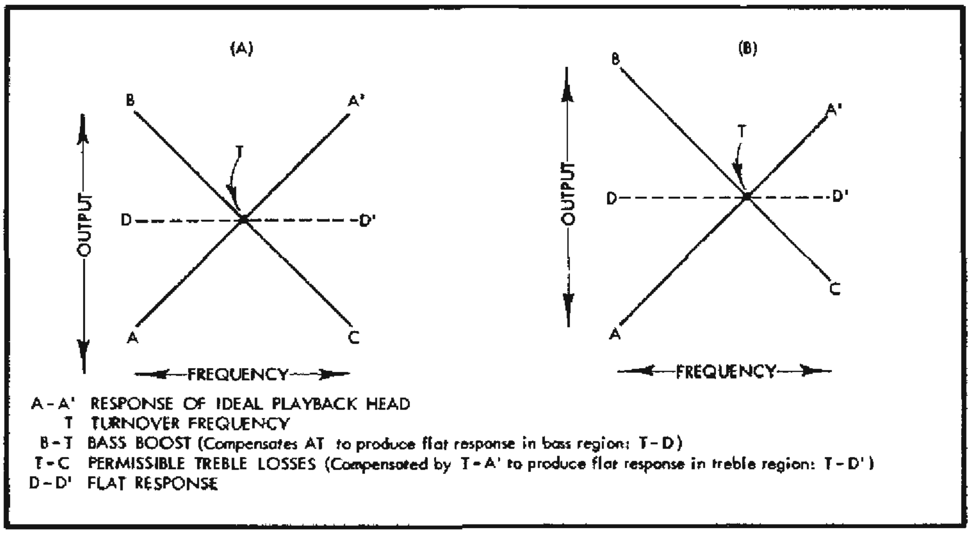

Fig. 7. Patterns of equalization in a tape machine.

It is possible to employ varying patterns of equalization, each with a different turnover frequency, entailing different amounts of treble boost and therefore different amounts of distortion.

This may be explained with the aid of *Fig.* 7. Output voltage of an ideal playback head rises steadily with increasing frequency, as shown by curve A-A′. This upward sloping line may be viewed as either treble boost or bass cut, depending upon our standard of reference, namely the turnover frequency, which is labeled T. The portion of the line above T may be said to represent treble boost, while the portion below T represents bass cut.

If the turnover frequency is low, as at (A), this signifies that bass boost begins at a low frequency. Thus the bass boost in playback, B-T, compensates the bass droop A-T of the playback head. And the rising response T-A′ of the playback head compensates the net recording loss (after treble boost) T-C. The large rise of T-A′ in (A)—resulting from the choice of turnover frequency—signifies that substantial treble losses are permissible in recording. This in turn means that less treble boost need be used in recording, which leads to less distortion.

(B) in *Fig.* 7 represents a different pattern of equalization, one with a high turnover frequency. Consequently the playback head supplies less treble boost compared with the scheme of things in (A). This signifies that less treble loss is permissible in recording, thereby necessitating more treble boost in recording and greater distortion. (On the other hand, the equalization pattern of (B) permits a better signal-to-noise ratio because more signal is recorded on the tape and because there is greater de-emphasis of the treble frequencies—the noise region—in playback.)

The pattern of equalization is not a matter for the tape recordist to decide. It is an industry decision. At the time of writing the question of suitable equalization patterns (turnover frequencies) for the tape speeds principally in home use, namely 7.5 and 3.75 ips, was still unsettled and undergoing discussion by industry committees. But it does not appear that the equalization patterns ultimately settled upon will vary greatly from those in present use. There is a good chance that the turnover frequency (at which bass boost commences) of 3180 cps commonly used at 7.5 ips will become an official standard. In the case of the 3.75 ips speed, turnover frequencies of either 795 cps or 1590 cps have often been used. At the time of writing there was a proposal before the industry to use a turnover of 1326 cps. This kind of compromise would permit playing older tapes with a fairly minor deviation from flat response; and this kind of deviation could be corrected fairly well by means of audio system tone controls.

Distortion and Bias Current

Figure 8 shows how harmonic and IM distortion vary with changes in bias current. We are concerned with the area to the right of the −7 db point; it is not practical to record in the area to the left—that is, at small values of bias current—because somewhat less signal is then recorded on the tape. Restricting ourselves to the practical area of operation, the following conclusions can be drawn from *Fig.* 8.

1. With increasing bias current, distortion declines steadily to a minimum level and eventually rises again.
2. IM distortion is greater than harmonic distortion.
3. The changes in IM distortion with changes in bias current are sharper than the changes in harmonic distortion.
4. The greater the signal level recorded on the tape the more critical is the bias setting for minimum distortion.
5. An increase in signal level produces the least increase in distortion when bias is set for minimum distortion.

It is apparent from the foregoing that proper bias setting is of great importance in minimizing distortion. On the other hand, as brought out before, it is necessary to take into account that treble losses increase as bias is increased, and that such losses grow more severe as tape speed is reduced.

Distortion and Tape Speed

To reduce treble losses and make it possible to achieve frequency response approximating high fidelity requirements, it is necessary at certain tape speeds to reduce bias below the point (*Continued on page 70*)

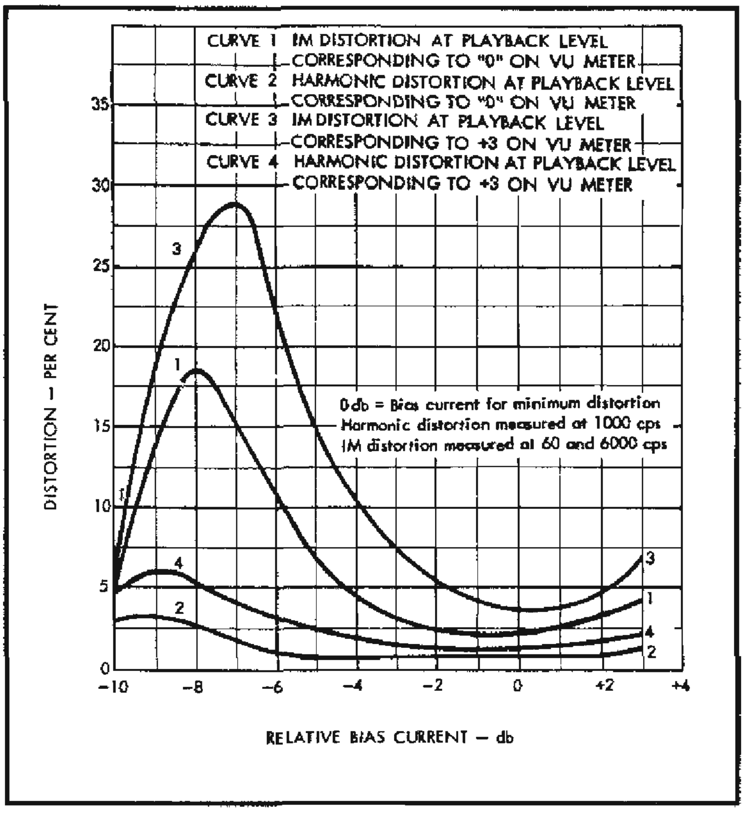

Fig. 8. Variation of tape distortion with changes in bias current.

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(from page 40)

corresponding to minimum distortion. To prevent excessive distortion, it therefore becomes necessary to reduce recording level somewhat.

At 15 ips it is generally feasible to set bias current for minimum distortion without impairing frequency response. At 7.5 ips it is usually necessary to settle for a value of bias current somewhat below the minimum distortion point. At 3.75 ips it is necessary to employ bias current well below that which achieves minimum distortion. And the situation grows considerably worse at still lower speeds.

One might ask: Why not compensate the increased treble losses at slow speeds by greater treble boost in recording instead of by a reduction in bias current? may be due to accumulation of tape oxide or other material on the head, or to a tape that is not sufficiently polished or lubricated.

Wow and flutter produced by the tape transport mechanism may also be considered forms of distortion. Wow is apparent as a quavering in the pitch of a The answer is that greater treble boost would tend to overload the tape.

Mechanical Distortion

Distortion may be due to mechanical rather than electronic factors. If the tape does not pass smoothly over the heads, the result may be what is known as modulation distortion. Friction between the tape and the head may cause the tape to undergo a sort of vibratory action, akin to the effect that a bow has upon a violin string. The result is that the frequencies being recorded or played back are modulated at the vibration frequency. Thus the vibration frequency or frequencies are impressed on the audio signal and are manifest as distortion. Friction between the tape and the heads prolonged tone. Flutter, consisting of speed changes that take place hundreds or even thousands of times per second, serves to modulate the audio signal, so that the flutter frequency or frequencies become the distortion products. Accordingly, the sound tends to take on a grainy quality. Æ

More About Tape Heads

Tape recorder performance depends very largely on the quality of the heads. Understanding their functioning and the factors which affect their performance will aid in selecting a recorder.

HERMAN BURSTEIN*

In an earlier article in this series—"How Many Heads for the Tape Recorder?"—tape heads were discussed principally in terms of their functions, relating to the advantages and disadvantages of using separate heads for record and playback or of using a single head for both purposes. In the present article, at the risk of a slight amount of repetition, we should like to go deeper into the subject of how the heads operate. It may be pointed out that while a good deal has appeared in the popular electronics literature to explain the operation of phonograph cartridges, comparatively little has been said about their important counterpart, tape heads. Yet a basic understanding of the playback head, as well as of the record and erase heads, can prove of substantial value to the audiofan concerned with the purchase of a tape machine, with replacing heads, with maintenance, or with simple repairs.

In most home machines the same head is used for both recording and playback. However, to facilitate discussion, we shall speak of record and playback heads as separate units.

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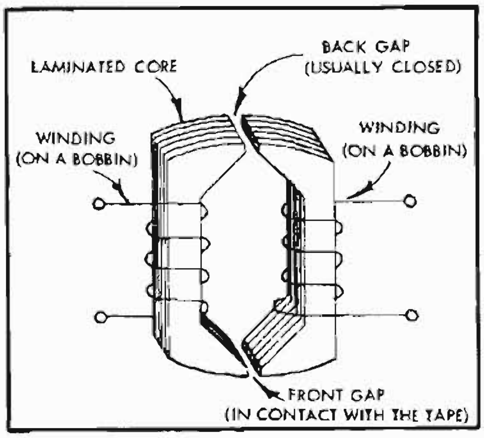

Fig. 2. Construction of a laminated tape head.

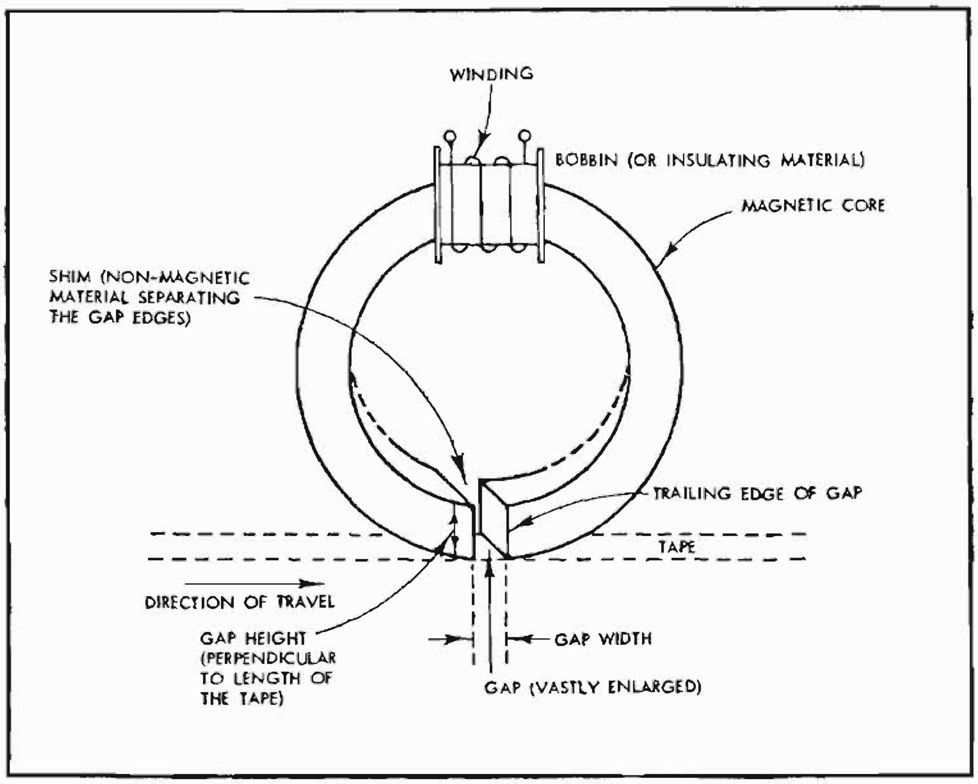

Fig. 1. Basic structure of the tape head.

Structure of the Heads

Record, playback, and erase heads all have three basic elements, shown in *Fig.* 1: a core of magnetic material, a gap in the core, and a winding around the core. The head is enclosed in a protective housing made partly of magnetic material to prevent the coil from picking up hum from external sources such as motors and transformers.

The tape and the head make contact at the gap. In the case of the record and erase heads, a magnetic field passing through the core enters the tape at the gap and either causes a signal to be recorded on the tape or erases a previously existing signal. In the case of the playback head, a magnetic field on the tape enters the core through the gap.

Cores are sometimes laminated, as in *Fig.* 2, and sometimes non-laminated, as in *Fig.* 3. The better, and usually more

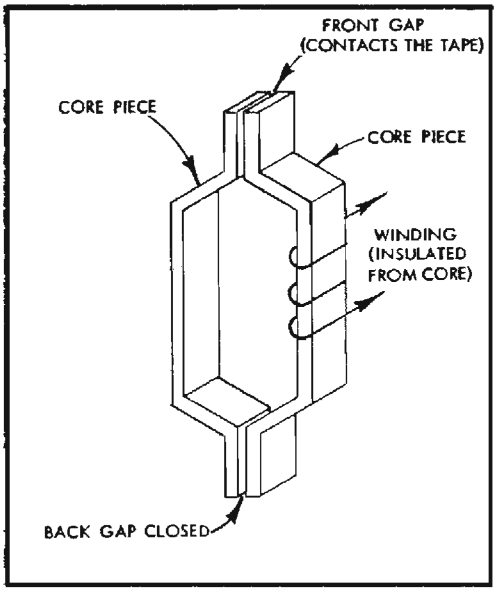

Fig. 3. Construction of a non-laminated tape head.

expensive, heads are generally laminated and have more signal output because they contain a greater volume of magnetic material. The laminations reduce certain magnetic losses (eddy-current losses), which can produce an appreciable drop in treble response.

For maximum efficiency and to cancel hum, most heads employ two windings, one on either side of the gap, as in *Fig.* 2. In a playback (or record-playback) head the windings are connected in series for maximum voltage, output, which is the important thing in playback in order to keep the signal level well above the noise and hum produced by the tape playback amplifier. In the case of record and erase heads, however, the windings are usually connected in parallel, because maximum current-carrying ability then becomes the important consideration in achieving efficient operation. This is the kind of thing to keep in mind if one is replacing a head and has the option of wiring the windings in series or parallel. *Figure* 4 illustrates the series connection, and *Fig.* 5 the parallel one.

The playback head should have a large number of winding turns in order to produce high signal output, which increases with number of turns. The record and erase heads, on the other hand, should have a relatively small number of turns to permit the easy passage of current through the winding. This is one of the conflicts involved in using the same head for recording and playback.

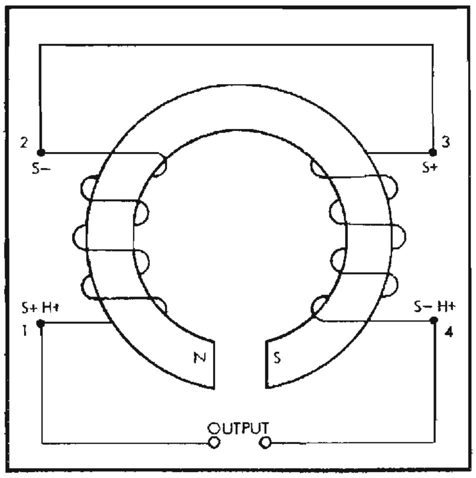

Fig. 4. Series connection of dual windings.

The playback head must have a very narrow gap in order to reproduce the high frequencies. Recording, however, does not require a narrow gap. While successful recording is possible with the extremely fine gaps used for playback, some as narrow as .00009 in., a relatively wider gap, about .0005 in., tends to be optimum. Erasure definitely requires a wide gap, about .0005 to .001 in. Thus the magnetic field emanating from the core can span a substantial portion of the tape and erase effectively.

Operation of the Record Head

Figure 6 illustrates what happens when audio current is fed to the wind-

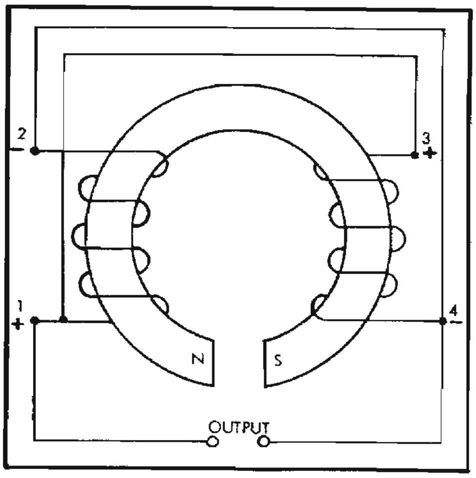

Fig. 5. Parallel connection of dual windings.

ing of the record head. This current produces a magnetic field in the core. Since the tape bridges the gap in the core, the magnetic field flows through the tape, which offers less resistance to this field than does the gap. The tape becomes magnetized in accordance with the variations of audio current. This magnetization continues until the instant the tape leaves the trailing edge of the gap (see *Fig.* 1); the magnetization remaining on the tape corresponds to the tape's magnetic condition at this instant. For optimum results, the trailing edge must be as straight and sharp as possible. Here is one of the principal differences between mediocre and high-quality heads.

The amount of audio signal that can be fed to a record head before tape distortion becomes excessive—about 3 per cent harmonic distortion—varies among heads of different manufacture. Should

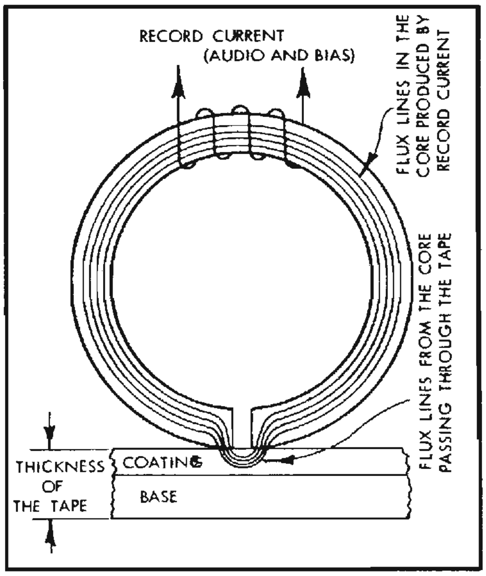

Fig. 6. Operation of the record head.

you substitute a head of a different brand for the one you have now, this factor must be taken into account. The manufacturer of the head can supply information on how much signal may be fed to the head and how this can be measured by the service technician or possibly yourself.

When audio current is fed to the record head, a high-frequency current, called bias, is also fed to the head, as shown in *Fig.* 7. Bias current plays a role analogous to that of a catalytic agent in a chemical process. In the absence of bias current, distortion on the tape would be intolerable and the signal level recorded on the tape would be very low. Up to a point, which ordinarily is not reached at 7.5 ips and lower speeds, the greater the bias current the less is the distortion. It might seem, therefore, that one merely has to crank up the bias current until distortion is at a minimum. Unfortunately, bias current makes the record head behave in the same manner as an erase head, particularly at high frequencies. In short, excessive bias spoils treble response. Therefore bias current must be chosen so as to achieve a satisfactory combination of good treble and low distortion.

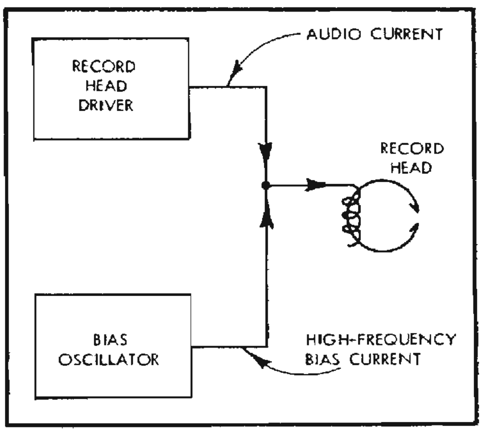

Fig. 7. Supplying audio and bias current to the record head.

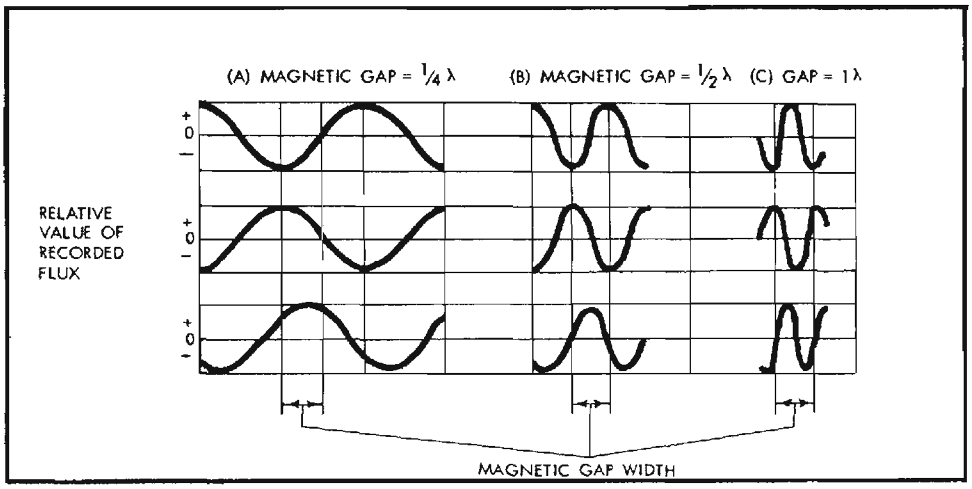

Fig. 10. Magnetic potentials scanned by the gap of a playback head for three wavelengths, (A), (B), and (C), at different phases of the waveform, (1), (2), and (3).

The bias frequency and the harmonics of the audio frequency tend to "clash"—beat is the technical term—resulting in audible noises. To minimize this, the bias frequency should be at least four or five times the highest audio frequencies that the tape machine is capable of recording with moderate losses. Thus if the upper limit is 12,000 cps, the bias frequency should be at least 50,000 to 60,000 cps. Before purchasing a tape recorder, it pays to check what the specifications say about bias frequency.

Operation of the Playback Head

Operation of the playback head is essentially the converse of the record head. The magnetic field on the tape enters one gap edge, flows through the core, and comes out the other edge to re-enter the tape. The field passing through the core induces a voltage in the winding, which is then fed to the playback amplifier.

Assuming that all frequencies are recorded at equal strength on a tape, it is in the nature of a playback head to produce increasing voltage output as frequency rises. As frequency doubles (goes up one octave), head output also doubles (increases 6 db). Hence the rising characteristic of the head is termed a 6-db-per-octave line. The rise continues until treble losses due to gap width take effect. The net result is shown in *Fig.* 8, based on a tape speed of 7.5 ips, a gap width of .00025 in., and the assumption of signals of equal strength on the tape at all frequencies.

Why does head output increase with frequency? A voltage is induced in the winding as the result of a *changing* magnetic field. The greater the number of changes per second, the greater is the induced voltage. The field is that which flows from the tape into the core of the head. This field changes along the tape in correspondence with the audio signal that was recorded. A high-frequency signal entails more changes (cycles) per second, or per inch of tape, than a low-frequency signal. Therefore the variations in magnetic field representing high frequencies induce a greater voltage than do the variations representing low frequencies.

The narrower the gap, the more ex-

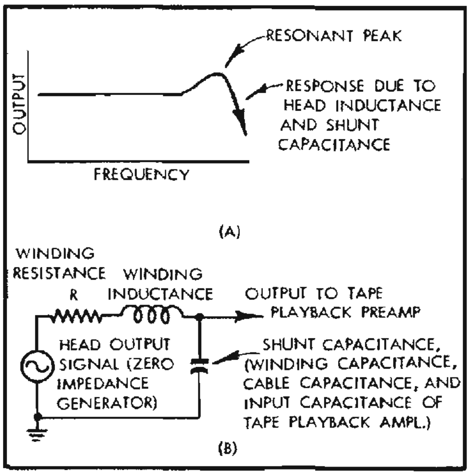

Fig. 11. Effect of shunt capacitance upon playback-head output. A shows frequency response characteristic and (B) is the electrical equivalent circuit of the head.

tended is the treble response of the playback head at any given tape speed. If tape speed is reduced, the gap must be proportionately narrower to maintain treble. Thus at 3.75 ips the gap must be half as wide as at 7.5 ips for equally good performance at the high end. Whereas a head with a gap of .0002 to .00025 in. is considered adequate for reproducing a tape at 7.5 ips, a gap of about .0001 in. is called for at 3.75 ips.

An approximation to the useful frequency response of a playback head is given by the formula $f = S/2G$, where f is frequency in cps, S is tape speed in ips, and G is gap width stated as a fraction of an inch. Thus at 7.5 ips a gap width of .00025 in., according to this formula, indicates useful response to about 15,000 cps.

The formula allows for the fact that "magnetic" width of the gap tends to be somewhat greater than the physical dimensions of the gap, thereby reducing high-frequency response. The sharper and straighter the edges of the gap, the less is the difference between the magnetic width and the physical width of the gap. In a high-quality head, the magnetic gap will tend to be about 10 per cent greater than the physical width. In a poor head, the difference may be much greater than this. It can be understood, therefore, why it is possible for a good head with a physical gap of .00015 in. to achieve better high-frequency response in playback than a poor one with a gap of, say, .0001 in.

Table 1 shows the approximate upper response limit for the gap widths most commonly encountered, at the tape speeds in common use. An important note of caution is in order here. The table only shows the maximum frequency that a head can *play back*. This does not signify that such a frequency can necessarily be recorded at the indicatetd speeds. There are severe recording losses that impose a limit on treble response; the higher the frequency or the slower the tape speed, the greater is the recording loss. To illustrate, the table shows that at 3.75 ips a modern head with a .00009 in. gap permits response to about 21,000 cps. However, high-frequency losses in recording are so great at this speed as to prevent such extended response in the present state of the art. (On the other hand, continued progress in the audio art may permit recording to 20,000 cps at 3.75 ips in a few years.)

Why does successful reproduction of the high frequencies require a narrow gap? To provide the basis for the answer, let us consider *Fig.* 9, which shows that when a sine wave is recorded on tape, this is equivalent to arranging the magnetic coating into a series of bar magnets. Each bar corresponds to half of a sine wave, and it has a north pole and a south pole. When a low frequency is

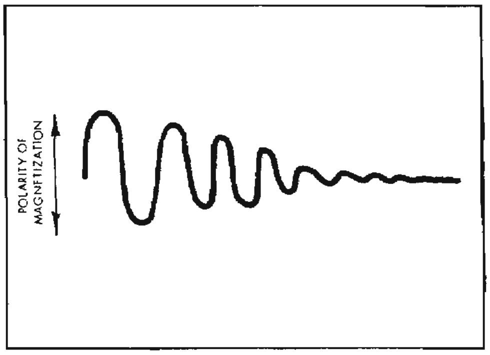

Fig. 12. Magnetization experienced by a tape particle during the erase process.

recorded, comparatively few bar magnets are recorded on each inch of tape. When a high frequency is recorded, many bar magnets are recorded in the same space; or, to put it differently, the magnets become shorter. In technical terms, the recorded wavelength is long at low frequencies and short at high frequencies.

Figure 9 shows the gap of the playback head in contact with the equivalent

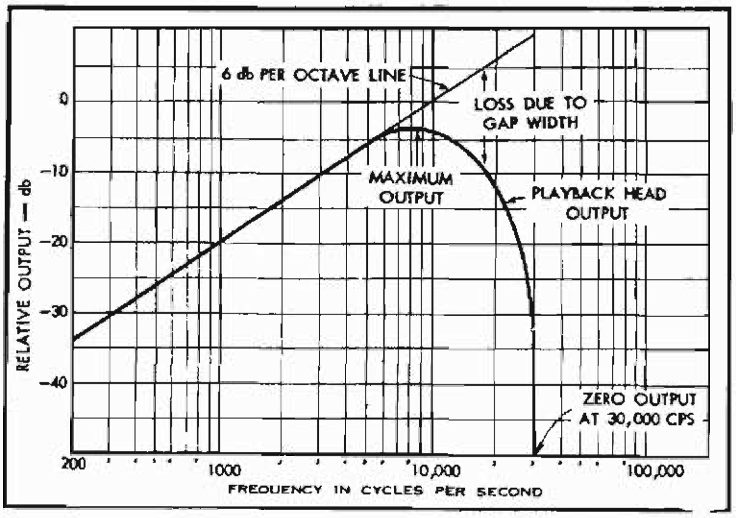

Fig. 8. Theoretical output of a playback head with a .00025-in. gap at 7.5 ips.

bar magnets on the tape. As the succession of magnets go by the head, they present a changing magnetic field, which induces a voltage in the winding of the head. However, it is not enough that there be a changing magnetic field. It is also necessary that there be a difference in the magnetic intensity of the field at each edge of the gap. As shown, one edge of the gap is half-way between the north and south poles of the first bar magnet, whereas the other edge is nearer to the south pole. Hence there is a difference in magnetic intensity, called a magnetic potential.

If frequency increases sufficiently, meaning if the recorded wavelength grows short enough, a magnetic potential will no longer be presented to the gap. This happens when the (magnetic) gap width becomes equal to one wavelength. When there is no magnetic potential across the gap, the head produces no signal output. Output of the playback head falls very rapidly as gap width approaches one wavelength.

This can be clarified with the aid of *Fig.* 10. This shows the relationship of the gap width to three successively higher recorded frequencies; i.e., to three successive-shorter wavelengths. In (A), the gap is equal to $\frac{1}{4}$ wavelength; in (B), to $\frac{1}{2}$ wavelength; in (C), to 1 wavelength. It may be seen in (A) and (B) that a magnetic potential will appear across the gap of the playback as various portions of waveform pass the gap. In (C), however, no matter where the gap is in relation to the waveform, both edges of the gap are always at the same stage of the waveform, so that a magnetic potential never develops across the gap. Hence the head delivers no signal.

The narrower the gap, the higher is the frequency at which the gap approaches one wavelength. Therefore the point at which response begins to fall off is moved upward in the frequency range.

Record-playback heads and, particularly, heads intended only for playback have a large number of turns in order to maximize signal output. This raises a problem of treble losses due to capacitance across the head, such as the capacitance added by the cable from the head to the playback amplifier, and the input capacitance of the amplifier. Moreover, the winding itself has capacitance, namely that between turns of the coil; the more turns, the greater is the winding capacitance. As shown at (B) in *Fig.* 11, the inductance of the playback head and the total shunt capacitance (of the winding, the cable between the head and the playback amplifier, and the amplifier) form a low-pass filter, producing a decline in treble response beyond the resonant frequency determined by the values of the winding inductance and of the shunt capacitance. The greater the winding inductance, which increases with turns, and the greater the shunt capacitance, the lower is the frequency at which the treble drop begins. This frequency can occur within the audio range. To minimize the danger of substantial treble loss, the cable from the head to the tape amplifier should be of minimum length and minimum capacitance per foot.

On the other hand, the resonant peak produced by the head inductance and shunt capacitance, shown at (A) in *Fig.* 11, can be put to work to *improve* treble response. This is done by adjusting the shunt capacitance so that the peak occurs at a suitable point, say at about 14,000 or 15,000 cps. Thereby response is given a lift at the top end of the useful audio range, with the fall-off occurring beyond 15,000 caps, where it does little or no harm. For the purpose of so adjusting the shunt capacitance, manufacturers of tape machines sometimes place a capacitor of proper value across the tape playback head.

Operation of the Erase Head

The erase head is powered by a high-frequency current, generally obtained from the same oscillator that supplies bias current to the record head *Figure* 12 illustrates what happens to a given particle of tape in the erasing process. At first the particle is heavily magnetized in a given "direction"—say, with its north pole on the right. This destroys

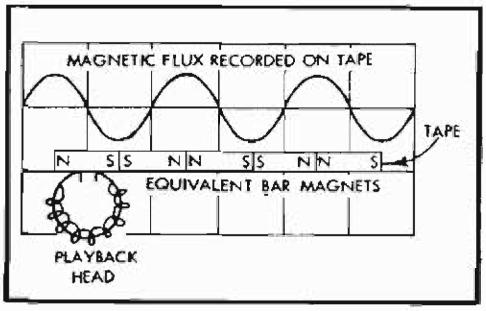

Fig. 9. Equivalent bar magnets produced by a sine wave recorded on tape.

any previously existing magnetic pattern on the tape. A brief instant later, the particle is magnetized in the opposite direction, with its north pole to the left. These alternate magnetizations grow weaker and weaker is the particle recedes from the head, until the magnetization reaches zero or very close to zero.

Since the erase head is powered by the same oscillator that supplies bias to the record head, the frequency of the erase current is typically between 30,000 and 100,000 cps. Erasure tends to be less effective at the higher frequencies. Therefore it would be desirable to use a low erase frequency, for example 30,000 cps. But this conflicts with, and must give way to, the desirability of a high bias frequency for recording purposes, namely to avoid audible beats between the bias frequency and the upper harmonics of the audio signals. Consequently the erase frequency tends to be upward of 50,000 cps.

A few of the least expensive home machines employ a permanent magnet instead of the electromagnetic type of erase head just described. The basic principle of operation is the same. A series of two or three magnets, in one housing, alternately differ in magnetic polarity. They subject the tape to a changing magnetic field, and one that gradually decreases in strength, as the tape moves past and

(*Continued on page 57*)

TABLE 1
APPROXIMATE TREBLE RESPONSE LIMITS FOR VARIOUS GAP WIDTHS AT TAPE SPEEDS IN COMMON USE

SPEED	GAP WIDTH					
ips	.0005	.00025	.00020	.00015	.00009	in.
15	15,000	30,000	37,500	50,000	83,000	cps
7.5	7,500	15,000	18,750	25,000	41,700	cps
3.75	3,750	7,500	9,375	12,500	20,800	cps
1.875	1,875	3,750	4,700	6,250	10,400	cps

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away from these magnets. In another form of permanent magnet erase head, a single magnet runs diagonally from the lower to the upper edge of the track, thus subjecting each particle on the track to a magnetic field which is changing in polarity and strength.

Permanent magnet heads generally do not erase as well as electromagnetic heads, and they tend to leave noise on the tape. Orientation of the head with respect to the tape is critical. The head must be automatically moved away from the tape when the recorder is in the playback mode. Hence the permanent magnet head is seldom used. Æ

Equalization

The fundamental facts about the elements which affect frequency response of tape recorders should be understood by any serious tape fan, along with the reasons for equalizing and the types of equalization required. Here they are—

HERMAN BURSTEIN*

AT VARIOUS TIMES in earlier articles we have touched upon the subject of equalization in tape recorders, for example in connection with frequency response, distortion, and signal-to-noise ratio. The purpose of the present article is to bring together the various aspects of the subject and present in one place a composite picture of this complex topic. Without an understanding of equalization the tapefan cannot have a full basic comprehension of the tape recorder.

Perhaps the best way to launch the discussion is to consider what the record-playback response of the tape recorder would be like if there were no frequency compensating circuits in it. A typical unequalized response curve appears in *Fig.* 1 for a machine operating at 7.5 ips, with bias current at a value normally used. The need for frequency-compensating circuits is instantly apparent, because in their absence there would be grave deficiencies both in bass and treble. In other words, bass boost and treble boost are needed.

The following discussion will deal with the factors responsible for the bass and treble losses; with losses that vary with wavelength and tape speed compared with those that vary purely with frequency; with the optimum location of compensating circuits as between record and playback; with the NAB equalization curve; with possible variations in equalization at a given speed; with changes in equalization as tape speed is changed; and with the basic types of equalization circuits employed (in elementary terms).

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Record and Playback Losses

Following are the factors that account for the departure of the curve in *Fig.* 1 from flat response.

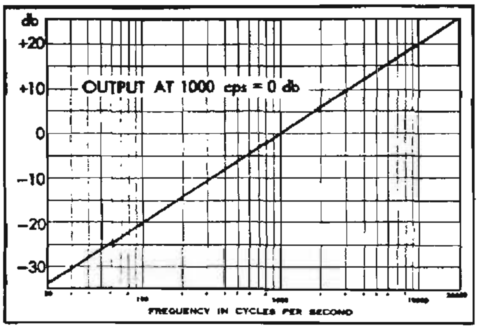

Fig. 2. Output of an "ideal" playback head, assuming equal recorded flux on the tape at all frequencies.

1. *Velocity Characteristic of the Playback Head.* Assuming a tape is recorded so that it contains equal amounts of signal (magnetic flux) at all frequencies, the response of an ideal playback head would be as in *Fig.* 2, that is, rising steadily with frequency. A tape head is called a "velocity device" because its output depends upon how rapidly the magnetic field presented by the tape to the head changes, in other words upon the velocity of this field. The higher the frequency, the greater is the rate of change of the magnetic field on the tape —i.e., the more cycles there are per second—and therefore the greater is the output of the playback head. This accounts for the sloping portion of the *left* side of *Fig.* 1, which is a straight line. (The line is perfectly straight only in theory. In practice, the line tends to wiggle a bit (contour effect) at the very low end and to decline more than is shown (wrap effect) due to interactions between the tape and the entire head. However, the line is sufficiently close to straight to be described in that manner.)

The following factors account for the

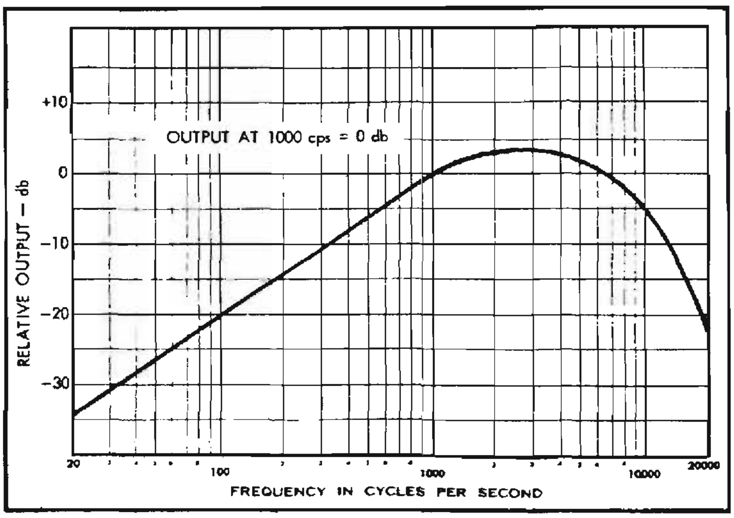

Fig. 1. Unequalized record-playback response of tape recorder, 7.5 ips, .00025-in head gap.

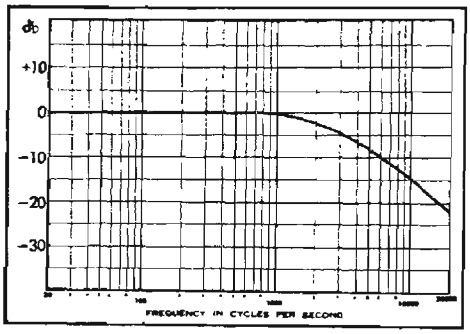

Fig. 3. Typical demagnetization loss in recording at 7.5 ips.

sloping portion of the *right* side of *Fig.* 1.

2. *Demagnetization Loss in Recording.* When a signal is recorded on the tape, the magnetic particles of the coating arrange themselves to form the equivalent of small bar magnets along the length of the tape. Each such magnet has a south pole and a north pole. The higher the frequency, the larger is the number of bar magnets formed in a given span of tape. That is, the shorter is each magnet. The north and south poles of a magnet tend to have a cancelling effect upon each other, and the closer the poles are to each other the greater is the cancelling effect, which is called demagnetization loss. Thus as frequency rises, demagnetization losses increase quite rapidly, resulting in treble losses such as shown in *Fig.* 3, based on a tape speed of 7.5 ips.

3. *Bias Erase in Recording.* Bias current, which is applied to the record head in order to reduce distortion and increase the amount of signal recorded on the tape, also causes this head to behave in the same manner, but to a lesser degree, as when the bias frequency current is applied (in much greater amount) to the erase head. In short, the record head behaves slightly as an erase head. The recorded frequencies most susceptible to erasure are the treble ones, because they do not penetrate the tape as deeply as the middle and low frequencies; that is, the high notes lie closer to the surface of the tape. The result is to produce a relative drop in treble response. *Figure* 4 shows typical treble loss at 7.5 ips due to the bias current normally employed at this speed.

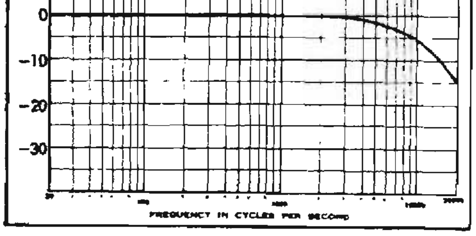

Fig. 4. Typical bias-erase loss in recording at 7.5 ips.

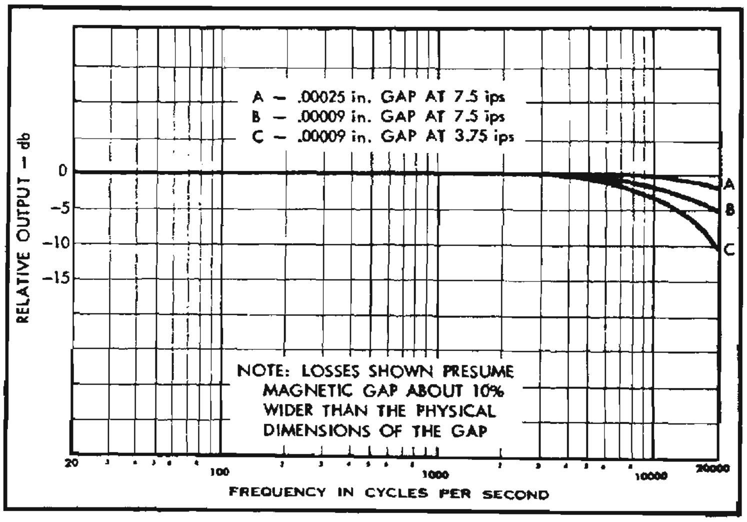

Fig. 5. Losses due to gap of the playback head.

4. *"Iron Losses" in the Record and Playback Heads.* The magnetic fields passing through the core of a tape head induce what are known as eddy currents in the head. Also, the core to an extent resists magnetization and demagnetization, this resistance being referred to as hysteresis. Both phenomena represent a dissipation of energy. This dissipation of energy increases with frequency, resulting in treble losses. However, in modern high-quality heads, the treble losses

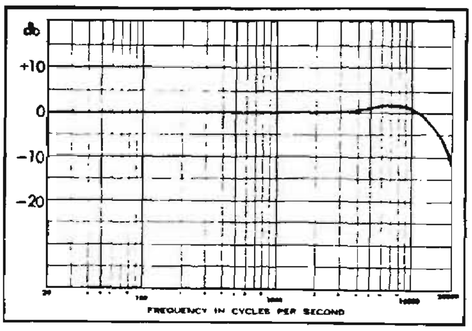

Fig. 6. Typical high-frequency loss due to excessive capacitance across the playback head.

due to eddy currents and hysteresis—collectively termed iron losses—are quite small and often entirely negligible, perhaps 1 db or so at 15,000 cps.

5. *Gap Loss of the Playback Head.* The upper response limit of the playback head depends upon the width of its gap. The narrower the gap, the higher are the frequencies that the head is capable of reproducing at a given tape speed. Curve (A) in *Fig.* 5 shows the declining treble response at 7.5 ips for a playback head with a gap of .00025 in.) which used to be fairly standard. Curve (B) shows the virtually negligible loss at 7.5 ips with a modern head having a gap of but .00009 in. On the other hand, even with a gap of .00009 in. there is a significant loss at 3.75 ips, as shown by Curve (C) in *Fig.* 5.

6. *Electrical Losses of the Playback Head.* The inductance of the playback head winding, together with capacitance across the head—capacitance of the winding itself, of the cable to the playback amplifier, and of the amplifier itself—can produce a significant drop in treble response, as illustrated in *Fig.* 6. Usually, however, in a well-constructed tape recorder the capacitances are kept low enough to avoid undesirable loss. Therefore it will be assumed in further discussion that treble loss due to this factor does not take place.

Figure 7 presents a composite picture of the various losses described above. It may be seen that when the various losses are added up the final result is the unequalized response curve of *Fig.* 1.

Wavelength Effect and Tape Speed

The preceding description of losses that take place in recording and playback has been couched in terms of the

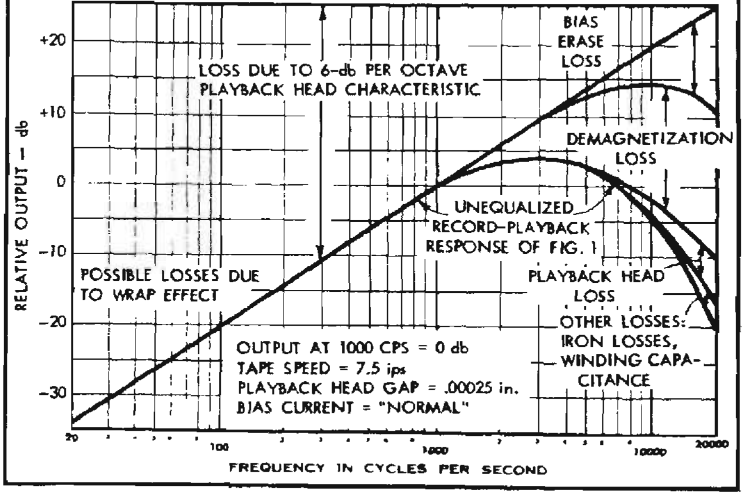

Fig. 7. Composite picture of factors responsible for unequalized record-playback response of Fig. 1.

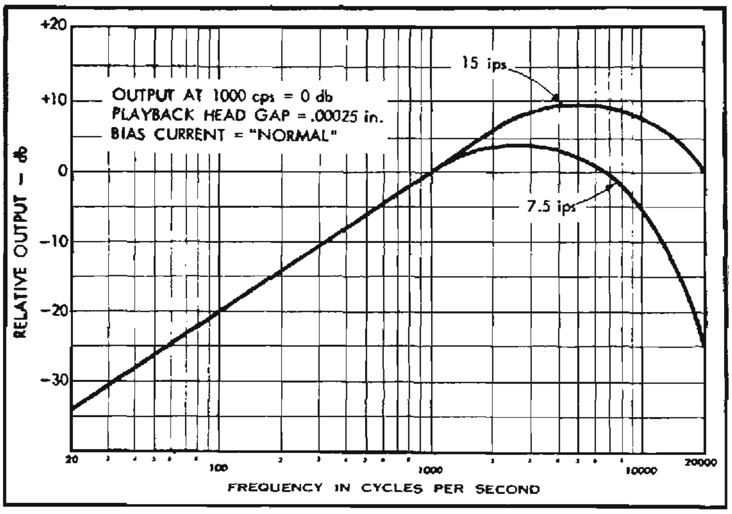

Fig. 8. Comparison of unequalized record-playback response at 7.5 and 15 ips.

variation of the loss with frequency at a given tape speed. Actually, however, a number of these losses vary with the wavelength of the signal recorded on the tape—that is, with the distance along the tape occupied by one cycle of the audio signal. The shorter the wavelength, the greater is the loss. Inasmuch as the wavelength grows shorter as frequency rises, at a given tape speed, the loss also increases with rising frequency. On the other hand, if tape speed increases, which also increases the wavelength, the loss diminishes; and vice versa with a decrease in tape speed. We see, therefore, that the type of loss in question does not remain the same at a given frequency. The amount of loss depends on both the frequency and the tape speed.

Treble losses that vary with wavelength—that is, with frequency and tape speed—are (1) demagnetization loss in recording; (2) bias erase in recording; (3) gap loss of the playback head. On the other hand, the iron and electrical losses of the record and playback heads vary purely with frequency, regardless of the speed at which the tape machine is operating.

To make clear how treble losses vary with tape speed, *Fig.* 8 shows the unequalized record-playback response of a tape machine at 7.5 ips and at 15 ips. When the speed is doubled, a given wavelength represents a frequency twice as great, that is, an octave higher. Hence the treble loss that occurs at a given frequency at the lower speed occurs at twice that frequency at the higher speed.

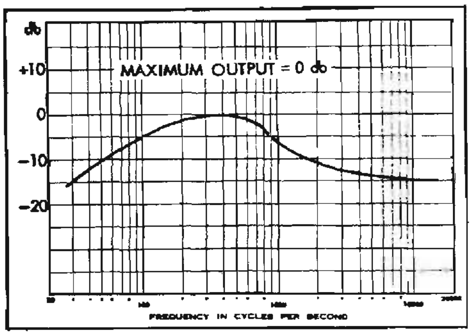

Fig. 9. Distribution of peak audio energy for a typical symphonic composition.

All other things remaining the same, a doubling of tape speed can improve performance, so far as treble response is concerned, from fair to excellent. For example, it could extend the upper response limit from 8000 cps at 3.75 ips to 16,000 cps at 7.5 ips.

Figure 8 shows us two important things with respect to equalization requirements. (1) The lower the tape

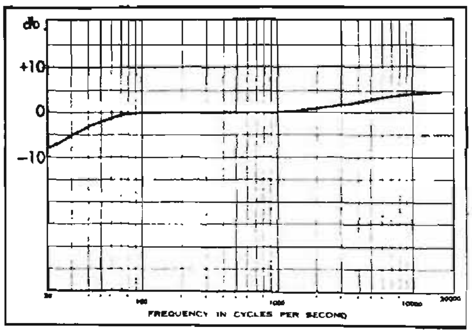

Fig. 10. Approximate relative permissible recording level at various frequencies at 7.5 ips.

speed the greater is the required treble boost. (2) On the other hand, the lower the tape speed, the less is the required bass boost. It may be seen in *Fig.* 8 that the point at which bass boost should commence is an octave lower at 3.75 ips than at 7.5 ips. This fact should be kept in mind in later discussion.

Location of Equalization Circuits

The designer of a tape recorder has the choice of incorporating the necessary bass boost and treble boost in the record amplifier, in the playback amplifier, or in both. The proper choice is that which takes into account not only considerations of frequency response but also considerations of distortion and of signal-to-noise ratio. Sometimes economy becomes important enough a factor to alter the picture.

The most logical place for bass boost is in playback. At 7.5 ips, well over 30 db of bass equalization is needed to achieve flat response at this end of the audio spectrum. At lower speeds, the amounts of bass boost required are still in the vicinity of 20 db or higher. Such quantities of bass boost, if employed in recording, would greatly overload the tape and produce tremendous distortion.

On the other hand, a slight amount of bass boost in recording is usually tolerable, because in most music and speed there is a drop in audio energy at the very low end. *Figure* 9 shows a typical distribution of peak audio energy for an orchestra playing a typical symphonic composition. Hence it is desirable to taper off the playback bass boost at the extreme low end, and supply bass boost in this region in recording instead.

The most logical place for treble boost is in recording. At 7.5 ips, over 20 db of treble boost is usually required to achieve flat response. As much or more is needed at lower tape speeds. If this were supplied in playback, it would inordinately accentuate tape hiss and noise of the playback amplifier. However, if considerable treble boost is used in recording, does this not overlead the tape and cause distortion? The answer is that two factors permit substantial treble boost in recording before the overload point is reached. (1) As indicated in *Fig.* 9, there is a substantial drop in audio energy at the high end of the audio spectrum, offsetting the rise in recording level due to treble boost. (2) For the same amount of distortion, one

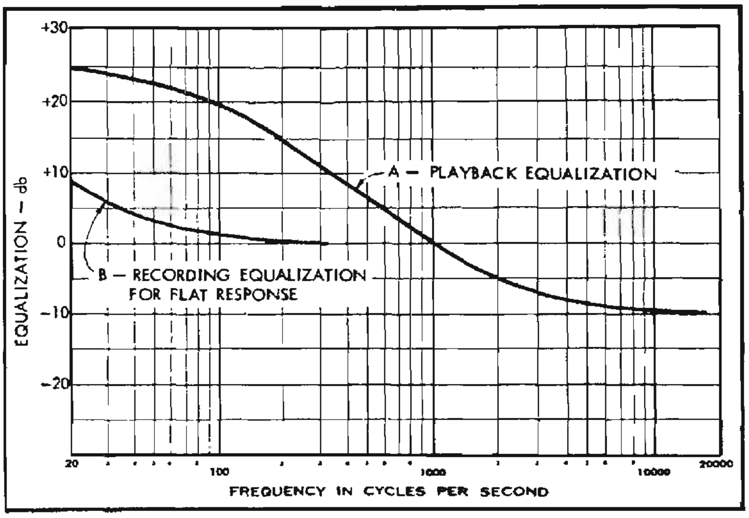

Fig. 11. NAB tape equalization.

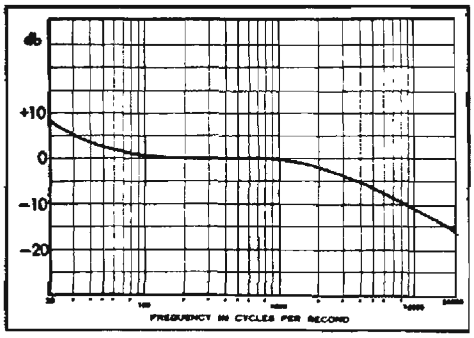

Fig. 12. The NAB recording characteristic (signal to be recorded on the tape).

can record at a somewhat higher level as frequency increase, as illustrated in *Fig.* 10.

To the extent that gap loss and perhaps iron losses occur, additional treble boost is required above the amount needed to compensate for demagnetization loss and bias erase. Such additional treble boost should be incorporated in the playback amplifier for two reasons: (1) to avoid carrying total treble in recording to the point where excessive distortion results; (2) because the gap loss and iron losses vary with the individual head used for playback, and it is not easy to anticipate when recording a tape on one machine—referring here primarily to commercial tapes—what kind of head will be used to play the tape on another machine.

While economy is a factor in any tape recorder, it becomes a dominant factor in low-price tape machines, and a compromise is then often made with optimum principles of equalization. Instead of providing nearly all the bass boost in playback and nearly all the treble boost in recording, the bass boost is divided equally between record and playback, and so is the treble boost. Thus the same equalization circuitry may be used for recording and playback, which simplifies circuitry and reduces cost.

NAB Equalization

NAB (formerly NARTB) equalization conforms to the principles for optimum location of equalization circuits, as follows.

1. It stipulates that bass boost shall take place principally in playback, as shown by Curve A in *Fig.* 11. This curve tapers off at 50 cps; if perfectly flat response is desired, the remainder of the bass boost is supplied in recording, as shown by Curve B.
2. It stipulates that treble boost shall take place principally in recording, except to the extent required to compensate losses of the playback head, namely gap loss and iron losses. The latter losses shall be compensated, if necessary, in playback.

NAB equalization does not provide for a specific record equalization (treble boost) curve. Instead, it simply states that record treble boost shall be such that, in conjunction with the playback equalization of *Fig.* 11, record-playback response shall fall within certain limits, as follows: flat within ±1 db between 100 and 7500 cps; not more than 1 db or 4 db down at 50 cps; and not more than 1 db up or 4 db down at 15,000 cps.

While the NAB standard does not provide for a specific *equalization* characteristic in recording, it does provide for a specific recording characteristic *on the tape,* assuming the signal fed into the tape recorder is of equal magnitude at all frequencies. This recording characteristic appears in *Fig.* 12. The characteristic of *Fig.* 12 and the playback equalization curve of *Fig.* 11 bear a complementary relationship to each other. That is, given one, the other can be deduced by allowing for the fact that output of the playback head rises 6 db per octave (per *Fig.* 2). Since it is much easier to measure playback equalization than the magnetic field on the tape, NAB equalization is stated as a playback equalization curve instead of in terms of the amount of magnetic flux to be recorded on the tape at each frequency.

NAB playback equalization was originally designated as a standard only for the 15 ips speed. However, with improvements in tape machines and in tapes, NAB equalization was also found practical at 7.5 ips, and by common acceptance it has become virtually standard for the latter speed as well. At speeds below 7.5 ips, it is desirable to employ less bass boost than specified by NAB. Thus at these lower speeds it is customary to find a similar playback bass boost curve, but commencing at a frequency that is substantially lower than in *Fig.* 11. At these lower speeds the principles of optimum equalization are followed for the most part, at least in the better tape recorders.

It would be possible for NAB equalization to be stated in terms of a specific recording equalization curve were it not for the following two factors: (1) At a given tape speed, the amount of bias current employed varies somewhat from one tape recorder to another. Accordingly, there is a variation in bias erase and in the amount of record treble boost that is needed. (2) Iron losses, if any, in the record head vary from one tape recorder to another. Since these are supposed to be compensated in the record amplifier, the amount of record treble boost that is required will vary from one tape machine to another.

In view of the fact that NAB equalization has been adopted for 7.5 ips, it becomes all the more difficult to prescribe a specific equalization characteristic for recording. At 7.5 ips the treble losses due to demagnetization and bias erase are much greater than at 15 ips, so that much more treble boost—at least 10 db more—is needed when recording at 7.5 ips than at 15 ips.

Variations in Equalization at a Given Speed

Referring to *Fig.* 1, it would appear that playback bass boost should start—i.e. attain a rise of 3 db—at about 1000

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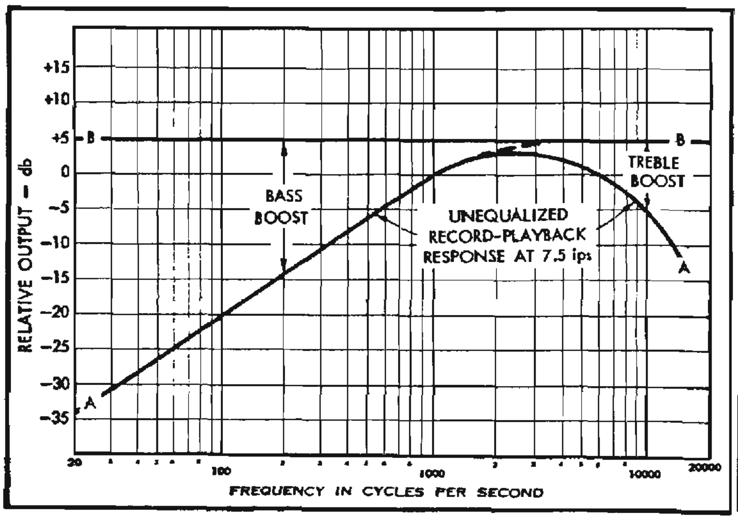

Fig. 13. Combination of relatively small amounts of treble and bass boost to achieve flat frequency response.

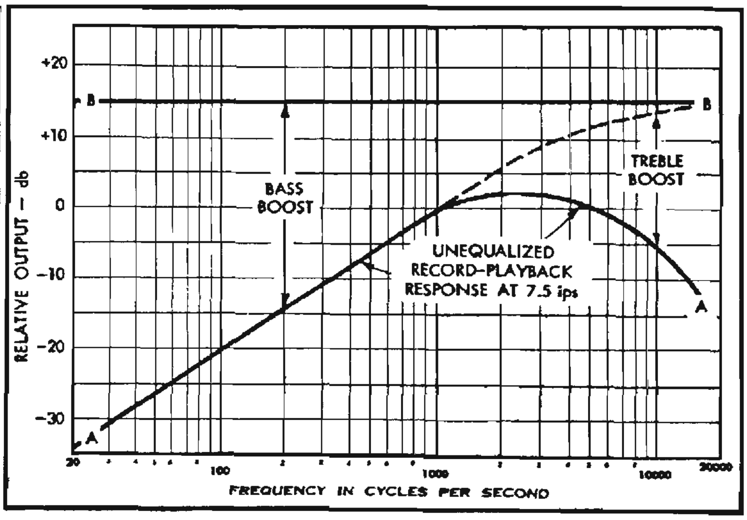

Fig. 14. Combination of relatively large amounts of treble and bass boost to achieve flat frequency response.

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cps. Yet, as shown in *Fig.* 11, NAB equalization calls for bass boost to start at 3180 cps. How does one reconcile this seeming contradiction? The answer, in broad terms, lies in the fact that, within a certain range, various playback equalization curves prove feasible at a given tape speed. In other words, the point at which bass boost reaches 3 db, known as the turnover frequency, can vary within a fair range and still permit relatively flat response through suitable adjustment of record equalization.

When the bass turnover frequency is changed, this in turn changes the requirements for record treble boost in order to achieve more or less flat response. If the bass turnover frequency is lowered, less record treble boost is needed. When the bass turnover frequency is increased, more record treble boost is required. In sum, the amounts of playback bass boost and record treble boost vary in the same directions. The advantage of more record treble boost is that more signal is recorded on the tape, resulting in a better signal-to-noise ratio; the disadvantage is that the greater amount of record treble boost raises the danger of excessive distortion.

Figures 13 and 14 serve to clarify how it is possible to employ various combinations of playback bass boost and record treble boost at a given tape speed. *Figure* 13 illustrates the situation where *relatively* little record treble boost and playback bass boost are employed. Curve A-A is the original unequalized record-playback response. Curve A-B represents record-playback response after application of treble boost in recording. The distance between Curves A-A and A-B represents the amount of treble boost that has been supplied. Curve B-B, which is a straight line, represents flat response after application of bass boost in playback. The distance between Curves B-B and A-B represents the amount of bass boost that has been supplied.

Now consider *Fig.* 14, where all the curves have the same meaning as in *Fig.* 13. Now, however, note that greater treble boost has been supplied in recording (distance between Curves A-A and A-B), so that more signal has been put on the tape, resulting in greater output at the high end in playback. And note that, in turn, greater bass boost is required in playback (distance between Curves B-B and A-B) in order to achieve flat response.

Change of Equalization with Speed

Possibly *Fig.* 13 or 14 or some intermediate combination of recording and playback equalization may be optimum in the sense of combining flat response with minimum distortion and maximum signal-to-noise ratio. At any given tape

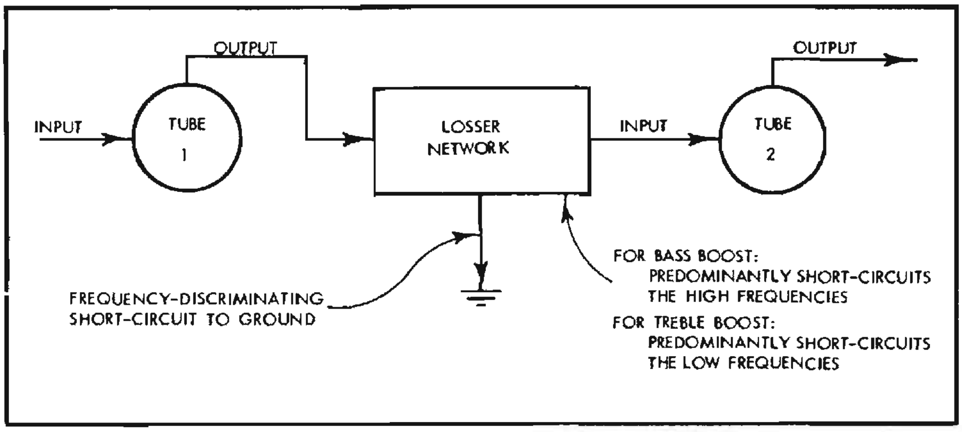

Fig. 15. How losser equalization works.

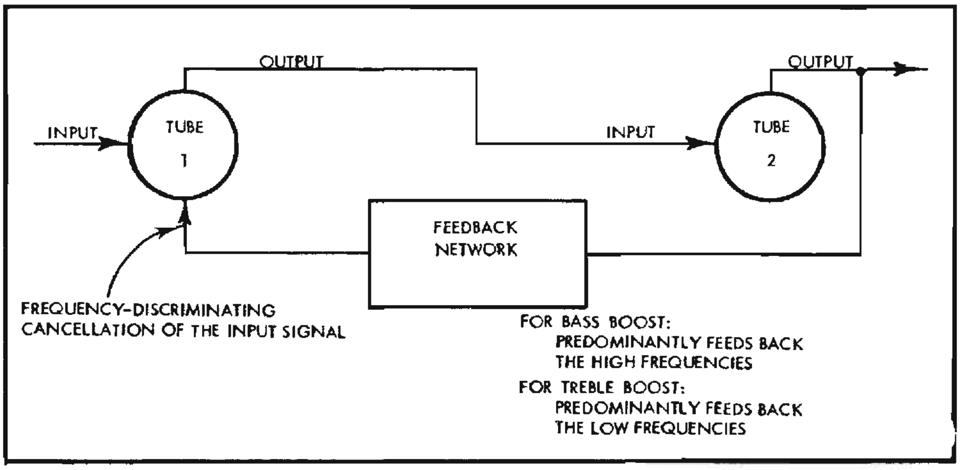

Fig. 16. How feedback equalization works.

speed there is only one optimum combination. Since a given playback curve implies complementary recording equalization that will achieve flat response, it may be stated simply that at each tape speed there is only one playback equalization curve which represents optimum performance.

The foregoing statement should not be taken too strictly. Thus it has been found that NAB playback equalization, considered ideal for 15 ips, can also be used very satisfactorily at 7.5 ips. True, a moderate departure from the NAB curve at 7.5 ips may yield better results in terms of distortion and noise; however, the improvement would be a slight one.

Going down the speed scale, namely to 3.75 ips and lower speeds, we arrive at situations where NAB equalization in playback would entail a serious sacrifice in performance. Thus a quality tape machine will accompany a change in tape speed with a change in playback equalization when the speed change entails going below 7.5 ips. In some tape machines, however, for reasons of economy, the same playback equalization is used for both 7.5 ips and 3.75 ips. The equalization employed may then be NAB or it may be an intermediate curve part way between the NAB one and the optimum curve for the 3.75 ips speed.

From earlier discussion it is obvious that as speed is reduced, greater treble boost is needed in recording to overcome the increased high-frequency losses. In other words, a reduction in tape speed also calls for a change in recording equalization. However, this is not always done. In some of the less expensive machines, the manufacturer may aim for response extending to about 15,000 or 16,000 at 7.5 ips, which means that without any change in record equalization response will extend to about 7500 or 8000 cps at 3.75 ips. Response extending to about 8000 cps surpasses the performance of most AM radios and permits a fairly good quality of reproduction, pleasing to many persons. Thus the manufacturer may consider that response to 8000 cps still gives the individual a good run for his money. (It isn't so many years since high-quality tape recorders offered response to only about 8000 cps at 7.5 ips.)

In a number of tape machines, despite their ability to extend response to 12,000 cps or better at 3.75 ips, there may be little or no change in record treble boost compared to that employed at 7.5 ips. Instead, these machines reduce the amount of bias current when the tape machine is shifted to the lower speed. This reduces the high-frequency loss due to bias erase. Some tape machines employ a combination of methods, namely a change in record treble boost plus a change in bias current. Generally, it is sought to avoid treble boost at 3.75 ips substantially greater than that at 7.5 ips, for there is too much danger of running into tape overload.

Decreasing bias current entails an increase in distortion. Hence in recording at 3.75 ips with a machine that employs the latter technique it is advisable to reduce the recording level by several db. Experience will tell the operator how much to reduce level.

In machines which do not reduce bias current but rely on an increase in record treble boost to achieve extended response at 3.75 ips, it is similarly advisable to reduce recording level to avoid possible overload at the very high end.

Equalization Circuits

Equalization circuits are of two basic kinds: losser and feedback. Losser equalization, illustrated in *Fig.* 15, involves the principle of short-circuiting all frequencies to ground, but with some frequencies short-circuited to a greater degree than others, so that *in relative terms* bass boost or treble boost is achieved. Thus if treble boost is desired, a loss is produced at all frequencies, but with the loss growing progressively less as frequency rises. Conversely, if bass boost is desired, the losser circuitry produces more of a short-circuit at high frequencies than at low ones.

In the case of feedback equalization, the principle employed, as shown in *Fig.* 16, is to take the signal from one stage of an amplifier and apply it to an earlier stage, thereby cancelling part of the original signal, so that over-all amplification is reduced. The signal fed back to the earlier stage goes through a network that favors either the low frequencies or the high frequencies. If bass boost is desired, the network admits the high frequencies more easily, thereby cancelling the highs to a greater extent and achieving bass boost in relative terms. If treble boost is needed, then the network admits the bass frequencies more easily.

There are pros and cons for both losser and feedback equalization circuits. Feedback circuits are claimed to achieve less distortion. Sometimes they may also serve important ancillary purposes, such as reducing high-frequency losses. On the other hand, it is claimed that losser circuits, with suitable design, can be about as distortion-free. Above that, they can generally provide more exact equalization, and this equalization tends to be more immune from the effects of tube aging. Æ

Checking Frequency Response, Equalization, and Azimuth

HERMAN BURSTEIN*

Checking the performance of a tape machine to maintain it in optimum condition requires making a series of tests. The methods of making them and of evaluating the results are all described here.

When sound reproduction is faulty, various checks of the tape machine's performance are obviously needed. More than this, the meticulous audiofan may wish to keep a step ahead of impaired performance by having his tape machine checked periodically, however well it may be working at the time. Accordingly, it is the purpose of this article and the following one to discuss basic tests of tape recorder performance. The present article deals with the inter-related factors of frequency response, equalization, and azimuth. The next article will deal with checks of other aspects of performance, including bias current, distortion, signal-to-noise ratio, record-level indication, wow and flutter, tape speed, erasure, and head height.

In these days when very good audio test equipment, such as vacuum tube voltmeters, oscillators, oscilloscopes, distortion analyzers, and the like, can be had in kit form at modest prices, it is not unusual to find the audiofan possessing some of these instruments or able to borrow them from friends. Therefore this article is aimed in part at the audiofan in a position to make instrument checks of his tape machine.

But this is not meant to suggest that the serious audiofan should go out and purchase or otherwise obtain access to test equipment if he has not already done so. In most instances the owner of a tape machine will rightfully turn to a qualified technician for a check of performance. Nevertheless, a comprehension of testing procedures may stand him in good stead, and therefore this article is aimed equally at the latter type of owner. If he knows what should be checked and how to check it, there is a better chance of his wants or needs being met by the technician, at least in part because the two speak the same language. The knowledgeable audiofan is in a position to ask the right questions and to profit by the answers.

The testing techniques to be discussed are basic procedures that apply to the majority of tape recorders in home use. It should be noted, however, that because of variations in the design of various brands and models of tape machines, the *specific* procedures may be somewhat different for certain tests. Hence the audiofan who plans to probe the operation of his machine is well advised to obtain a service manual from the manufacturer.

Frequency Response

The effects of azimuth alignment, bias current, dirty heads, tape-to-head contact, and other factors upon frequency response have been discussed in earlier articles. It is presumed that all these have been attended to before checking frequency response, or else that they will be attended to as the result of checking frequency response.

In checking frequency response we may be concerned with (1) over-all response, namely the result of recording and playing back audio signals; (2) playback response only; (3) with both over-all response and playback response. If a tape machine is used only for playback of recorded tapes, over-all response is obviously not a consideration. On the other hand, although a machine is used for recording as well as playback, one will still be interested in playback response in itself if he plans to play recorded tapes. It is quite possible for a tape machine to have relatively flat record-playback response, yet fall quite wide of the mark in playing recorded tapes.

The best course in checking playback response is to use a test tape. It is not sufficient to check playback equalization, namely the frequency response of the playback amplifier, to ascertain whether this equalization conforms to a standard curve. By playing a test tape one takes into account not only the playback equalization but also the characteristics of the playback head, which tend to produce treble losses, and other factors that may affect frequency response, such as the cable from the head to the playback amplifier.

Using a test tape, it is quite difficult and generally unsatisfactory to check

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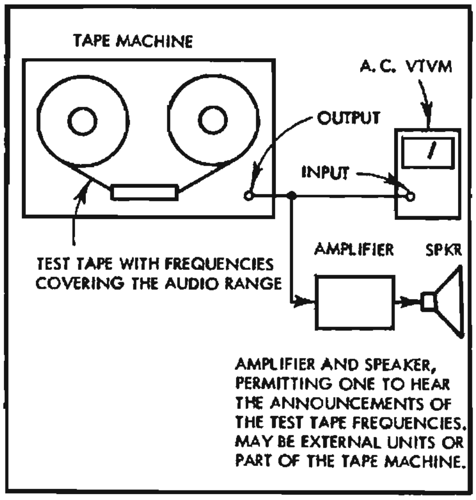


Fig. 1. Set-up for measuring playback response of a tape machine.

playback response by ear, because the apparent frequency response will depend upon the sound level at which the speaker is operated, the listener's hearing acuity at various frequencies, and the frequency characteristics of the speaker system and other components in the audio chain. Therefore the preferred measuring device is a VTVM (vacuum tube voltmeter), connected to the output of the tape machine as in *Fig.* 1. An oscilloscope is also generally satisfactory. Although it does not give as precise a reading, usually its indication can be interpreted with an accuracy of 1 db or better. A volt-ohm-milliameter should not be used because of its inferior frequency-response characteristics.

If there are tone controls in the tape machine, it is advisable, if feasible, to measure the output signal prior to the tone controls. Otherwise, set the controls to the position designated as flat, if there is such a designation. In the absence of a setting marked flat, it becomes necessary to measure response at various settings of the controls in order to ascertain the position nearest to flat response.

It is of course necessary to identify the frequencies on the test tape. Voiced announcements of these frequencies appear on the tape and may be heard through an amplifier and speaker connected to the output of the tape machine, in parallel with the connection to the VTVM. The capacitance of the cable from the tape machine to an external amplifier, as well as the input capacitance of the amplifier itself, may produce a change in the playback response of the tape machine, principally a loss of high frequencies. If the tape machine has a low-impedance output, such losses will not occur. However, a short cable is always the safest course.

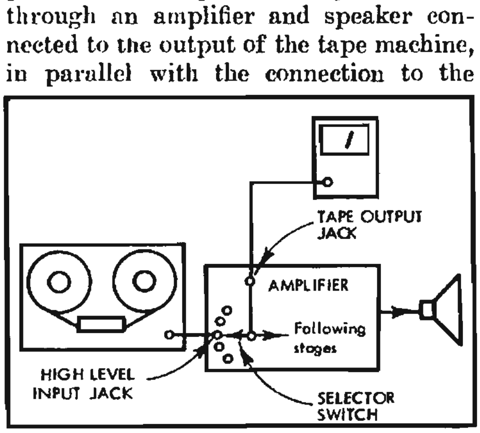
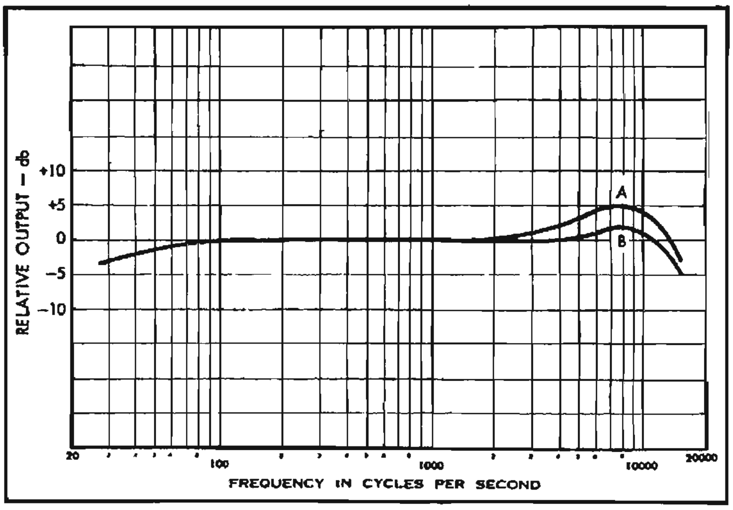


Fig. 2. Method of connecting a VTVM to the tape output jack of an amplifier to measure frequency response.

If an amplifier and speaker are needed to monitor the test frequencies in playback, this problem may be solved in several ways: (1) Most home machines contain an internal amplifier and speaker. (2) Some machines provide a monitor output jack connected in parallel with the regular output jack; hence the VTVM (or oscilloscope) can be connected to one jack and the external amplifier and speaker to the other. (3) In many audio control amplifiers, the incoming signal is switched directly to a tape output jack; the VTVM or oscilloscope can be connected to this jack, as illustrated in *Fig.* 2. (4) If there is no other alternative, one can make a splice permitting both the VTVM and the external amplifier to be connected to the output jack of the tape machine; it is probably easiest to splice into the leads to the VTVM inasmuch as these are usually not shielded.

Fig. 4. Illustration of differences in treble response.



Use of a test tape implies a standard of equalization universally or widely accepted by the industry. The situation is quite clear in the case of the 15 ips and 7.5 ips speeds, where NAB equalization is prevalent for consumer use. However, the situation is still unsettled for the 3.75 ips speed, at least at the time this was written. Two different equalization characteristics have been used at this speed, and a third has been suggested (details on these three characteristics appear later in this article). Until the industry agrees upon a 3.75 ips standard, the individual wishing to use a test tape for this speed is perhaps safest in using the one put out by Ampex, inasmuch as most commercial tape recording is done with Ampex equipment, so that Ampex equalization tends to serve as an unofficial standard.

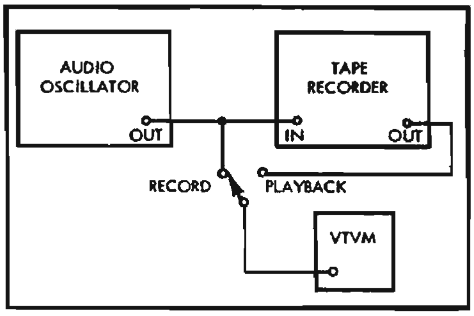


Fig. 3. Set-up for checking record-playback response.

Record-Playback Response

As stated before, one may be interested purely in over-all response, without caring whether the playback response conforms to a standard characteristic. On the other hand, if playback response does conform to a standard, then a measurement of over-all response implicitly becomes a check upon whether the tape machine is *recording* in conformity with the standard characteristic; in other words, whether the signal impressed upon the tape has a frequency characteristic such that flat response will result when the tape is played upon any machine incorporating the standard playback characteristic.

Figure 3 shows an arrangement for checking over-all response, using an audio oscillator as the signal source for feeding in a number of frequencies at constant level, and a VTVM to measure the relative magnitudes of these frequencies in playback. The VTVM is used during recording to ensure that a constant signal level is fed to the tape machine. In playback it is switched to the output of the machine to measure response.

When the machine has a record-playback head instead of separate record and playback heads, checking frequency response becomes somewhat laborious, be-

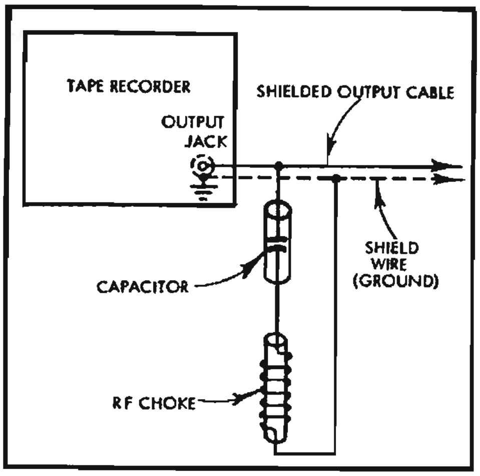


Fig. 5. Method of connecting a bias trap across output.

cause after various frequencies are recorded, the tape must be rewound and then played back. Accordingly, it may be desired to check a minimum number of frequencies spanning the audio range. However, these frequencies should be no more than about an octave apart. The following ten frequencies would provide a fairly satisfactory measure of response: 30, 50, 100, 200, 500, 1000, 2000, 5000, 10,000, and 15,000 cps.

A more thorough check of frequency response, one that is more likely to uncover significant peaks or dips, might use the following frequencies: 30, 50, 70, 100, 200, 300, 500, 700, 1000, 2000, 3000, 5000, 6000, 7000, 8000, 9000, 10,000, 12,000, and 15,000 cps. Note the emphasis on the 5000 to 10,000 cps region. In the attempt to achieve response out to 15,000 cps, some machines peak excessively in the region of 5000 to 10,000 cps. This is illustrated in *Fig.* 4. Curve A shows over-all response that is only 3 db down at 15,000 cps but at the cost of a 5 db peak at 8000 cps. Curve B is smoother, having a peak of only 2 db at 8000 cps, although response is therefore down 6 db at 15,000 cps. By and large, Curve B provides more accurate reproduction and better listening.

It is vital that the test frequencies be recorded at a level at least 20 db below the maximum permissible recording level as indicated by the record-level indicator. The procedure would be to feed in a tone, say 1000 cps, that causes the indicator to read maximum—the eye to close in the case of an electronic indicator, or the pointer to read 0 VU or slightly higher in the case of a VU meter. Then the signal of the audio oscillator would be reduced 20 db (or more) as measured by a VTVM connected directly to the oscillator. This signal reduction is imperative to avoid overloading the tape at high frequencies due to the large amount of treble boost incorporated in the record amplifier. (In the case of program material, the highs are usually much lower in level than the mid-frequencies, so that tape overload is thereby avoided.)

When checking frequency response of a tape machine with separate record and playback heads, there is no problem in keeping track of the frequency being tested. But there is a problem of frequency identification when checking a machine that uses a record-playback head, because all the frequencies must be recorded, the tape rewound, and then played back. One method of frequency identification is to intersperse voiced announcements with a microphone, and to monitor the tape playback with a speaker. However, this is laborious. Another technique is to "mark" several key frequencies, say 100, 1000, and 10,000 cps. This can be done by turning the oscillator on and off several times when each of the key frequencies are recorded, causing the VTVM pointer to fluctuate in playback as an indication that a key point has been reached in the schedule of recorded frequencies. Monitoring the tape with an oscilloscope can serve in identifying the test frequencies.

While testing frequency response is on the whole much easier in the case of a machine with separate record and playback heads, because one can record and play back simultaneously, there tends to be one drawback. This concerns pickup of bias current at the output jack of the tape machine. Hence the VTVM may be reading not only audio signal but also bias signal, preventing an accurate measurement of frequency response. Bias current, which typically lies between 50,000 and 100,000 cps, has radio frequency characteristics and therefore is apt to appear at various places, including the output jack. This tends to be especially true in compact home machines, where the record and playback amplifiers are very close together and it is difficult to shield adequately against radiation of the bias frequency. The fact that it is necessary to check frequency response at a level at least 20 db below maximum permissible recording level aggravates the problem.

To prevent bias current reaching the VTVM in significant quantity, one can connect a simple, inexpensive bias trap across the output jack, as illustrated in *Fig.* 5. This consists of an r.f. choke and a capacitor in series between the high side of the output jack and ground. If the resonant frequency of these two components corresponds to the bias frequency, the latter will be shorted out to ground. A garden variety 2.5 millihenry choke and a capacitor in the range of 1000 to 4000 μμf (micromicrofarads) will cover the range of 50,000 to 100,000 cps, wherein the bias frequency probably lies. Given the bias frequency (as stated by the manufacturer of the tape machine), one can calculate the required capacitor from the formula $C = 1/\pi^2 f^2 L$, with C representing the capacitor in farads, f the frequency in cps, and L the the inductance in henries of the choke. This formula can be stated in more convenient terms as approximately $C = 25{,}000/f^2L$ with C in μμf, f in kilocycles (thousands of cycles), and L in millihenries.

If the calculated capacitance calls for a non-standard value of capacitor, one may try using several capacitors of the nearest standard value; due to their tolerance, one or the other may fall sufficiently wide of nominal value to achieve maximum attenuation of the bias current. In most tape machines it is possible to change the bias frequency by adjusting the slug in the oscillator transformer. One could adjust the oscillator frequency to coincide with the resonant frequency of the bias trap, but this is unwise. Changes in bias frequency can change the amount of bias current going through the record head and thereby alter the distortion and frequency response of the tape recorder. Also, changes in bias frequency (upward) tend to decrease the

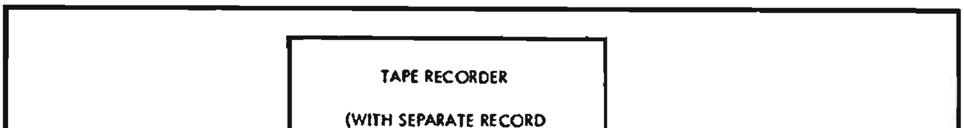
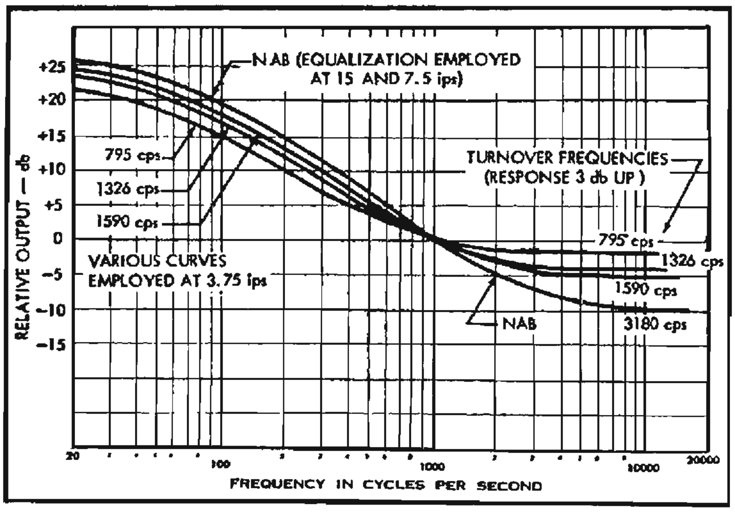


Fig. 6. Set-up for checking record-playback response by comparing playback signal with original source.



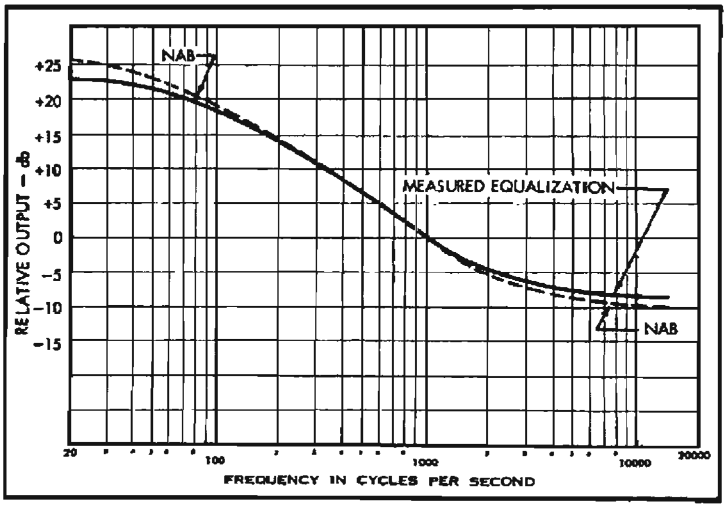


Fig. 7. (left) Playback equalization curves. Fig. 8. (right) Typical measured playback equalization into a tape machine claiming adherence to the NAB curve.

effectiveness of the erase head. One will probably find that even though the resonant frequency of the bias trap is not exactly the same as the bias frequency, nevertheless sufficient attenuation will take place to prove satisfactory.

In the absence of test equipment, one can nevertheless make a fairly good evaluation of a tape machine's over-all frequency response by recording the music from a phonograph disc of good quality and wide range, and then comparing the tape playback with the disc. If the tape machine has separate record and playback heads, one can immediately compare the incoming signal (from the record) with the playback signal, provided that the control amplifier of the audio system has a tape-monitor switch, as illustrated in *Fig.* 6. Otherwise, and in the case of a machine having a record-playback head, it is necessary to record the tape, rewind it, and compare the tape playback with a replay of the disc.

With a little experience, one can learn to synchronize the two. Play a few measures of the recorded tape and stop the machine at an easily recognized point in the music. Start the record, and when it reaches the same point in the music, start the tape machine again. Switch between the tape machine and the phono disc by means of the selector switch in the audio system. If the record and the tape are not fully synchronized, either slow down the record by pressing a finger against the edge or slow down the tape by briefly turning the transport motor off and then on.

Equalization

As brought out in earlier articles, the tape recorder employs large amounts of equalization to compensate for bass and treble losses that occur in the record-playback process. If a test of tape recorder performance reveals that frequency response is deficient in playback or over-all, then one of the things we wish to check in order to learn the cause is the equalization in the record and playback tape amplifiers.

A check of playback equalization can enable us to ascertain fairly well whether the tape machine meets a standard playback characteristic, such as the NAB curve. *Figure* 7 shows the NAB playback curve, commonly used at 15 and 7.5 ips, as well as three curves that have been or are being used at 3.75 ips. If the measured equalization shows insufficient cut (referred to 1000 cps) in the upper treble range, there may be a very good reason, namely that the remaining treble cut is produced by the playback head and possibly other circuit elements. If the measured equalization shows insufficient bass boost, again there may be good reason. For one thing, the NAB standard permits response to be 4 db down at 50 cps. For another, the playback head may behave in a manner that causes some bass boost; a small head tends to behave in the same manner as its gap at very low frequencies, where the head as a whole is small relative to the wavelength recorded on the tape. *Figure* 8 shows the measured equalization that one might typically find in a machine of good quality, operating at 7.5 ips. There is somewhat less bass boost than called for by the NAB curve, and there is somewhat less treble cut (referred to 1000 cps) to compensate for high frequency losses due to the playback head. On the other hand, note that between 100 and 10,000 cps the measured equalization corresponds to the NAB curve, which presumes the use of an "ideal" playback head.

If the measured playback equalization comes within ±2 db of the NAB curve or other given curve between 100 and 10,000 cps, this may be considered excellent; within ±3 db is good. At 50 cps, the measured bass boost should not be more than 4 db below that of the standard curve. At 15,000 cps, the measured treble cut should not be more than 4 db in excess of the standard curve; taking into account the likelihood of head losses, the measured treble cut should probably be no greater than the standard curve.

The technique commonly used to measure playback equalization is shown in *Fig.* 9. The signal is inserted into the playback amplifier through the playback head in order to take into account high-frequency losses due to winding capacitance of the head, as well as other circuit capacitances. The signal is taken from a voltage divider made up of R_1 and R_2 in order to present a very small signal (on the order of 1 millivolt or less) to the head and thereby prevent overloading the playback amplifier. The gain control of the audio oscillator should be ad-

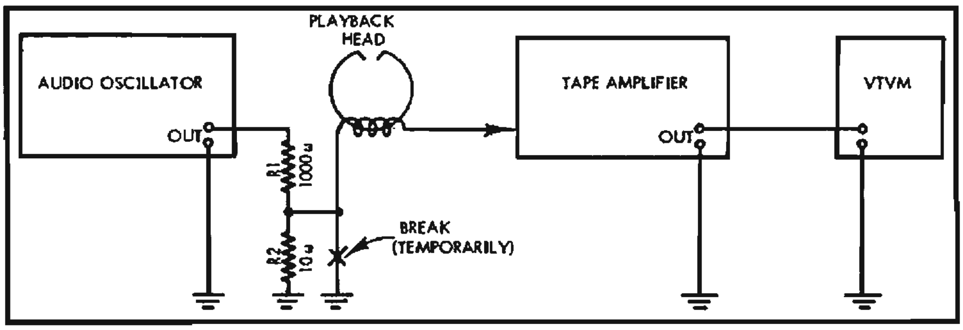


Fig. 9. Set-up for measuring playback equalization.

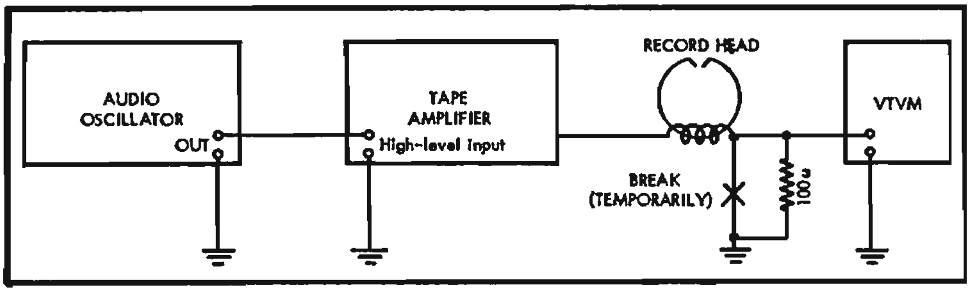


Fig. 10. Set-up for measuring recording equalization.

justed so that the playback amplifier delivers no more than about 1 volt output. Maximum output will occur at the bass end, with decreasing output as frequency rises. In the case of NAB equalization, there is 36 db of emphasis of the extreme bass relative to the extreme treble. In other words, if signal output is 1 volt below 50 cps, then the output will be only about 16 millivolts at 15,000 cps. Hence a sensitive VTVM is required. If a sufficiently sensitive VTVM is not available, one may increase the signal input into the head at a mid-range frequency, say 500 cps, and take into account the amount of signal increase in measuring output of frequencies from 500 cps upward. But, as stated before, the signal input should not cause signal output to exceed 1 volt.

Figure 10 shows the procedure commonly used to measure record equalization, which may be compared with the record equalization curve—treble boost—specified by the manufacturer of the tape machine. (At a given speed, record equalization varies somewhat from one tape machine to another according to the manufacturer's decision as to the bias current that achieves the optimum combination of low distortion and extended treble response.) A low-value resistor, typically 100 ohms, is inserted between the ground lead of the record head and ground. As signals of varying frequency are fed into the record amplifier, voltage measurements are taken across the resistor to indicate the equalization characteristic.

If the VTVM does not have sufficient sensitivity, it may be feasible to use a 1000 ohm resistor instead. The important thing is that the resistance be small at all frequencies compared with the impedance of the head and other circuit elements (including plate resistance of the tube that drives the record head and the "constant current" resistor between the plate of this tube and the head). To be on the safe side, however, a 100-ohm resistor and a sensitive VTVM should be used. Before taking measurements of the equalization charactristic, it is necessary to remove the oscillator tube. Otherwise one will be measuring bias current, which is several times as great as the audio current that flows through the record head.

If record or playback equalization departs significantly from the prescribed curve, it may be possible to correct the situation readily by means of a control incorporated in the tape amplifier for this purpose. Such controls ordinarily are found only in machines of semi-professional and professional caliber. The less expensive machines generally incorporate fixed instead of variable equalization, and it becomes necessary to replace components, namely a resistor, capacitor, or inductance, in order to change equalization. This is usually a task for a service technician.

If playback response has been checked out as satisfactory (including equalization of the tape amplifier plus characteristics of the head and other circuit elements), and if bias current has also been checked out as correct (discussed in a following article), but it appears that record equalization requires adjustment, the proper procedure is to make the adjustment such that over-all response is as flat as possible. This may be somewhat more laborious than conforming the record equalization to the manufacturer's specified curve, particularly if the tape machine employs a record-playback head, so that it is necessary to rewind the tape before it can be played and checked for over-all response. However, adjusting on the basis of over-all response allows for slight deviations in bias current and for the characteristics of the particular head and tape one is using, whereas conforming to the manufacturer's record equalization may result in performance that, although good, is not as good as could be.

To avoid the possibility of overload in checking record equalization, the signal input to the record amplifier should be kept low enough to prevent the record-level indicator from exceeding a normal reading at any frequency.

Azimuth Alignment

To assure that the gap of the head is at right angles to the length of the tape, many tape machines provide for facile azimuth adjustment by simply turning one of the screws that fastens the head to the tape deck. Turning the screw up or down causes the head to tilt in one direction or the other. There are three ways in which the operator can check for azimuth alignment of the playback head or of a record-playback head:

1. The simplest and least expensive is to make the adjustment by ear when listening to a commercial prerecorded tape known to have wide frequency range. The head is tilted for maximum brilliance of sound.

2. The standard procedure is to employ an azimuth alignment tape, of which there are a number on the market. These tapes contain a high frequency tone, such as 7500 or 10,000 cps. A VTVM is connected to the output of the tape machine, and the head is adjusted for maximum output while playing the test tape. One should be on the watchout for "false peaks"—minor peaks on either side of the major peak indicated by the VTVM. One can also adjust on the basis of listening to the azimuth tone.

3. A laboratory technique, employed in making an azimuth test tape, consists of the following complex and time-consuming "absolute" procedure. Using an audio oscillator as the signal source, a high-frequency tone is recorded by means of the playback head one wishes to align. This tape is played back with the *base* instead of the coating against the head. A weak but measureable playback signal will be obtained. The playback head is adjusted for maximum output. Then the head is returned to a point half-way, as nearly as one can judge, between its first position and the second position; a half-way adjustment is called for because the azimuth error in playback is in the opposite direction of that in recording, as illustrated in *Fig.* 11. The procedure is repeated a number of times, until it is found that adjusting the head in playback results in no signal increase.

If the tape machine employs a separate record head, the playback head is aligned first. Then the record head is adjusted so that maximum output is obtained when recording and simultaneously playing back a high-frequency tone. Æ

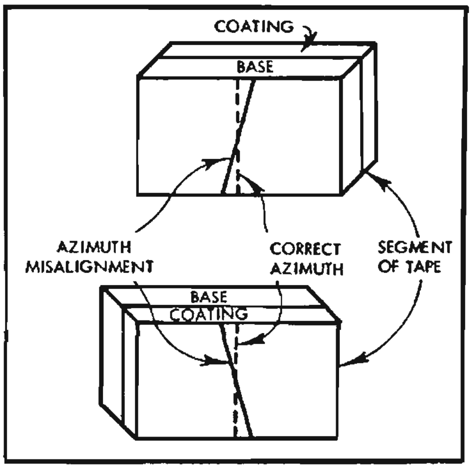


Fig. 11. Reversal of direction of azimuth misalignment when tape is played with base instead of coating against the head.

Checking Other Aspects of Performance

Correct recording bias is a key to maximum treble response at minimum distortion. Equally, a key to achieving the best performance of your tape machine is an understanding of the how and what-to-do of tape speed variation. This article covers these topics—and much more.

HERMAN BURSTEIN*

The preceding article discussed techniques of measuring frequency response, equalization, and azimuth. The present article is devoted to techniques of measuring other important aspects of tape machine performance.

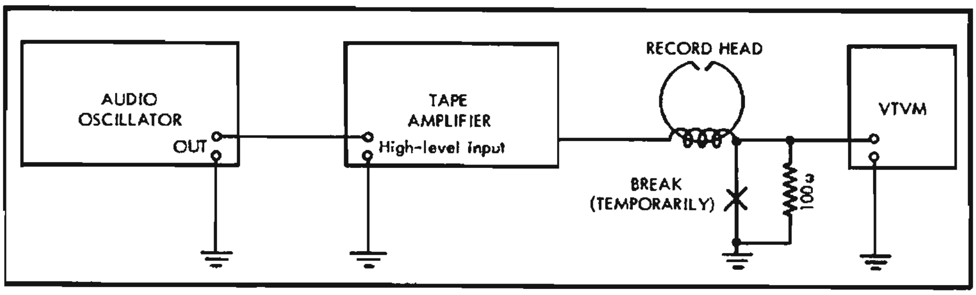


Fig. 1. Setup for measuring bias current.

Bias Current

It is vital that bias current be set at the correct value in order to achieve the best practical compromise between extended treble response and low distortion, a compromise that varies with tape speed.

The method employed to check record equalization, shown in *Fig.* 1, is also frequently used to measure bias current. A 100-ohm resistor inserted in the ground lead of the record head is generally a suitable value for the purpose. If the meter has insufficient sensitivity, it is often feasible to use a 1000-ohm resistor instead. The problem is to ensure that the resistor in series with the head offers negligible resistance to current compared with the impedance of the head. A typical record-playback head, with an inductance of about 0.5 H, has an impedance exceeding 150,000 ohms at the bias frequency, so that the additional impedance presented by a 1000-ohm resistor is then negligible. On the other hand, a head designed only for recording may have an inductance of but a few millihenries, so that an additional impedance of 1000 ohms becomes significant.

The resistor in question should of course be accurate, preferably to 1 per cent. Similarly, the meter should be accurate. Current is calculated by means of Ohm's Law, namely $I = E/R$, where I is current in amperes, E is voltage in volts, and R is resistance in ohms. For example, if one measures 70 millivolts (.07 volt) across a 100-ohm resistor, then $I = .07/100 = 0.7$ ma (milliamperes). It happens that a bias current of 0.7 ma is typical of a number of machines in home use. However, the required value, depending upon the head, can deviate a good deal from this. Hence the optimum value should be obtained from the manufacturer of the tape recorder, or possibly from the manufacturer of the head if the two are not the same. Moreover, one should obtain the optimum value for the speed at which one plans to do the most recording, inasmuch as the optimum value is different at the various machine speeds. To illustrate, at 7.5 ips the optimum value may be 0.7 ma, whereas it may be 0.6 ma at 3.75 ips.

As components in the bias oscillator circuit warm up, they tend to change value somewhat, and the frequency and magnitude of the bias current tend to change accordingly. Thus it is desirable to measure and adjust bias current only after the tape recorder has warmed up for a period, say for 15 or 20 minutes. (In use, it is similarly desirable to provide a warmup period before making a recording.)

Another technique for adjusting bias current involves measuring the change in recorded signal amplitude as the bias is varied. While this method can be used with machines having a combination record-playback head, it is such a tedious procedure that it is primarily recommended for machines with separate re-

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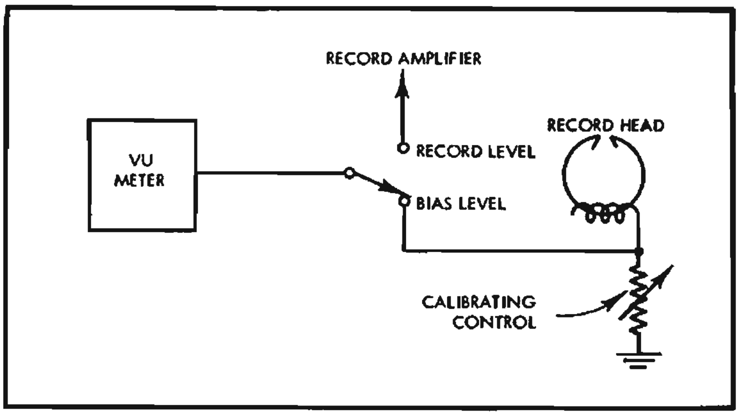


Fig. 2. Checking bias current with a VU meter.

cord and playback heads. At a given frequency, usually 1000 cps, bias current is adjusted until maximum signal amplitude is recorded on the tape. This is determined by measuring the playback signal. The gain controls of the record and playback tape amplifiers are not varied during this procedure.

It is often recommended that instead of adjusting bias current for peak output, one should set it at about "0.5 db above peak." This means *increasing* bias current until the signal amplitude (1000 cps) drops 0.5 db. Note carefully that it is the signal amplitude which is to drop 0.5 db, not the bias current. At this point slight changes in bias current, upward or downward, have a minimum effect on distortion and frequency response. An authority has stated: "For optimum results in recorders with wide frequency range at low tape speeds, it seems desirable to set bias at not under 0.5 db above peak. . . . This offers close to optimum distortion characteristics, while producing minimum change of relative response with bias change."[1]

Still another technique for gauging and adjusting bias current relies upon an instrument for measuring distortion, either intermodulation or harmonic distortion. First record at a relatively high level so that variations in distortion due to bias current changes may be readily perceived. Next adjust bias for minimum distortion. Then high-frequency record-playback response is checked at a much lower recording level, at least 20 db below maximum permissible level. Bias is reduced until treble response is considered adequate. Satisfactory treble response also entails adjustment of the amount of treble boost used in recording, assuming that the record amplifier contains a control for varying the treble boost. It is not advisable to go much above 20 db treble boost at 15,000 cps, for this raises the danger of overloading the tape at high frequencies on program material.

At a tape speed of 15 ips, it will probably be found that the bias corresponding to minimum distortion also permits response substantially flat to 15,000 cps without undue treble boost in recording. But at 7.5 ips and lower speeds, it will be found that, in order to obtain satisfactory high-frequency response, it is necessary to reduce bias current below the quantity corresponding to minimum distortion. It is then necessary to ask oneself whether and to what extent it is worth achieving extended treble response at a cost of increased distortion. Through trial and error, one can arrive at the point which carries flat response a fairly long way, say to 10,000 or 12,000 cps at 7.5 ips, without an undue increase in distortion. But trying to reach "all the way out," namely to 15,000 cps, may cause an increase in distortion disproportionate to the added realism of reproduction achieved by extending flat response from 12,000 cps out to 15,000 cps.

It is necessary to take into account that the amount of bias current suitable for one brand of tape may not be the optimum for another brand. To illustrate the point, following are the results of a test conducted to determine the amounts of bias current producing minimum distortion for four brands of conventional tape. Using Tape A as a reference, bias current for minimum distortion was 0 db for Tape A, +0.75 db for Tape B, −0.50 db for Tape C, and 0 db for Tape D. While the differences in bias current, expressed in decibels, appear small, nevertheless they are great enough to have appreciable effect upon distortion and upon frequency response above 10,000 cps. Accordingly, it is desirable to adjust bias current on the basis of the brand of tape one plans to use for most recording purposes.

The individual seeking maximum performance from his tape machine will wish to check bias current periodically because its magnitude may change as the result of aging of the oscillator tube and of other components in the oscillator circuit. Relatively slight changes in bias current can produce relatively large changes in distortion and frequency response. In tape recorders of professional and semi-professional quality, it is usually the practice to incorporate a switching arrangement that permits the VU meter to check the bias current. The meter does not provide an absolute reading but does indicate in relative terms whether bias current is at correct level. As shown in *Fig.* 2, a calibrating control is incorporated to cause the meter to read 0 VU (or some other designated figure) when bias is correct.

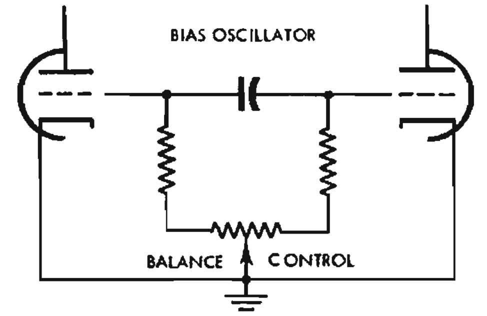
Bias Waveform

It is important to check not only the magnitude of the bias current but also its waveform. Obvious distortion can readily be detected by connecting an oscilloscope across the record head. Harmonic distortion in excess of 5 per cent is apparent to the eye. Ideally, the bias waveform should be a perfect sine wave. If harmonics of the bias frequency are also present, they will produce noise in recording. Therefore some tape machines include a control for balancing the oscillator tube to achieve minimum distortion, as shown in *Fig.* 3. This adjustment can be performed by ear on a machine having separate record and playback heads. With no audio signal fed in, one simultaneously "records" and plays back a blank tape, meanwhile adjusting the balance control for minimum noise. It is advisable to use a test tape that has been thoroughly erased by means of a bulk eraser, so that noise due to the bias current waveform will be readily apparent and not masked by tape noises.

Bias Frequency

If for any reason the bias frequency should change radically, there will tend to be the following deleterious effects: (1) If the frequency is too low, there

Fig. 3. Oscillator circuit incorporating a balance control for minimizing noise.



[1] C. J. LeBel, "More on recorder bias," *Audio Record*, March–April 1956, p. 7 (formerly published by Audio Devices, Inc., New York City).

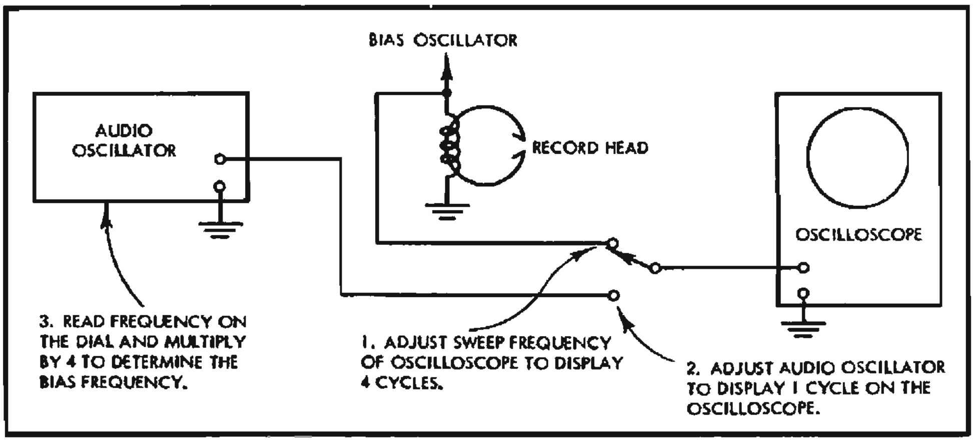


Fig. 4. Setup for measuring bias frequency.

will tend to be audible beat notes between the bias frequency and the harmonics of the higher audio frequencies; the bias frequency should be at least four to five times the fundamental of the highest audio frequency to be reproduced. (2) If the bias frequency is too high, the effectiveness of the erase head (which is supplied current by the same oscillator that feeds the record head) is usually diminished. Thus in rerecording a tape, one may hear some of the first recording, particularly the lower frequencies, coming through the second recording. (3) With a change in frequency, there may be a change in the amount of bias current reaching the record head; an increase in current will reduce distortion and restrict high-frequency response; a decrease in current will increase distortion and extend high-frequency response, possibly producing an undesirable peak.

An audio oscillator and an oscilloscope may be used to check the bias frequency, as illustrated in *Fig.* 4. Connect the oscilloscope to the record head and obtain a display of four cycles. Then connect the oscilloscope to the audio oscillator and adjust the latter's frequency until one cycle appears on the oscilloscope. The frequency indicated by the dial of the oscillator is one-fourth of the bias frequency. If the audio oscillator goes to 100,000 cps, then one may adjust the oscilloscope for a reading of one cycle when connected to either the record head or the oscillator. The accuracy of the method described depends upon how precisely the oscillator is calibrated. However, a slight error is unimportant, and almost any audio oscillator will be sufficiently accurate for the purpose.

Distortion

Whereas such audio components as control amplifiers, power amplifiers, and tuners are commonly checked for intermodulation distortion as well as harmonic distortion, the general practice is to test tape recorders only for harmonic distortion, although an IM test is usually quite revealing.

Figure 5 shows the setup for checking total harmonic distortion (THD) of a tape machine having separate record and playback heads. A signal from an audio oscillator is fed into the tape recorder, and the output of the latter is measured by a harmonic distortion meter. The procedure is essentially the same in the case of a machine that uses a single head for both record and playback, except for the fact that the recording of the test signal and its measurement take place at different points of time; after recording the test signal it is necessary to rewind the tape and play back the signal into the distortion meter.

The test frequency is usually 400 cps or thereabout for a variety of reasons. (1) Peak audio energy of most sound sources occurs in the neighborhood of 400 cps. Hence, at a given recording level, a test in this region is indicative of the maximum distortion likely to occur. (2) It is desirable to use a relatively low frequency in order to be able to record on the tape a large number of the harmonics due to distortion. Thus if the test frequency were, say, 5000 cps, and if the tape machine had a range extending to 15,000 cps, only the second and third harmonics could appear.

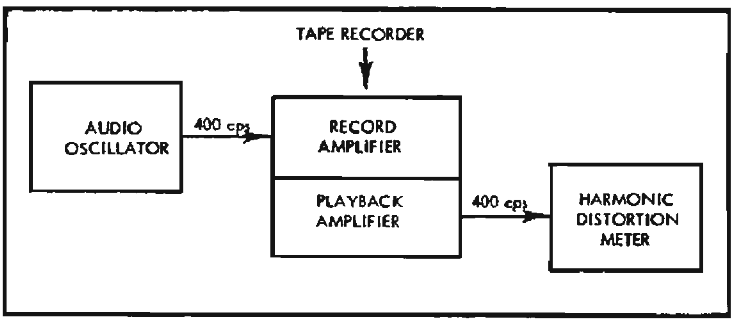
Testing harmonic distortion at the higher frequencies—above 5000 cps or so—is not apt to be very meaningful. On the one hand, the pronounced treble boost in the record amplifier tends to exaggerate high-frequency distortion. In recording natural sound sources, this high-frequency emphasis tends to be offset by a decline in audio energy of the source; but there is no such offset when dealing with a test signal. On the other hand, to the extent that the tape recorder has limited high-frequency response, say to 15,000 cps with a very sharp decline thereafter, harmonic distortion shows up only to a slight extent if at all. One can easily establish this fact by recording a high frequency such as 10,000 cps at a very high level at 7.5 ips and viewing the playback waveform on an oscilloscope. Since the harmonics are beyond the reproducing ability of the tape recorder, the playback waveform will appear as an undistorted sine wave no matter how high the recording level.

The fact that distortion in the high frequency range fails to show up on a harmonic distortion test does not mean there is no problem here. Overloading at high frequencies can create havoc in terms of intermodulation distortion. To illustrate, a 10,000-cps note and a 12,000-cps one, if over-recorded, may intermodulate to produce spurious distortion products that are audible.

Measurement of intermodulation distortion offers a much more sensitive test of the performance of a tape recorder. At a recording level where a harmonic distortion test may reveal only 2 or 3 per cent THD, the IM test may reveal 10 per cent, 20 per cent, or even more IM distortion. *Figure* 6 shows the setup for checking IM distortion, requiring only the use of an IM analyzer; this assumes that the IM analyzer supplies the frequencies, usually about 60 cps and 6000 cps, needed for the test.

Record Level Indicator

The test of distortion must have reference to the recording level as indicated by the record-level indicator. If the indicator is of the magic eye type, maximum permissible recording level corresponds (or should correspond) to eye closure, that is, minimum fluorescent shadow. If the indicator is of the neon lamp type, maximum level is denoted by ignition of the lamp. If the indicator is a VU meter,



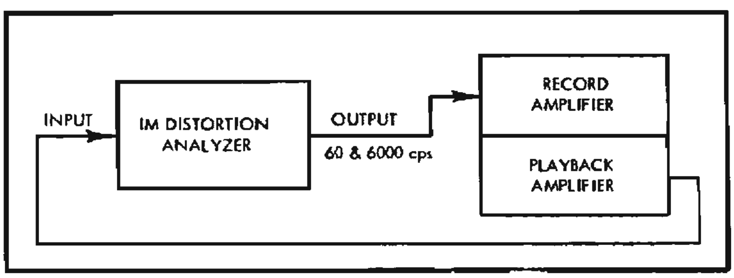


Fig. 5 (left). Measuring harmonic distortion. Fig. 6 (right). Setup for measuring intermodulation distortion.

maximum recording level is generally 6 to 8 db above the 0-VU reading. For example, in the case of a number of professional tape recorders, when the meter reads 0 VU on a steady signal this is intended to correspond to a recording level that produces 1 per cent total harmonic distortion. The level that produces 3 per cent THD is about 6 db above 0 VU. If the recording level that produces 3 per cent THD is considered to be the maximum permissible amount, then there is a 6 db allowance for the mechanical lag of the VU meter. Thus to measure distortion at maximum permissible recording level, the input signal should be 6 db higher than that required to drive the meter to 0 VU.

One can obtain test tapes containing a signal recorded at a level producing a stated amount of harmonic distortion. Thus Ampex produces a test tape with a 250-cps tone recorded at a level resulting in 1 per cent harmonic distortion. Dubbings Sales Corp. in the past produced a tape with a 400-cps tone recorded at a 2 per cent harmonic distortion level; although not presently in production, it may still be available from some mail order houses and other distributors of audio supplies.

Such tapes can be used to check the calibration of the record-level indicator with reasonable accuracy. To illustrate, assume the use of the Ampex test tape. Play the tape and measure the level of the output signal. Using an audio oscillator or test record, record the same frequency (250 cps) or a nearby one on another tape and adjust the recording level until the same playback level is obtained as with the test tape. Then one is recording at a level producing approximately 1 per cent harmonic distortion, assuming bias current is at "normal" value.

If the tape recorder is rated on the basis of 3 per cent harmonic distortion, the input signal can be raised about 6 db to obtain an increase from 1 to 3 per cent harmonic distortion. At this point the magic eye indicator should close or the neon lamp should ignite. In the case of the VU meter, the 0 VU indication should have been reached at the 1 per cent harmonic distortion level or earlier.

If the tape recorder is rated on the basis of 2 per cent harmonic distortion, the input signal can be raised about 3 db to obtain an increase from 1 to 3 per cent harmonic distortion.

Of course, the more direct method of checking calibration of the record level indicator is to measure distortion at the point where the magic eye closes or the neon lamp ignites or the VU meter reads 0 VU.

Signal-To-Noise Ratio

Signal-to-noise ratio is measured on the basis of maximum permissible recording level, namely that which produces 1, 2, or 3 per cent harmonic distortion. Assuming that 3 per cent harmonic distortion is considered acceptable, as is commonly done, and that the record-level indicator is correspondingly calibrated, the procedure for measuring signal-to-noise ratio is as follows: A frequency between 250 and 1000 cps —usually 400 cps—is recorded at maximum permissible level. The playback signal level is measured. The tape is rewound and the process repeated, except that this time no audio signal is recorded. Again the playback signal is measured. Now the output is due to noise and hum of the record amplifier and of the playback amplifier (largely the latter in most cases), noise produced on the tape as the result of distortion in the bias-current waveform, tape hiss, and imperfect erasure by the erase head. The ratio of the first playback signal (with an audio input signal) to the second playback signal (without an audio input signal) constitutes the signal-to-noise ratio of the tape recorder. In a high quality machine, the predominant contribution to noise will be tape hiss.

If one is measuring the signal-to-noise ratio of a tape machine designed only for playback and not for recording, the measurement has to be made on the basis of a test tape. As stated before, test tapes are available carrying frequencies recorded at a given level of harmonic distortion. Comparison of the output level when playing the test tape and when playing a blank virgin tape yields the signal-to-noise ratio. If the test signal is at a level corresponding to 1 per cent harmonic distortion, the signal-to-noise ratio at the 2 per cent distortion level can be approximated by adding 3 db; at the 3 per cent distortion level, by adding 6 db.

In a high-quality tape machine the signal-to-noise ratio will range upwards of 50 db, or a voltage ratio exceeding 300:1. Therefore the measuring instrument has to be a vacuum-tube voltmeter of high sensitivity. Typically, the output of a tape machine (before the power output stage, if any) is about 1 volt. Hence if the noise is 50 db lower, the output voltage is only about 3 millivolts. If the machine has a signal-to-noise ratio as high as 55 db, as some do, then, based on a maximum signal output of 1 volt, the noise content will be less then 2 millivolts.

Wow and Flutter

A quantitative measurement of wow and flutter requires equipment unlikely to be in the possession of the home recordist and therefore will not be discussed here. However, a simple and effective test can be made by recording and playing back a tone in the vicinity of 3000 cycles, where the ear is highly sensitive to changes in pitch. The signal source can be an audio oscillator or a phonograph test record. However, inasmuch as the phonograph or the record or both may be a source of wow and flutter, an audio oscillator is the better source. Wow will be apparent as a quavering effect. Flutter will be apparent as graininess or coarseness of the tone.

In the case of tape machines designed for playback only, there are a number of test tapes which can be used for checking wow and flutter by ear. These tapes incorporate a frequency in the range of 2000 to 5000 cps.

Tape Speed

Tape speed may best be measured by means of a tape stroboscope, such as shown in *Fig. 7*, consisting of a wheel that is pressed against the moving tape and therefore moves at the same speed as the tape. The wheel contains a number of bars along its circumference, and these bars are viewed under a 60-cycle light source; for greatest clarity, a neon or fluorescent lamp is desirable, although the ordinary incandescent lamp will do. If tape speed is exactly correct, the bars on the stroboscopic wheel will appear to be standing still. If speed is slow, the

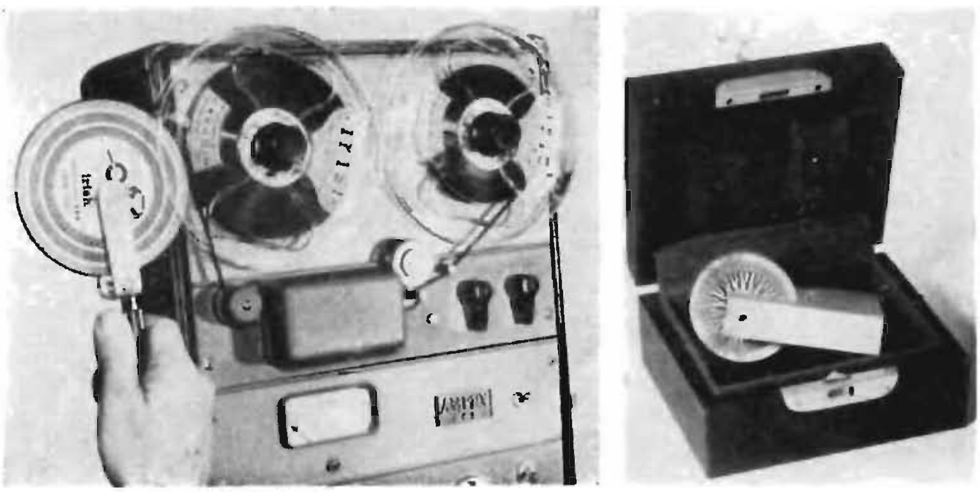


Fig. 7. Two stroboscopes for measuring tape speed.

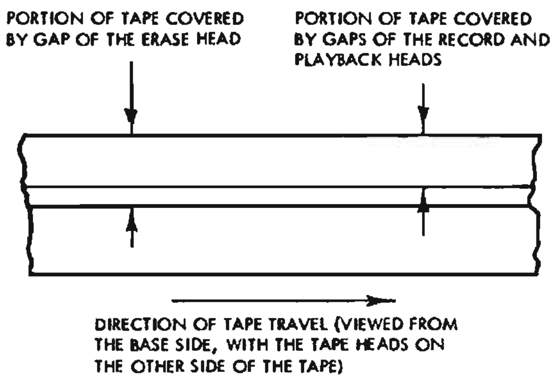


Fig. 8. Portion of the tape spanned by the gaps of half-track heads.

bars will appear to be moving backward, that is, in the direction opposite to tape motion. If speed is fast, the bars will appear to be moving forward, or in the same direction as the tape.

Apparent motion of 72 bars per minute past a given point, say the tip of a pencil, corresponds to a speed error of 1 per cent. Fair to good home machines may be expected to have speed errors not exceeding 2 per cent, namely 144 bars per minute. Good to excellent home machines should have speed errors not exceeding 1 per cent. Machines of semi-professional quality will have speed errors under 0.5 per cent, while professional ones will have speed errors not exceeding 0.2 per cent.

Erasure

The ability of the erase head to perform effectively may be tested by recording program material at a level great enough to produce maximum *permissible* indication on the record-level indicator, putting this tape through the recording process once more but without a signal input, and then playing the tape. A properly operating erase head will leave none of the first signal, at least not enough to be heard even at a high playback level.

However, if the tape has been recorded at excessively high level, one that produces a great deal of distortion, even a very good erase head may not achieve adequate erasure. It may then be necessary to put the tape through the erase procedure twice, or else use a bulk eraser.

Inability of the head to erase a normally recorded tape may be due to the following factors:

1. *Insufficient erase current.* High-frequency current through the erase head can be measured by the same technique that is used to measure bias current through the record head. As was illustrated in *Fig.* 1, a low-value resistor, say 100 ohms, is inserted between the ground lead of the head and ground. Voltage across the resistor is measured, and current is computed by Ohm's Law. The measured voltage divided by the resistor equals the current. Thus if one volt is measured across 100 ohms, current is 1/100 ampere, or 10 millamperes. Erase heads in many home machines typically use between 10 and 15 milliamperes of current. Some, however, use a good deal more.

2. *Improper vertical positioning of the erase head.* If the gap of the erase head does not fully span the recorded area, because the head is positioned too low or too high, complete erasure cannot take place. One can tell if this is the cause because the signal remaining on the tape will then have full frequency content. But if incomplete erasure is due to other causes, the remaining signal will consist mostly of low frequencies, which are the most difficult to erase.

3. *Improper azimuth alignment of the head.* If the gap departs considerably from correct azimuth, which is a position at right angle to the length of the tape, erasure may be affected. A slight azimuth error, however, will usually have little, if any, effect.

4. *Frequency of the erase current.* As pointed out earlier, the erase head tends to become less effective as the erase current increases in frequency. Means of checking this frequency have already been described.

5. *A defective head.* An erase head with shorted turns will not function properly. The test here consists of head substitution. A head of poor design will give inadequate performance. Here one may try using a different brand of head, assuming it can be mounted on the tape deck. Some manufacturers produce two kinds of erase heads, one with a single gap and the other with two gaps side by side so that the tape is twice exposed to an erasing field in a single pass. Substitution of the two-gap head may achieve the desired improvement.

Head Height

The erase, record, and playback heads must be vertically positioned so that their gaps all span the same portion of the tape, that is, the same track. This is illustrated in *Fig.* 8 for half-track (mono) recording. The same principle applies to two-track and four-track stereo heads.

As pointed out in the preceding section, if the erase head is out of vertical alignment, incomplete erasure results. If the machine has separate record and playback heads, there are two additional problems should these heads be out of vertical alignment with each other: (1) Playback signal will be less than maximum, with consequent deterioration of the signal-to-noise ratio, if the playback gap does not span as much of the recorded track as possible. (2) There may be crosstalk—signal pickup from an adjacent track—because the playback head partially spans or comes too close to an adjacent track.

In the case of half-track heads, visual alignment can often suffice. This means aligning by eye the edge of the tape with the edge of the gap. To maximize the chances of accurate alignment, the position of the tape relative to the head should be that which occurs in the normal "dynamic" state—that is, when the tape is moving past the head. To obtain the dynamic position, place the transport in motion, then shut off the power (if necessary, by removing the power plug from the wall outlet) so that the tape is stationary against the head. If pressure pads are employed, remove the pressure pad holder after shutting off the power. Then visually align the gaps to the tape.

To check the height adjustment of the playback head, one can play back a commercial prerecorded tape and move the head up and down. When the output is maximum, as measured by a VTVM, playback-head vertical alignment is correct. It is advisable to use a test tape with a low-frequency signal, such as 400 cps or less, so that slight changes in azimuth as one moves the head do not appreciably affect the playback level.

If the tape machine uses a separate record head, the latter can be aligned to the playback head by adjusting the record head's height for maximum signal output while simultaneously recording and playing a low-frequency signal. Then the erase head height can be adjusted for maximum erasure.

In the case of stereo heads, involving two gaps one above the other, as shown in *Fig.* 9, vertical positioning is more critical than for half-track heads. Visual alignment is probably a preliminary step, with further steps more or less mandatory. The checks already described can be used. One can further check on the basis of crosstalk. When playing a commercial prerecorded two- or four-track tape containing program material, if the playback head is improperly positioned to the extent that it impinges on an adjacent track, the signal of the adjacent track will come through with full frequency content. If the playback head is incorrectly positioned so that it does not actually span

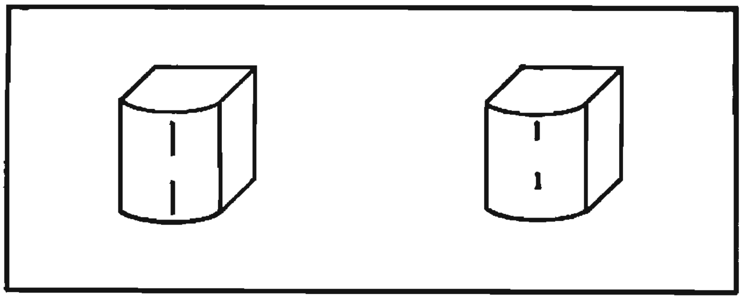


Fig. 9. Gaps of two- and four-track stereo tape heads.

part of an adjacent track but comes very close, then the crosstalk will consist predominantly of low frequencies, which extend beyond the track to a greater extent than the high frequencies. On the other hand, if the crosstalk consists predominantly of high frequencies it results in a tinny sound coming through from the adjacent track. This is indicative of crosstalk within the head, which has nothing to do with vertical position. Such crosstalk is in the nature of transfer of the signal from one section of the stereo head to the other because of magnetic coupling.

More precise and more direct methods of checking and adjusting head height are available, as follows:

1. *Test Tape.* RCA test tape No. 12-5-64T is illustrative of a tape containing a section—Part 1—specifically designed for height alignment of four-track heads. As shown in *Fig.* 10, the entire width of Part 1 of this tape is recorded except for a .043-inch band erased at the location of track 3, which is called A-2. The recorded signal is 1000 cps at 3.75 ips and 2000 cps at 7.5 ips. The instructions state: "In order to adjust the head assembly height, move the head up or down until the audible tone or the output meter reading on the A-2 track is at a null or minimum. Note that a head height either up or down from this proper position will cause an increase in signal level."

2. *Visual Indication.* Reeves Soundcraft Corp. makes a product called Magna-See that enables one to view the signal recorded on the tape. This has a number of uses, including checking head height, azimuth, and head wear. To use Magna-See, the tape is recorded at slightly higher volume than normal, the tape is immersed in a bath of Magna-See solution, and the tape is allowed to dry out, whereupon the recorded track becomes visible.

Fig. 11 shows how a two-track stereo recording would appear after exposure to Magna-See. One would first adjust the height of the record head on the basis of several trials with the aid of Magna-See. If the tape recorder has separate record and playback heads, then the playback head would be aligned on the basis of maximum output when simultaneously recording and playing a low-frequency signal. Or, one could perhaps temporarily connect the playback head to the record amplifier and make a recording through this head; then view the results by means of Magna-See. To align the erase head, one could erase a tape that has been recorded with an aligned head, and check the results with Magna-See to determine whether the recorded sections have been erased excessively or insufficiently.

Test Tapes

As indicated in this article and the preceding one, various test tapes are available to facilitate checking and adjusting tape machines. Some test tapes have but a single purpose, while others have several. Typically, the single-purpose tapes are for azimuth alignment. Thus Audio Devices produces an "audio alignment tape," containing tones of 1000, 5000, and 7500 cps when played at 7.5 ips. The two lower tones are of 30 seconds duration, while the highest lasts 60 seconds. The purpose of the lowest tone is to set level. A preliminary adjustment of azimuth is made with the 5000 cps tone, and a final adjustment with the 7500 cps tone.

Ampex's No. 5563 test tape is an example of a multi-purpose tape, to be used at 7.5 ips. It contains a 10,000 cps frequency for azimuth alignment, a 250-cps tone for setting playback level (adjusting the VU meter in playback), another 250-cps tone at a lower level to be used as a reference for a frequency response check, and a series of tones for checking frequency response at 10,000, 7500, 5000, 2500, 1000, 400, 200, 100, and 50 cps. Similar tapes are produced by Ampex for the 3.75 and 15 ips speeds.

One of the most comprehensive test tapes that has been made, intended specifically for use by the audiofan, is the one put out several years ago by Dubbings. Although no longer manufactured, it may still be available from some audio supply houses. This tape contains a series of beeps five minutes apart for checking tape speed; a 400-cps tone recorded at a level corresponding to 2 per cent harmonic distortion; a high-frequency tone for azimuth alignment; a series of tones for checking frequency response; a tone for checking wow and flutter by ear; and a series of progressively quieter signals for checking signal-to-noise ratio (when the signal on the tape is no greater than system noise, this indicates the signal-to-noise ratio, identified by a voiced announcement).

A number of tapes have appeared for testing stereo machines. For example, RCA Victor's 12-5-64T is a quarter-track tape, with recordings on two of the four tracks. Among the tests is one for adjusting the height of the head (as previously described); another is a tone that permits one to phase the

Fig. 11. Appearance of two-track stereo tape after exposure to Magna-See.

speakers—the tone appearing midway between the speakers when phasing is proper. Another stereo test tape is the Audiotester 30-208, which includes the sound of a metronome for stereo balancing. The Sonotape Stereophonic Alignment Tape includes a test for correct track placement; that is, namely sound from the left coming from the upper track and sound from the right coming from the lower track when the tape moves from left to right; tests for frequency and loudness balance between tracks; tests for proper speaker location based on degree of separation and mixing of sounds from each track.

A word of caution is in order with respect to test tapes, particularly those bearing on azimuth alignment and frequency response. The heads of a tape machine tend to become magnetized gradually, causing partial or complete erasure of high frequencies on a tape passing over the head. Therefore it is recommended that the tape heads be demagnetized before a test tape is placed on the machine. Æ

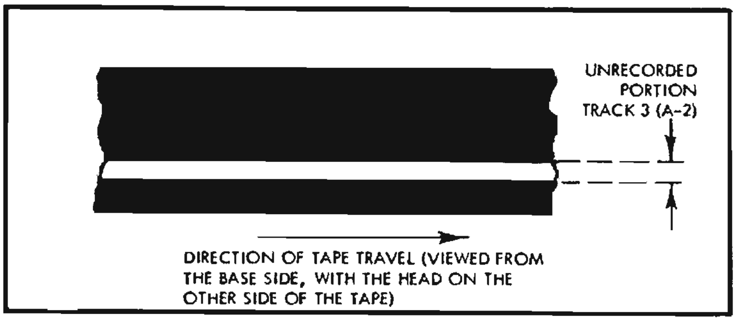


Fig. 10. Portion of RCA test tape No. 12-5-64T used for height alignment of four-track stereo tape heads.

Level-Test Tapes Aid Program Reproduction

JAY C. ABBOTT*

Wide differences in level from tape to tape have plagued the broadcaster as well as the serious tape recordist for a long time. Here is a proposal, which, if universally adopted, may well eliminate the problem.

THE ALMOST UNLIMITED level range of modern recording tape unfortunately is a contributing factor towards sadly disappointing the listener when tapes which sounded brilliant at the time of recording are played over radio or transferred to disks.

The lack of positive levels that would apply to all tape recorders, regardless of make or working condition, would, from the experience of this writer, seem to be the factor needing correction.

Perhaps this would then be a good time for tape recordists to give thought to the plight their work has brought to the broadcasting and recording industry.

For example, the working day of the station engineer finds him faced with an endless stream of records and tapes which he must get on the air, often while dubbing as announcer, phone-answerer and general office boy. Many of these same factors apply to commercial record companies who have daily production quotas to consider.

While commercial records in general are recorded within basic db level ranges, the tapes an engineer can encounter in a day might vary from a whisper to the roar of a hundred jets on take-off.

Considering that a radio station disk jockey, who nowadays often doubles on the controls, might be clear across the room engaged in the lawful pursuits that management has also found to occupy his time, it is no wonder that tapes containing a wide range of sound levels often come over the air with disastrous results.

Record engineers have the struggle of compressing tape sounds within the db range their equipment is capable of transferring to records. Within either industry the transfer of program material tape-to-tape, to the air, or record surfaces often ends up in a game of Russian Roulette due to the lack of known levels.

Level-test tapes made at the time of recording can do much to correct this gap. There is a twofold purpose in having level test tapes: the first being to establish the general level used by the operator in recording his program; the second is to establish a known working level of the recorder being used. It is amazing the variance of zero db levels that one can encounter going from one tape recorder to another, even of the same manufacture.

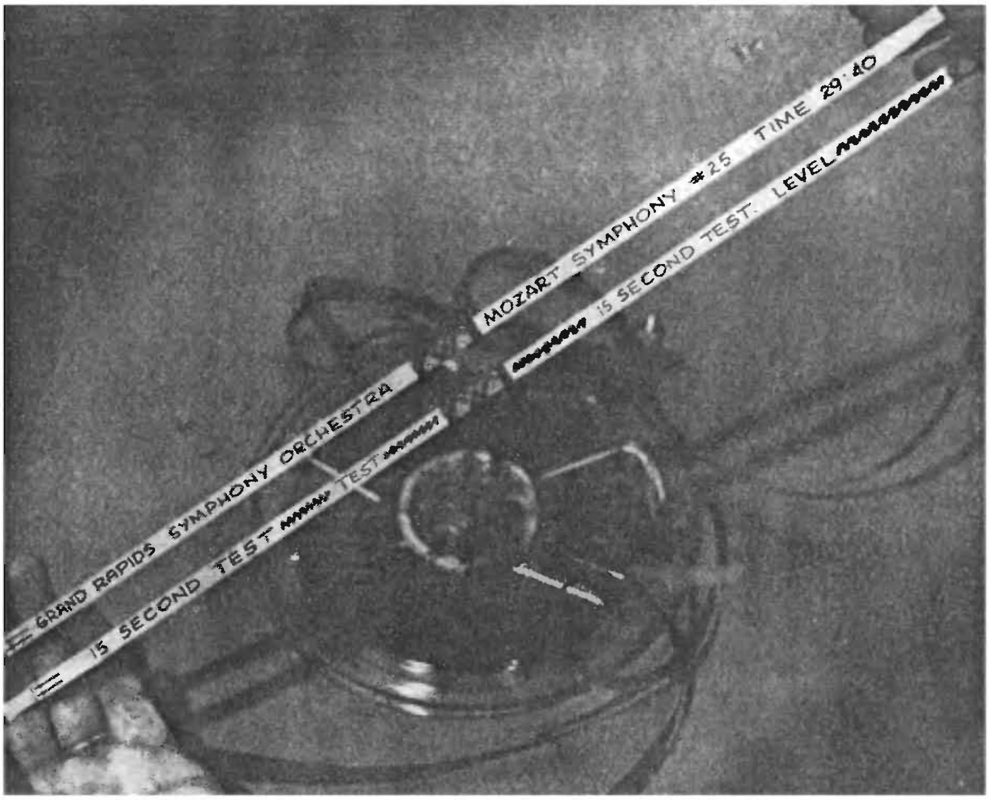


Fig. 1. Marking the level-test tape.

Factors controlling this variance can be bias level at the moment of recording, tube and component conditions, tape and surface condition of the record and playback heads. Level-test tapes allow for all these variances and permit, when duplicating or broadcasting, operation at the highest possible signal level without fear of overload or change in volume level during the run.

Consider for a moment an engineer faced with putting a live tape recording on Ravel's *Bolero,* on the air or a record master. This dynamic concert favorite opens with a soft roll of the snare drum and plucking of the harp. For five minutes this beat is exchanged among the various instruments of the orchestra with little increase in overall volume. Then the roof begins to bulge and the operator is faced with a db meter that doesn't seem to know any limit to its rise. At the end of twelve minutes the operator generally has overcome the single effect the conductor and orchestra have labored so hard to create—that of a constantly rising crescendo of sound, climaxed with a loud discordant noise marking the end.

With level-test tapes the operator can within seconds adjust his equipment so as to completely preserve the musical text indicated by the composer and win for himself the applause of listeners everywhere.

The problem of established levels for tape recording was encountered by the

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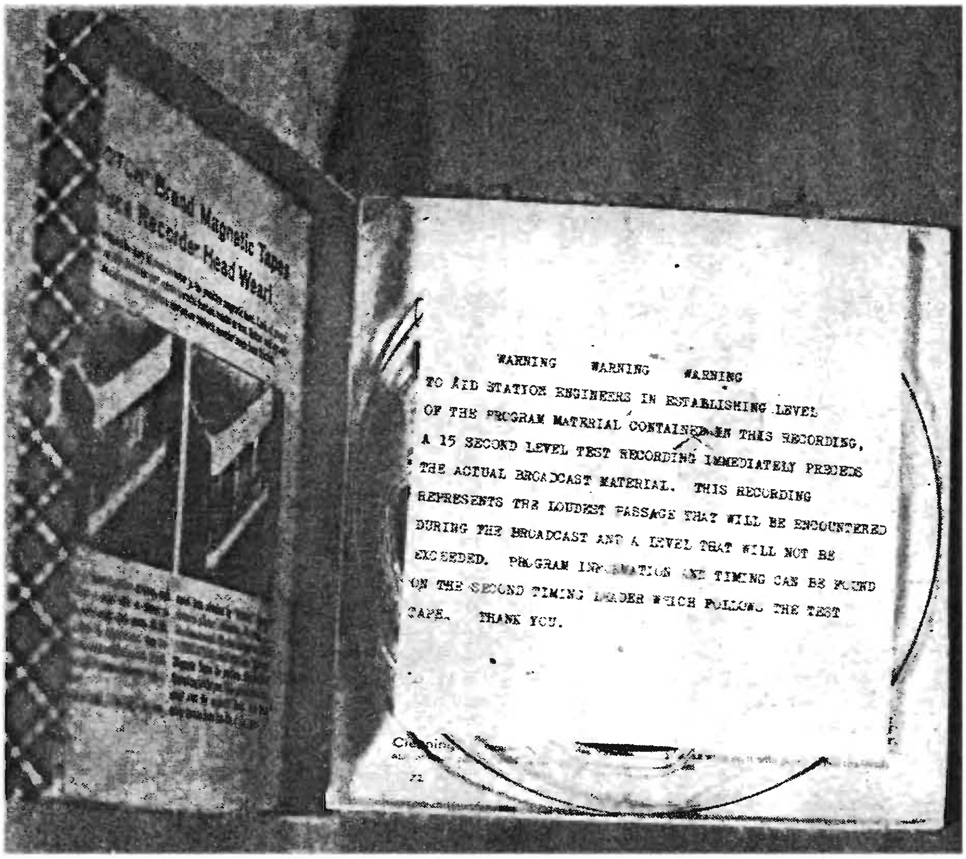

Fig. 2. Flyer is packed with tape to inform broadcast engineers about test tape. This will prevent him from inadvertently putting the test tape "on the air."

writer during production of a radio series featuring a 90-piece symphony orchestra. A group of this size is capable of producing levels that can exceed even the db level limits of tape. Add to this orchestra a 600-voice festival choir and the stage is set for a tape broadcast that would tax the ability of all but the most experienced engineer.

Let an inexperienced person happen to be at the controls for such a broadcast however, and you can imagine the disaster that would result—*and did!* This then is the reason for the effort that went into formulating this proposed idea for level-test tapes.

The solution which is offered here to correct the problem of tape level has been taken in part from a practice followed by news services in the transmission of wirephotos throughout the world.

Preceding each picture transmission, a brief interval is taken to transmit a sending level to which each set on the network is then adjusted by the receiving operator. Through this method the operator in Grand Rapids, Michigan; Spokane, Washington; or Atlanta, Georgia received identical reproductions of the transmitted image. All variations are eliminated.

Following the suggestion of creating a tape recording test signal we turned to the musical score being recorded and decided to employ the loudest passage contained in the recording and to copy this signal for approximately 15 seconds. By inserting this special recording as a leader to the actual recording the station engineer or recording studio could set their equipment for the loudest known signal they would encounter in the program which followed.

Later another procedure was developed which accomplished the same results without the time-consuming process just described. The revised system however requires audio signal generating equipment and for the benefit of those lacking such equipment or for those who would find such practice not practical "in the field," the first procedure is described in full.

To create a level-test tape without the aid of audio signal generating equipment the recordist selects that passage he knows to contain the loudest db level range. This is then recorded for approximately 15 seconds, making sure that the controls are not changed after being set. An error at this stage can throw the entire procedure off balance and destroy the intent of level-test tapes.

We first employed this system in preparing symphony recordings and the enthusiasm with which it has been received by station engineers has been most encouraging. The finished product on the air has become a true presentation of the program material contained on the tape with absolutely no evidence of overloading or a rush to adjust receiver volume. All of the dynamic range so favored by musicians is preserved especially when heard on FM transmission.

While most stations prefer full-track recording, when necessary or the occasion demands, anything will do. Here again, level-test tapes will permit an adjustment to be made to correct for the natural loss that is encountered when half-track recordings are being played on full-track equipment. The second track naturally being blank.

In practice the system offered here is achieved in the following manner, however, the individual recordist can easily alter the system to fit his individual needs.

While these notes refer to symphonic recordings, all procedures also apply to recordings of popular music, group or single performers, any recording where it is desired to preserve the full range of tone used by musicians, actor or speaker to present his performance to the public.

The symphonic series I have mentioned is first taken on what is called the music master tape, which is edited to the time limits of the program. Voice announcements are then added to, and in cases over, the music on a first duplication process.

Either during initial editing or when dubbing voice, careful note is made of the dynamic range of the symphony, concerto or suite being worked. The selected passage is duplicated making sure the controls are set so as to duplicate exactly the volume intensity.

During the second editing on what has now become the "program master" tape, the selected passage representing the loudest part of the program is attached to the lead of the tape with appropriate leader strips.

The program master tape is now ready for duplication in the required number to accommodate stations carrying the symphony broadcast. The same test level added to aid radio stations now serves to assist the duplication service, if employed, in quickly and accurately adjusting their equipment. Further, by recording at the highest possible level through the test level system, a considerable drop in tape hiss has been noticed in the duplicated tapes.

Where master tapes are released for use and duplication not employed the same procedures would apply with the test strip applied to the beginning of the program.

By specific instruction duplicated tapes are returned with the test level strip intact. However leader strips must be placed between the test and program material and so noted. (See *Fig.* 1.) The first leader strip is carefully marked in crayon or ink, "15 second test-level recording." On the second leader strip immediately preceding the actual broadcast required program information and timing is noted.

As a final step in packaging the program a flyer is attached so as to lie over the program reel. (See *Fig.* 2.) Sug-

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gested copy for this flyer might be as follows.

WARNING WARNING WARNING TO AID STATION ENGINEERS IN ESTABLISHING LEVEL OF THE PROGRAM MATERIAL CONTAINED IN THIS RECORDING, A 15 SECOND LEVEL-TEST RECORDING IMMEDIATELY PRECEDES THE ACTUAL BROADCAST MATERIAL. THIS RECORDING REPRESENTS THE LOUDEST PASSAGE THAT WILL BE ENCOUNTERED DURING THE BROADCAST AND A LEVEL THAT WILL NOT BE EXCEEDED. PROGRAM INFORMATION AND TIMING CAN BE FOUND ON THE SECOND TIMING LEADER WHICH FOLLOWS THE TEST TAPE. THANK YOU.

On those recording machines that do not employ db level meters but favor other devices to indicate peak recording levels, the same means that have been outlined can be employed. For "magic eyes," the eye would just approach overlap, or for neon lamp the overload lamp would just flicker.

The principal fault with the original procedure was the time-consuming factor of finding and reproducing a duplication from the loudest musical passage. Further the operator had to find a passage where the sound was sustained long enough to give the operator time to set his equipment.

Returning to the original idea employed by wire-photo transmission where a single note is used to set level it was decided to experiment with the possibility of employing a generated tone signal that could be set to zero db level.

In practice the tone system for preparing level-test tapes has proved quite simple and is a procedure that does not necessarily have to be performed on the spot or in the field.

First a series of duplications were made and on completion and with the controls still at the same level, input connections were transferred to an audio generator. The note selected for this test was 400 cycles and the generator output control adjusted until the recording meter reached the 100 per cent or zero db level. 10 to 15 seconds of tone was then recorded for each program. These test tone tapes were then affixed to the program tapes as previously outlined.

In each case and regardless of the program material the tone level test satisfactorily performed the task of establishing a positive known playing level. Satisfied that this revised system could be depended upon a second series of broadcasts were released with the tone test system. A call from the first station confirmed that this system could be depended upon and further the steady tone signal permitted an even faster setting of controls.

Where the operator is not planning duplication of the program, he should record a test tone signal at zero db on the machine used to make the original recording and attach it ahead of the program.

If a recording has been made where inadvertently the level creeps into the red zone or, say plus 2 db in level, then the operator must select this point in preparing his test tape and not zero db. If overlooked or a regular zero db tape is attached then overload can certainly be expected. As with the musical level test tapes, operators should make a definite point of indicating the presence of such material on tapes released for record or air-time use.

When a large number of duplications are being prepared, sufficient tape can be recorded as zero-level test tape to supply the current production, providing the test tape is recorded at the same time to preserve the original intent of this system—to present a true picture of recording factors at the time the production was undertaken.

The procedures outlined in this article represent a start in what is hoped will be an aid to those engaged in the recording field. Experience undoubtedly will show the system can be modified or simplified and still achieve the same results. For example, tape manufacturers might come out with a line of colored timing strips which could be used to denote or identify test recordings to the industry. Possibly a 15 second test may be felt to be too long. Indeed a host of other suggestions may be forthcoming.

Meanwhile, however, the challenge is given to recordists to recognize the plight their work has brought to the broadcasting and record fields and what, if willing, they can do to overcome the problem. Local experience with this system has conclusively proved that in giving engineers a break in their daily work they have more than reciprocated in presenting the program material so offered. Æ

Stereo Considerations

HERMAN BURSTEIN*

For those considering the purchase of a stereo tape machine here is an informed discussion of the merits of four-track tape systems.

Most of what has been said in previous articles applies to mono and stereo tape operation alike (unless specifically directed at one or the other). However, stereo recording and playback entail certain special questions, which are the concern of the present article. The topics to be discussed here are four-track versus two-track stereo, the tape cartridge, coordination of the two channels, and conversion for stereo.

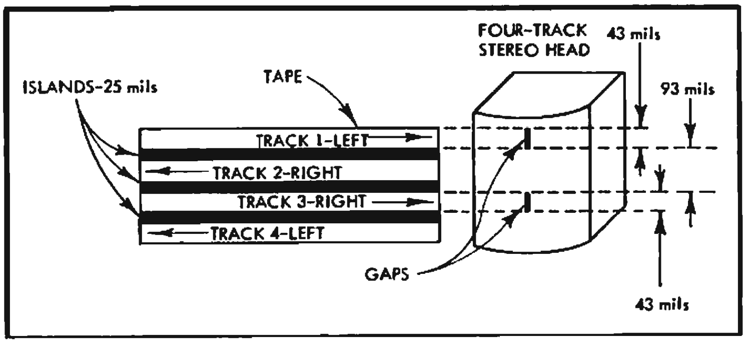

Fig. 2. Four-track stereo tape.

Four-Track Stereo

When stereo tape was first introduced, it employed the same track arrangement as half-track mono tape, shown in *Fig.* 1A, except that the lower track (for a tape running from left to right) was used for the second channel. Originally, a staggered-head arrangement, shown in *Fig.* 1B, was used to record and play 2-track stereo tape, but this eventually gave way to a single in-line head, shown in *Fig.* 1C. The staggered arrangement employed two conventional mono heads, spaced about 1¼-in. apart, and positioned so that the gap of one head spanned the upper track while the gap of the other spanned the lower track. Use of separate heads permitted individual adjustment of azimuth of each gap, assuring maximum treble response on each channel; and it avoided the problem of crosstalk between heads, namely the appearance of the left signal in the right head and vice versa.

With improvements in manufacturing techniques, the in-line head proved to be a reliable and not overly expensive device. Thus the staggered-head arrangement, innately a clumsy one, became obsolete, and so did the tapes that had been recorded by this method. It was not feasible to use an in-line head to play a tape with a displacement of 1¼-in. between channels, corresponding to a time difference of 1/6 second at 7.5 ips and 1/3 second at 3.75 ips.

Two-track stereo tape ran into problems of tape economy and convenience

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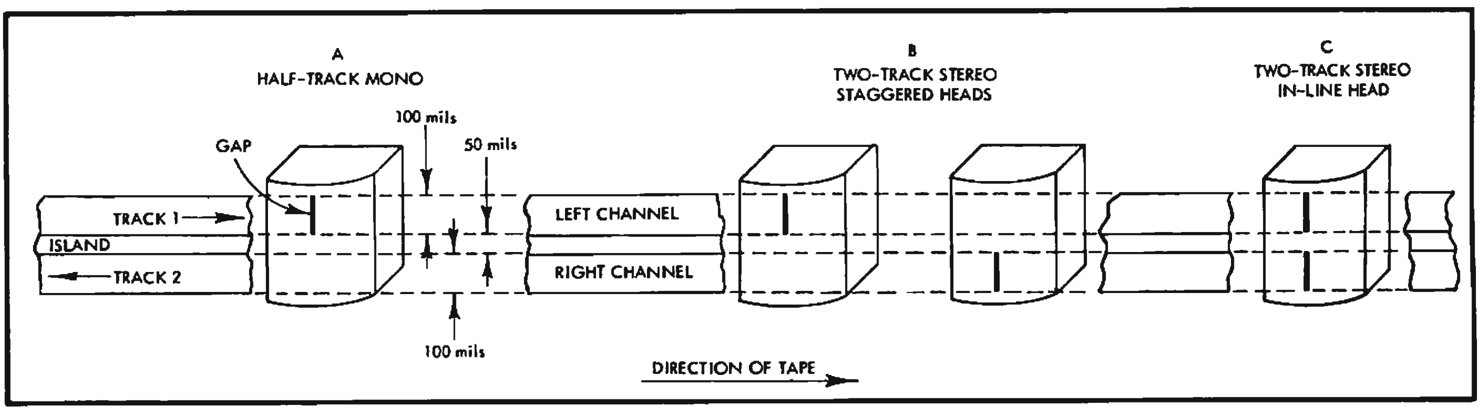

Fig. 1. Head configurations for half-track mono and two-track stereo tape.

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of operation. In the case of mono half-track operation, one could record or play the tape in one direction, reverse the reels, and promptly continue operation in the other direction. But two-track stereo permitted the tape to be used only in one direction, which was wasteful of tape, particularly for commercial applications. In the case of prerecorded tape, the tape itself presents a major item of cost, whereas in the case of a phonograph disk the vinylite material is a matter of a few cents. Moreover, after a two-track tape has been recorded or played, it is necessary to rewind it in order to get it back on its original reel; this is not the case for mono half-track tape, where half the width of the tape (approximately) is recorded in one direction and the other half in the opposite direction.

The problems of tape economy and convenience of operation were solved by four-track stereo recording, as shown in *Fig.* 2. Tracks one and three are recorded (and played) in one direction, and, after the reels are reversed, tracks four and

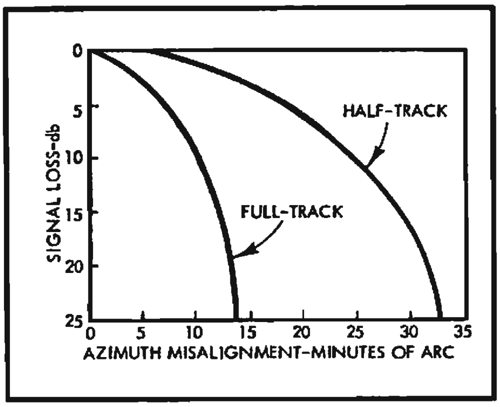

Fig. 3. Signal losses due to azimuth misalignment at 7500 cps at a tape speed of 7.5 ips.

two are recorded in the other direction.

The major disadvantage of four-track compared with two-track tape is a reduction in signal-to-noise ratio. The tracks of the former are about half the width of the latter, so that there is a proportionate reduction in the amount of signal that is recorded. Consequently the signal level obtained from the four-track tape in playback is about 6 db less than the signal from a two-track tape. This means that the ratio of audio signal to noise and hum produced by the tape recorder electronics is decreased 6 db.

However, ways are being found around this problem. For one thing, tape electronics today tend to be less noisy than those of yesteryear due to improvements in circuit design and in tubes or transistors. For another, it is possible through skillful design to produce playback heads with increased output for a

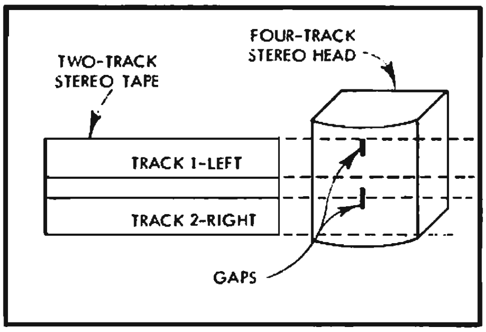

Fig. 4. Mismatch between the lower gap of a four-track stereo head and the lower track of a two-track stereo tape.

given amount of signal recorded on the tape. Third, there have been continual improvements in tape quality, and it is reasonable to believe that time will bring improvements with respect to the amount of signal that can be recorded on the tape without increasing distortion or other undesirable effects. In sum, one may look forward to eventually achieving a signal-to-noise ratio on four-track track that approaches the ratio achieved in the past on two-track tape.

On the other side of the coin, four-track stereo tape has two positive advantages over two-track tape. One is the fact that azimuth alignment becomes less critical as track width is decreased, thereby reducing treble losses due to slight departures from exact azimuth. *Figure* 3 suggests the benefits obtained from narrowing the track. It compares azimuth losses for a half-track mono recording with those for a full-track mono recording. Obviously, much greater azimuth misalignment, in relative terms, is tolerable for the narrower track. The benefits obtained by going from two- to four-track stereo are comparable with those indicated in *Fig.* 3.

The second advantage of four-track tape lies in the greater separation between the two gaps of the in-line head. Hence there can be greater separation between the two sections of the head, resulting in less crosstalk. Comparison of *Figs.* 1C and 2 shows that there is 50 mils (thousandths of an inch) separation between the gaps of a two-track stereo head, compared with 93 mils between the gaps of a four-track head.

Although it is indicated that four-track stereo tapes will supersede two-track stereo tapes, there will remain the problem of playing valued two-track tapes purchased or recorded in the past. Therefore the manufacturers of tape machines have sought to make it possible to play two-track stereo tapes with four-track heads. The problem lies in the fact that the lower gap of the four-track head does not lie fully within the recorded area of the lower track of a two-track tape. This is made clear in *Fig.* 4. The fact that part of the lower gap spans unrecorded space means less output on the lower track, with a consequent reduction in signal-to-noise ratio.

Some tape machine manufacturers have chosen to accept this limitation on signal-to-noise ratio of one of the channels. Others, however, have incorporated a mechanical device for shifting the head up and down. For four-track tape, running from left to right, the head is shifted up. It is shifted down for two-track tape. There is some danger of impairing azimuth alignment as the head is shifted up or down. Consequently in a few high-price machines a separate head has been introduced for playing two-track tapes.

Most home tape machines use the same head for record and playback. Such machines, perforce, permit four-track recording as well as playback. However, the higher-price tape machines usually employ separate record and playback heads. Some of them use two-track record heads, while others provide four-track record heads. It would appear, however, that eventually all home machines will permit four-track recording.

(As a side note, it is of interest to observe that the four-track head makes possible four-track *mono* operation, thereby doubling the playing time obtainable from a reel of tape. The recording or playback sequence is: tracks one, four, three, and two. A number of tape machines, through extra switching facilities, take advantage of this opportunity. Switching must do the following: (1) In recording, it must channel the input signal first to one section of the head (for tracks one and four) and then to the other section of the head (for tracks three and two). (2) At the same time, in recording, it must shut off the bias current to the record head section not in use and it must shut off the erase current to the erase head section not in use. (3) In playback it must channel the signal first from one section of the head and then from the other section to one of the playback amplifiers. If one uses

Fig. 5. The RCA tape cartridge.

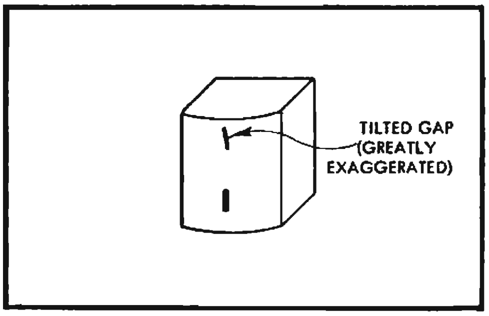

Fig. 6. Stereo head with tilted gap.

double-play tape at 1.875 ips, it is possible to record as much as 17 hours of program material on a 7-in. reel.)

The Stereo Cartridge

It has long been the goal of a section of the tape industry to simplify the playing of prerecorded tape to the point where this is just as easy as playing a phonograph disc. To this end the tape cartridge has been introduced. First on the scene was the RCA cartridge, shown in *Fig.* 5, which houses the tape in a plastic container with apertures that permit the tape to contact the heads, capstan, and guides. It is merely necessary to position the cartridge on a tape machine designed for the purpose and push a button, whereupon the machine takes over without the need for the operator ever to touch the tape. Some cartridge players are designed to stop the tape after it has played in one direction, while others will reverse the tape and play it back in the opposite direction against another head, after which the tape is automatically stopped.

The RCA cartridge can hold up to 600 feet of tape, which at 7.5 ips affords a maximum playing time of 32 minutes if the tape is operated in two directions. But to be competitive with the stereo disc, the stereo cartridge must be able to provide up to an hour of program material. Therefore it is necessary to reduce the speed of the tape cartridge to 3.75 ips. In sum, the RCA tape cartridge and the 3.75 ips speed go hand in hand.

Fortunately, improvements in tape heads, in other components, and in techniques have made it possible to obtain good fidelity at 3.75 ips. Thus it appears that the tape cartridge will prove to be a suitable medium for popularizing prerecorded tape. While the 3.75 ips speed may not be suitable (yet) for truly high fidelity, it is still good enough to provide pleasurable reproduction of music to the many persons who own moderate-price sound systems and who do not demand the ultimate in available quality.

On the other hand, for those who demand the best it appears that for some time to come the 7.5 ips speed will be used, combined with four-track stereo recording on open reels.

At 3.75 ips, it is possible today, owing to playback heads with extremely fine gaps, to preserve frequency response substantially out to 15,000 cps, closely rivalling the performance at 7.5 ips in this respect. Still, in terms of distortion and signal-to-noise ratio, 3.75 ips recordings lag behind those at 7.5 ips. To achieve response out to 15,000 cps or thereabouts at 3.75 ips, it is necessary to reduce bias current fed to the record head below the value employed at 7.5 ips, thereby reducing treble losses due to bias erase. The decrease in bias current results in an increase in distortion. The increase in distortion can be offset by lowering the recording level. But the latter measure means less signal on the tape and therefore a lower signal-to-noise ratio in playback. In practice, the course usually followed is to accept some increase in distortion and some decrease in signal-to-noise ratio, rather than just one or the other.

As stated just before, response to 15,000 cps is feasible at 3.75 ips. But to achieve such response, greater attention must be paid than at 7.5 ips to factors that can adversely affect treble response: too wide a gap in the playback head; incorrect azimuth alignment; poor tape to head contact because of dirt, brittle tape, improperly adjusted pressure pads, etc.; excessive bias current to the record head; improper record or playback equalization.

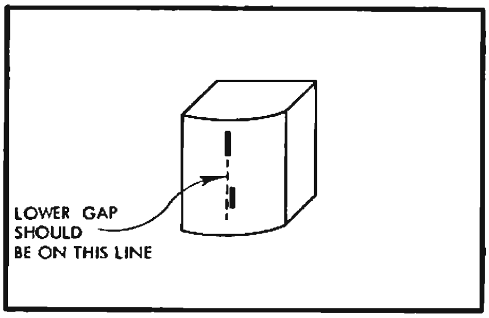

Fig. 7. Stereo head with displaced gap.

Although the 3.75 ips speed has only quite recently, and after much striving, proven capable of good quality, the 1.875 ips speed is hard on its heels in vying for serious consideration. The further reduction in speed would make prerecorded tapes still more economical and would permit tape cartridges and cartridge players to be more compact. Along this line CBS and the 3M Co. recently announced a tape cartridge designed to be operated at 1.875 ips. While commercial production was estimated to be several years distant, demonstrations to the trade were convincing as to the possibilities of good results at this speed. Moreover, the machine designed to play this cartridge incorporated a changer mechanism, putting tape fully on a par with the phono disc for simplicity and convenience of operation.

It may be added that a number of open reel tape machines already incorporate the 1.875 ips speed. While they do not claim high fidelity performance at this speed, the results are surprisingly good. For example, they can reproduce music quite satisfactorily for background or party purposes, where the presence of competing sounds makes it pointless to strive for high fidelity. But the 1.875 ips cartridge proposes to go a major step forward by lifting the quality at this speed to meet at least minimum high fidelity requirements.

Coordination of Channels

A unique problem of stereo tape machines is that of properly coordinating the two channels in various respects. This problem may lie with the manufacturer of the machine, with the user, or partly with both.

1. *Co-Linearity of the Stereo Head Gaps.* One of the problems of manufacturing a good stereo head is to insure that the gaps are in exactly the same straight line. If one gap is tilted with respect to the other, as illustrated in *Fig.* 6, then it is not possible to achieve correct azimuth on both channels simultaneously; hence high frequency response will suffer on one channel or the other, or both. If one gap is displaced with respect to the other, as illustrated in *Fig.* 7, then the time relationship between the left and right signals will be altered. Some experts have claimed that extremely small changes in the time relationship can significantly alter the stereo effect.

2. *Equal Playback Levels.* One section of a stereo playback head may produce a few db more signal output than the other for the same amount of signal level recorded on the tape. Or one playback amplifier may have more gain than the other. To determine the relative playback levels on each channel, set the playback gain controls at the position most apt to be used, and play a full-track test tape. Compare the signal levels with a VTVM or by ear, assuming in the latter case that the channels of the rest of the audio system are balanced right through to the speakers. Adjust one of the gain controls on the tape machine for equal volume on both channels. If the tape machine does not have separate playback gain controls for each channel, then it becomes necessary to use the input level sets, if any, on the audio system amplifier to equate the signals. If there are no input level sets, then the balance control of the stereo amplifier must be used for this purpose. It is then necessary to take note of the balance control setting which achieves signal equality on tape playback.

3. *Equal Recording Levels.* Some tape machines use a single switched record level indicator for both channels, while others use separate indicators for each channel. The fact that both indicators

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give the same reading does not necessarily signify that the same level is being recorded on each channel of the tape. For the same signal input, there may be differences in recording level due to variations between the sections of the stereo record head. Or, for the same signal input, the recording level indicators may each give a slightly different reading. To check for equal recording level and the relative indications by the record level indicators, the following procedure can be used.

Assume that the position of the playback gain controls for equal signal output has already been determined. Feed the same signal, say from a mono phonograph disc, into each recording input. Adjust the input gain controls for equal indications on the record-level indicators. Play back and compare the signal outputs with a VTVM, or by ear. If these signal outputs differ substantially, repeat the process, but after reducing the recording gain control setting for the channel with the louder signal. Continue this procedure until the playback signals on the two channels appear equal. Now note the relative indications on the record-level indicators, and be guided accordingly in the future.

4. *Matched Frequency Response.* A check for reasonably similar frequency response on each channel can be made quite easily. Record a high quality mono disc on both channels; the disc should be one that substantially covers the audio range. Then in playback compare one channel with the other by switching between the left signal and the right signal, as most stereo amplifiers enable one to do. If there is a significant difference in frequency response between channels, this can be due to such factors as differences in equalization, in bias current, in playback head gap width, and in azimuth alignment. The last three factors named will primarily affect treble response.

5. *Common Bias Frequency.* It is im-

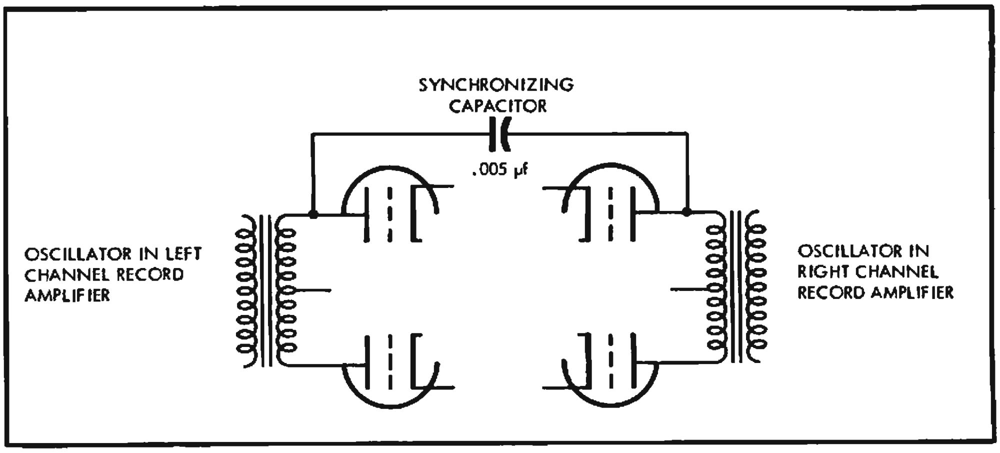


Fig. 8. Synchronizing the frequencies of two bias oscillators.

portant that the bias frequency be the same on each channel. Bias current passing through one section of an in-line head tends to leak through to some degree to the other section. Hence there are two bias currents through each section, although of different magnitude. If the frequencies of these two currents are different, there will be resultant beat frequencies that are recorded on the tape. If the stereo tape recorder employs separate record amplifiers for each channel, each with its own bias current supply, it becomes necessary to synchronize these two frequencies so they are the same. This is a simple matter, at least for the audio technician. As shown in *Fig.* 8, a small capacitor can be connected from the plate of one bias oscillator to the plate of the other for synchronization. This assumes that the two bias frequencies were originally fairly close together, say within about 10,000 cps of each other.

6. *Crosstalk.* Coordination between channels in this case means keeping the left signal in the left channel and the right signal in the right channel. Crosstalk can occur because of improper vertical positioning of the head or because of construction of the head. In modern high quality heads, crosstalk within the head has been reduced to negligible proportions by shielding between sections and by other design factors. Such crosstalk as does occur consists primarily of the higher audio frequencies, so that crosstalk due to the head characteristically has a tinny sound.

Fig. 9. Four-track conversion kit.

Conversion for Stereo

Converting a tape machine for stereo purposes may mean either (1) converting from mono to stereo or (2) converting from two- to four-track stereo. In the latter case, the conversion is usually quite simple, involving the replacement of the two-track head by a four-track one in the same mounting fixture, and possibly the addition of an electronic component or two. *Figure* 9 shows a conversion kit put out by Ampex that enables owners of its two-track stereo machines to convert a four-track stereo for playback.

Most tape recorder manufacturers offer a conversion kit. If they do not, it is possible to purchase a stereo head from one of several prominent manufacturers of tape heads, such as Brush, Nortronics, Shure, and Viking. Sometimes the head is available in a variety of mounting styles so as to fit the mounting fixture in a particular machine. For example, as shown in *Fig.* 10, Nortronics makes three mounting styles, which among them will fit the majority of tape machines on the market. For all other machines, Nortronics has a fourth mounting style, with an accompanying mounting bracket, as shown in *Fig.* 11. In the last instance it is necessary to attach the mounting bracket to the tape deck with self-tapping screws. The bracket has slotted mounting holes to permit proper vertical positioning of the head relative to the tape. Azimuth adjustment is performed by bending the bracket. It is usually desirable when using an "outboard" head installation of this kind to also install a tape guide post, such as that in *Fig.* 12, to insure proper passage of the tape across the head.

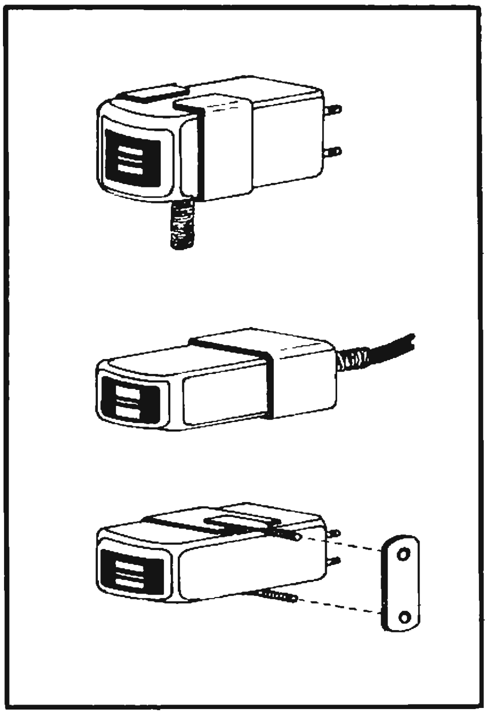


Fig. 10. Various head mounting methods for four-track stereo conversion.

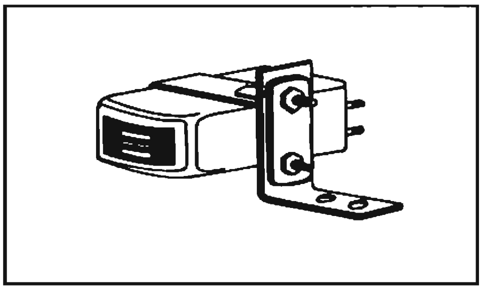


Fig. 11. Mounting the head "outboard".

If the tape machine is a mono device, then it is necessary to install not only stereo heads (including a stereo erase head), but also additional electronics for the second channel. If the tape machie is intended for playback only, at least for stereo, it is likely to be unnecessary to purchase a second playback amplifier because most stereo amplifiers provide an input for accommodating the signal directly from a tape head. On the other hand, if it is desired to record as well as play stereo tapes, then a second tape record amplifier, which incorporates the required amplification and equalization for the second channel, must be acquired. Such tape amplifiers are available from several manufacturers. As discussed previously, when separate record amplifiers are used for each channel, it is necessary to synchronize their bias oscillators to avoid beat notes. Æ

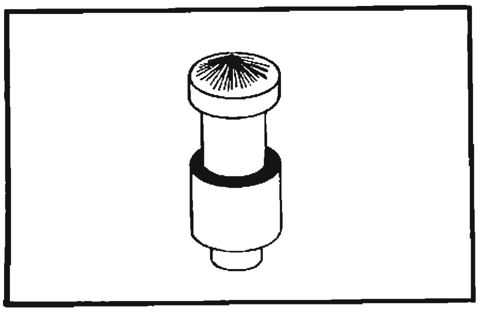


Fig. 12. Guidepost used in conjunction with "outboard" head mounting.

Tape Indexing Nomograph

JERRY LERNER*

The index counter is a useful device for conveniently locating a specific section of tape. Unfortunately it does not indicate either elapsed time or time remaining on the reel—unless used with the following nomograph.

FOR ANYONE MAKING serious use of a tape recorder, an index counter is unquestionably a great convenience. It enables the operator to locate desired sections of tape easily and rapidly, without the bother of either attaching a marker to the tape itself or going through endless repetitions of the stop-listen-rewind cycle. The value of this device is best measured by noting that all of the professional and semi-professional tape recorders come equipped with an index counter, and even the less expensive machines usually have one.

While serving perfectly well for location of the various selections on a length of tape, on most machines the reading of the index counter is *not* proportional to the amount of tape used. This means it does not directly measure either elapsed time or time remaining on the reel. In general, the index counter reading is proportional to the number of revolutions of the feed reel. Since the effective diameter of this reel changes with time as the tape runs out, the effective circumference of the reel also changes, so that one revolution of the supply reel does not necessarily correspond to a fixed length of tape. In fact, the amount of tape used per revolution varies by a factor of about three-to-one between the beginning and the end of the reel for the conventional 5- and 7-inch sizes. Corespondingly, towards the end of a reel of tape the index counter is going three times as fast as it was at the beginning, even though the tape speed is constant.

Offhand it might seem that the engineers in charge of designing tape machines hooked up the index counter at the wrong place. There is good reason, however, for connecting the counter to the reel drive rather than to the tape drive. In record or playback position there is a precise mechanism for pulling the tape past the heads at constant speed. In fast forward or fast rewind this mechanism is generally disconnected from the tape. Instead, power is applied directly to the reels. If a footage counter is to be useful under all circumstances it would have to be connected to the tape at all times and operate without slipping even when the tape is running at high speed. This would surely be very hard on the tape.

The nomograph on the opposite page enables the operator to convert the index counter readings into "elapsed time" or "time remaining" with little effort. It is based on the assumption that the conventional 5- or 7-inch reels are used, though it is appropriate for any reel in which the ratio of full diameter to empty diameter is about three-to-one. For reels with other ratios a formula is given below. Assuming that the counter is reset to zero at the beginning, the operator need only know the index counter reading for the full reel and the total duration of the reel.

A straight line connecting the index counter reading for the full reel (left scale) and passing through the existing index counter reading (center scale) intersects the right hand scale in a point that gives the percentage of the total time used up and the percentage of the total time still remaining on the reel. For example, on some tape recorders an 1800-foot reel of tape (1½ hours at 3¾ ips) totals 1200 on the index counter. If the index counter is reset to zero at the start and reads 600 after a selection has been recorded, then 63 per cent (about 57 minutes) has been used up and there is a little over one-half hour of tape remaining. An operator who expects that only about half of the tape has been consumed is in for a surprise. In fact, if 1200 is the full tape reading, then 450 on the index counter corresponds to the midpoint of the tape.

The nomograph can be useful in many situations. It's most obvious function is to determine directly such things as the duration of a selection, the time required to reach a certain passage, the amount of time remaining on a roll of tape, etc. The nomograph also has other, less obvious applications.

Suppose you borrow a tape from a friend, and want to locate a particular selection. On his machine the index reading is 650–930, with a reading of 1800 for the entire reel. Where does it begin on your machine? From the nomograph you determine that at 650 (with a total of 1800) the percentage of tape used is 48. Draw a line from 48 per cent to the point on the left scale corresponding to the full tape index counter reading for your tape recorder. Since the running time is the same on both machines, the place where this line crosses the center scale is the index counter reading for the beginning of the selection on your machine.

Another common situation is the following. You have a two-track tape recorder and wish to play the section 550–720 of Side 1 on a reel whose total reading is 800 on the index counter. Naturally, the reel is wound so that it is all ready to play Side 2. You could rewind the entire reel, then interchange the full and empty reels, re-thread the tape, reset the index counter, and then run it up to 550. A faster method is to determine from the chart that 550 corresponds to 80 per cent of the full reel. The complement of 80 per cent is 20 per cent, which corresponds to an index reading of 110. Reset the index counter to zero, run the tape until the counter reads 110, then interchange the partly loaded reels and you are at the correct place.

For those who do a lot of recording it might be a good idea to prepare a chart from the nomograph (or the formula) giving the direct correspondence between index counter reading and actual playing time for your particular machine and the tape length and speed you use most frequently. Pasted on the inside of your tape recorder cover, this table provides pertinent playing-time information at a glance.

In using the nomograph, the tape length, tape speed and tape thickness are quite immaterial. The only pertinent factors are the full reel and partly-full reel index counter readings, the total tape time, and the 3:1 ratio of full to empty reel diameters, For other reels or for pre-recorded tape the ratios of di-

* *348 Niagara, Park Forest, Ill.*

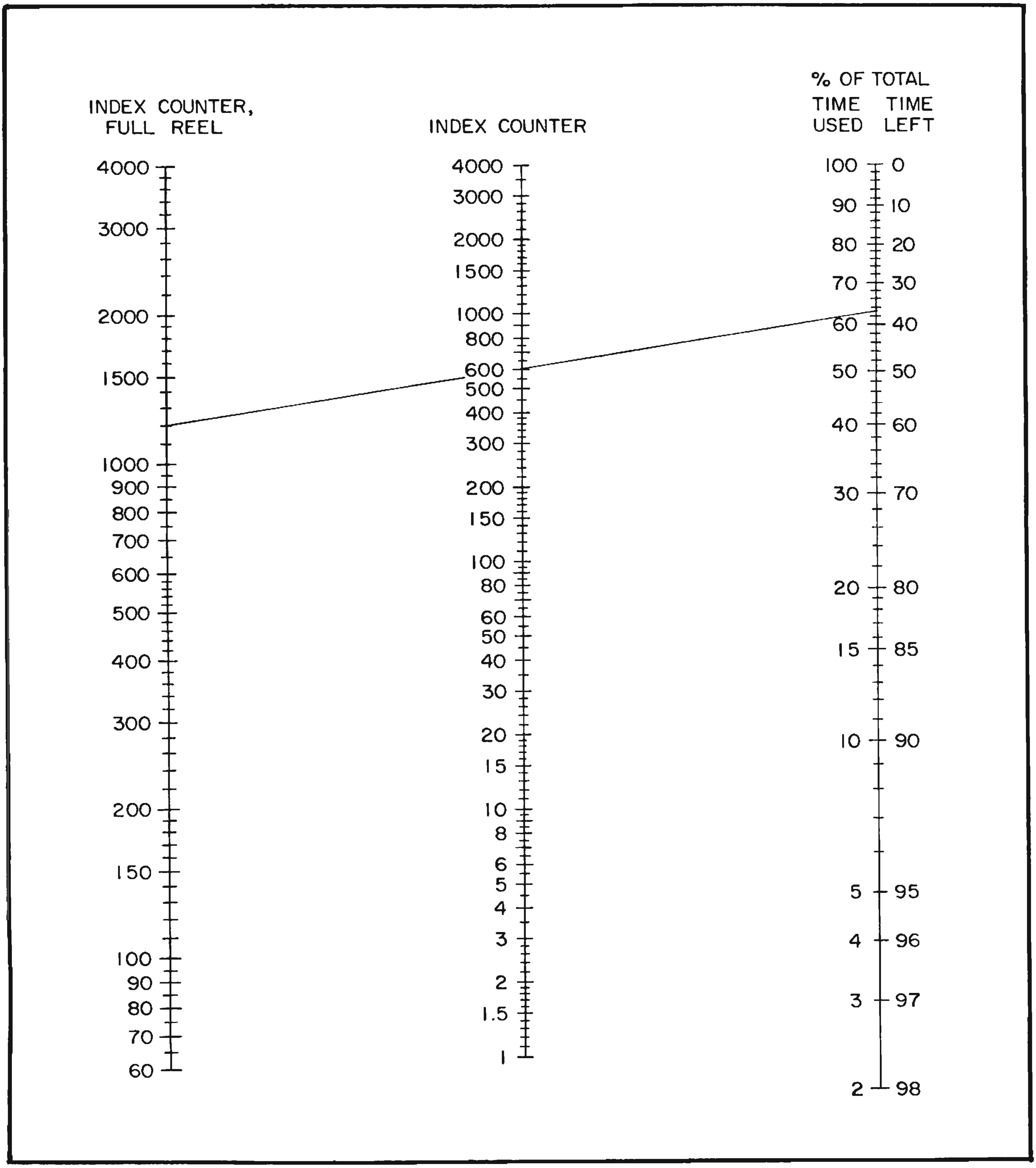
ameters may be different. The appropriate formula is then:

$$t/T = n\left[\frac{2-n(1-r)}{1+r}\right]$$

where t = elapsed time,
T = total time,
n = ratio of present index counter reading to total index counter reading,
and r = ratio of inner diameter to outer diameter (both n and r are always less than one).

There are a few tape recorders in which the index counter is connected to the take-up reel rather than to the feed reel. The nomograph can be used for these machines with a slight modification. The index counter reading is subtracted from the full tape index reading and used for the scale reading on the center line. Also, the calibrations of the "Tape Used" and "Tape Left" scales are interchanged.

It should be noted that values for the playing-time obtained from the nomograph are at best approximate. The index counter readings are generally not accurately reproducible, primarily because the tape is not always wound with uniform tightness. Other factors, such as slippage of the counter drive, fluctuations in tape thickness, etc., also influence the index counter readings. Æ



Nomograph for determining elapsed and remaining time on reel

Characteristics of Tape Noise

WILLIAM B. SNOW*

Tape noise is a fundamental limitation in all recording processes. Here are some criteria for judging a tape recorder with respect to noise.

Noise is a fundamental limitation in all recording processes. Unless a low noise level is achieved, true high-fidelity sound reproduction is impossible because low passages will be heard against a background of interfering and unwanted sounds. Low noise level with consequent wide dynamic range is a characteristic of modern magnetic tape recording

Signal-to-Noise Ratio

Signal-to-noise ratio in magnetic tape recorders is ordinarily expressed as the ratio of rms single-frequency signal at the level yielding 3 per cent harmonic distortion, to total noise measured over the complete reproducing frequency range. The 3 per cent point represents the maximum permissible recording level for signal peak amplitudes, and is usually measured at 250 cps.

When the signal-to-noise ratio is expressed as a single number in this manner for magnetic tape recorders, it essentially represents a signal-to-hum ratio. Hum reduction is particularly difficult with tape recorders because the magnetic reproducing head must be mounted near motors and power transformers which produce magnetic fields from which the head must be shielded. In addition, the playback equalization for magnetic recording necessitates maximum gain at low frequencies. It was felt that a somewhat detailed examination of the noise from a tape recorder would be of interest. A Movicorder tape recorder was employed operating at a speed of 7.5 inches per second.

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Tape Noise Frequency Analysis

First, a portion of tape containing a 250-cps tone recorded at maximum level was reproduced to give a reference output reading. Then, erased tape was reproduced without alteration of the playback amplifier gain while noise output was measured through the electrical filters of two types of frequency analyzer.

Figure 1 shows the results of the noise measurements made with a narrow band (25 cps) and an octave band analyzer. The usual signal-to-noise ratio described above is shown by the line at "Over-all" to be 52 db. With the octave filters, noise was checked for three conditions: tape erased in the machine (Curve A), bulk-erased tape (Curve B) and tape stopped showing only playback amplifier noise (Curve C).

It can be seen that the over-all level is mostly accounted for by the noise in the two lowest octave bands. Above 300 cps the levels are much lower. At low frequencies two sets of "spikes" are shown, measured with the 25-cps band analyzer which could separate the individual hum components. The noise in the two lowest octave bands is contributed almost entirely by the hum components, 60 and 120 cps, and is essentially unchanged when the tape is stopped. Above 200 cps, however, the noise comes principally from the tape, and residual electrical noise (Curve C) is negligible in comparison to it. The amplifier has the capability of playing much quieter tapes in the future as they are developed. Small difference between noise for bulk-erased and machine-erased tape indicates good balance in the erase oscillator.

Comparison With Room Noise

It is important to the success of magnetic recording that the signal-to-noise ratio at higher frequencies is much greater than the usual single number discussed above. The octave-band levels are roughly constant and are about 75 db below the standard 3 per cent distortion level. *Figure* 2 has been prepared to explain the significance of this. Rather than ratios, this figure shows actual sound levels as measured in a room with a sound level meter and analyzer. They have been plotted in the special form of "masking level"; that is, the level which noise from the reproducing system must attain if it is to be detected in the presence of the room noise. If it falls below

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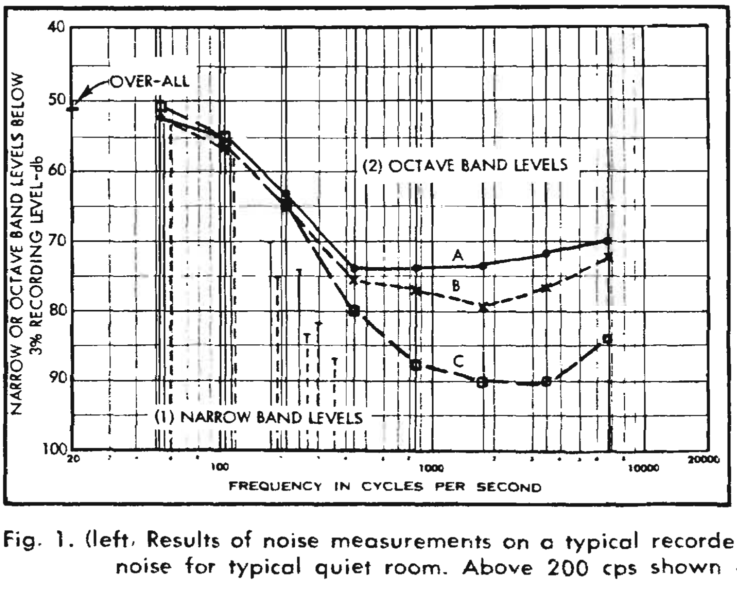

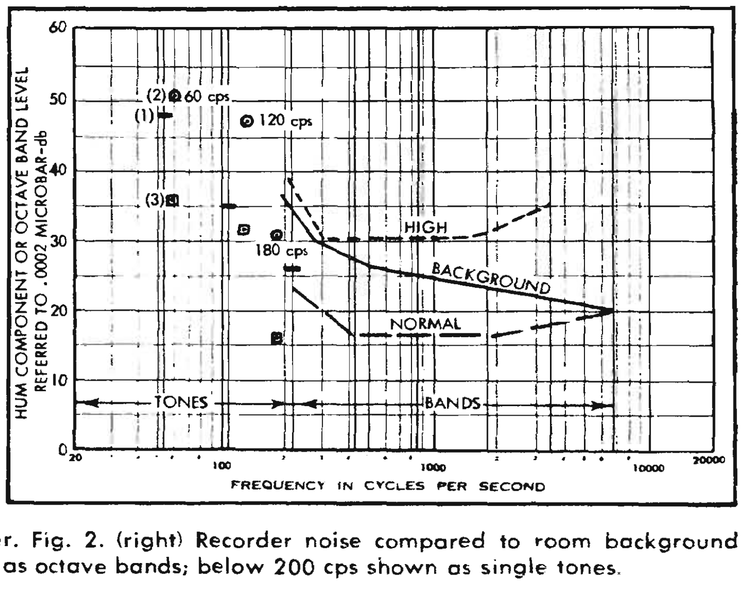

Fig. 1. (left) Results of noise measurements on a typical recorder. Fig. 2. (right) Recorder noise compared to room background noise for typical quiet room. Above 200 cps shown as octave bands; below 200 cps shown as single tones.

TAPE NOISE

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this value, it will be "masked" by the room noise and will not be heard .

A typical quiet, listening room background noise condition (1) is shown by the solid lines. This is given in terms of *tones* in the hum region, and *octave band noise* above 200 cps. Of course, the noise level in rooms varies, but the shape remains much as shown.

Figure 2 also shows two reproduced noise conditions during playback taken from Curve A and the "spikes" of *Fig.* 1. The one labelled High (2) is for a volume control setting giving maximum octave band program readings of 95 db in an ordinary living room—a level used mainly to impress friends with the height of the Fi. The one called Normal (3) represents a reproducing level 15 db lower, which is more representative of a usual domestic listening intensity. What stands out is the fact that the shape of the noise curve from the reproduction matches the shape of the background noise with which it competes. At the High setting noise is audible at both high and low frequencies to about the same degree during completely quiet intervals of program. At the Normal setting the reproduced noise would not be audible in this "typical" room, although it might be in very quiet rooms. But, even in a completely quiet room, the *shape* of the reproduced noise curve would remain satisfactory, because the actual minimum sensitivity of hearing nearly parallels the room noise curve.

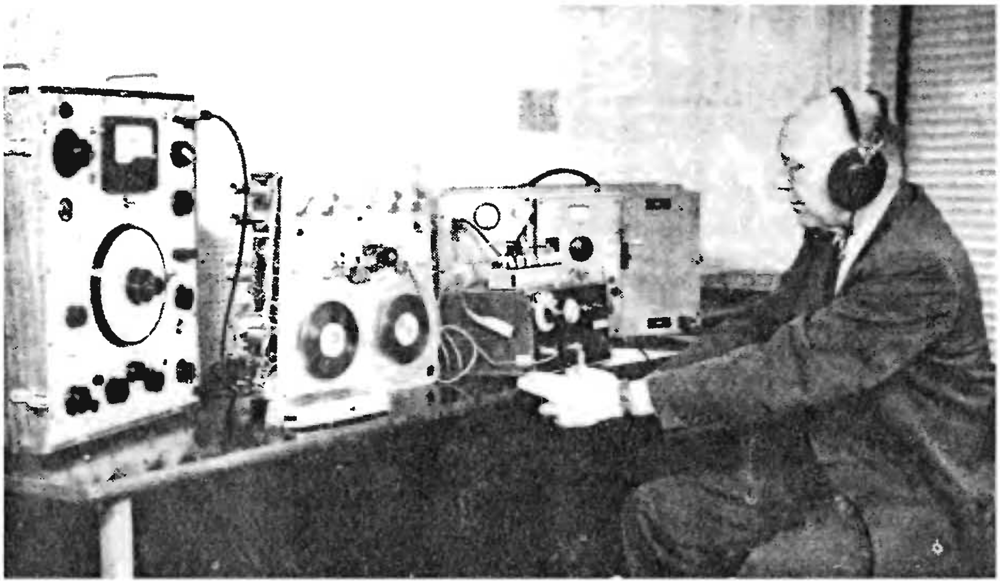

Fig. 3. The author shown measuring signal-to-noise ratio of a Movicorder.

Weighted Signal-to-Noise Ratio

The effects described above are recognized in communication and noise measurement, where frequency "weighting" of the response of the measuring instrument is used to reduce the contribution by low frequencies. Usually an unweighted and a weighted value are given, and this might be a useful concept for tape reproducers. In the case of the results just presented, for example, and using the A weighting scale of a standard sound level meter, the signal-to-noise ratios measured were

Unweighted, C-Scale	52 db (as given above)
Weighted, A-Scale	74 db

These two numbers show that the most intense noise components are at low frequencies where more can be tolerated, while the high-frequency contribution is a great deal lower. If only the unweighted number is given, a recorder with poor tape or poor construction could measure nearly as well, yet give prohibitively annoying high-frequency noise. Thus by adding one additional number, obtained with a relatively simple addition to standard measuring equipment, a great deal of meaning could be added to signal-to-noise ratio specifications. In the critical listening region good tape recorders give, not the 50 db ordinarily thought of, but 70 db and more. Æ

Understanding the Tape Oscillator

Incorrect bias can increase distortion, reduce the amount of signal recorded, and also decrease high frequency response. Here is an explanation of biasing which will answer many questions as to how and why.

HERMAN BURSTEIN AND HENRY C. POLLAK*

Only in one respect—the oscillator—do tape amplifier circuits differ radically from the circuits normally found in control and power amplifiers. Otherwise, the tape amplifier employs similar means for the similar tasks of amplifying signals, controlling gain, shaping frequency response (equalization), minimizing noise and hum, and performing various switching functions.

Accordingly, the technician or audiofan conversant with audio amplifier circuits should not find tape electronics presenting essentially different problems, except for the oscillator. Therefore it is the purpose of this article to provide a basic understanding of the oscillator circuits commonly found in tape recorders. Such an understanding will facilitate the work of the individual seeking to restore a tape oscillator to correct operation, to improve its performance, or to build a tape amplifier capable of recording satisfactorily.

Functions of the Oscillator

The oscillator operates only when the tape recorder is in the record mode and supplies high-frequency current, also known as bias current, to the record and erase heads. The frequency is usually between 40,000 and 100,000 cps. In a few recorders employing a permanent magnet for erase or in special machines such as tape duplicators, where erase is not required, current is supplied only to the record head.

Bias current in the record head serves two vital purposes. It increases the amount of signal recorded on the tape. It reduces distortion. Unfortunately, as bias current is increased above a certain point, high frequency response deteriorates. Hence one must guard not only against insufficient bias current, which results in excessive distortion and poor signal-to-noise ratio, but also against too much bias current, which produces severe treble losses. The slower the tape speed, the greater are these high-frequency losses.

Bias current requirements of record heads are usually quite modest, on the order of 1 ma for many heads. In contrast, erase heads require a good deal more current in order to perform effectively. A typical erase head may require from 15 ma upwards.

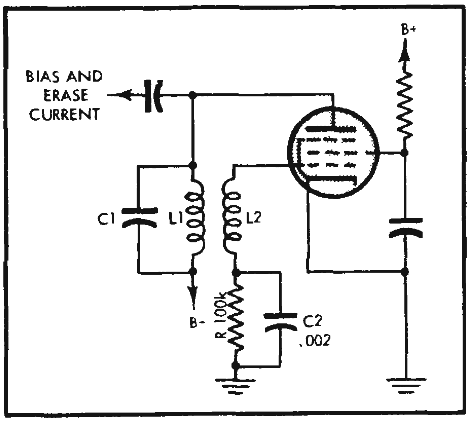

Fig. 1. Single-ended oscillator employing plate-to-grid feedback.

Oscillator Operation

Most oscillators employed in tape recorders operate by applying positive feedback between appropriate tube elements, usually between plate and grid, in an amount sufficient to sustain oscillations in a tuned circuit consisting of a coil and capacitor. The values of the coil and capacitor essentially determine the frequency of oscillation.

The operation of a tuned-circuit oscillator is a complex process, with many things happening at once. A complete description requires tracing over one cycle of oscillation the phase relationships between voltage and current in electromagnetic and electrostatic fields and in a tube circuit. Instead of going through such an analysis, this article will attempt to provide a simpler, basic insight into how an oscillator works.

A fundamental explanation can be based around *Fig. 1,* a simple oscillator similar to that actually found in many moderate-price tape recorders. To understand why oscillation takes place, it is helpful to consider first just the tuned circuit, comprising *C1* and *L1*. Assume that for some reason the upper plate of *C1* is charged, that is, contains more

(*Continued on page 32*)

* Authors of *"Elements of Tape Recorder Circuits."* Gernsback Library.

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electrons than the lower plate. Seeking equilibrium, electrons tend to flow from the upper to the lower plate through the path afforded by *L1*. This flow creates an electromagnetic field about the coil and, by Lenz's Law, induces a voltage across the coil of a polarity such as to prevent electrons from flowing rapidly through the coil. Thus the electron flow *gradually* reaches a maximum and then starts to slow down as the charges on the upper and lower plates approach equilibrium. However, when the rate of discharge of electrons from the upper to the lower plate begins to slow down, the field of *L1* begins to collapse. By Lenz's Law, a voltage is again induced which opposes the change in electron flow through the coil. Thus the collapsing electromagnetic field promotes the continued flow of electrons from the upper to the lower plate. In this manner the lower plate collects not just enough electrons to restore equilibrium with respect to the upper plate (zero voltage across the capacitor); rather, it accumulates an excess of electrons compared with the upper plate.

Eventually the coil's field has fully collapsed so that no more electrons arrive at the lower plate. Now this plate has an excess of electrons; in other words, the capacitor has an electrostatic field, which is the counterpart of the coil's electromagnetic field. Therefore, electrons begin to flow from the lower to the upper plate through the coil. As before, an electromagnetic field is built up around *L1* and, when this field collapses, it results in the continued accumulation of electrons on the upper plate of *C1*, so that the original state of matters is restored: an excess of electrons exists on the upper plate. This completes one cycle of oscillation.

Assuming no resistance in the coil and no load, the tuned circuit produces a perfect sine wave, eminently desirable for tape recording purposes to achieve a minimum of noise. In practice, this is, of course, impossible; some distortion is always present. However, oscillator waveform distortion and resulting noise are kept to negligible quantities in high-quality tape recorders.

The frequency of oscillation—or the time required for one cycle—essentially depends upon the values of *L1* and *C1;* to some extent it is also governed by the slight amounts of inductance and capacitance found in the tube and other components associated with the tuned circuit. The coil and capacitor values, *in conjunction with each other,* determine how long it takes for the electromagnetic field of *L1* to build up and die

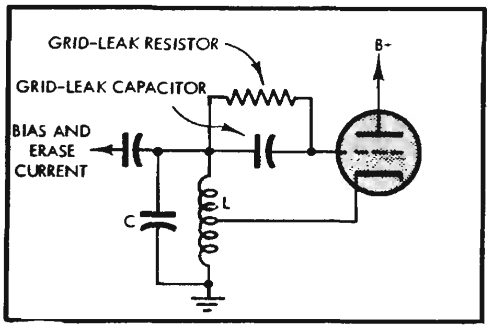

Fig. 2. Single-ended oscillator employing cathode-to-grid feedback.

away and for the electrostatic field of *C1* to do the same. The larger the inductance of *L1*, the longer its field takes to grow and fall. Similarly, the larger the capacitance, the longer it takes to discharge electrons from one plate to the other. At the oscillation frequency, the charge or discharge rates of the two components are equal, and they work in unison: the electromagnetic field stores energy for the same period that the capacitor is able to deliver it, and in turn the capacitor stores energy for the same period that the coil is able to deliver it.

Another way to appreciate why a circuit such as *Fig. 1* oscillates at one particular frequency is to consider the impedance between the plate side of the tuned circuit and ground. (It should be recognized that the bottom of the tuned circuit is effectively at ground so far as a.c. is concerned because of the filter capacitor associated with B-plus.) Maximum impedance of the tuned circuit occurs at the frequency where the reactances of *L1* and *C1* are equal. For any other frequency, the impedance is less, so that either the coil or capacitor tends to serve as a shunt to ground. Consequently, alternating current developed through oscillation tends to be shunted to ground except at the frequency where impedance is maximum.

Once started, oscillation in a tuned circuit would theoretically continue forever were it not for various losses, in-

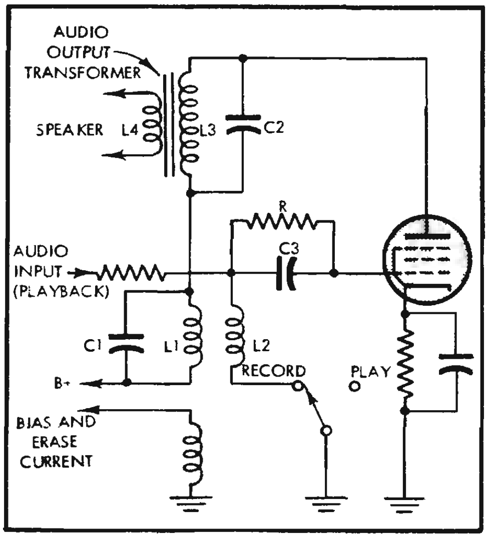

Fig. 3. Use of the audio output tube as an oscillator in the record mode.

cluding those due to coil resistance, capacitor leakage, and the load presented by the tape recorder heads and other circuit elements. For oscillation to be sustained, the tuned circuit needs outside aid. This is similar to the child on a swing, who keeps moving as the result of a moderate systematic push from someone on the ground.

The tuned circuit receives systematic aid from the tube circuit with which it is associated. When the upper plate of *C1* is to be charged, matters are arranged so that tube current increases, thereby sending more electrons to this plate. Conversely, when the lower plate of *C1* is being charged, tube current decreases, sending more electrons to this plate (from the viewpoint of a.c., a decrease in tube current is in effect a flow of electrons from B-plus toward the tube).

The purpose of *L2* in *Fig. 1* is to vary the grid voltage in a manner which causes tube current to assist the oscillation process. The changing electromagnetic field of *L1* cuts across *L2* and, by transformer action, induces a voltage across *L2*—that is, between grid and ground. The windings of *L2* are so connected to grid and ground that when tube current is increasing the grid end of *L2* goes positive, which causes a further increase in tube current. This of course is positive feedback. Similarly, when tube current is decreasing, the grid goes negative, resulting in a further reduction in tube current.

The cumulative increase or decrease in tube current which takes place due to positive feedback approaches an end when the charge on either plate of *C1* approaches maximum. There is a slowing collapse of the magnetic field around *L1* and eventual reversal of this field as *C1* approaches maximum charge and then begins to discharge. This results, through transformer action, in a decrease in grid voltage (positive or negative as the case may be) and eventual reversal of grid polarity.

Though belated, an explanation of how oscillation gets started is now appropriate. Assuming that B-plus has been applied to the circuit and current supplied to the tube heater, initially there is zero voltage between grid and cathode. Due to the random motion of electrons emitted from the cathode, a minute voltage will appear at the grid. Assume that at a given instant this voltage is positive-going. Therefore the current through the tube increases. This increase in tube current results in a charge on *C1*, a change in the electromagnetic field of *L1*, positive feedback at the grid, a further increase in tube current—and the process of oscillation is on, as already described.

Grid-Leak Bias

The purpose of grid resistor *R* and

grid capacitor C_2 in *Fig.* 1 is to provide the oscillator tube with the required negative grid bias. The amount of grid bias depends upon the magnitude of oscillation needed—within the tube's capabilities. When the grid goes positive and draws current, the resulting electron flow charges the top of the capacitor. The only path for the capacitor to discharge through is the grid resistor. As electrons leak slowly from top to bottom of the capacitor through the resistor, this flow causes a negative d.c. voltage to appear at the top of the resistor. This voltage also appears at the control grid.

The negative grid-leak bias reduces the transconductance (g_m) of the tube, and thereby the gain around the complete oscillation loop. If the loop gain is greater than 1, as it must be for oscillation to start, the amplitude of each successive oscillation will be greater than the previous one. This causes the grid to swing more into the grid current region on each positive half cycle, resulting in more grid-leak bias. But the bias affects the transconductance markedly; the greater the negative bias, the lower the g_m. So each positive grid swing results in added negative bias, reducing the g_m, and hence the gain, until the loop gain is exactly 1. The amplitude of the oscillations will remain at this value very closely.

The self-regulation of the grid-leak bias system is not perfect, but is sufficient to make the oscillator relatively insensitive to line voltage variations, changes due to normal heating of the components, and tube aging.

The grid capacitor loses some of its charge during every cycle, but unless the oscillations are getting smaller, each positive grid swing recharges the capacitor, thus maintaining the bias voltage. The time constant of the grid-leak capacitor and grid resistor (product of R times C_2) determines how long the capacitor can discharge through the resistor before the voltage has dropped appreciably. This time constant should be about 5 to 10 times the period of one cycle of oscillation to maintain grid bias adequately. For example, if the oscillator frequency is 50,000 cycles per second, one cycle is 1/50.000 second, or 20 microseconds (μsec); 10 times this amount is 200 μsec. The time constant of the 100,000 ohm resistor and .002 microfarad capacitor in *Fig.* 1 is 200 μsec.

Although grid-leak bias keeps the amplitude of oscillations from being extremely great, it is very desirable also that feedback be limited so that the tube operates within the linear portion of its characteristic in order to maintain an oscillation waveform with minimum harmonic distortion. In the case of *Fig.* 1, feedback is controlled by using a proper ratio of turns and the right amount of coupling between L_1 and L_2, thus limiting the voltage fed back to the grid.

Oscillator Variations

There are several variations of the single-ended oscillator of *Fig.* 1. In a popular variation the oscillator coil is in the grid cathode circuit, as in *Fig.* 2. For positive feedback to occur here, it is necessary that the grid go positive relative to the cathode when tube current increases, and negative when current decreases. Positive grid-cathode voltage, in turn increases tube current, and negative grid-cathode voltage decreases it. The cumulative buildup or decrease in tube current is controlled by the tuned circuit so as to sustain oscillation.

Assume that the current flowing from ground through the lower part of oscillator coil L and then through the tube is momentarily increasing. This increasing current induces a voltage across the grid-ground portion of the coil such as to oppose the increase. That is, the inductive reactance of the coil causes a voltage drop across it, causing the

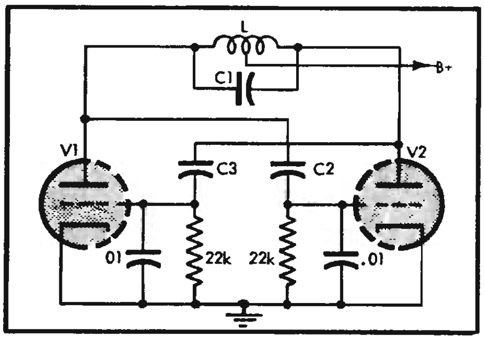

Fig. 4. Typical push-pull oscillator.

cathode end to go positive with respect to the ground end. By autotransformer action, the positive-going voltage at the cathode appears as a still more positive voltage at the grid, causing a further increase in tube current. Thus, as in *Fig.* 1, positive feedback is present. The voltage between grid and ground causes capacitor C to charge, making the grid-end positive and the ground-end negative. Since the tube current cannot increase without limit, the tube current eventually reaches a maximum, that is, a steady value. As a result, there is no longer an induced voltage due to tube current increasing. Consequently the grid-to-ground voltage decreases and capacitor C discharges upward through the coil.

As the grid-to-ground voltage decreases, the grid-to-cathode voltage decreases and so does the tube current. This induces a voltage in the lower portion of the coil, this time negative at the top and positive at the ground end. As before, autotransformer action causes the grid to go more negative with respect to the cathode, further reducing tube current, making the grid still more negative, and thus assisting the upper plate of the capacitor to go negative with respect to the lower plate. It should be kept in mind that the process of positive feedback and the turning points from increasing to decreasing tube current are under control of the tuned circuit, which determines the rate of increase and decrease in tube current and thus the frequency of oscillation.

Finally, it may be pointed out that while the locations of the grid-leak resistor and capacitor are different in *Fig.* 2 than in *Fig.* 1, the action is exactly the same.

Double-Purpose Oscillator

The majority of moderate-priced tape recorders contain a small speaker and a power amplifier, usually single-ended, for playback purposes. As a measure of economy, a number of these machines convert the audio output tube to an oscillator in the record mode. In a few instances, a similar double function is served by other tubes. For example, in one recorder the playback input stage becomes an oscillator when recording.

Figure 3 shows a circuit in which the audio output tube doubles as an oscillator. L_1 and L_2 constitute the oscillator coil, providing plate-to-grid feedback. The primary of the audio output transformer is in series with L_1. Capacitor C_2 across the output transformer primary offers a low-reactance path at the oscillator frequency between the plate of the tube and the primary of the oscillator coil. Similarly, L_1 of the oscillator coil offers a low-reactance path at audio frequencies between B-plus and L_3, the output transformer primary.

Push-Pull Oscillators

The great majority of professional and semi-professional tape recorders and a fair number of moderate-price ones employ a push-pull oscillator, customarily using the two halves of a dual triode such as a 12AU7 or 12BH7. While one triode is in the positive half of its oscillation cycle, the other is in the negative half. Thus, symmetrical forces are at work, reducing even-harmonic distortion. Distortion in the bias waveform is a source of noise. The greater the demands upon the oscillator to provide enough current for adequate erasure, the greater is the likelihood of distortion. Because of its lower distortion for the same output, the push-pull oscillator is favored.

Figure 4 shows a typical push-pull oscillator. Feedback is from the plate of V_1 to the grid of V_2 through capacitor C_2; and from the plate of V_2 to the grid of V_1 through capacitor C_3. C_1, as well as C_2 and C_3, together with coil L essentially determine the resonant frequency.

Assume the grid of V_1 is positive-going at a given instant. This produces a

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negative-going voltage at the plate of V_1, which is transferred to the grid of V_2, causing the plate of V_2 to go positive. The voltage fed from the plate of V_2 to the grid of V_1 is therefore of the same polarity as the original signal on this grid; feedback is positive. The same is true for the voltage fed from the plate of V_1 to the grid of V_2. However, the voltages on the grids of V_1 and V_2 are of opposite polarity, so that one triode is in the positive half-cycle while the other is in the negative half. B-plus is supplied to the plates of V_1 and V_2 through the center-tapped coil, *L*. The grid resistors and grid capacitors of each triode produce a negative d.c. bias in the same manner as in *Fig.* 1.

The feedback capacitor of each triode forms a voltage divider in conjunction with the grid capacitor. Voltage divider action limits the amount of feedback to the grid, preventing the tubes from being driven excessively. In *Fig.* 4, the .01-μ grid capacitor has a reactance much smaller than 22,000 ohms at the bias frequency, roughly 50,000 cps, so that the voltage divider consists principally of the feedback and grid capacitors.

Sometimes, to supplement the d.c. bias obtained by grid-leak action, cathode bias is also used. That is, instead of connecting the cathodes of the triodes directly to ground, they are both connected to ground through a common resistor with a value of several hundred or a few thousand ohms. Æ

Evaluating Signal-to-Noise Ratio

HERMAN BURSTEIN*

Signal-to-noise ratio is an indicator which can help the audiofan select compatible components.

Along with good frequency response and low distortion, a high signal-to-noise ratio is one of the pillars of high fidelity. Of the three, signal-to-noise ratio probably gets talked about least. Yet it would be difficult to argue that noise is much less a consideration in pleasurable listening than are distortion and frequency response. Anyone who has had to endure an insistent hum or frying sound during the soft passages of a musical program will attest to this.

Therefore in assembling, enlarging, or otherwise modifying an audio system the audiofan and those who give him counsel—the salesman, the technician, and well-meaning friends—should carefully evaluate the signal-to-noise ratio of each component. This is not simply a matter of reading a component's specifications and deciding on this limited basis whether the signal-to-noise ratio is high enough. Such a procedure is apt to contain a booby trap. This is certainly not meant to imply that manufacturers' specifications are exaggerated. Quite the contrary. The writer's experience in testing many pieces of equipment is that manufacturers' ratings are usually met or exceeded. However, signal-to-noise specifications are based on the premise that maximum signal will have a certain value or that the maximum input signal will have a certain value. But in practice the actual output or input signal may differ appreciably from the presumed value. Taking this factor into account plus other factors bearing on the gain and distortion characteristics of the component in question, one often finds that the rated value of signal-to-noise varies appreciably from the *effective* value.

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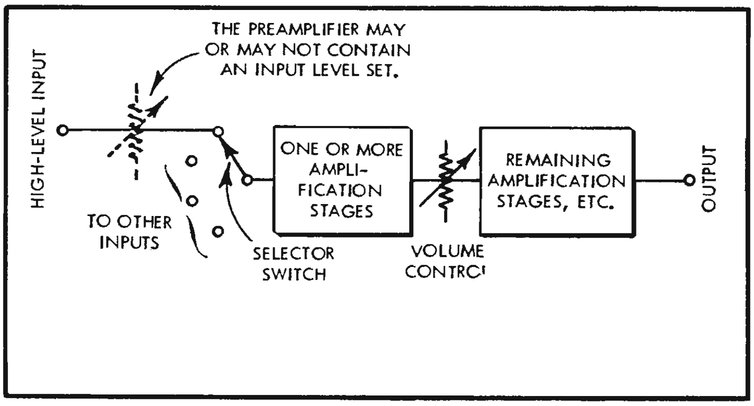

Fig. 2. Configuration wherein the high-level input signal is reduced after in level after amplification.

Meaning of Signal and Noise

Signal denotes the *desired* audio information, usually music or speech but sometimes other sounds, such as that of racing cars, jet airplanes, rockets, waterfalls, breaking glass, and so on.

Noise refers to *undesired* sounds produced by, or in, the audio equipment. Its chief forms are hum and hiss. Hum is 60 cps (the line frequency) and its harmonics, principally 120 cps (ripple in a full-wave detector) and 180 cps (odd harmonic distortion of a power transformer). Hiss consists of random frequencies throughout the audio spectrum. Because there are more frequencies in the upper octaves than in the lower ones, and because each frequency contains about the same amount of energy, this form of noise has the high-pitched characteristic that we call hiss. Noise takes additional forms, such as clicks, pops, crackles, squeals, and so forth.

Distortion also consists of undesired sounds. How then do we differentiate between noise and distortion? The difference lies in the fact that noise is always present, whereas distortion arises only as a consequence of audio signals. In sum, noise consists of undesired sounds whose existence and level is independent of the audio signal.[1]

Accordingly, the signal-to-noise ratio designates the magnitude of the desired audio signal relative to the magnitude of the undesired noise, expressed in decibels.

In measuring signal-to-noise, consideration is sometimes given to the ear's reduced sensitivity to low and high frequencies compared with the mid-range. Therefore the various noise frequencies are given varying weights in accordance with the ear's sensitivity to them. The weighted value is higher than the unweighted value. For example, if noise consists principally of 60-cps hum and if account is taken of the ear's lower sensitivity to 60 cps than to 1000 cps, the weighted signal-to-noise ratio is a "better" figure than the unweighted ratio.

Most often, however, the signal-to-noise ratio is stated on an unweighted basis, and the following discussion shall proceed on this basis. There are good reasons for preferring the unweighted rating in evaluating a component. For one thing, while hearing sensitivity varies

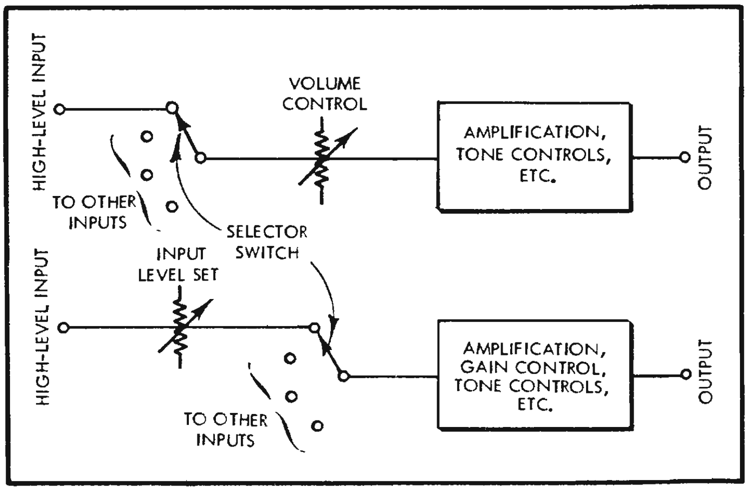

Fig. 1. Configurations wherein the high-level input signal is reduced in level prior to amplification.

[1] There is a minor exception. Tape modulation noise varies in intensity with the level of the audio signal. It is caused by variations in thickness of the tape's base and/or coating and by variations in the magnetic oxide. Modern high quality tapes, however, produce very little noise of this sort.

with frequency, there are considerable differences between individuals as to the degree of variation. Second, even though a given noise frequency may be completely inaudible, it may nevertheless produce distressing results. For example, assume that a defective power amplifier pulses strongly at the rate of 10 cps (motorboating). The resulting large excursions of the speaker are of course inaudible in themselves, but they may take the cone so far out of its linear travel-path as to cause considerable and audible distortion of program material. Similarly, oscillation at a supersonic frequency will be inaudible but can give rise to distortion of audio frequencies.

High Fidelity Requirements

Using program peaks as a reference level, it may be said that noise should be at least 55 db below the peak audio signal to justify the designation high fidelity. In other words, the minimum signal-to-noise ratio for high fidelity is about 55 db. Inasmuch as the dynamic range—distance between the maximum and minimum levels—of discs, tapes, and FM seldom exceeds 55 db and is often 50 db or less, a signal-to-noise ratio of 55 db may be considered acceptable.

Assume that the signal source has a 55 db signal-to-noise ratio. This does not mean that it is purposeless having a preamplifier (and power amplifier) with a signal-to-noise ratio greater than 55 db. Quite possibly, the principal noise components of the preamp fall in a different part of the audio spectrum than do the chief components of noise in the signal source, so that noise of the preamp may be distinct unless its ratio is appreciably higher than 55 db.

A ratio of 60 db offers a worthwhile bit of margin, and higher than 60 db is all to the good. Ratios of 65 and up put the listener really "in the clear."

It is desirable that each component in an audio chain have a higher signal-to-noise ratio than the preceding component so that performance of the earlier component will not be degraded. To illustrate, assume that a preamp can achieve an enviable 70-db ratio at the magnetic-phono input. It would be a shame to be profligate with such performance by feeding the preamp signal into a power amplifier having only a 60 db ratio.

Power Amplifiers

The signal-to-noise ratio of a power amplifier is almost invariably stated on the basis of the amplifier's maximum rated output. For example, if the amplifier can produce up to 60 watts at low distortion, the manufacturer will specify a signal-to-noise ratio on the basis of 60 watts output; a typical rating might be 80 db. However, the rated signal-to-noise ratio may be much higher than the effective ratio, as we shall soon see.

(It is difficult to obtain a precise quantitative statement of the effective ratio, but for purposes of discussion we shall put numbers on the factors we are discussing.)

The effective signal-to-noise ratio must take into acount: (1) the efficiency of the speaker which is used; (2) the level of room noise, which may or may not mask amplifier noise; and (3) the maximum power to which the amplifier will actually be driven, which is dependent in part upon speaker efficiency and room noise, and in part upon room size and the audiofan's preference as to volume level.

The efficiency of typical speakers varies roughly from 1 to 20 per cent, a range of 13 db. The less efficient the speaker, the more it cuts down the noise issuing from the amplifier. But it also cuts down the audio signal, so that the amplifier must be operated at greater output in order to obtain the desired acoustic level. Therefore we have more audio signal at the amplifier output but the same amount of noise as before, which means an improvement in effective signal-to-noise. The converse is true for an efficient speaker.

Using the minimum audible sound at 1000 cps as a 0 db reference, room noise may vary from a low of possibly 20 db (night-time in the country) to about 50 db (daytime in the busy city). Forty db is fairly typical. The higher the noise level, the more likely it is that the audiofan will want to operate his sound system at higher power in order to get above the room noise. Correspondingly, the effective ratio goes up.

Depending on speaker efficiency and room noise, plus the listener's taste in volume level, a 60-watt amplifier might be operated so that program peaks require from as much as 60 watts to as little as 1 watt or less. Assume that a maximum of 1 watt is used because of a 20 per cent efficient speaker, low room noise, and the desired acoustic level. Then the effective signal-to-noise ratio is 18 db below the rated value: 18 db is the ratio of 60 watts to 1 watt. Thus a rated figure of 80 db becomes an effective figure of 62 db, which does not offer very much margin above the 55 db minimum. An effective ratio of 62 db for a power amplifier may fall below the effective value for a fine preamplifier.

To make all the more clear why there is a decrease of 18 db in the effective value, let us approach the above example in another way. Noise is rated 80 db below 60 watts, which means that noise output is .0000006 watt, or .6 microwatt. Since maximum output signal is 1 watt, the ratio between signal and noise (1 watt/.6 microwatt) is 62 db.

On the other hand, assume that a 1 per cent efficient speaker is substituted in our example. This speaker produces 1/20th as much volume as the 20 per cent efficient speaker, so that maximum power required from the amplifier rises to 20 watts; 20 watts of signal compared with .6 microwatt of noise yields an effective ratio of 75 db, which is very respectable.

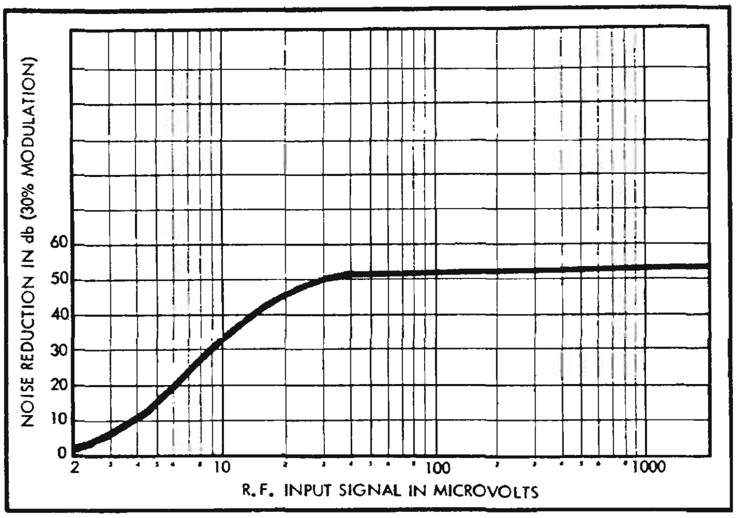

Fig. 3. Quieting characteristic of an FM tuner.

In sum, the less one employs of maximum power, the more one must downrate the amplifier's rated signal-to-noise ratio in order to arrive at the effective value.

Another factor to be taken into account, and again difficult to put into quantitative terms, is the relative smoothness of the speaker system to be used in conjunction with a given power amplifier. A speaker that peaks at the hum frequencies will deteriorate the amplifier's ratio. To illustrate, a certain large horn-loaded speaker has prodigious but smooth ability to reproduce bass. Connected to almost any high-fidelity amplifier, this speaker reproduces little or no audible hum. Another speaker, contained in an enclosure of less than 2 cubic feet, has about the same over-all efficiency in the mid- and treble range but is far less worthy a reproducer of bass. Nevertheless, because of an unlovely peak, it delivers audible hum when connected to all but the very finest power amplifiers.

Similarly, amplifier hiss is more apparent when the speaker has rough rather than smooth response in the treble range.

Preamplifier: High-Level Input

Preamplifiers generally have a rated output of at least 2 volts. Usually they can deliver 2 volts at less than 1 per cent IM distortion. Some can do so at .1 per cent IM or less. A few can come up with even 3, 4, or 5 volts without exceeding .1 per cent IM.

Two volts is enough to drive virtually any power amplifier to full output. If the preamp can provide more than 2 volts at low distortion, and if it has sufficient gain to do so on the basis of the incoming signal level, the effective signal-to-noise ratio is higher than the rated value. To illustrate, assume that on high level input the preamp is rated at 70 db. Assume that the preamp can produce 4 volts before distortion rises appreciably above .1 per cent IM, and that its sensitivity is rated at 100 mv for 2 volts output. Accordingly, an input of 200 mv is needed for a 4 volt output. If the high-level signal source, say an FM tuner, can deliver at least 200 mv on peaks (most FM tuners can furnish 1 volt or more on peaks), the preamp will have no problem producing an output of 4 volts. Therefore the effective signal-to-noise ratio can be based on an output of 4 volts rather than 2 volts. Since 4 volts is 6 db more than 2 volts, we can add 6 db to the 70 db for a total effective value of 76 db.

Of course, an output of 4 volts is not actually usable, because this would drive any power amplifier into clipping. But the 4 volts can usually be reduced at the input of the power amplifier via a level set. Preamp audio signal and preamp noise are simultaneously reduced, so that the preamp's effective signal-to-noise ratio remains the same as calculated on the basis of an output of 4 volts.

However, if the power amplifier lacks an input level set, the preamp's signal must be reduced at some stage within the preamp. If this reduction occurs at the very input via the volume control or a level set, as shown in *Fig.* 1, the effective value cannot exceed the rated value. But if the volume control follows a stage of amplification, as shown in *Fig.* 2, the effective value of signal-to-noise can exceed the rated value to the extent that the control reduces the noise of the first stage, which is ordinarily the principal source of preamp noise.

Now assume that we have a preamp with a rating 70 db based on an output of 2 volts, and that its sensitivity is rated at 200 mv input for 1 volt output. Further assume that we plan to use an FM tuner having a ratio detector *not* followed by a stage of audio amplification, and that the peak output of the tuner is apt to be about 200 mv on weak stations (output of a ratio detector varies with signal strength). Therefore the most signal we can expect out of the preamp is 1 volt, and we must calculate the effective signal-to-noise ratio on the basis of an output of 1 volt. Since a 1 volt output is 6 db less than 2 volts, we subtract 6 db from 70 db, leaving an effective signal-to-noise ratio of 64 db.

Some people might quarrel with downrating the preamp, feeling that the problem actually lies with the tuner, which has relatively low output as tuners go. This is a matter of viewpoint, but the writer feels that the blame really belongs to the preamp because of its low sensitivity. Another preamp, as in our earlier example, would have sufficient sensitivity to convert a 200 mv input to appreciably more output than 1 volt.

Until now we have presumed that 2 volts are needed to drive the power amplifier. In most cases this is unrealistically high, not only because a somewhat smaller voltage will ordinarily drive the amplifier to full output but also because we do not want ordinarily to utilize more than a fraction of the amplifier's maximum power, even on signal peaks. To that degree that less than 2 volts is needed to drive the power amplifier on signal peaks the preamp's effective signal-to-noise ratio is increased, assuming that the preamp's output will be reduced by an input-level set in the power amplifier.

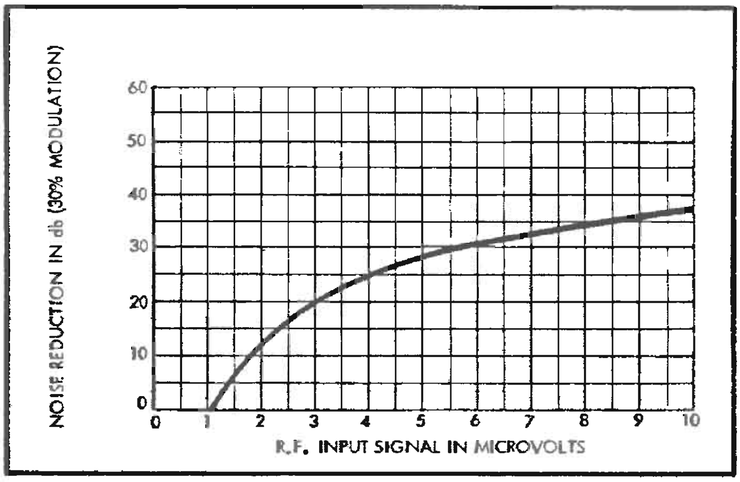

Fig. 4. Quieting characteristic of an FM tuner over a limited range.

For example, assume that a 40-watt power amplifier requires 1 volt to be driven to full output, but that 10 watts peak output will be quite adequate for the speaker that is used. For 10 watts output, only .5 volt input is needed (input voltage varies as the square root of the output wattage). Assume that we are dealing with the same preamp as in our first example, which has an effective signal-to-noise ratio of 76 db—based on 4 volts output which can be reduced to 2 volts by the power amplifier's input-level set. Since we can tolerate a reduction to .5 volt output, which is 12 db below 2 volts, we can add 12 more db to the effective value, for a total of 88 db.

On the other hand, the absence of an input-level set in the power amplifier can cause the preamp's effective value to be less than its rated value. For example, assume as before that the preamp's rating is 70 db based on an output of 2 volts, and that we desire only .5 volt to drive the power amplifier. Further assume that the preamp's volume control is located right after the input jack. Therefore the only way to reduce the preamp's output signal to .5 volt is by cutting the input signal, using either the volume control or the preamp's input-level set if there is one. The preamp's output noise is unaffected by these maneuvers, but its output signal is now .5 volt instead of 2 volts, a reduction of 12 db. Therefore the effective value is 12 db below 70 db, or 58 db.

Preamplifier: Magnetic Phono Input

To calculate the effective signal-to-noise ratio on magnetic phono input, we must know how much peak signal can be expected at 1000 cps from the magnetic cartridge to be used. The effective ratio can then be calculated on the basis of input signal to the preamp or on the basis of output signal from the preamp. The first method is used when the preamp's signal-to-noise ratio is stated in terms of input signal and the second method is used when the ratio is stated in terms of output signal. To illustrate:

1. Assume that the cartridge to be used is rated at 5 mv output at 5 cm/sec groove velocity, which is a fairly typical rating. Peak levels on a phono disc are apt to reach around 15 cm/sec at 1000 cps, so that the cartridge's peak output may be estimated at 15 mv. Assume that the preamp's rated signal-to-noise ratio on magnetic phono input is stated as 60 db below 10 mv input signal (at 1000 cps). Accordingly, the effective signal-to-noise ratio is 63.5 db, because 15 mv is 3.5 db higher than 10 mv. On the other hand, if one were to use a low-output cartridge with a peak output of, say, 5 mv,

the effective value would be only 54 db, because 5 mv is 6 db less than 10 mv.

2. Assume again that the cartridge has a peak output of 15 mv. Assume that the preamp is rated at 60 db below an output of 2 volts. This translates into a noise output of 2 mv. Next we must consider the preamp's sensitivity on magnetic phono input, which we shall assume to be 4 mv for an output of 2 volts at 1000 cps. Therefore the gain is 54 db. Accordingly, a 15-mv input translates into an output of 7.5 volts. The ratio of 7.5 volts of signal output to 2 mv of noise output indicates an effective signal-to-noise ratio of 71.5 db. True, the preamp is apt to run into excessive distortion because it can't turn out 7.5 volts. However, we assume this output level only for calculation purposes. In practice, we would turn down the phono level set, or else the volume

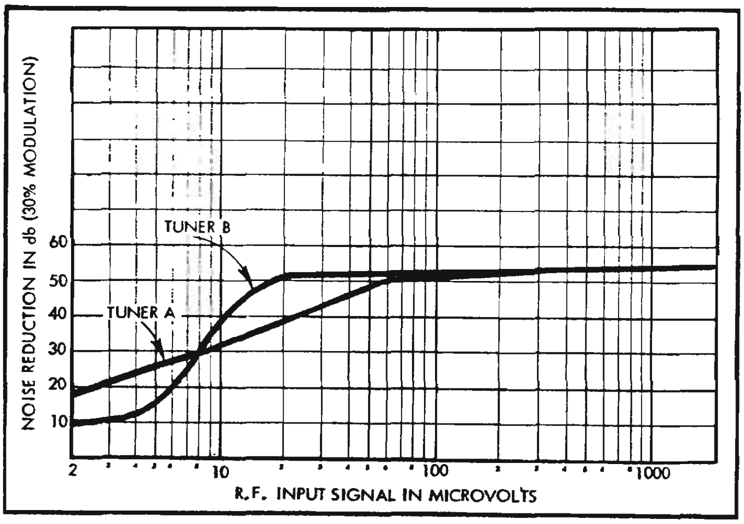

Fig. 5. Comparison of the quieting characteristic of two different FM tuners.

control, to prevent signal output from climbing up to 7.5 volts. Since the noise figure is principally due to the first stage of phono amplification, turning down the phono level set or the volume control serves to reduce noise.

There is a limit to the increase of the effective value as cartridge output increases. This limit, which varies with the design of the preamp, lies in the fact that excessive input to the phono preamplification stage will cause excessive distortion. Therefore some preamps incorporate two input jacks for magnetic cartridge, one marked "high" and the other "low." The "high" jack is for high-output cartridges; the cartridge signal is attenuated by a voltage divider before it reaches the first phono amplification stage. If it is necessary to use the "high" jack, one should take into account the degree of signal attenuation in calculating the effective signal-to-noise ratio. For example, assume that the peak output of the cartridge is 50 mv, and that the effective value based on this much signal is 80 db. But suppose that the cartridge signal must be fed into the "high" jack, where it is reduced 10 db. Therefore the effective value comes down from 80 db to 70 db.

Preamplifier: Tape Head Input

Evaluation of signal-to-noise ratio proceeds exactly the same way for tape head input as for magnetic phono input. However, it is necessary to recognize the following fact of life. The effective value tends to be at or below the minimum amount consistent with high fidelity, namely 55 db, because of the very low output of a tape head. At 1000 cps a tape play-back head operating at 7.5 ips on a quarter-track basis—the generally accepted mode of operation today for home use—and reproducing a tone recorded at a level producing 3 per cent harmonic distortion (commonly accepted as the maximum tape recording level tolerable to the ear), will typically deliver no more than 1 or 2 mv. A high-inductance head[2] (usually 1 to 2 henries) will produce more signal than the head having a typical inductance of about .5 henry; the difference between them is 3 to 6 db of signal output.

For the following discussion, let us suppose that the peak output at 1000 cps is 1 mv. Assume that the rating of a preamplifier for tape-head input is 54 db below an output of 2 volts. Accordingly, noise output is 4 mv. Assume that the gain on tape-head input is rated at 60 db at 1000 cps. Therefore a 1 mv input produces an output of 1 volt. The effective signal-to-noise ratio—1 volt signal/4 mv noise—is only 48 db.

In the case of another preamplifier, equivalent noise input is rated at 3 μv for tape-head input. With a 1 volt input from the tape head, the effective value—1 mv/3 μv—is 50 db.

On the other hand, if a high-inductance playback head is employed, the peak signal input may be from 3 db to 6 db greater, with a corresponding improvement in the effective value. If high-output tape is used for recording, the increased output in playback may add another 6 to 8 db to the effective value.

[2] The term "high-impedance head" is usually employed.

FM Tuner

More often than not, FM tuner specifications say nothing about signal-to-noise ratio. Occasionally one may find a specification such as the following: "Noise and hum are 60 db below full output." Full output is the peak audio signal, corresponding to 100 per cent modulation (± 75,000 cps) of a strong incoming signal. Full output is typically between 1 and 3 volts.

In the absence of a specification, how does one evaluate the signal-to-noise ratio of an FM tuner. One way is to write to the manufacturer. Another is to refer to his chart that shows the tuner's quieting characteristic as the r.f. signal is increased; this chart is frequently supplied in the instructions or service manual accompanying the tuner. *Figure* 3 shows a typical quieting characteristic. The principal purpose of such a chart is to show how small an r.f. signal is required to produce a signal-to-noise ratio of 20 db or 30 db, based on 30 per cent modulation. By following the quieting characteristic to its extreme, one can discover the maximum quieting, typically 50 to 55 db. Since this is based on 30 per cent modulation, we can add another 10 db to arrive at the effective signal-to-noise ratio based on 100 per cent modulation. The tuner represented in *Fig* 3, which shows about 55 db maximum quieting, has an effective value of 65 db on a strong incoming signal.

Unfortunately, the quieting chart does not always extend far enough to indicate maximum quieting. For example, the chart of *Fig.* 4 shows quieting only for r.f. signals up to 10 μv. From the slope of the characteristic at 10 μv, it is obvious that maximum quieting is appreciably higher than the maximum value appearing within the confines of the chart. But how much higher? Write to the manufacturer.

Quieting characteristics are usually based not on 100 per cent modulation (± 75,000 cps) but on 30 per cent modulation (± 22,500 cps), in accord with the IRE standard. This entitles the tuner to be credited with an effective signal-to-noise ratio 10 db higher than maximum quieting, as in the case of the tuner represented in *Fig* 3. In the absence of information to the contrary, one may *suppose* that the reference is 30 per cent modulation, but one can't really be sure. For example, the quieting chart for a certain tuner expressly states that the reference level is ± 75,000 cps modulation. In the case of another tuner, discussion by the writer with the chief engineer revealed that quieting is based on 100 per cent modulation. In these cases, the effective value is the same as maximum quieting.

Evaluation of a tuner's signal-to-noise

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ratio should also take into account the slope of the quieting characteristic. The steeper the slope the more quickly the tuner attains maximum quieting and the more opportunity one has of enjoying quiet reception of weak r.f. signals. *Figure* 5 compares the quieting characteristics of two tuners. Tuner *A* can claim greater sensitivity because it attains 20 db quieting at 3 μv input, compared with 6 μv for Tuner *B*. But Tuner *B* may provide better listening on weak stations because it reaches near-maximum quieting at 20 μv while Tuner *A* does not attain near-maximum quieting until 60 μv.

The foregoing discussion has indicated that typical FM tuners have effective values of 60 to 65 db. On the other hand, the signal put out by the broadcaster seldom is better than 55 db. This does not mean that a signal-to-noise ratio better than 65 db is wasted on an FM tuner. For one thing, an occasional FM program, particularly a live broadcast, comes through with better than 55 db. For another, it may be expected that the quality of FM broadcasts will steadily, if slowly, improve, particularly as the result of FM stations having to introduce much better transmission equipment in order to cope with the exacting technical requirements of multiplex.

Tape Recorder

A tape recorder that can achieve a signal-to-noise ratio of 55 db or more, based on a maximum recording level producing 3 per cent harmonic distortion at 400 cps may be rated excellent in view of the present state of the art. Relatively few tape recorders achieve such performance. Quite a number do not exceed 50 db, and some fall below 45 db. Those that attain 55 db are usually the ones with separate record and playback heads, so that the playback head can be designed for maximum signal output (high-inductance head), without having to make the compromises required when a head has to serve for recording as well as playback (low inductance is preferable for recording).

Depending on how the specification reads, the rated value may not be the same as the effective value. A number of the less expensive machines are rated on the basis of a recording level that produces 5 per cent harmonic distortion (at 400 cps), which is very close to tape saturation and produces too much IM distortion for high fidelity purposes. Such a machine should be de-rated about 6 db to put its ratings on the basis of a 3 per cent harmonic distortion level. Thus if a tape recorder is rated at 54 db on the basis of 5 per cent distortion, its effective value, presuming the desire to make reasonably clean recordings, is only 48 db.

When a tape recorder has a VU meter, sometimes the rating is stated as a given number of db below 0 VU. Then we must know what recording level is represented by 0 VU. In some high quality machines, 0 VU represents a recording level that produces only 1 per cent harmonic distortion at 400 cps. This provides a margin of approximately 6 db to allow for the mechanical lag of the pointer on transients; that is, when the meter reads 0 VU on a transient, the actual level may be several db above 0 VU. If such a machine's signal-to-noise ratio happens to be rated on the basis of 0 VU, the effective value is 6 db greater than the rated value, because there is about 6 db difference between the recording levels that produce 1 and 3 per cent distortion.

If 0 VU corresponds to 3 per cent distortion, the effective value is the same as the rated value. If 0 VU corresponds to 5 per cent distortion, we should subtract 6 db from the rated signal-to-noise ration. Æ

Professional Tape Reversing Mechanism

NORTH C. HAM*

Playing a tape in the reverse direction involves more than just changing directions; the difference in torque between the take-up and supply-reel motors must be compensated for.

OBSOLESCENCE OF ELECTRONIC EQUIPMENT is always a vexing problem. A particular case in point is my acquisition of a Berlant Concertone approximately five years ago: a 20/20 TWR custom recorder adapted for 2-track stereo record-playback. The vexing decision was how best to "appreciate" the investment so that it could accommodate my backlog of favorite 2-track recorded tapes and also adapt to the presently available 4-track tapes.

The eventual decision made, considering the many possible combinations, was to modify the original tape deck to provide the following features:

1. *Retain the original 2-track stereo record and playback features at speeds of 7½ and 15 ips.*
2. *Retain the use of 7 and 10½-in. reels.*
3. *Accommodate the new 4-track prerecorded tapes.*
4. *Include an automatic tape direction reverse for full continuous playback-rewind without jeopardizing wow and flutter.*
5. *Include a form of d.c. dynamic braking for reel stopping.*

1116 N. 29th Place, Phoenix, Arizona.

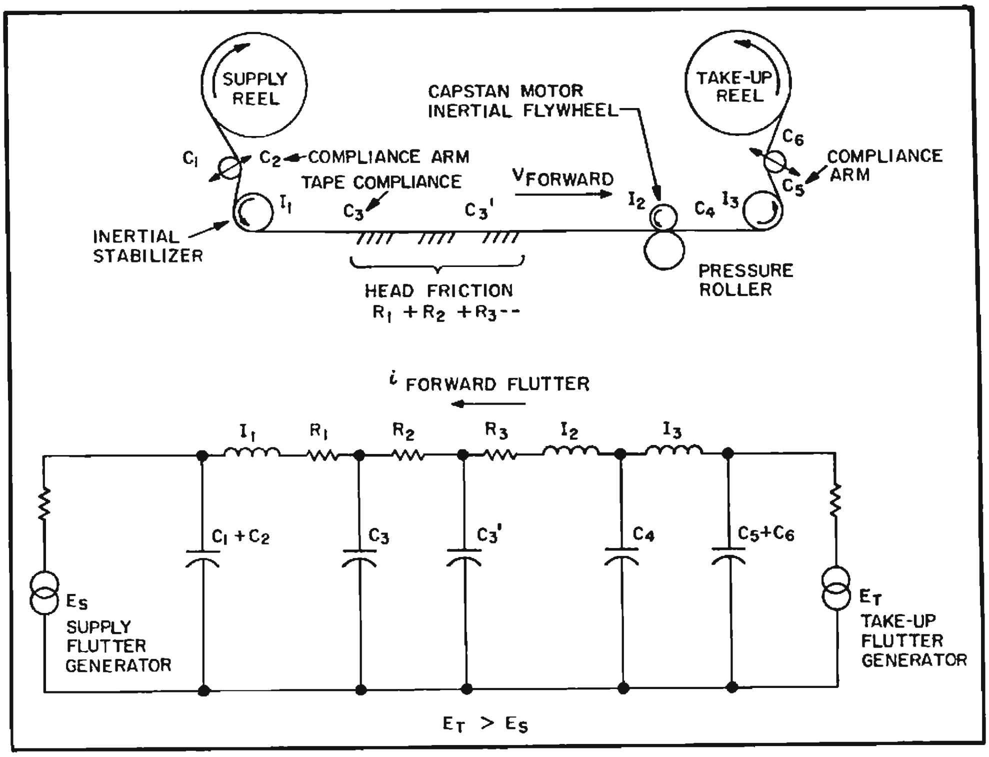


Fig. 2. Mechanical arrangement and electrical analogue for forward direction.

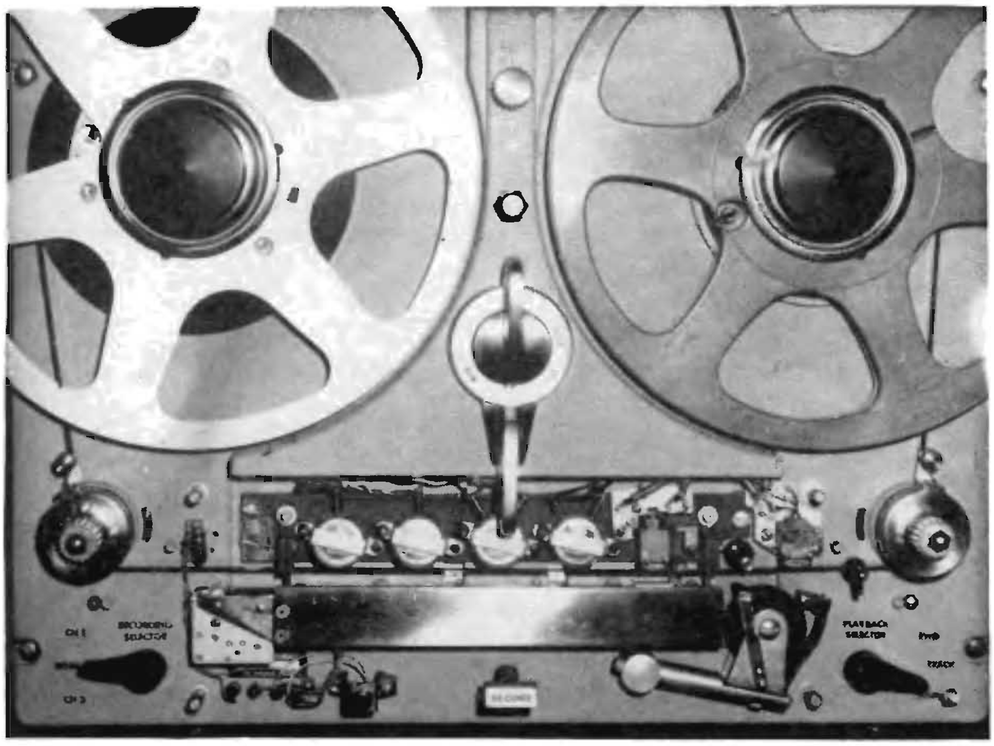


Fig. 1. Top view of tape deck with head covers removed.

The first two mentioned features were easily accomplished by retaining the original mechanical structure and head placement composed of two ½-track erase heads, one 2-track record head, and one 2-track playback head. This total of four heads in a linear array is the original Concertone design.

The third and fourth features were accomplished by using two 4-track playback type heads displaced vertically and at right angles to the tape travel, and separated from each other on each side of the capstan drive-motor assembly. The 4-track heads are automatically switched dependent upon the tape direction. Tape head No. 1 matches track 1 and 3 for the forward direction and head No. 2 matches track 2 and 4 for the reverse or rewind mode. The pickup coils of head No. 2 are actually inverted relative to the matching coils on head No. 1 since the tape reels are not flipped for part two. This is necessary to maintain the proper left and right microphone placement between the part one and part two tape programs. Automatic reversing is accomplished by a light-beam sensing circuit together with a latching relay used as a memory device.

The scheme is basically an inertialess and non-pressure method and will operate reliably regardless of the reel size.

The last feature was obtained by switching 30 v. d.c. through the reel motor fields during the stopping period. A time-delay relay removes this current after a definite braking time. The original felt pressure-braking pads are retained and are useful as a backup and in maintaining a taut tape to reduce spillage during editing.

The result of these changes and additions is a tape recorder-playback mechanism with the following specifications:

	½-Track	2-Track	4-Track
Record	yes	yes	no
Playback (Forward)	yes	yes	yes
Playback (Reverse)	yes	—	yes
Tape Speed 7½ and 15 ips	yes	yes	yes
Reel Sizes up to 10½ in.	yes	yes	yes
Wow and Flutter: Less than	.15% rms	.15% rms	.15% rms

A. Dynamic electrical braking plus mechanical brake.
B. Automatic shutdown after tape completion or breakage.
C. Selection of either 2-track or 4-track playback.
D. Fast reel spooling, forward and rewind.
E. Automatic reverse mode indication and disable.
F. Automatic forward play reset.

An important feature of the mechanical design, concerning head placement, was of great value in achieving the reversing feature. The Concertone TWR tape transport configuration utilizes the arrangement shown in *Fig.* 1. A supply reel pays off the tape which passes over a combination alignment guide, spring loaded compliance arm, and tape stabilizer inertial roller, then passes over the head assemblies. Then the tape is pulled by passing between a capstan with a constant tangential velocity and a rubber pressure roller. Following the capstan, the tape is again fed over another tape guide, compliance arm, and roller prior to being wound upon the take-up reel. Both the supply and take-up reel motors are energized during operation, but rotate in opposite directions, with the take-up motor having greater torque than the supply motor.

The simplified mechanical arrangement and the electrical analogue are shown in *Fig.* 2. The two inertial stabilizers located at both sides of the head assemblies, and the capstan flywheel, form a filter system which reduces the amount of flutter induced into the tape by the mechanical system. Because of the multiple-head arrangement used on this particular mechanism, the total tape friction is the sum of all the head and pressure-pad bearing frictions upon the tape. The total friction was reduced by replacing the original pressure pads with Teflon material risers interspersed between the heads. The risers, being above the head gap surface, provide adequate tape wrap. Also, the low coefficient of friction of Teflon results in reduced tape friction. This reduced friction allows the mechanical filter to have a sharper cutoff and greater attenuation near the filter cutoff frequency.

The resultant combination thus effectively filters out the flutter components of the 60 cps and its harmonics. The expression for the resonant frequency near cutoff is:

$$f_r = \frac{1}{2\pi\sqrt{IC}}$$

where I = total inertia elements
C = equivalent compliance.

This frequency is generally very low and approaches ¼ cps. The effective flutter-generator source is the difference in torque produced by the take-up and supply motors. The take-up motor has the greater torque because of the shunting impedance around the supply motor (see schematic, *Fig.* 6). Since the take-up motor is located at the right, during forward tape travel, the resultant is that the effective flutter current (a.c. component) in the electrical analogue, flows away from the higher potential towards the supply motor or against the steady d.c. current (constant tape travel). (See *Fig.* 2.) The heads, during this tape direction, are located as the head frictions R_1, R_2, R_3, and so forth. It is apparent that the flutter current at this point is low due to the two series filter components, I_3C_5 and I_2C_3, located between this point and the flutter driving generator.

In the case of the reverse tape direction, see *Fig.* 3, the reverse exists in that the motor at the left is now the take-up motor and its greater torque designates it as the driving flutter generator. Hence,

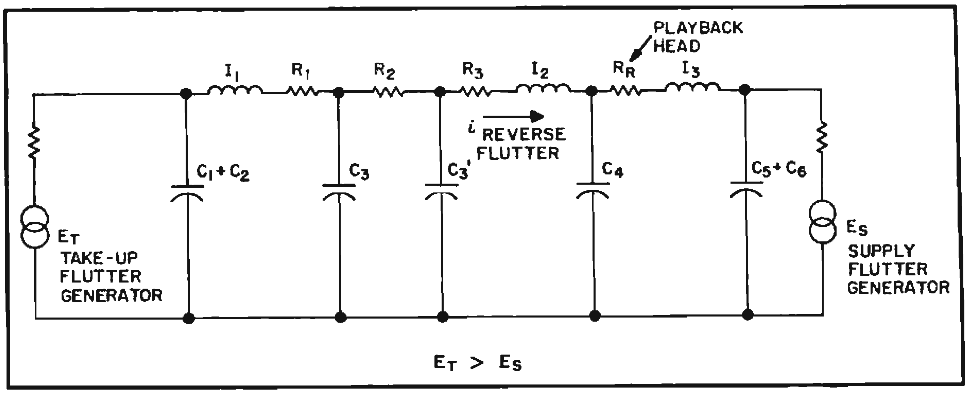


Fig. 3. Electrical analogue for reverse direction.

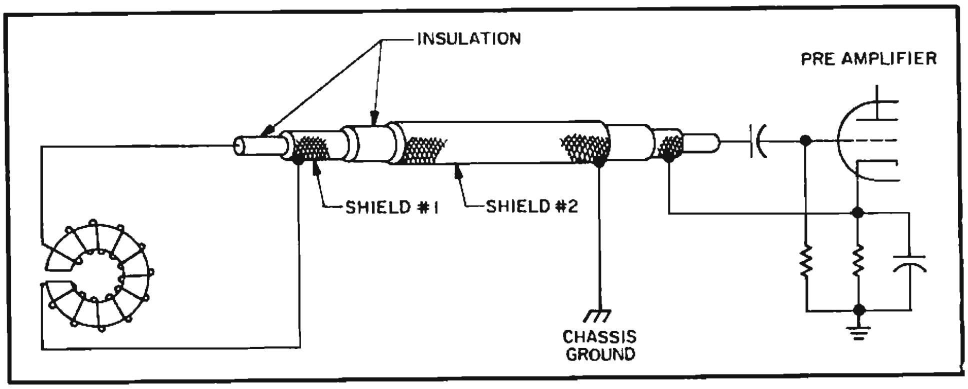


Fig. 4. Cable connection between playback head and preamp.

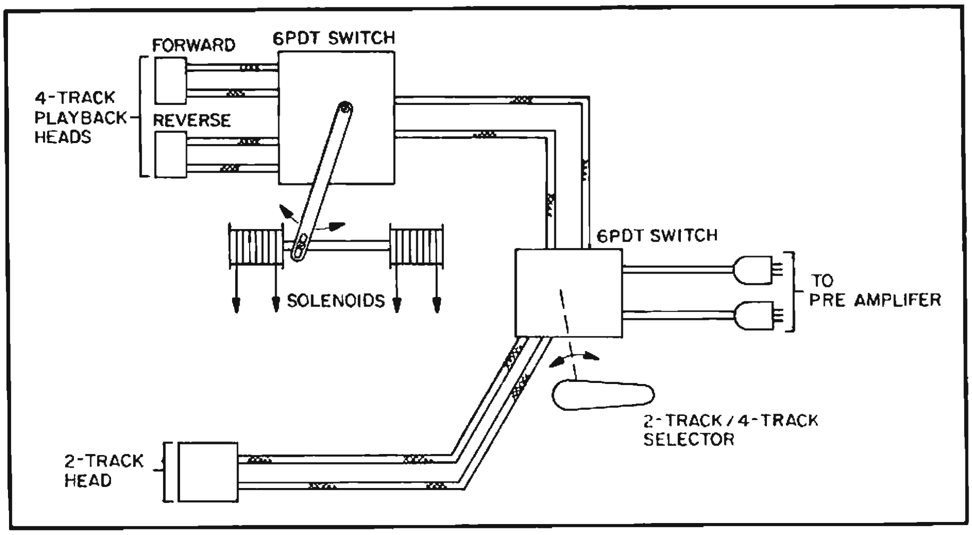


Fig. 5. Switching arrangement between the playback heads.

to reduce the flutter current, the ideal place for locating the reverse playback head would be at the point R_R to benefit from the series components, I_1C_3 and I_2C_4. This fact was proven quite dramatically when the head was placed at this location (as against the initially tried location near R_1)—the reduction in flutter was considerable!

The original control circuitry needed only minor changes to accommodate the reverse-direction mode. Essentially the forward-play circuit remains unchanged, retaining the feature of a transient boost of current through the take-up motor for the first four seconds to overcome the starting inertia. This is achieved by the action of relay K_1 and the discharge of the 80-μf capacitor. Also, the cutoff switch (for end of tape or breakage) and the fast and slow capstan motor-speed switches remain the same. The heavier lines in the schematic diagram indicate the original wiring.

One major change to the original circuit concerns the modification of the capstan motor. The motor was disassembled and an extra connection made to the starting windings of the capacitor induction motor so that the polarity of this field may be reversed relative to the starting capacitor. This results in six separate cables emanating from the motor frame.

The remaining circuitry can be subdivided into the following main sections:

1. The memory device, latching relay K_2, which remembers whether the mode is forward or reverse play.
2. The sensing circuit for reversing the tape direction composed of the photocell, lamp, and relay amplifier.
3. The reverse transient surge-current circuit for momentarily increasing the torque of the reverse mode take-up motor composed of thermal relay K_7 and 100-ohm resistor, R_3.
4. The power supply for the transistor circuit and d.c. braking circuit which is automatically removed by the time relay K_4.
5. The solenoids, K_8 and K_9, for selecting the proper playback head by the associated 6PDT rotary wafer switch.

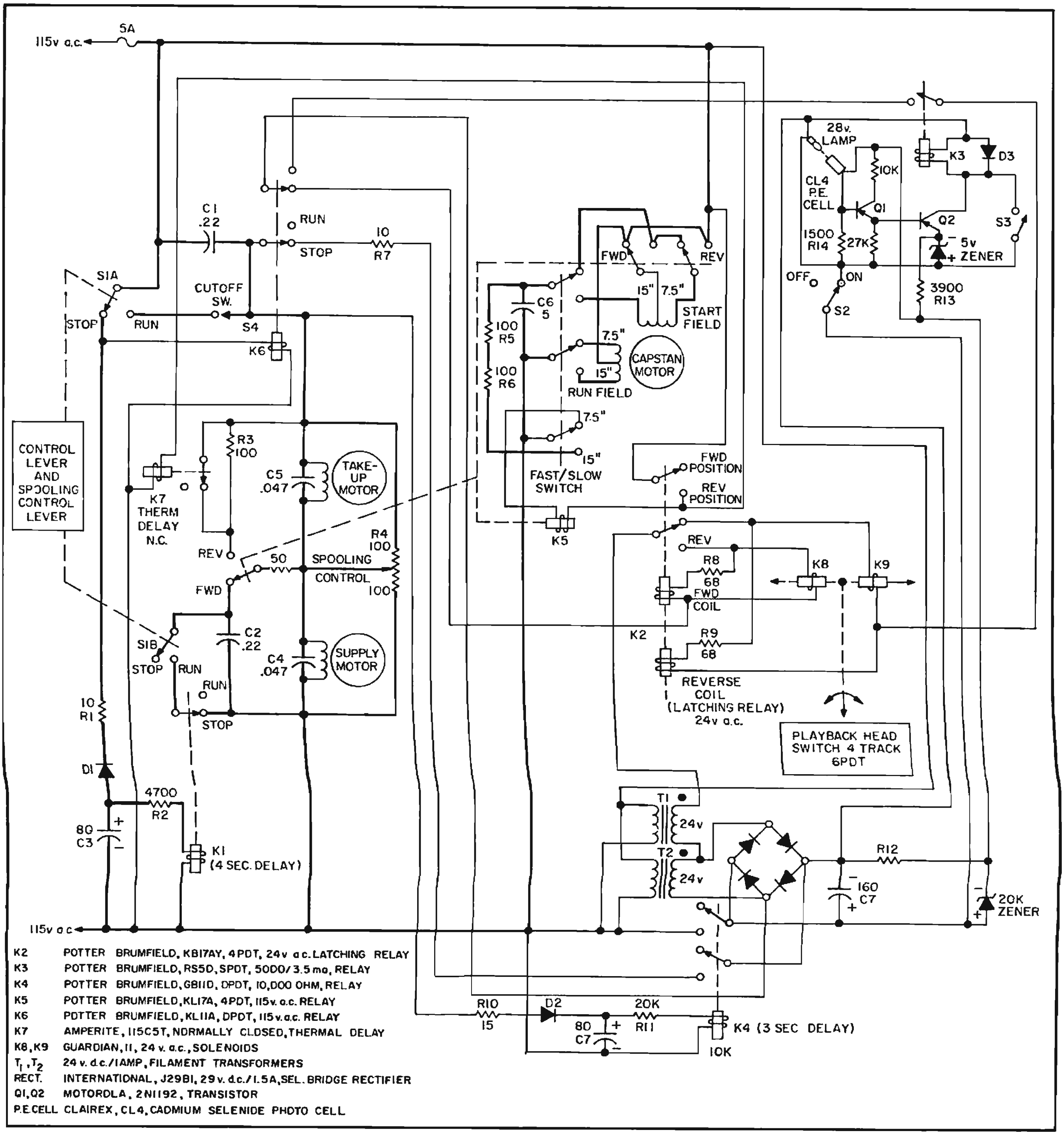


Fig. 6. Schematic of control circuit.

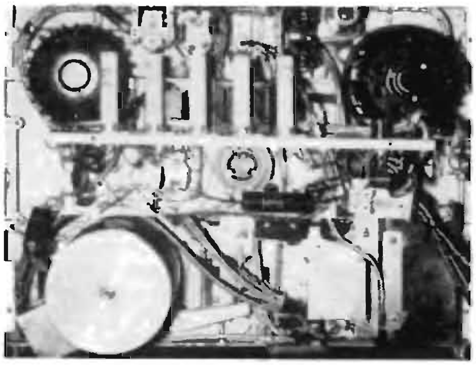


Fig. 7. Bottom view of the deck with chassis cover removed.

When the line voltage is first applied, the latching relay, K_2, will be in the forward (tape direction) mode either initially or automatically by the action of relay K_6 which applies current to the proper latch coil through the crossed connection of the latch relay pole and coils. The latching relay, because of its memory latch, maintains all forward relays and solenoids in the correct forward direction mode. When the tape-drive control lever is placed into the RUN position, normal forward mode operation results. This mode is also automatically achieved whenever the lever is returned to the STOP position; hence, the forward direction is reset by simply stopping the tape mechanism regardless of the particular direction of play at that time.

The sensing device for reversing is mounted at the left end of the tape pressure mounting rack so that, as the tape travels from left to right in the forward direction, the last program selection will be properly completed before reversing. (See *Fig.* 1.) Various methods can be used which trigger the sensing device by allowing the lamp to impinge upon the photocell mounted on the opposite side of the normally opaque tape—methods such as a small punched-out hole, or a clear portion of tape (either spliced in or removed oxide).

Switch S_3 is used to energize the transistor relay, K_3 thus permitting manual tape direction reversal at any time desired for the purpose of either program selection or the application of the reversing trigger spot.

An additional switch, S_2 allows the reversing light and transistor circuitry to be switched OFF whenever the reverse mode is not desired, such as during 2-track playback and recording. The lamp provides an automatic indication of this mode by allowing the light to pass through a colored plastic bezel.

After the tape has finished its last selection in the forward direction, and the (photocell) sensing circuit has been triggered, the sequence of operation is as follows: 1) The transistor relay, K_3, closes which immediately places the latching relay, K_2, into the "Reverse" memory position; 2) Simultaneously the reverse solenoid, K_9, receives an impulse that rotates the head selection switch to connect the proper pickup head to the preamplifier; and 3) The capstan motor is reversed and a current surge, for 4 seconds, is applied to the supply-reel motor which now becomes the take-up motor. The impulse-operated solenoid and latching relay scheme was used to eliminate the need for continuous energizing current. This accounts for the 48-v. a.c. being applied to the 24-v. solenoids and the 68-ohm dropping resistors, R_8 and R_9, for the 24-v. latching relays.

After the final selection has been completed, and the tape completely rewound, tape tension no longer causes the cutoff switch, S_4, to remain closed; and hence the reel motors are automatically de-energized and the tape motion stops. Placing the control lever into the STOP position permits the completed tape reel to be changed and automatically resets the tape mechanism for the forward playback of the new tape reel.

The STOP mode, either actuated by the control lever or the tape spooling control lever, automatically removes the 115-v. a.c. potential from the reel motors by switch, S_{1A}, and simultaneously applies 35 v. d.c. to these motors by the energizing of relay K_6. 115 v. a.c. also is removed from the normally energized relay, K_4, and, after a 3-second delay, capacitor C_7 discharges below the relay threshold removing the d.c. from the reel motors. The time constant comprised of C_7 and R_{11} determines the delay time. The series resistor R_7 determines the magnitude of the braking current and is selected as a compromise for fast braking of small reels, without causing tape stretch, and braking of the 10½-in. reels without causing loops.

The supply voltage for the transistor driver circuitry is regulated to 20 v. d.c. by employing a 20-v. zener diode and dropping resistor R_{12} to limit the diode current. The Q_1 base resistor, R_{14}, is adjusted for holding the relay open under ambient light conditions and proper relay closing dependent upon the lamp intensity. The diode, D_3, across relay K_3, is a surge-current protector for Q_2.

The SPDT switching of the 48-v. a.c. potential by relay K_6 insures proper mode operation and eliminates oscillation in the relay loop.

The various playback heads are connected to wafer switches for 2-track, 4-track, and 4-track forward-reverse selection. The original preamplifier playback-head cable was a triple coaxial arrangement to reduce circulating hum loop currents. (See *Fig.* 4.) This scheme was retained for reducing hum currents and, as a consequence, required a 6PDT wafer type switch. *Figure* 5 illustrates the over-all switching scheme for selecting between either 2-track or 4-track playback and the automatic switching for the 4-track forward-reverse modes (spring detent on wafer switch removed). The switches are enclosed within shielded boxes to reduce capacitance coupling of hum currents to the high-impedance cables.

Mechanically, the various relays, transformer, power supply, solenoids, and so forth were mounted on sheet-metal plates dispersed around the tape deck structure. (See *Fig.* 7.) The upper left plate comprises the time-delay relay circuitry-K_4 and associated rectifier and capacitor-resistor time constant. The

(*Continued on page 57*)

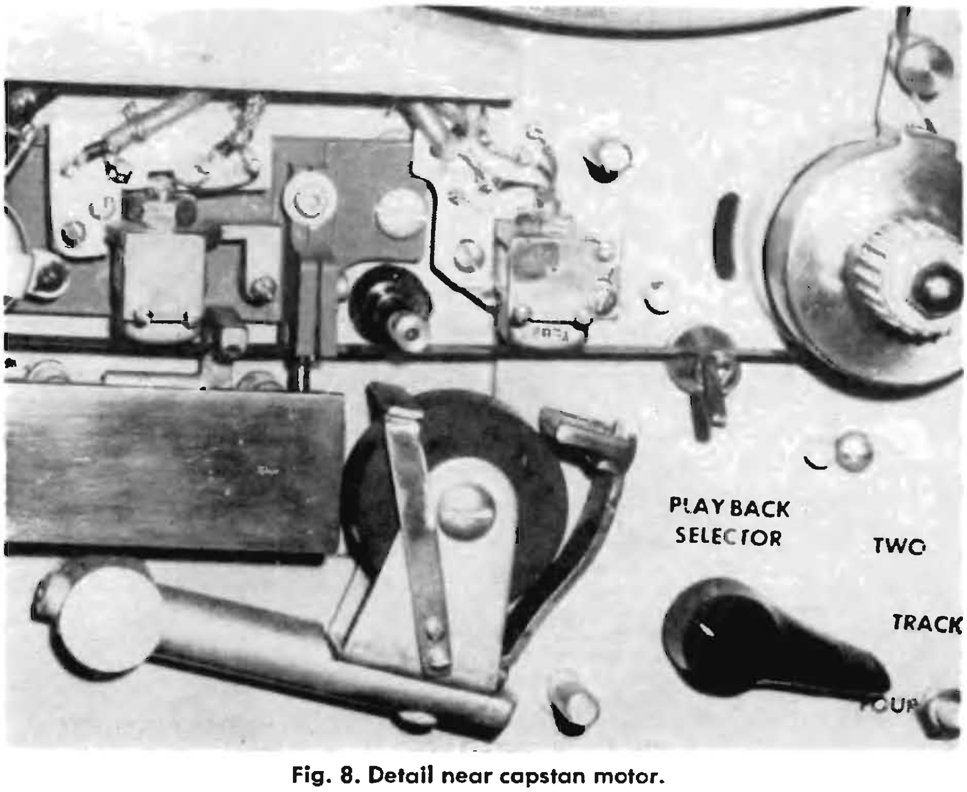


Fig. 8. Detail near capstan motor.

TAPE REVERSING MECHANISM

(*from page 26*)

transformers and the power supply are mounted on the control chassis below the delay relay. The upper right plate contains the latching relay and thermal-delay relay. On the right, below the reeling motor, are mounted the transistor relay, K_3, and relay K_6. At the bottom, near the center, are the two switching solenoids and head-selection switch for the 4-track heads. The box at the lower left contains the 2-track or 4-track selector switch. Reversing relay, K_5, is mounted within the main chassis just below the d.c. power supply.

Power and connections to the transistor circuit and relay are made through feedthrough terminals at the lower portion of the deck. (See *Fig.* 1.) The manual reversing switch is mounted on a bracket adjacent to these terminals. Flexible leads transfer the power to the transistor circuitry terminal board mounted upon the tape tension rack assembly, which moves up and down by the action of the tape-drive control lever. At the opposite end of this rack, near the capstan, there is an "L" shaped metal pull-down hook that automatically pulls the tape away from the heads during fast reeling. The oxide-contacting surface of the hook utilizes a glass rod. (See *Fig.* 8.)

The additions, mounted on the front of the tape deck, are enclosed during normal operation. An additional tape guide was placed to the right of the reverse playback head to help align the tape travel during reverse play mode.

The Nortronics TLB-2 4-track stereo heads are used on a specially fabricated bracket. The bracket was constructed to allow adjustment in three planes. These particular heads require a magnetic shield to reduce induced hum flux. A metal magnetic shunt plate was positioned to distort the hum flux to the minimum value for each head. These were mounted on the pressure rack and pressure roller arm to allow automatic retraction during reeling and threading operation.

Another variation would be to install a second photocell setup at the right side of the mechanism, or logic circuitry with the present device, to automatically place the playback mode back into the forward direction for repeat of the complete tape program. Thus, if 10½-in. reels were used with 4-track information on ½-mil Mylar tape, it would require over four hours of playing time before the program material would repeat itself when played at the speed of 7½ ips. All that is required for this feature is to effectively place the latching relay back into the forward mode. Momentarily switching S_{1A} to the STOP position and back to the RUN position would accomplish this. The operation, when switching to the opposite mode, requires a gradual slowing down and change in direction so that no loops or tape stretch would result to even ½-mil tape. Æ

Is Magnetic Tape Long Lived?

JOHN T. MULLIN*

Fig. 1. Checking 1949 tapes of Bing Crosby and Al Jolson radio programs he recorded, Mullin finds no signs of aging. Tapes were stored in original, unsealed boxes for more than 15 years in garages in California heat and Minnesota cold. When played, sound reproduction was excellent. *(Photo courtesy 3M.)*

No one knows how long good quality tape will retain all the signal, but this pioneer has tapes which are 15 years old, and still going strong!

No one knows how long a good quality magnetic tape can last without measurable deterioration.

As a matter of fact, and I speak from a background of 20 years of experience in professional recording and playback, there is excellent evidence that good tape will retain high quality recordings indefinitely.

Of course, magnetic tapes were not developed for general use in the United States until 1946 so that a first impression is that our experience can go back only that far. However, magnetic tape engineers and technicians in our laboratories for years have been testing various tapes on a 24-hour basis, punishing them with continuous wear equivalent to 100 years of normal use without appreciable change in the performance of these tapes.

In 1949, while I was recording engineer for Bing Crosby's radio shows, I taped Crosby singing a duet with Al Jolson. On the same tape is Mary Martin at a time when she was doing the musical, South Pacific." The tapes were "Scotch" Brand No. 111 manufactured by 3M Company. The tape was the same tape that was available in the stores for home recording. The recorder I used was the first Model 300 to leave the Ampex factory.

In the more than 15 years between 1949 and 1965, those tapes were stored in home garages in California, where summer heat raised inside temperatures as high as 120 degrees above zero, and in Minnesota, where the winter cold sometimes reaches 35 below.

No particular pains were taken to care for these tapes. The tapes were not played periodically or even run through a recorder to relieve tension. They were stored in their original boxes and the boxes were not sealed.

Yet, when I played the tapes a few weeks ago, the audio levels peaked at the same levels as when the tapes were recorded more than 15 years ago. Instead of being brittle, the tapes were still pliable on the reel. There was no snapping or pulling as a layer of tape came off the roll toward the playback head and there was no flaking; that is, no flakes of iron oxide fell from the tape. There were, in fact, no signs of aging except that the tape reels are the old-fashioned kind and the boxes are dirty and worn from years of storage under adverse conditions.

Most important, sound reproduction was excellent. I could detect no loss of quality at all.

Those of us who developed 3M Company's Professional Mastering Recorder, a machine that records master tapes from which disc records and pre-recorded tapes are made, have also made tests of machine reaction to tapes of various age. Segments were taken from a number of old tapes and were spliced together and run through test machines to determine, among other questions, the effects of age on magnetic tape. Here too we found that good quality magnetic tapes last indefinitely.

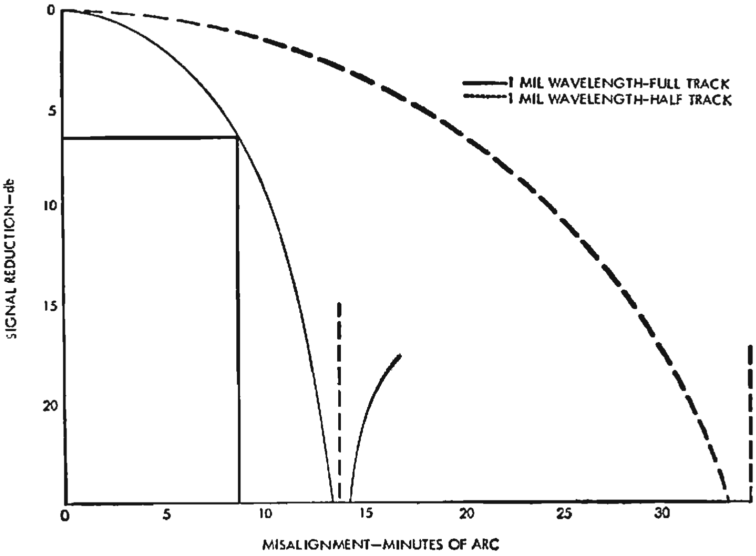
What about Lubrication?

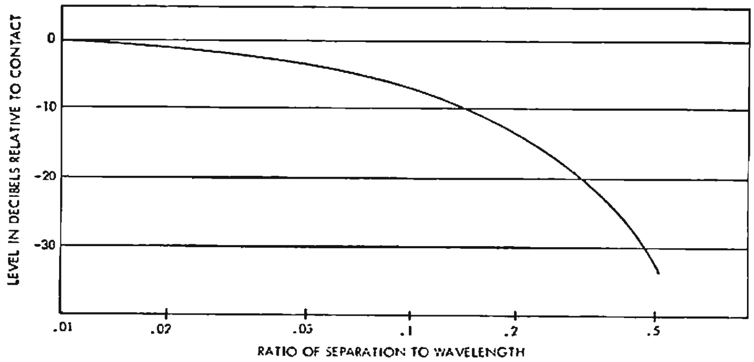
There are those who believe that relubrication of magnetic tapes will keep them smooth and will reduce the amount of oxide that wears off on tape heads. In my experience, this has been an unnecessary and unimportant step with the tapes I have used. "Scotch" Brand tapes are manufactured with a built-in, dry, Silicone lubrication which lasts as long as the tape itself, which is to say that Silicone lubrication lasts indefinitely.

When slight ruboff does occur from extended use, the wear products from the more rugged oxide coating take on the consistency of a fine transitory powder rather than a gummy "balling" consistency of conventional coating wear products. This design feature of the tape virtually eliminates oxide building at the head gap, which is a common cause of output loss due to head-to-tape separation. Such tapes, therefore, can be used time after time and can be rerecorded after many years of age with-

Fig. 2. Tape-punishing setup at 3M testing laboratory in St. Paul. Tapes are fashioned into continuous loops and are run 24 hours a day, giving wear equivalent to 100 years of normal use without appreciable change in performance of tapes. *(Photo courtesy 3M.)*

*Minnesota Mining and Manufacturing Co., St. Paul, Minn.





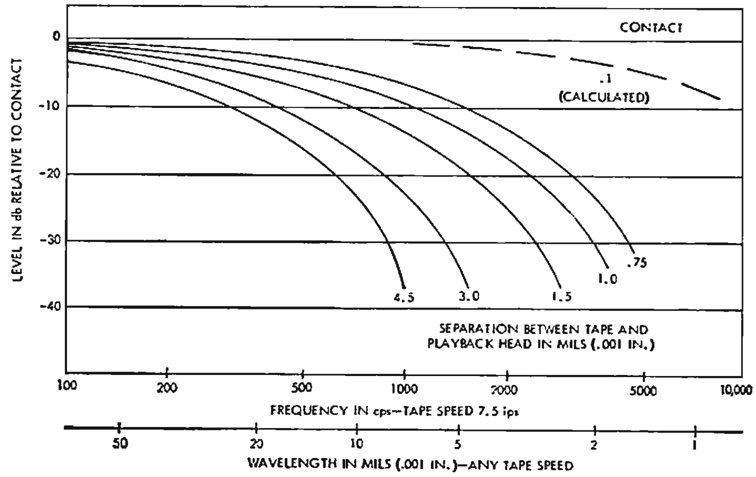


Fig. 3. (above) High-frequency signal reduction due to misalignment.

Fig. 4. (above right) Signal attenuation caused by poor contact in playback universal curve for any speed, frequency and separation.

Fig. 5. (right) Signal attenuation caused by poor contact in playback as function of frequency (or wavelength) for various separations.

out appreciable loss of their ability to retain and faithfully reproduce the recorded signal.

Some lubricated tapes, such as "Scotch" Brand No. 151, use oxide formulations similar to conventional magnetic tapes. They differ only by the addition of an external lubricant layer. These tape constructions also provide the benefits of the more dependable oxide dispersions used in advanced instrumentation and computer tapes. These heavy duty dispersions offer lower static charge buildup and greater wear resistance, making them particularly suitable for continuous loop applications.

Some Rules for Preserving Tape

There are, of course, some common sense rules to consider in the handling of tapes to allow them to perform to their utmost capability. The rules are perhaps most important in the use of tape for very sophisticated applications such as those in areospace and computer work.

If the tape is wound badly, for example, the result would be bad tape tracking, resulting in poor frequency response due to distorted, rippled or warped edges. Damaged tape will not meet the playback head properly.

For reasons of interchangeability, an attempt is made to align the head gaps on all recording machines exactly perpendicular to the tape. This is very important. For example, in a full width (¼-in.) recording of one-mil wavelength, a misalignment of only eight minutes of arc will reduce the output by about six decibels (see *Fig.* 3).

Intimate contact must be maintained between tape and head gap for good frequency response. Loss of contact between head and tape, due to specks of dust, splicing adhesive lodged on the recording or playback head, scratches in the head surface or foreign matter on the tape, however slight, has a profound effect on high frequency output (see *Fig.* 4 and 5).

Storing tape in original box protects it from dust and from physical damage to its edges. Normally, cleaning of tape is not necessary but if there is excessive dust on the tape, the reels may be vacuumed and the tape can be cleaned by wiping it with a clean, lint-free, dry cloth while rewinding.

Avoid accidental exposure of the tape to magnetic fields. Weak magnetic fields will increase print signal and strong fields within a few inches of the tape may cause erasure. Don't store tape in cabinets with magnetic door latches if the tape is likely to come in contact with the magnetic latch mechanism.

In general, occasional use of recorded tapes, or simply re-winding them, improves their resistance to aging in prolonged storage.

Most cases of tape distortion can be traced to excessive winding tension, uneven winding or both. A high speed wind is usually soft enough due to entrapped air.

If you know you're going to store tapes for an unusually long time, use a polyester-backed tape. Polyester has 50 per cent better resistance to temperature change and 15 times better resistance to humidity change than acetate-backed tapes. Under ideal conditions, magnetic tape should be stored at room temperature—between 60- and 80-deg. F, with relative humidity controlled between 40 and 60 per cent. Temporary variations beyond these limits, though, generally are not harmful.

If some tapes become brittle during storage under severely hot, dry conditions, they can usually be returned to a condition that will allow playback of the information. Simply storing a brittle tape, out of its box or container, at proper environmental conditions for 24 hours should bring the tape back into balance. A simple way to restore moisture to dry tapes is to leave a lightly moistened sponge or blotter with the reel of tape in a closed container for 24 hours, being careful to keep the moisture from coming in direct contact with the tape.

A tape exposed to extreme cold should be allowed to return to normal room temperature before it is played.

Finally, occasional cleaning of the recording head, capstan, tape guides and other parts of your machine will help assure utmost wear life for your tapes. Many tape recording engineers use Freon TF or ethyl-alcohol or similar cleaning agents but, for your own machine, see your recorder operating manual.

Each of these points makes its contribution toward excellence in recording and playback but, in my opinion, the place to start is with the tape. Make sure it's a quality tape. Æ

New Tape Types

LARRY ZIDE

Everyone who uses a tape recorder uses raw tape at some time or other. With so many types of coatings, base material, and thickness, the recordists should become thoroughly familiar with all of their characteristics.

RAW TAPE IS THE LIFEBLOOD of tape recording. That statement, on the surface, is so obvious that it would seem to need no restating, yet how many tape fans are truly aware of how vital a link in the overall picture raw tape actually is?

Tape manufacturers are offering an almost bewildering array of tape types—each designed (it says in the advertising) to best do a specific job. Well, some do, and some don't. Or, to be more accurate, some do their special job to the detriment of another parameter that may be important to you. Only that isn't mentioned in the ads.

Consider that tape is actually a three-part affair. There is the base material, the magnetic oxide coating and finally, the binder that holds oxide to base. Each of these basics is subject to wide technical variations in type or quality. And, it is these factors that determine the ultimate value of a specific tape for a specific need.

This is not an article to determine if one company is making a superior product. Our investigations indicate quite clearly that no manufacturer has a monopoly on superiority. At the same time, none of the manufacturers we have surveyed has come up with defective merchandise either. Any of the recognized brand names offer a product that is qualitatively acceptable. Further, we found that any of the manufacturers could be depended upon to hold a tight line of consistency from reel to reel of a particular tape type. No tape ran over 0.75 dB reel-to-reel variation. Some ran as little as 0.25 dB. So, from best to worst is no more than a discountable 0.5 dB.

Tape researchers are constantly striving to improve their product. That reel of standard Scotch 111 or Audiotape is not the same stock that you may have bought a few years back. All of these companies have been quietly improving the basic product.

Regular readers of AUDIO have seen reports of tape recorders with legitimate 25,000 Hz response at 7½ inches per second. These figures are being achieved with *standard* tapes. The special extended high-end stocks have not been used (more about them, presently). Such high-end response would not have been possible a few years ago. True, improved tape heads play an important part, but the simple fact is that tape, and not the machines, is the limiting factor in recording. Manufacturers are close, but they have not yet reached the theoretical limitations for any given tape speed.

So, sound quality is one tape factor. And what is involved in tape sound quality? Frequency response, noise level, distortion, and print-through. Each must be minimized to improve the qualities of the finished recording.

All of the recently released formulations are offering improved potential for high-frequency response. There seems to be no clear-cut superiority here that would tend to suggest that any one manufacturer has a better product on this score alone.

Noise level is something else again. This is one area where manufacturers have been busy. Increased noise suppression means better dynamic range and this is something demonstrable. Audio Devices, Scotch, Kodak, and Sony-Superscope are offering new "low-noise" tapes. Just what are these?

Audio Formula 15 and Scotch #201-202-203 exemplify these new tapes. Both offer about a 5-dB improvement over "standard" tapes when it comes to reduction of background noise. And both claim, and apparently succeed, in holding other tape parameters to equal or better values. But neither tape will provide superior performance to standard tapes without changes in recording practice. Full advantage of low-noise characteristics requires an increase in recording bias of about 20 per cent. Further, an increase in recording level of about 2 dB is called for.

The Sony-Superscope low-noise tape is just being introduced. At present we have no technical details on it. However, it is likely that it will require the same adjustments to achieve the same results.

Kodak does not market a specific low-noise tape. Their claim is that their standard formulation has certain low-noise characteristics, yet conforms more closely to standard bias and level considerations. Our limited tests tend to confirm this.

These low-noise tapes offer other side benefits to the user of top-notch recording equipment. As side effects of their characteristics they allow lower distortion recording (sometimes significantly so). Used on unmodified machines they will perform no worse than standard tape and may show slight increase in dynamic range and high-end response.

Are they of value to the average recordist? That depends on your definition of average. Many home type machines will derive no benefit. Some will offer slight improvement. And a few will be able to achieve the full advantages that these new tapes offer. Certainly the professional will find increased dynamic range and reduced distortion of real value.

Is it really necessary to write again about print-through *vs.* tape thickness, time, and temperature? 1.5-mil thickness should always be used except, and only except, when greater time is the vital factor. And then, avoid oven-like storage.

Most modern tapes have print-through characteristics that are considerably better than those of the recent past. And most of the manufacturers offer low-print tapes for applications where this item is a specific problem. In addition to the companies already mentioned, Burgess and BASF offer formulations that minimize print-through of signal. Using these tapes does result in a slight loss of signal output too, but the print-through loss is greater than the gain loss thus improving the over-all ratio.

By the same token there have been several new high-output tapes introduced. These increase the output for a given signal all right, but they also increase the distortion. And, they may increase the print-through.

Tape Backing Material

Plastic and Mylar®* have been been around for a long time and really

(*Continued on page 60*)

*TM Registered, DuPont Co.

TAPE TYPES

(*from page 23*)

need little reintroduction. Plastic is, of course, cellulose acetate; Mylar is the DuPont trade name for polyester film. We have some thoughts on the relative merits of these that may differ from what is usually considered.

Mylar is stronger than plastic. This is true. But Mylar, if pushed beyond endurance, stretches into a useless string. Plastic merely breaks, and can be respliced.

Mylar is a staple material. It does not absorb or lose water to the atmosphere. So it has a degree of *permanence* not ascribed to plastic. Plastic can lose water; further, it can lose that water faster from the reel edges than from the center so it cups and makes poor head contact.

This is theory. And in practice these tapes will conform to their type. However, for all practical purposes, allowing for careful storage of acetates, Mylar offers no real long-term storage advantages. *Why?*

There are many factors that limit the archival value of magnetic film. Stock deterioration is *not* the primary one. The very advantage of Mylar can be turned against it in long storage. Mylar is exceedingly smooth. It presents unique problems of emulsion adhesion. Time can cause the separation of oxide and base. And this is more a problem with Mylar than with plastic. The Library of Congress, in a recent report on "Preservation and Storage of Sound Recordings" concludes its study of tape materials by stating that 1.5-mil Mylar seems to be the best for long-term storage but that some doubt exists about base-to-coating adhesion.

This confusion of choice (Mylar is higher priced than plastic) is further confounded by the recent introduction of two new materials. The first of these is tensilized cellulose triacetate.

As can be guessed from the generic name, this material is closely related to our old standby. Only Eastman Kodak offers this material. Their trade name is Durol. What does it offer over regular plastic?

Durol's great virtues seem to be two-fold. First, significantly greater break resistance over standard plastics (though not as good as Mylar). Second, exceptional stretch resistance—half that of plastic, as much as a twentieth that of Mylar. This is important for it means that a broken tape may mean no wow distortion or loss. Durol tapes cost little more than standard plastics but it must be remembered that their keeping qualities are similar to the plastics.

PVC—polyvinyl chloride—is a new-old name in tape stocks. Only BASF and Burgess offer PVC-based tapes. We were not able to receive complete technical data on this stock in time for this report. However, PVC exhibits a strength that is similar to plastic; it is resistant to aging in the same way as Mylar and is extremely supple, thus providing excellent head contact. However, stability and oxide-adhesion questions remain unanswered fully.

We have touched on this question of oxide-to-base adhesion. That, in part, is the job of the binder. This is one component about which we have been able to find out absolutely nothing. Binders are carefully guarded proprietary secrets. Needless to say, all manufacturers claim perfection for their binders but we suspect that there may well be long-term differences between them.

Several manufacturers are offering triple-play tapes; that is, 3600 feet on a seven-inch reel. At least two—Reeves and Ferrodynamics—are coating their super-thins with high-output tape. The result, they claim, is no loss of output due to the use of thinner coatings. We have no figures on the print-through.

The aforementioned Ferrodynamics also adds to all of their tapes, at no additional cost, a strip of head cleaner tape plus metallic tabs fore and aft each reel. Irish offers a very wide range of regular and economy tapes. Finally, Sony-Superscope is the only one we know of offering a 3¼ inch reel that will fit the standard wide-pin professional machines. For owners of battery portables and big studio-type machines, this can fill a real need.

These then are the new types of tape to have appeared recently. Again, it needs to be said that the so-called standard tapes of all these companies, Audio Devices, BASF, Burgess, Ferrodynamics, Irish, Kodak, Reeves, Scotch and Sony-Superscope are vast improvements over that which has only recently been available. Remember, finally, to do as the professionals do. Find the tape that best suits your needs. Set the machine bias to suit that product—and stick with it. Æ

Tape Duplication at Ampex

Making high-qualiy recorded tapes commercially isn't as simple as dubbing a record at home. One of the industry's leading practitioners in this art tells how his company does it. But don't expect to do likewise—like us, you probably do not have a 120-ips machine, nor even a 60 or a 30.

ED ZDOBINSKI*

In a sense, recorded-tape duplication begins when the first master tape is cut in the recording studio. This original recording is the closest link between subsequent duplications—both disc and tape—and the live recording session.

In making disc recordings, an electrical transcription is first produced from the master tape. The electrical transcription is then used to produce the recorded discs. At this point, if the particular selection is chosen by Ampex for tape duplication, the master tape is loaned to Ampex's recorded-tape production center in Hackensack, New Jersey.

Ampex's tape facility is the largest producer of recorded tapes in the world. Five tape-production lines, each consisting of a master recorder and ten slave recorders, are in operation two shifts each day. Between 4000 and 6000 recorded tapes are turned out each day. Selections vary from educational language tapes to popular music to classical music.

Because of the recording company's need to preserve the original master tape, Ampex creates a second master using precise recording techniques and equipment. Making the second master from the original is probably the most critical phase of recorded-tape production. Original master tapes are monitored to establish side timings, duplication levels and necessary equalizations. Each original master is then copied on four-track master tape at 15 inches per second using the highest quality Ampex professional tape recording equipment.

Second Master Tape

The result is a second master tape as close to being identical with the original master as possible. From the second master, as many as 20,000 consumer duplicates must be made. The original master is then returned to the appropriate recording company. Labels which Ampex records include Atlantic, Command, Warner Brothers, London, D.G.G., Kapp, Mercury, and Verve.

The highly precise recording techniques used in producing the second master tape are also used in making the duplicates. As in mastering, tape duplication demands the best in tape, tape recorders, and slaves. The master recorder runs at 120 ips. In producing 7½ ips tapes, slaves run at 60 ips; for 3¾ ips tapes, slaves run at 30 ips. Recording is constantly monitored during duplication and the final product is spot-checked.

From the duplicating area, tapes are transferred to the packaging area. Here, tapes are placed in boxes, labeled and vacuum-wrapped to prevent deterioration or contamination in shipment or storage.

Finally, Ampex recorded tapes are shipped to one of our regional distribution depots (Los Angeles, Elk Grove Village, Ill., and Hackensack, N.J.) Æ

Ampex Engineering Manager—Tape Duplication

Left, mastering section, where duplicates are made of the original masters. These duplicates serve as the actual working masters from which as many as ten final "release" tapes are made. Below, one of five tape-mastering lines at the Ampex Stereo Tape production facility in Hackensack, N.J., where more than 3000 recorded tapes are produced daily. The second master is placed on the Ampex professional magnetic tape duplicator (left) which is electronically connected to 10 slave recorders which produce copies on blank tape.

The most generally agreed upon objective of high fidelity is to recreate the realism of live music performances in the home listening environment. Various acoustic characteristics and psychoacoustic phenomena have been identified over the years that appear to be responsible for the emotional impact and excitement that we experience at a live performance. However, past limitations in the tape recording process and software formats for traditional delivery systems, that is records and tapes, have imposed serious restrictions on one of the major factors involved — dynamic range. This article will explore the significance of dynamic range in music and review the use of noise-reduction technology and dynamic range expansion to improve the realism of music reproduction for increased listening pleasure.

Since the late 1890s when Thomas Edison and Emile Berliner first began to record musical performances on cylinders and discs, there has been a persistent disparity between the quality of "master" recordings produced with the performing artist and that of mass-produced copies made from the masters. Steady advances in recording technology, however, have improved every link in the chain of music recording and reproduction, narrowing the gap between live and recorded performances.

A major milestone in the history of recorded music was the introduction of the long-playing microgroove record in the 1940s, which increased the playing time of records so that compositions of considerable length could be recorded uninterrupted. Another major advance followed in the form of magnetic tape recording that provided a means of editing recorded performances and producing a master tape from which an unlimited number of vinyl pressings could be made. Unfortunately, magnetic tape recording of analog signals introduced its own set of problems which detracted from the fidelity of recorded music, the most notable being tape hiss, but also including wow and flutter and other forms of distortion.

The realism of recorded music was dramatically enhanced in the 1950s with the introduction of stereophonic sound, which brought much of the perspective of the hall or stage to the home. The "three-dimensional" character of sound produced by multi-channel signal processing touched off a great deal of research that, in the late 1960s, led to a development that promised an even greater increase in realism of recorded music — quadraphonic sound. But, the audio industry experienced a marketing disaster with quad due, in large part, to its inability to agree on hardware and software standards. The potential benefits of four-channel sound were never fully realized, even though the concept had considerable technical merit and is still being explored.

Over the years, the quality of music-reproduction hardware (amplifiers, record playing equipment, and speakers) has surpassed that of available music software — records, tapes, and radio broadcasts. In response to growing demands for recordings with improved sound quality, so-called "audiophile" records were introduced during the 1970s in the form of direct-to-disc records and digitally mastered records. The major contribution made by these technical innovations was the elimination of tape hiss and the various forms of distortion associated with analog master tape recordings. Their superiority over conventional records was immediately obvious; however, even these fine recordings are too often marred by the presence of record surface noise and restrictions on the dynamic range that could be captured on and retrieved from a vinyl disc.

We are still faced with the challenge that has been the underlying motive for the steady stream of technological advances in sound recording and reproduction — *to recreate the excitement and emotional impact of a live performance in the home listening environment.* Our ability to meet this challenge is certainly enhanced by analyzing and understanding the acoustic and psychoacoustic factors that characterize "live performance" sound, so that appropriate consideration will be given to these characteristics when attempting to make improvements in the music recording and reproduction process.

Characterizing Live Performance Sound

Our appreciation and enjoyment of music, whether live or recorded, is strongly related to three major factors that characterize live performance sound: Tonal balance, spatial perspective, and dynamic range.

Tonal Balance

The tonal balance of music has received the greatest amount of attention over the years. It has long been appreciated that the low-frequency content of music should be kept in balance with the high-frequency content, consistent with that which occurs in live performance. Today, most recording and reproduction equipment is capable of handling frequencies that extend well beyond the audible range of 20 Hz to 20 kHz. Electronics are readily available that offer responses over this frequency range with variations smaller than those which can be detected aurally. Covering a frequency range necessary for quality music reproduction is no longer a significant technical challenge for records, phono cartridges, and tape recorders. Also, many fine speakers are now available that are capable of uniformly reproducing most all frequencies in the audio spectrum. Further improvements in frequency response characteristics are unlikely to produce greatly significant improvements in the perceived tonal balance quality of recorded music.

Spatial perspective is a somewhat elusive quality of sound involving a complex combination of geometric and temporal factors. In the geometric dimension, our perception of the spatial character of sound seems to center around a *panoramic* sound field in which individual instruments can be localized. Giving breadth to the musical performance, the sense of spaciousness involved is extremely important in creating the illusion of "being there" at a live performance. Stereo sound reproduction represented a major step forward relative to recreating the spatial perspective of live music. Also, special speaker designs have been developed in an attempt to produce a combination of direct and reflected sound similar to that which exists in live performances. Finally, recent developments have resulted in microphones and signal-processing techniques which are claimed to recreate the sound field geometry present during the original performance.

The temporal aspect of spatial perspective is primarily related to what is usually described as a sense of *ambience.* What a listener hears at a live performance is a composite of multiple sound waves, each arriving at slightly different times because of the many different sound transmission paths that are involved. This dimension of live music has been the subject of considerable research and experimentation over the years involving the use of reverberation and delay-line devices to create auxiliary acoustic signals that are delayed in time from the master audio signal. The most recent developments that address the temporal character of sound in music reproduction are electronic "time delay" or "ambience recovery" systems that employ either analog or digital signal processing to create auxiliary signals with variable amounts of time delay. The primary function of such devices is to simulate the ambience of a wide range of acoustic environments — from small, intimate rooms to large, highly reverberant halls.

There is still much to learn about the geometric and temporal characteristics of sound. Some of the approaches taken to introduce spatial perspective in music reproduction have

resulted in an unnatural sound quality. Others have improved the sense of realism. More substantial advances in creating a realistic spatial perspective undoubtedly are yet to come.

Dynamic Range

Dynamic range is the difference between the sound levels during the loudest (fortissimo) and quietest (pianissimo) music passages. Giving depth to the musical performance, its existence during a live performance is as apparent to the listener as its absence in a recording. Exposure to the dynamics of live music has caused us to appreciate the extremely large amplitude differential between the whisper of a lone flute and the thunderous finale of a symphonic work. It's not too unusual to experience a 90-dB dynamic range in a live performance but, unfortunately, more than one-third of this range (and its associated effect on realism) has traditionally been lost before the music signal gets through the recording and reproduction process.

In order to store the music information on magnetic tape or vinyl discs, it traditionally has been necessary to compress or otherwise modify the amplitude of the recorded signal so that (1) the signal strength during loud passages stays below saturation and tracing distortion levels for tape recording and disc mastering, respectively, and (2) the signal strength during quiet passages stays sufficiently above magnetic tape noise levels and record surface noise levels. Basically, the problem has to do with the *signal-to-noise ratio* (S/N) limitations of tape and disc recording processes. The S/N ratio has to be greater than the desired music dynamic range if loud music passages are to be recorded without distortion and quiet passages are to be heard clearly above the background noise on the tape or record. Hence, an S/N ratio of 60 dB may be required to provide a clean dynamic range of 50 dB with 10 dB safety margin shared between the top and bottom ends of the dynamic range.

During tape recording, a common method for restricting the dynamic range is to "gain ride"; that is, the recording engineer manually adjusts the levels while the recording is being made — reducing them during fortissimo passages and increasing them during pianissimo passages. This same result is frequently accomplished automatically by using *limiters* that prevent high-level signals from exceeding a preset level or by using *compressors* that gradually reduce level when loud passages occur and increase level during quiet passages. Without the aid of a tape noise-reduction system and excluding allowance for signal peak "headroom," the dynamic range capability of a professional studio tape recorder is typically 60 dB at the 15 ips speed and somewhat higher at 30 ips. The comparable figures for high-quality audiophile open-reel tape recorders operating at 7.5 ips is 50 dB, and for a good cassette recorder, about 45 dB applies. The psychoacoustic impact of such dynamic range restrictions is to make the music sound "flat" or "thin." The sharp edge of percussive attacks is blurred, the contrast between loud and quiet instruments is muddled, and the overall definition is obscured, thereby diminishing the excitement and realism of the recorded performance.

A similar problem occurs during disc mastering, since the maximum dynamic range that can be stored on a vinyl disc is about 55 dB for conventional pressings and up to about 65 dB for the very finest pressings. Again, a substantial loss in excitement, emotional impact and realism results from the dynamic range restrictions of conventional discs — and they are plagued as well by annoying record surface noise, even for the best direct-to-disc and digitally mastered records.

Since dynamic range has a tremendous effect on our enjoyment of music, it should be recognized for its importance and receive more attention in the music reproduction process. Further improvements in characteristics like frequency response are not likely to significantly increase the overall quality of music reproduction. On the other hand, a modest increase in dynamic range will be immediately perceived as making substantial improvements in the realism of music reproduction. Consequently, dynamic range represents very fertile ground to be explored relative to coming closer to our goal of recreating the live-performance musical experience in the home.

Understanding Dynamic Range

The range of sound pressure that humans are capable of perceiving is extremely large. For example, the sound pressure is about one million times greater at a level that causes pain or discomfort (about 120 dB) as it is for the threshold of audibility (0 dB). This means that the human auditory process handles approximately twice the dynamic range in dB (or 1000 times in sound pressure) of a professional studio tape recorder.

Relative Sound Pressure	Sound Pressure Level	Typical Sound Sources or Environments
	130 dB	Artillery Fire (Close Proximity)
1,000,000	120 dB	Jet Aircraft (Close Proximity)
	110 dB	Orchestra/Band (Audience)
100,000	100 dB	Train/Propeller Airplane (Interior)
	90 dB	Bus/Truck (Interior)
10,000	80 dB	Automobile (Interior)
	70 dB	Average Street Noise
1,000	60 dB	General Business Office
	50 dB	Private Office
100	40 dB	Residential Living Room
	30 dB	Suburban Bedroom
10	20 dB	Recording Studio
	10 dB	Sound-proof Room
1.0	0 dB	Total Silence

Fig. 1 — Typical sound sources or environments for a range of sound pressure levels measured in dB as well as indicated relative to 0 dB.

Our hearing mechanism responds to changes in sound intensity (pressure) in a roughly logarithmic manner, rather than in an absolute way. For this reason, and as a matter of convenience, the decibel scale is used to describe Sound Pressure Level (SPL), as follows:

$SPL = 20 \log_{10}(P/P_o)$

where the reference pressure P_o is defined as the threshold of audibility corresponding to 0.0002 μbar, and a μbar (microbar) is the pressure of one millionth of an atmosphere. Hence, a range of 0 dB to 120 dB covers the entire range of sound amplitude that is of interest, much like the range of 20 Hz to 20 kHz that applies to the audible frequency range of sound.

Typical sound sources or environments are presented in Fig. 1 for a 130-dB range of sound pressure levels. A doubling of sound pressure corresponds to a 6-dB increase in SPL, while 20 dB represents an order of magnitude increase. While the levels of SPL are approximate for each sound source indicated, a comparison of relative SPLs is very revealing. For example, the background noise level in a residential living room is 10 times (or 20 dB) higher than the level in a recording studio. Similarly, the noise level that exists in a general business-office environment is 10 times higher than the living room level.

Returning to music, it is possible for peak sound pressure levels to momentarily reach 120 dB during transients. If these peak levels are produced in a studio, concert hall, or home environment, the "noise floor" of 20 dB to 40 dB, depending on its spectral distribution, can reduce the perceived dynamic range of music to about 90 dB. Therefore, a music recording and reproduction system need only be required to handle a maximum dynamic range of 90 dB to properly represent the characteristic of live performance sound.

Noise Sources and Masking Effects

It is important to note that there are two separate and distinct fundamental sources of noise which detract from the fidelity of recordings — those that are introduced while recording a master tape and those that are associated with disc mastering and playback. Direct-to-disc, digital recording, and tape noise reduction applied to conventional analog recording are techniques that address the issue of eliminating (or at least reducing) the introduction of noise prior to disc mastering. Unfortunately, full benefit of these advanced recording techniques is lost to a great extent when the recording is transferred to a vinyl disc.

With the masking effect of tape hiss removed, the surface noise of the record becomes all the more objectionable. It appears that the elimination of a major noise source, such as tape hiss, serves to highlight another — record surface noise generated as a result of the interaction between the stylus and the record groove. Any roughness of the vinyl surfaces of the groove walls causes extraneous stylus motion, creating the familiar record surface noise which is inseparable from the music on conventional vinyl discs. Of course, the opposite is also true. If the surface noise of discs is eliminated, any imperfections in the master tape relative to hiss or other forms of noise and distortion become more apparent. Hence, it is clear that noise in *both* the master tape and the recorded disc has to be eliminated, or at least dramatically reduced, to significantly improve the quality of recorded music.

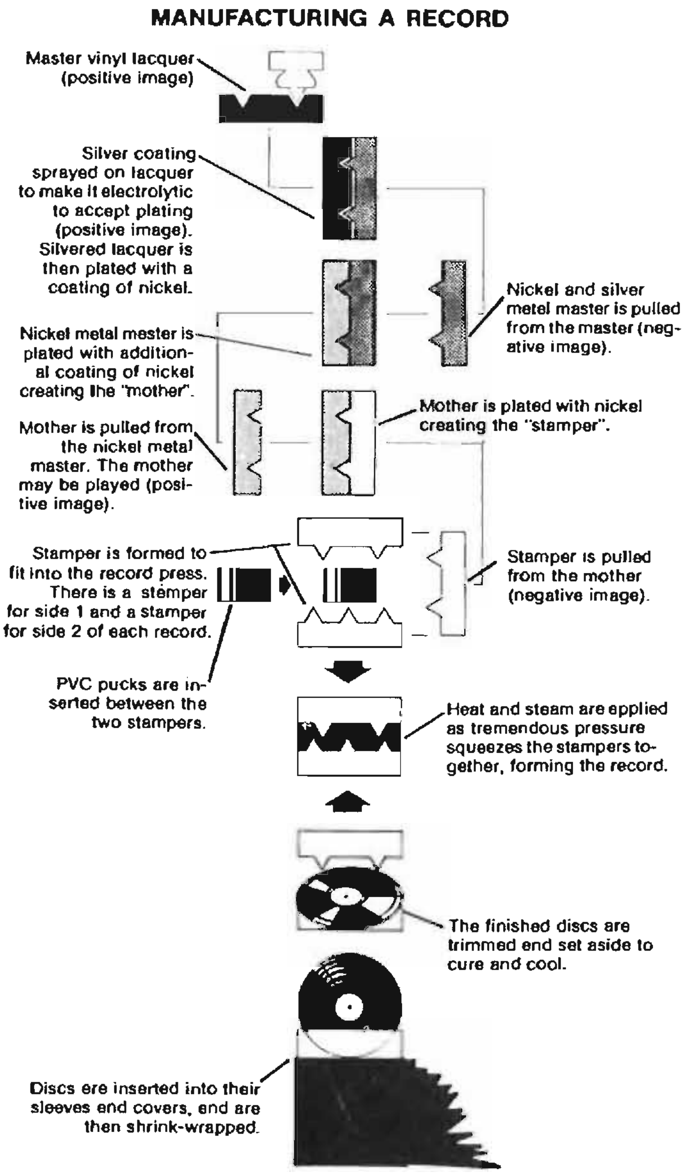


Fig. 2 — Illustration of the various stages of the record manufacturing process. (Courtesy of Nautilus Recordings.)

Evolution of Recording Technology

Since the mid-1940s, there have been major technological advances in sound recording. Prior to the development of magnetic tape recording, records were made in the old 78 rpm format involving the cutting of a single-channel (monaural) groove in the surface of a master recording blank. The fidelity and time per side (four minutes per side on a 12-in. disc) were limited, and any performance errors required recutting the master. With the advent of the long-playing microgroove record came extended playing time per side, improved pressing quality (accomplished by using vinyl instead of shellac), and increased durability — all resulting in a better overall value to the record-buying public.

The Record Manufacturing Process

The beginning point in the record manufacturing process is the production of a master disc or lacquer. This is accomplished by using a master tape recording (or the real-time output of a mixing console in a direct-to-disc recording session) to provide an electrical signal that is fed to a heated cutting stylus which engraves the surface of the lacquer with traces representing the musical waveform. The diagram presented in Fig. 2 illustrates each subsequent step of the process involved. Since the quality of a lacquer may deteriorate if it is not plated shortly after being cut, the multi-stage electroplating process is generally scheduled to occur immediately after the master cutting session. The (positive image) lacquer is plated to produce a (negative image) metal master. Then a (positive image) metal "mother" is made, from which (negative image) metal stampers are made. The stampers are used to press (positive image) vinyl discs that are a replica of the original lacquer.

From each master lacquer, only one metal master can generally be made. The number of mothers available from each metal master is limited, as are the number of stampers from each mother and the number of pressings from each stamper. These limitations are such that only about 20,000 to 25,000 high-quality pressings can be produced from each lacquer. Multiple lacquers, therefore, are necessary to produce larger quantities of quality pressings.

Analog Tape Recording

For the last several decades, it has been common practice in producing records to temporarily store the musical performance on magnetic tape. A typical recording session involves the recording of multiple "takes" of a musical selection in order to provide enough material to create something which approaches "the perfect performance." This is accomplished by an editing engineer who listens to the various takes and chooses the best parts of each one, then combines them together (by physically cutting and splicing the magnetic tape). This results in an edited recording of the musical selection. The practice is so common that few musical selections (notably those recorded "live" before an audience) listened to on a record today represent a continuous performance by the musicians or performing artist. Rather, we generally hear a composite of several performances where any mistake or imperfection is removed in the editing process. Consequently, the ability to edit represents one of the major advantages of using a magnetic tape recorder in producing a record.

Another advantage of tape recording is that the master (edited) tape can be used to cut as many lacquers as are required to meet consumer demand for the record album. Hence, there is no limitation to the quantity of records that can be produced from a master tape.

Although records produced from analog master tapes represent the best sound recordings available until recently, there have been serious technical limitations inherent in the process. Various forms of distortion and noise are introduced into the recording because of the mechanisms that are operative in the recording process. For example, analog tape recording suffers from such problems as tape hiss, wow and flutter, frequency response non-uniformity, modulation noise, multi-channel crosstalk, print-through, and transient distortion. Of particular significance is the background noise that appears as tape hiss in the recording, which ultimately gets transferred to the master disc and vinyl pressings produced from the tape.

As the magnetic tape passes by the recorder head of a tape transport during the recording mode, the electrical signal provided by the microphone pickup of the music performance causes a reorientation of the magnetic particles on the tape to produce a magnetic replica of the music waveform. In the playback mode during the cutting of a master disc, the magnetic tape passes by the recorder head but, in addition to "reading" the music signal, the head also "reads" the random distribution of magnetic particles on the tape and this is perceived as tape hiss. Consequently, during the cutting of a master disc, both music and tape hiss signals are communicated to the cutting stylus, thereby introducing background noise in the master disc and all records that are subsequently pressed. As discussed earlier, the presence of noise in analog tape recording results in dynamic range being restricted to about 60 dB for professional studio tape recorders, while for audiophile reel-to-reel and cassette tape recorders, the dynamic range is restricted to about 50 dB and 45 dB, respectively. A trade-off exists, consequently, between the advantages of tape recording (editing and unlimited production capabilities) and the undesirable restriction it imposes on dynamic range.

Tape Noise-Reduction Systems

A number of techniques have been employed since the early 1970s to improve the dynamic range capability of analog tape recorders. These "noise-reduction systems" process audio signals in various ways to increase the S/N ratio of a tape recording system, resulting in an increase of its usable dynamic range.

Two of the most successful approaches to tape noise reduction have been those developed by Dolby Laboratories and dbx, Inc. Neither one can improve the quality of an existing audio signal, that is, any noise existing in the audio signal is processed unaltered. Their function is solely to reduce the amount of noise added during the recording process. Hence, they might appropriately be called *noise-prevention systems*.

Both Dolby and dbx operate effectively as *companders* — a signal processing system involving *encoding* of the audio signal during recording and *decoding* the signal during playback. However, their operation is based on completely different principles.

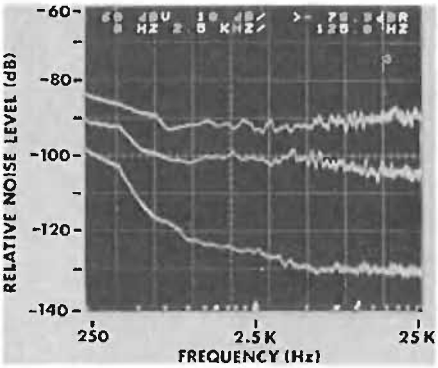
The Dolby Type A ("professional") noise-reduction system is actually a dynamic equalization system that operates over four separate bands of the audible frequency range. During recording, increasing amounts of pre-emphasis are applied to the audio signal in these frequency bands as the signal level approaches predetermined reference levels. During playback, complementary amounts of de-emphasis are applied to the signal in the four frequency bands. Recognizing that the behavior of the Dolby system is nonlinear relative to amplitude, the identical reference or "threshold" levels must be employed during recording (encoding) and playback (decoding) to avoid mistracking distortion. The Dolby Type B ("consumer") noise-reduction system operates over a single frequency band in the high-frequency region above 1 kHz where tape hiss is normally encountered. During recording, all signals having high-frequency amplitudes below a reference level are subject to pre-emphasis, with subsequent complementary de-emphasis applied during playback. In the course of attenuating the high-frequency content of the audio signal during playback, tape hiss is reduced.

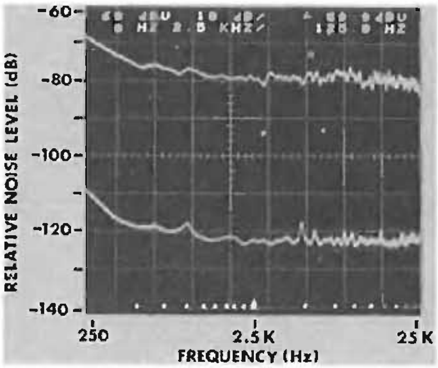
The dbx Type I ("professional") and dbx Type II ("consumer") noise-reduction systems operate on all frequencies and all amplitudes of the audio signal. The dbx system functions as a linear decibel compressor/expander that is neither frequency selective nor level sensitive, and thus it does not involve the use of reference levels or calibration test tones. During recording, the amplitude of the audio signal is compressed while, during playback, the amplitude of the signal is expanded in a complementary fashion. The basic idea behind the dbx tape noise-reduction system is to keep the music signal level sufficiently higher than the offending noise level (e.g., tape hiss) during recording (encoding) so that, in the course of expanding the signal during playback (decoding), tape hiss is reduced, thus extending the S/N ratio of the tape recording system.

By processing all the frequencies in the audio signal, the dbx system eliminates any anomalies that may be introduced by limiting the companding action to a limited frequency band. Precision level-sensing circuits are used to control both the encode (compression) and decode (expansion) modes of operation during recording and playback, respectively. Employing a voltage-controlled amplifier (VCA) and a level detector in each stereo channel, the compression/expansion gain instructions given to the encode and decode VCAs are equal and opposite (mirror image) as long as the two level detectors track accurately.

All recording systems have some frequency-dependent phase shift which can change waveforms considerably even though the sound is not audibly degraded. This could introduce an error signal that is added to the music signal prior to decoding. The dbx system circumvents this problem by employing wide-range rms (root-mean-square) level-sensing circuits that respond to the signal energies, regardless of their phase relationships, in creating a command signal to drive the VCAs. Using a 2:1 compression during recording and a 1:2 expansion during playback, this linear decibel compander provides accurate results, even with extremely sharp music transients, over a wide (100 dB) range of signal level.

The decode process in tape noise-reduction systems should be a mirror image of the encode process, and the accuracy of the system in providing precise mirror imaging determines the fidelity of the signal processing. Any extrane-





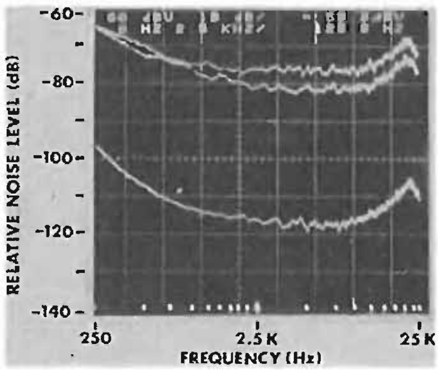


Fig. 3 — Constant 125-Hz bandwidth frequency spectrum of Ampex 456 tape using a Studer/Levinson A-80 professional studio recorder operating at 15 ips. Noise level is shown relative to 0 VU for no noise reduction (upper trace), Dolby Type A noise reduction (middle trace), and dbx Type I noise reduction (lower trace). The total wide-band (20 Hz to 20 kHz) noise level of -65 dB is reduced to -76 dB by the Dolby noise-reduction system and to less than -100 dB (at the limit of the measurement instrumentation) by the dbx noise-reduction system.

Fig. 4 — Constant 125-Hz bandwidth frequency spectrum of Scotch 206 tape using a Pioneer RT-707 audiophile open-reel recorder operating at 7.5 ips. Noise level is shown relative to 0 VU for no noise reduction (upper trace) and for dbx Type II noise reduction (lower trace). The total wide-band (20 Hz to 20 kHz) noise level of -59 dB is reduced to -96 dB by the dbx noise-reduction system.

Fig. 5 — Constant 125-Hz bandwidth frequency spectrum of Maxell UD XL-I tape using a Tandberg TCD-340A cassette recorder operating at 1⅞ ips. Noise level is shown relative to 0 VU for no noise reduction (upper trace), Dolby Type B noise reduction (middle trace), and dbx Type II noise reduction (lower trace). The total wide-band (20 Hz to 20 kHz) noise level of -48 dB is reduced to -56 dB and -84 dB by the Dolby and dbx noise reduction systems, respectively.

ous signal introduced into the system subsequent to encoding but prior to decoding is decoded along with the original audio signal. This, as well as any mistracking errors between the encode and decode modes of operation, can introduce distortion or artifacts in the form of unnatural sounds. The nature of the artifact is different for each type of companding system, but its character is frequently described as a "pumping" sound or "noise modulation." Obviously, one would like the companding system to be totally free of any forms of distortion, but reality is such that a trade-off is frequently involved between the elimination of tape noise and distortion of the companded signal. Fortunately, psychoacoustic masking provided by the music signal itself is generally adequate to make the artifact inaudible. The masking effect of a given sound is greatest upon offending sounds that are of somewhat higher frequency. Hence, a music signal having frequency content that encompasses the frequency range of the artifact will generally be processed with complete fidelity. The enhanced music experience that is provided by increased dynamic range and reduced background noise makes the trade-off heavily in favor of using these types of tape noise-reduction systems.

To illustrate the effect of tape noise reduction on the performance of a range of tape recording systems, background noise spectra were measured with and without Dolby and dbx noise-reduction systems in the circuit. The results are presented in Figs. 3 to 5 for a professional studio, audiophile open-reel, and high-quality cassette recorder, respectively. The frequency spectrum of noise was measured using an automatic spectrum analyzer for a constant 125-Hz bandwith. The noise spectra in the figures indicate the noise level relative to 0 VU for frequencies ranging between 250 Hz and 25 kHz. The total wide-band noise was also measured in each case for all frequencies in the audible frequency range (20 Hz to 20 kHz). Since Dolby noise-reduction systems comparable to dbx Type II systems are not readily available, only noise spectra with and without dbx Type II noise reduction are shown in Fig. 4 (as well as later in Figs. 6 and 7).

A summary of the total wide-band noise levels for professional studio, audiophile open reel, and cassette tape recorders, with and without tape noise reduction, is presented in Fig. 6. The chart also indicates the approximate increase in dynamic range provided by Dolby and dbx tape noise-reduction systems for each type of tape recorder. The information on this chart may be combined with that previously given to tabulate the approximate usable dynamic range capabilities of various types of tape recorders, with and without noise reduction, as presented in Fig. 7. Usable dynamic ranges in excess of 80 dB are possible for each type of tape recorder using dbx noise reduction. Professional studio recorders equipped with dbx noise reduction meet the objective of 90-dB dynamic range and, therefore, they can properly represent the music dynamics of live performance sound.

It is interesting to note that, with its wide-band companding action, the dbx noise-reduction system makes the dynamic range capability of open-reel tape recorders approximately equal to that of 14-bit and 16-bit digital recording systems (discussed later). This is a particularly important observation in view of the cost penalty associated with digital recording equipment.

Direct-to-Disc Recording

During the 1970s, a "new" recording approach was introduced for disc mastering. It involved cutting a master disc in real time — the same way that 78-rpm discs were produced from about 1898 until their demise in the early 1950s. Instead of storing the musical performance on magnetic tape, the electrical signal from the mixing console is amplified and fed directly to the cutting stylus of a mastering lathe — hence, a "direct-to-disc" recording.

In the many years since 78-rpm discs were cut direct to disc, advances in the design of disc-cutting equipment have allowed a return to the direct-cutting method to produce recordings that sonically are superior to conventional records produced from master analog tapes. The performance advantages of direct-to-disc records result from the elimination of problems associated with magnetic tape recording, such as tape hiss, channel crosstalk, distortion due to magnetic oxide saturation, print-through, wow and flutter, etc. Records produced in this fashion have greatly increased clarity of complex musical passages, particularly those that involve loud bursts of percussion.

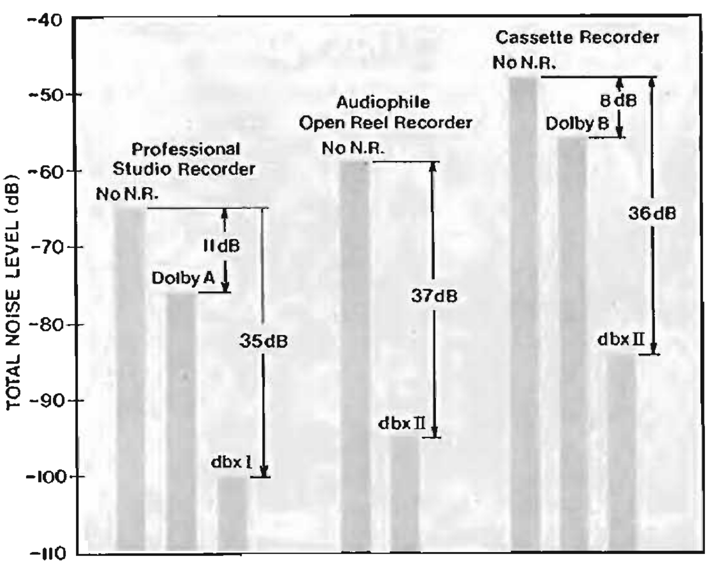
While their sonic superiority over conventionally produced records is immediately apparent, direct-to-disc records must still contend with the dynamic range limitations and surface noise problems of vinyl discs. Furthermore, direct-to-disc records have other potential disadvantages, including (1) inability to edit, since the master disc is cut in real time during the musical performance; (2) decreased playing time per side since the cutting lathe must be controlled manually, rather than by a computer as is normally done, and (3) limited quantity of records that can be produced (since there is no master tape to cut additional master discs as required), which results in a premium price. The trade-offs involved with records produced in this fashion may not be worth it for those who are bothered by the dynamic range limitations of vinyl discs and the annoying record surface noise that may be still present.

Digital Recording

Digital recording involves the storage and retrieval of a musical performance on magnetic tape with the continuous analog audio signal replaced by a series of binary numbers. Basically, when music is recorded digitally, its waveform is sampled electronically many times a second (such as 50,000 times/sec) to produce a sequence of numbers that, as a function of time, represents a piece-wise replica of the original analog music waveform. This sequence of binary numbers (which are comprised solely of a combination of "zeros" and "ones" — the language of the digital computer) are generated by an analog-to-digital converter at the input to the tape recorder.

When the tape is played back (for the purpose of cutting a master disc, for example), the binary numbers are "read" by the recorder head, and the sequence is translated into an analog electrical signal by use of a digital-to-analog converter. One of the most significant aspects of digital recording is that the only information the playback system can recognize is the sequence of binary numbers recorded on the tape. As the magnetic tape passes by the recorder head, the head ignores the random distribution of magnetic particles on the tape which, for an analog recording, is perceived as tape hiss. Only information about the pure music signal is passed on to the master disc, and noise comparable to tape hiss on analog recording is non-existent. Furthermore, digital recording is not plagued by other forms of distortion inherent in the analog recording process, such as wow and flutter, crosstalk, or print through, although it is susceptible to a type of distortion known as quantization noise.

Fig. 6 — Comparison of total wide-band noise levels of tape recording systems relative to 0 VU, with and without Dolby and dbx noise reduction, for professional studio, audiophile open-reel, and cassette tape recorders. The approximate dynamic range increase provided by Dolby and dbx tape noise-reduction systems is indicated for each type of tape recorder.



Noise Reduction	Recorder Type: Professional Studio	Audiophile Open Reel	Cassette
None	60 dB	50 dB	45 dB
Dolby	71 dB	N/A	53 dB
dbx	95 dB	87 dB	81 dB

Fig. 7 — Approximate usable dynamic range capabilities of professional studio, audiophile open-reel, and high-quality cassette tape recorders, with and without Dolby or dbx tape noise-reduction systems.

A 16-bit digital system is capable of recording music with a 90-dB dynamic range over the complete audio frequency range when a sampling rate of about 50,000 samples per second is used. Similarly, an 85-dB dynamic range is available from a 14-bit digital system. Editing is accomplished electronically rather than by physically cutting and splicing the tape. Digital tapes are immune to degradation caused by long-term storage of magnetic tape, and they can be duplicated through many generations without loss of sound quality. The only drawback to digital recording at its present state of development appears to be the relatively high equipment cost involved and potential problems caused by lack of standardization. Nevertheless, from a purely technical point of view, the performance capabilities of digital recording are extremely impressive and suggest the degree of sound quality that ultimately may be made available to the listening public when technical standards of delivery systems are agreed upon and digital playback systems of reasonable cost are developed.

Dynamic Range Expanders

Putting aside for the moment the concept of producing recordings with full dynamic range and inaudible background noise, it is appropriate to explore the possibility of enhancing the value of existing record and tape collections by restoring at least a portion of the dynamic range that existed in the original live performance. Since recordings played in the home, or those played in broadcast studios and transmitted to the home, have their dynamic range limited to something generally less than 60 dB (representing a loss of at least one-third of the potential 90-dB dynamic range), it seems logical to introduce some form of dynamic range expansion to counteract the compression that exists in tapes, records, and broadcasts. The general function of such devices, known as *dynamic range expanders*, is to make loud passages louder and/or make quiet passages quieter, resulting in a spreading out or expansion of the dynamic range of the music signal.

Unlike the situation which exists with companders, where the expansion process is a mirror image of the compression process, dynamic range expanders operate in a "single-ended" fashion, processing the music signal according to its particular design or user concept, rather than providing expansion that is the exact converse of the compression process.

Since the amount and nature of the compression that existed during recording and/or broadcast is not generally known, dynamic range expanders typically offer a variable range of expansion capability. The user selects the degree of expansion that provides an overall pleasing effect, avoiding excessive expansion that can introduce undesirable artifacts.

Dynamic range expanders based on somewhat different operating concepts are available from companies like dbx and MXR. Most of these devices process the wide-band audio signal by sensing the overall average level of the music signal and increasing the level (making the music louder) when a preset threshold level has been exceeded. More sophisticated expander designs separate the music signal into multiple frequency bands so that the degree of expansion that occurs in a given frequency range depends on the level of signal in that range, thus preserving the integrity of tonal balance and the timbre of individual instruments during complex musical passages.

The amount of expansion that is appropriate will vary according to degree of compression that resides in the audio signal as well as on individual music tastes. Excessive expansion can lead to artifacts frequently described as "pumping" or "breathing" noises. To avoid this situation, expansion should be limited to less than a factor of 1.5 for most popular or rock music, while factors of 1.3 or less are generally preferred for classical music. Properly utilized, dynamic range expanders can restore a significant portion of the original dynamic range that is lost in the recording process, dramatically increasing the excitement, realism, and enjoyment of conventional recordings and broadcasts.

dbx Encoded Discs

The two major problems with vinyl discs that have stayed with us over the years are restricted dynamic range and record surface noise. In cutting a master disc, a signal level that is too "hot" can create a condition that will cause tracing distortion by the cutting and/or playback stylus. If the level of the signal gets too low, it may be obscured by the record surface noise. These two conditions place upper and lower bounds on music signal levels that can be stored on a vinyl disc, resulting in a maximum dynamic range of 50 dB for conventional pressings and up to 65 dB for the very best pressings.

Record surface noise is generated as a result of the interaction between the playback stylus and the record groove. Modulations in the groove cause the stylus to undergo complex motions that are translated by the phono cartridge into an electrical signal representing the musical waveform. The stylus tip is less than one-thousandth of an inch in diameter, yet it must travel up, down, and sideways thousands of times a second to follow the undulations of the groove. Any roughness of the vinyl surfaces of the groove walls cause extraneous stylus motion, creating the familiar record surface noise which is inseparable from the music on a conventional disc.

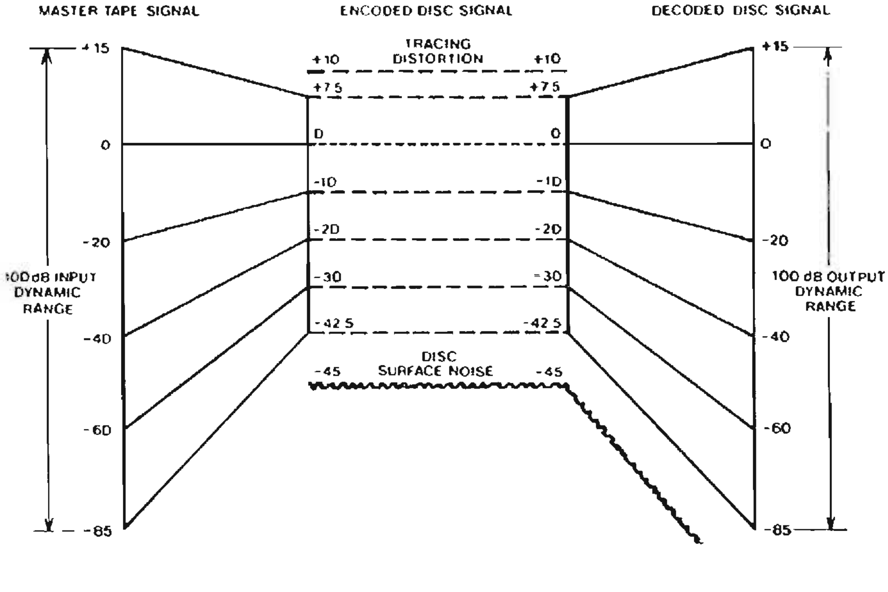
One solution to the problem of restricted dynamic range and surface noise of vinyl discs is available through the application of dbx noise-reduction technology. The dbx Type II noise-reduction system, previously described relative to tape noise reduction, can be employed in record mastering and playback to render record surface noise virtually inaudible, while dramatically increasing the dynamic range of the reproduced music signal. The operation of this noise-reduction process applied to discs is illustrated in Fig. 8 where, for purposes of convenience, a master tape signal having a 100-dB dynamic range is assumed to exist.

The dbx linear decibel compression/expansion (companding) system operates as follows. The music signal from the master tape (or directly from a studio console) is encoded (compressed) during the cutting of the master disc and decoded (expanded) during playback. The dynamic range of the music signal is linearly (in dB) compressed by a 2:1 factor when cutting the master disc, which means a music signal having a 90-dB dynamic range is reduced to 45 dB. This fits comfortably within the maximum dynamic range storage capability of vinyl discs. During playback through a decoder, the signal picked up by the phono cartridge is linearly expanded by a 1:2 factor so that the dynamic range of the original music signal is completely restored. And, as a result of downward expansion during decoding, the surface noise on dbx Encoded Discs is approximately 30 dB lower than on conventionally-recorded discs.

The frequency spectrum of record surface noise for a conventional disc and a dbx Encoded Disc is shown in Fig. 9 for a constant 125-Hz bandwidth analysis. Measurements were made on a disc that was cut with an unmodulated groove (no signal). About 30 dB of noise reduction is provided by the dbx Encoded Disc for frequencies below 10 kHz, which encompasses the frequency band of greatest concern relative to record surface noise. The total wide-band noise of -57 dB is reduced to -85 dB, in this particular case, for an overall reduction in surface noise of 28 dB.

There are a number of side benefits to dbx Encoded Discs. During cutting of an encoded master disc, the compressed signal reduces the demands on the cutting stylus, resulting in

Fig. 8 — Diagram depicting the combination of signal compression during encoding and expansion during decoding that results in surface noise reduction and dynamic range retention on dbx Encoded Discs.



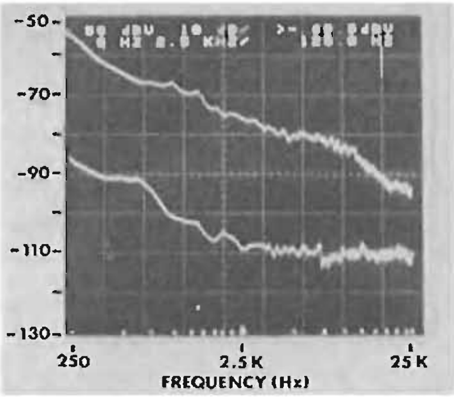


Fig. 9 — Constant 125-Hz bandwidth frequency spectrum of record surface noise, with and without dbx Type II noise reduction, as obtained from a vinyl disc cut with an unmodulated groove (no signal). Noise level is shown relative to 0 dB (7 cm/S peak velocity at 1 kHz) for no noise reduction (upper trace) and for dbx Type II noise reduction (lower trace). The wide-band (20 Hz to 20 kHz) noise level of -57 dB is reduced to -85 dB by the dbx noise-reduction system.

smoother grooves for less distortion. Minor ticks and pops that result from groove surface imperfections, as well as groove echo, are largely eliminated along with the record surface noise. Also, since narrower grooves can be used for the encoded signal, cutting of the master disc can be done further away from the center of the disc, which reduces inner-groove distortion. Finally, since the decoder is connected in the stereo playback system after the phono preamp stage (in the tape monitor loop of a receiver or preamp, for example), turntable rumble and even preamp electronics noise (hum and hiss) for all practical purposes disappear.

dbx Encoded Discs are compatible with conventional record playing equipment, although they require a dbx decoder to be employed during playback. However, they are incompatible with conventional discs since their sound quality is unacceptable without decoding. A decode function has been incorporated in audiophile versions of dbx tape noise-reduction systems for the past six years, so owners of these components are already equipped to play dbx Encoded Discs, and new decode-only units are now also available.

A library of encoded discs is under development, with some record companies remastering a wide range of titles from their catalog in the dbx format. The sound of these records is virtually indistinguishable from that of the master tape from which the record is made. It should be noted that, with the masking effect of record surface noise removed, any imperfections in the master tape (e.g. hiss or other forms of distortion) may become all the more apparent. Hence, by eliminating noise in both the master tape and the recorded disc, the quality of recorded sound can be brought several steps closer to that of a live performance.

Analog tape recording with noise reduction, digital recording, and direct-to-disc recording concepts, when combined with dbx Encoded Disc technology, can produce extraordinary sonic results. With their ability to reproduce music having the dynamic range of a live performance against a background of silence, dbx Encoded Discs reprsent a very significant breakthrough in recorded disc technology.

Summary and Conclusions

Recognition of the major acoustic factors that characterize live-performance sound leads to the conclusion that dynamic range is an extremely important consideration relative to providing realistic music reproduction in the home. Our ability to recreate the excitement and emotional impact of a live performance is enhanced by employing recording and reproduction systems capable of handling up to a 90-dB dynamic range. The weakest link in the music system "chain" is the software employed — records and tapes — which, until recently, has been saddled with severe dynamic range restrictions. Records, prerecorded tapes, and broadcasts have sounded lifeless in comparison to live performance because they have been missing more than a third of the original dynamic range of the music.

The human auditory process includes the ability to partially ignore the distraction of background noise in recordings and broadcasts. Having trained ourselves to listen to recorded music with this "psychoacoustic filter" operative, we are pleasantly surprised (if not startled) when we hear recordings in which background noise (such as tape hiss and record surface noise) is absent. The auditory mechanism that preconditions us to accept (or at least live with) the degradation of sound quality caused by the existence of noise is suddenly disturbed, and our emotional reaction is one of excitement and total involvement.

It is now possible to come substantially closer to the long sought after objective of enjoying music reproduced with sound quality that closely approximates the live performance experience. In addition to being able to use dynamic range expanders to increase the realism of existing recordings, advanced recording techniques, which include analog tape recording with noise reduction, direct-to-disc, and digital recording, are now available which allow records to be mastered that, on playback, have a dynamic range approaching 90 dB. New recorded disc technology, involving the dbx Encoded Disc concept, provides a practical and affordable means of enjoying this full dynamic range from a vinyl disc played on conventional record playing equipment in a stereo system that incorporates a dbx disc decoder. While a totally digital audio system theoretically should provide as good or better performance, there undoubtedly will be a substantial cost involved in both recording and playback equipment, and its wide-spread availability may be a number of years off. Recorded music with full dynamic range against a background of virtual silence is available to everyone now, providing a dramatically increased music listening experience.

Acknowledgements

The author wishes to thank Mr. Les Tyler, dbx Chief Engineer, for conducting laboratory experiments that resulted in the tape and disc noise measurements contained herein. The measurements were made using a Model 7530A spectrum analyzer manufactured by the Rockland Systems Corp. A

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Test Tape for Checking Head Position

C. G. McPROUD

Changing heads, replacing one type with another, or any other activity involving the moving of the heads on a tape recorder points up a need for a test tape to ensure proper positioning.

A FEW MONTHS AGO, the author had occasion to change a head on an Uher 4000-S tape recorder (AUDIO, October, 1965) and needed some sort of alignment-indicating tape to make sure that the new head, a Nortronics 1202, was in the correct location. Such a tape is commercially available from RCA under the designation 12-5-64T, and can be obtained from RCA Victor Record Division, 6800 E. 30th St., Indianapolis, Ind., we have since learned, but at the time we were unaware of it.

However, in talks with our associate, Larry Zide, we came up with a tape which most anyone with two properly head-positioned tape recorders can make for himself, provided one of the recorders is full-track and the other quarter track, with three heads.

The whole idea is borrowed from movie practice, in which an optical track was used for the same purpose. This track consisted of an area which was unmodulated in the proper position for the sound projector light slit, and the remainder of the track was modulated with a "buzz," essentially a low-frequency square wave. When the film was played on a properly aligned projector, nothing was heard. If there was any misalignment, you heard the buzz.

We decided upon a track which would have 1000-Hz modulation on the proper track positions, and a 150-Hz square wave on the balance of the tape. To produce such a tape, we first recorded the 150-Hz square wave on a full-track machine, thus giving us the "buzz"-track background. The next step was to record 1000 Hz on both channels of a four-track stereo recorder, with the normal erase head "cleaning off" the 150-Hz square wave prior to the recording of the 1000 Hz.

While this did exactly what we thought it would, we had overlooked the fact that the erase-head track is about 8 mils wider than the recorded track, so it left a clear space on each side of the 1000 Hz, as shown in *Fig.* 1, which is a simulated representation of the final result. This resulted in a playback of the 1000 Hz in a satisfactory manner, but permitted a slight deviation from the correct position before the error became noticeable. Obviously, the tape could weave some 4 mils either way before the background buzz was heard.

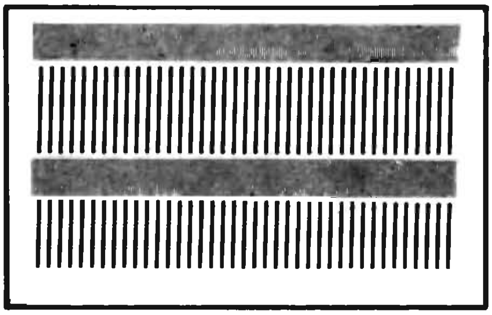


Fig. 1. First attempt to prepare the test tape resulted in tracks of 1000 Hz with a small clear space between them and the "background" of the 150-Hz square wave pattern.

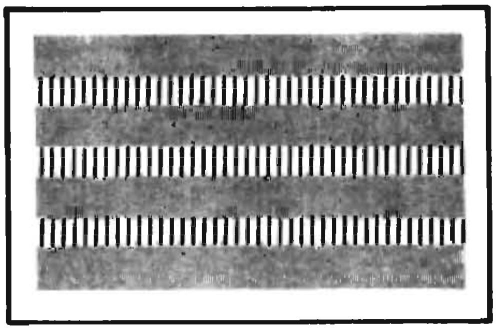


Fig. 2. By using the record head as an erase head for one pass, and then actually recording 1000 Hz with the same head on the second pass, the spaces between the tone and the buzz are eliminated. Needless to say, the erase head was not energized to make this recording. Then, reversing the reels and repeating the performance gave a four-track 1000-Hz tone, with the buzz background occupying only the island spaces.

The next step, logically, was to disconnect the erase head and connect the record head to the erase oscillator, using the record head as the erase head and thus cleaning off *only* the area properly occupied by the desired track position. A second pass through the machine with the record head connected in the normal manner and the erase head still disconnected gave a similar result to that in *Fig.* 1 but without the spaces between the 1000 Hz and the 150-Hz buzz.

Reversing the tape and making two more passes through the machine, repeating the last two steps, will provide the result shown in *Fig.* 2.

Thus we have a test tape which will reproduce only 1000 Hz when the heads are in the correct position, but any deviation will show up as a buzz superimposed over the 1000 Hz.

Modifications

While it may be argued that for the single purpose of track positioning the test tape could have accomplished the same objective if we had only erased the 150-Hz buzz track, with the record head serving as the erase head, and leaving the track position completely blank. This would undoubtedly have been preferable since it would be easier to hear the buzz in the absence of the 1000-Hz tone. However, this would have eliminated a second benefit of the test tape—that of checking the balance between the two tracks and the associated amplifiers.

The tape described offers the experimenter a simple and workable means of checking the head position after a change, and can be made on any standard four-track recorder in which the erase head may be disconnected and the record head used temporarily as the erase head.

In this latter connection, it is likely that the erase/bias oscillator may not furnish sufficient bias to the record head to erase the buzz without some adjustment, but some recorders have this facility. It is usual for the record head to have higher inductance than the erase head, and more erase current may be required than the erase oscillator can furnish. This can be remedied by interposing an amplifier between the oscillator and the record head. Then monitor on the play head and gradually increase the erase current until the track is silent. This does require a machine with three heads, but the typically ingenious tape enthusiast should have no trouble in obtaining the desired result. Æ

build your own

Solid-State Flutter Meter

BY: ARTHUR E. GLADFELTER*

This article describes an instrument capable of measuring the Flutter and Wow in sound recorders. Construction, calibration, and method of flutter measurement are included. Measured flutter data from several tape recorders are compared with the manufacturer's specifications.

In Four Parts

Part 1—How it works—and what it does

OBTAINING LOW FLUTTER AND wow in a tape recorder is just as important as obtaining low distortion, wide frequency response and a good signal-to-noise ratio. There is not a single element associated with the tape drive that cannot introduce flutter. If you doubt this, record a frequency of about 2 to 3 kHz on the best audio recorder you can obtain. It will be very obvious during a listening A-B test; the signal being played back has some "wiggles" and "garble" the original did not contain. The disturbance, or flutter, that is perceived is actually frequency modulation of the input signal. The modulation is caused by minute variations in the tape speed.

Any mechanical element within the tape transport that is associated with tape motion can cause flutter. Some of the more common causes of flutter include:

1. Eccentricities and mechanical unbalance of reels, rotating guides, capstan, pressure roller, pulleys, and bearings.
2. Irregularities in the surface and variation in compliance of tape, belts, and rubber rims.
3. Changes in motor torque caused directly by the motor and indirectly by variations in power-line frequency and voltage.
4. Variation in friction; such as tape-to-head friction, variation in bearing torque, and so on.
5. Cohesion between successive tape layers and irregularities in width of tape.

Why should a flutter measurement be made when the manufacturer states, for example, it is less than 0.1 per cent?

Senior Electrical Engineer, The Bendix Corporation, York Division, York, Pennsylvania

Front view of the author's flutter meter.

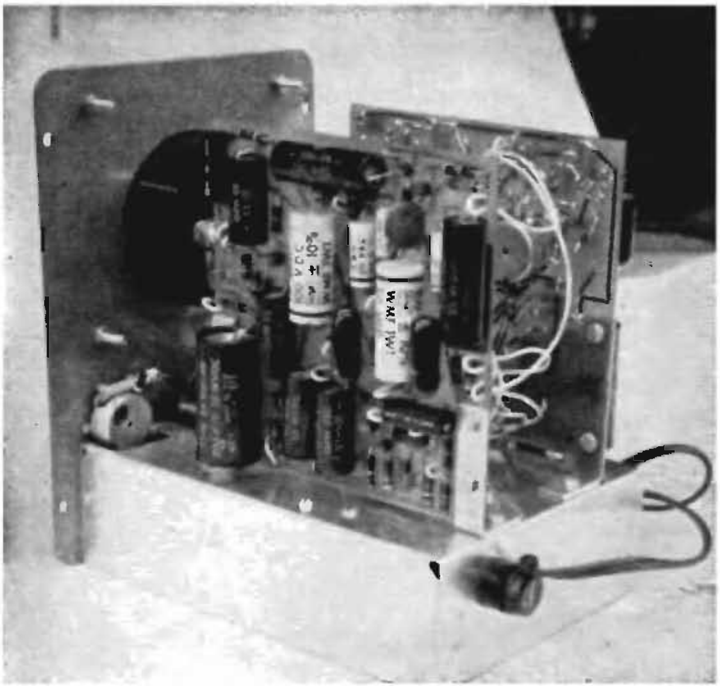
From a reputable firm, this figure would probably be correct—at least when the recorder was factory fresh. But what is the flutter after several thousand hours of operation, or does the flutter change after the machine has been running for several hours and becomes warm? How often should the iron oxide, dust, dirt, and the like be cleaned from various elements? How does flutter vary from start to end of a reel? Answers to these questions can be obtained from accurate flutter measurements. In addition, various mechanical adjustments (hold-back tension, takeup tension, pressure pads, and so on) can be made in an attempt to obtain minimum flutter.

Definition Of Flutter

Per cent of flutter is defined as a ratio of the root-mean-square deviation in frequency of the tone to the mean frequency of the tone.[1] If the deviation is in a sinusoidal pattern, the per cent flutter, f_k, is:

$$f_k = \frac{(100)\ (d\ f_o)}{\sqrt{2}\ \ (f_{avg})} \qquad (Eq.\ 1)$$

Where df_o is the maximum deviation from the average frequency, f_{avg}. The flutter rate is the number of times per second the frequency deviates. Each complete excursion from maximum to minimum and back again comprises one cycle. A flutter rate of about 1 to 5 H_z



Rear view of the completed flutter meter.

is heard as wow, while above 5 Hz the ear recognizes the variation as flutter.

The instrument used for the measurement of flutter contains a carrier oscillator. The oscillator signal is recorded on the tape via the record circuitry contained within the tape machine. The signal derived from the playback head is processed by an FM receiver that measures the frequncy modulation of the 3000-Hz carrier; that is, the signal passes through a band-pass amplifier, limiter, and discriminator. Following the discriminator, various filters are used to measure the flutter and/or wow spectrums separately. The deviation in frequency and consequent per cent flutter, will correspond to an a.c. voltage, the amplitude of which is measured and displayed by a microammeter. An output jack is provided for observing flutter components with an oscilloscope.

Block Diagram Description

The various front panel functions and basic circuits utilized within the flutter meter can be explained by referring to *Fig.* 1. A stable 3000-Hz tone (f_o in Eq. 1) is generated and applied to the recorder input. The oscillator input (J_3) amplitude is about 0.75 V rms and can be applied to the recorder high-level input. During playback, the recorder output is applied to the flutter meter input. (J_1) Following a level control (R_1) the signal passes through a three-stage band-pass amplifier. In addition to providing sufficient gain to drive the limiter, the amplifier improves the signal-to-noise ratio by providing more rejection to the frequencies on either side of 3 kHz; i.e., any components from 60 Hz on the low end, or from the bias oscillator on the high end. The symbols Q_1, Q_2, etc. on the block diagram refer to the various transistors in the over-all schematic of *Fig.* 2.

The output signal from the 3-kHz band-pass amplifier is attenuated and then applied to the "level set" position on the "range" switch. When in this position the voltmeter monitors the input amplitude. R_1 can then be used to adjust the incoming signal amplitude to a level sufficient to drive the limiter. The limiter, or more correctly a Schmitt trigger circuit that is utilized as a limiter, converts the incoming sine wave to a square wave. The amplitude and symmetry of the limiter output are virtually unaffected by a wide range (at least 30 dB) of input amplitudes, such as might be caused by tape dropout.

The limiter output or discriminator input will be a 3-kHz square wave that is deviating, at the flutter rate, above and below 3 kHz. The discriminator, which is really the "heart" of the system, converts the frequency changes into amplitude variations. This conversion from frequency to amplitude allows conventional voltmeter techniques to be used to display the percentage of flutter. The discriminator output amplitude will increase for frequencies above 3 kHz and decrease for frequencies below 3 kHz.

The discriminator output amplitude for 0.30 per cent flutter will be of the order of 13 to 14 millivolts rms—exact figures will be given in the calibration procedure. As a result of this relatively low level, a low-frequency amplifier with response deliberately limited to about 1000 Hz is used following the discriminator. The band-width is restricted to 1000 Hz to further attenuate the 3 kHz components that are not rejected by the discriminator and also to assure that the amplifier is not saturated by the 3-kHz signal. Of course, the 1000-Hz low-frequency amplifier, with a voltage gain of about 35 dB, could be eliminated and the sensitivity of the voltmeter circuitry increased. However, if this approach had been used, the signal levels passing thru the following low-pass and band-pass filters would be 35 dB lower, with the possibility of hum pickup,—basically because of the larger number of passive components within the filters and also because of the higher impedance levels in the filter than in the 1000-Hz low-pass amplifier. With 35 dB of amplification prior to the filters no shielding is required, nor is the parts layout critical.

The 250-Hz low-pass filter following the 1000-Hz low-frequency amplifier is used to further attenuate the 3-kHz components and to fulfill requirements established by the I.R.E., (now I.E.E.E.), that all flutter rates up to 200 Hz should be considered. It should be mentioned that tape recorders may have flutter-rate disturbances introduced by high-frequency frictional vibration up to and beyond 2 kHz[2]. The upper rate limit of 200 Hz appears to be an outcome of the limit that was used for the 96-Hz potential flutter and its second harmonic (192 Hz) caused by sprocket-holed disturbances in 35-mm film-drive systems. However, to adhere to previously established standards, a 250-Hz low-pass filter is used, to obtain relatively flat response up to 200 Hz.

For the purpose of analyzing the cause of flutter in a recording system, it is often desirable to make measurements in several ranges of flutter-frequency. By knowing the flutter rate, and then comparing this with the known rotational rates of the elements in the drive system, it is easier to correlate a given flutter disturbance with its mechanical cause. This then is the purpose of the 6-Hz low-pass and 6-Hz high-pass filters that are inserted betwen the 250-Hz low-pass filter and the dual emitter-follower-voltmeter. S_1 can then select any one of three bands that cover 0.5 to 6 Hz, 6 to 250 Hz or 0.5 to 250 Hz.

To prevent appreciable loading on the three filters, a dual emitter follower is used to increase the input impedance of the voltmeter circuitry. In addition, the "range" rotary switch S_2 is incorporated in the emitter-follower output circuitry. The voltmeter is a two-stage amplifier, with a microammeter in the feedback network. This will respond to the average voltage (or deviation in frequency); however, the meter is calibrated to read the rms value with a sinusoidal flutter rate. The voltmeter low-frequency response must extend to below 0.5 Hz, thus eliminating many existing rms voltmeters as a substitute indicating device.

The remaining block in *Fig.* 1, the power supply, provides +33 V d.c. at about 53 mA. All blocks, except the filters and attenuators, are connected via decoupling networks to the regulated power supply.

Before leaving the block diagram discussion, one further item should be mentioned. In some systems, a low-pass filter with a cutoff frequency slightly above 3 kHz is inserted between the limiter output and the discriminator input. The purpose of the filter is to remove the odd harmonics (9 kHz, 15 kHz, etc. and so on) from the average 3 kHz or the "carrier" frequency. A low-pass filter was not used because of the inherent filtering from the discriminator tuned circuits and also because of the decreasing amplitude of the higher harmonics. That is, the third harmonic will be reduced 10 dB, the fifth harmonic 14 dB, and so on, relative to the amplitude of the 3-kHz carrier. The over-all system tests, to be described later, justify the omission of a 3-kHz low-pass filter.

Circuit Description and Design Consideration

First consider the 3000-Hz oscillator, Q_{17}, which is of the Colpitts[3] configuration. Feedback from the tank circuit (L_5, C_{33}, and C_{34} in series) is provided by C_{33} and C_{34}. The base bias current is supplied by R_{65} and R_{66}. The Colpitts scheme was selected in preference to a Hartley because of the ease in adjusting the feedback; that is, the feedback fraction could be changed during the design phase merely by selecting C_{33} and C_{34}. In contrast, the Hartley scheme would have required a tapped inductor.

It is not necessary to have the oscillator frequency precisely 3000-Hz. In fact, the frequency can be within a few per cent of 3000 Hz as long as the exact frequency is known and used during the calibration procedure. It is, however, desirable to come as close as possible to 3000 Hz if for no other reasons than to: 1. Conform with existing standards. 2. Simplify the calibration procedure. 3. Assure the fact that the frequency is at the design center frequency of the discriminator, even though d.c. coupling is not maintained in the amplifiers following the discriminator.

Using the values in the schematic diagram of *Fig.* 2, the frequency of oscillation will be about:

$$f = \frac{1}{2\pi\sqrt{\dfrac{L_5 C_{33} C_{34}}{C_{33} + C_{34}}}}$$

$L_5 = 37\text{mH}$

$C_{33} = 0.10\mu\text{F}$

$C_{34} = 0.33\mu\text{F}$

$f = 3.06\ kHz$

The components in the tank circuit have a tolerance of ±10 percent. Even if the tolerance on the tank circuit components is reduced to ±5 percent (or even ±2 percent) the error in the oscillator frequency could be ±5 percent (or even ±2 percent). For this reason, ±10 percent capacitors are used, and L_5 is tailored to obtain exactly 3000 Hz. L_5 is made by winding 2400 turns of No. 36 wire on a 2 watt, 240 k ohm resistor, as shown in *Fig.* 3. Using an accurate frequency detector, such as an electronic counter, turns are removed (about a turn per cycle) until the frequency is exactly 3000 Hz. Of course, a variable inductor can be used, but was ruled out because of the cost *vs.* size compromise for many readily available inductors and, to a lessor extent, because of the change in inductance with temperature.

The frequency stability is almost entirely dependent on changes in the capacitance of C_{33} and C_{34} with temperature. Changes in h_{fe} have virtually no affect on the frequency. For example, decreasing the temperature of Q_{17} to +32° F increased the frequency from 3000 to 3004 Hz. Increasing Q_{17} temperature to +150° F decreased the frequency to 2995 Hz. Similarly, variations in the oscillator d-c supply voltage (although a regulated power supply provides +25 v.d.c at 3.5 mA to the oscillator) from +16 to +28 V d.c., or even variations in the output load (at J_3 to a dead short, provide about 2-Hz change in frequency. While it is not important that the oscillator output waveform be extremely low in distortion, the total harmonic distortion measured 1.4 percent.

The signal from the recorder output is applied, via J_1 and R_1, to a two-stage R-C coupled common-emitter amplifier (Q_1 and Q_2) which is directly coupled to an emitter follower (Q_3). With R_1 in the maximum-gain position, the input

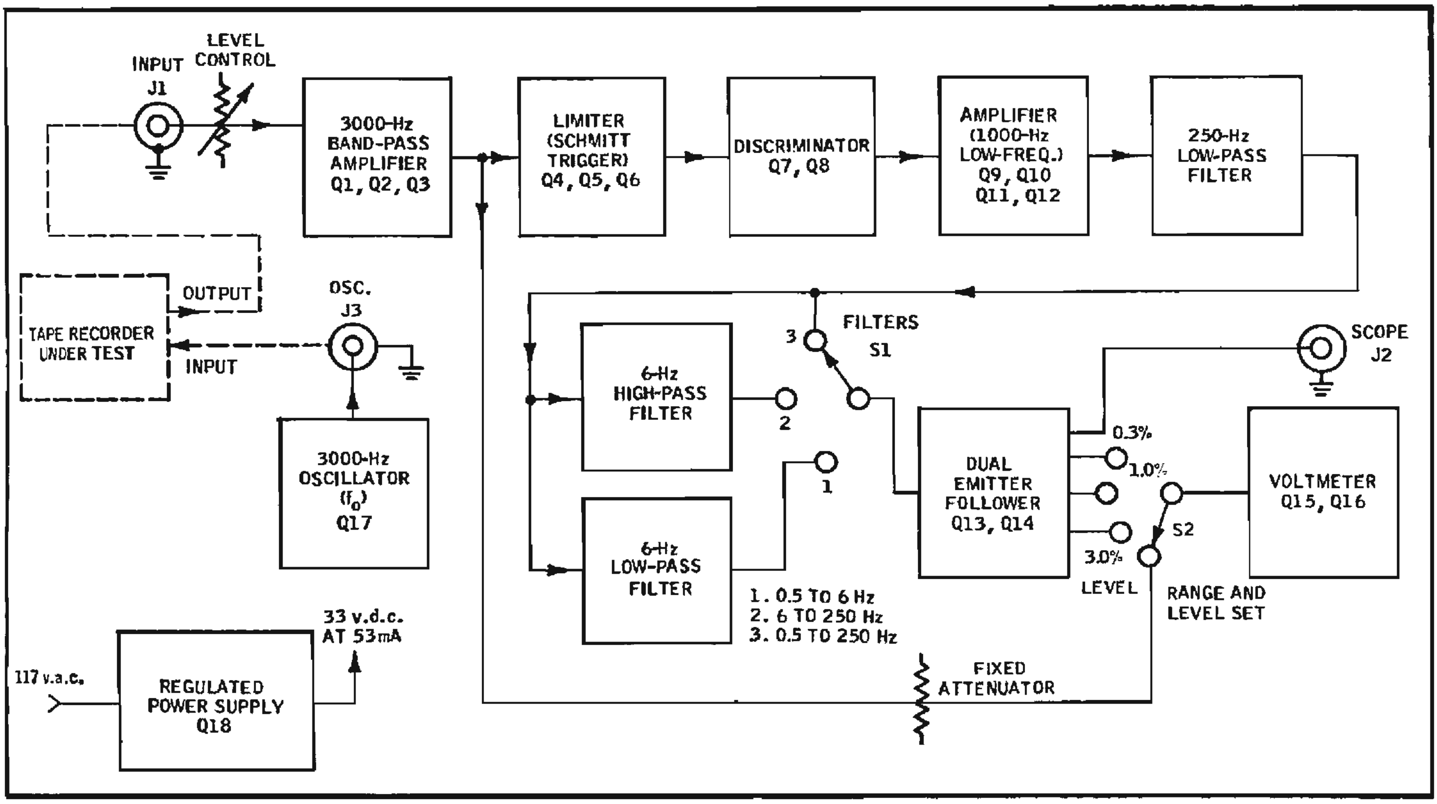


Fig. 1. Block diagram of the author's flutter meter.

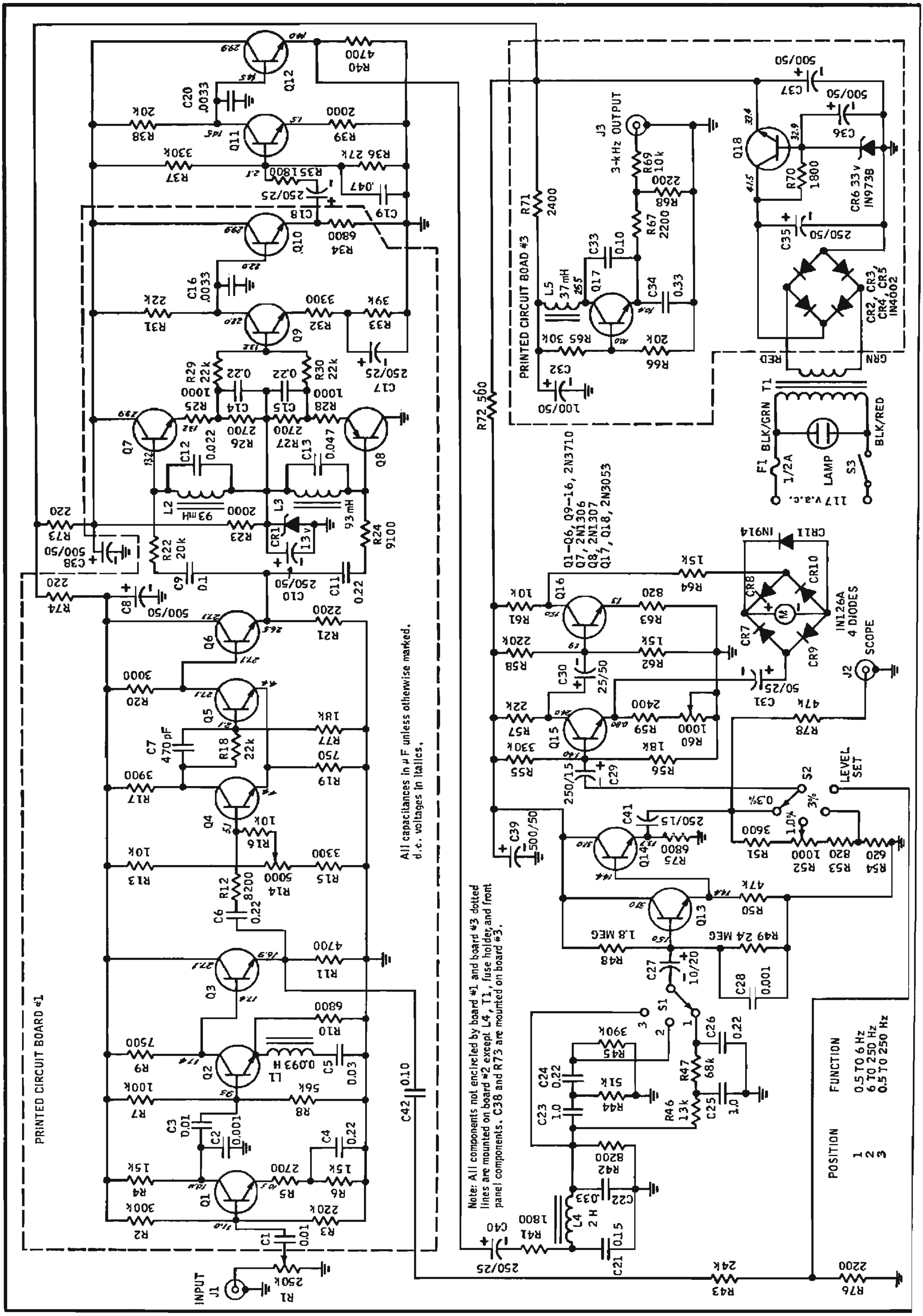


Fig. 2. Complete schematic of the flutter meter and its self-contained power supply.

The Parts List

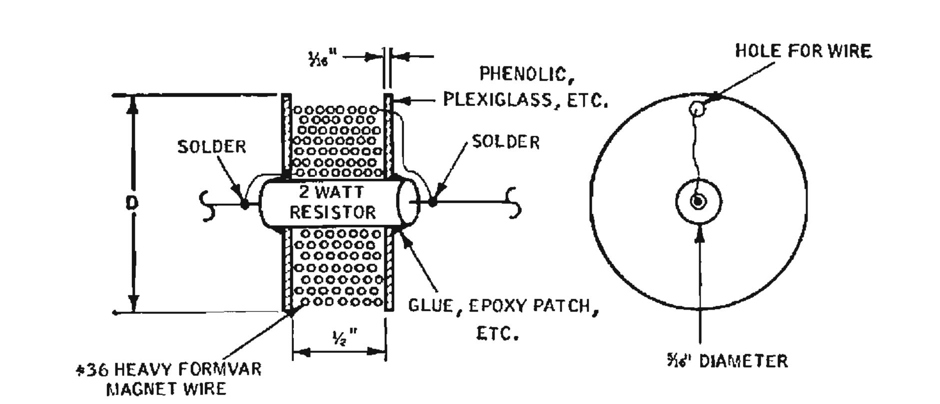
Recognizing the interest this project may instill in our readers, and yet being unable to include the entire article in one issue, we are furnishing the complete parts list with this installment so that readers may begin to collect all the required material in readiness for the concluding installments.

PARTS LIST

Fixed Resistors—½-watt, 5 per cent

R_2	300 k
R_3, R_{68}	220 k
R_4, R_6, R_{62}, R_{64}	15 k
R_5, R_{26}, R_{27}	2700
R_7	100 k
R_8	56 k
R_9	7500
R_{10}, R_{34}, R_{75}	6800
R_{11}, R_{40}	4700
R_{12}, R_{42}	8200
R_{13}, R_{16}, R_{61}, R_{69}	10 k
R_{15}, R_{52}	3300
R_{17}	3900
R_{18}, R_{29}, R_{30}, R_{31}, R_{57}	22 k
R_{19}	750
R_{20}	3000
R_{21}, R_{67}, R_{68}, R_{76}	2200
R_{22}, R_{38}, R_{66}	20 k
R_{23}, R_{39}	2000
R_{24}	9100
R_{25}, R_{28}	1000
R_{33}	39 k
R_{35}, R_{41}, R_{70}	1800
R_{36}	27 k
R_{37}, R_{65}	330 k
R_{43}	24 k
R_{44}	51 k
R_{45}	390 k
R_{46}	13 k
R_{47}	68 k
R_{48}	1.8 Meg
R_{49}	2.4 Meg
R_{50}, R_{78}	47 k
R_{51}	3600
R_{53}, R_{63}	820
R_{54}	620
R_{56}, R_{77}	18 k
R_{58}, R_{71}	2400
R_{65}	30 k
R_{72}	560
R_{73}, R_{74}	220
Variable Resistors	
R_1	250 k, linear. Mallory U-46
R_{14}	5000, linear. Mallory U-14
R_{61}, R_{60}	1000 linear. Bourns E-Z Trim, 3067-P
Fixed Resistors, 2 watt, 10%	
3	270 k (for coil forms)

Capacitors	
C_1, C_3	.01 μF, ±20%, disc ceramic, Sprague 5GABS10
C_2, C_{28}	.001 μF, ±10%, disc ceramic, Sprague 10TSD10
C_4, C_6, C_{11}, C_{14}, C_{15}, C_{24}, C_{26}	0.22 μF, ±10%, 100V. Sprague 192P22492
C_5	.03 μF, ±10%, 100V. Sprague 65P30352
C_7	470 pF, ±10% Sprague 192P47192
C_8, C_{37}, C_{38}, C_{39}	500 μF, 50 V. Sprague TVA1315
C_9, C_{33}, C_{42}	0.1 μF, 100 V. ±10%. Sprague 192P10492
C_{10}, C_{17}, C_{18}, C_{40}	250 μF, 25 V. Sprague TVA1208
C_{12}	.022 μF, ±10%, 100 V. Sprague 192P22392
C_{13}, C_{19}	.047 μF, ±10%, 100 V. Sprague 192P47392
C_{16}, C_{20}	.0033 μF, ±10%, 100 V. Sprague 192P33292
C_{21}	0.15 μF, ±10%, 100 V. Sprague 192P15492
C_{22}	.033 μF, ±10%, 100 V. Sprague 192P33392
C_{23}, C_{25}	1.0 μF, ±10%, 100 V. Sprague 118P10502S2
C_{27}	10 μF, ±10%, 20 V. Tantalum; Sprague 109D106X9025C2
C_{29}, C_{41}	250 μF, 15V. Sprague TE-1138
C_{30}	20 μF, 50V. Sprague TE1305
C_{31}	50 μF, 25 V. Sprague TE1209
C_{32}	100 μF, 50 V. Sprague TVA1310
C_{34}	0.33 μF, ±10%, 100 V. Sprague 118P33492S2
C_{35}, C_{36}	250 μF, 50 V. Sprague TVA1312
Transistors	
Q_7	2N1306 RCA, TI
Q_8	2N1307 RCA, TI
Q_{17}, Q_{18}	2N3053 RCA
All others (14)	2N3710 TI
Diodes	
CR_1	13 V. ±5% 400 mW Zener, 1N964B
CR_2, CR_3, CR_4, CR_5	100-V, 1.0A. Motorola 1N4002
CR_6	33V. ±5% 400 mW Zener, 1N973B
CR_7, CR_8, CR_9, CR_{10}	1N126A
CR_{11}	1N914
Inductors	
L_1, L_2, L_3	93 mH, 210 ohms (see text)
L_4	2 H, 170 ohms max (see text)
L_5	37 mH, 130 ohms (see text)
Miscellaneous	
S_1	1-pole, 3-pos. rotary switch, Centralab 1461
S_2	2-pole, 5-pos. rotary switch, Centralab PA-1002
S_3	SPST toggle switch
T_1	Transformer, 32-V. sec, 100 mA. Allied Radio, 64-U-732
Lamp	Drake Type 6073-534 "Glo-Lite"
Meter	100 μA, Simpson Model 1329
Chassis	5" x 7" x 2" Bud AC-402
Cabinet	8" H x 6⅜" W x 8" D Bud WA-1540
J_1, J_2, J_3	Phono jacks, Switchcraft Type 11
Misc: Knobs 3	Harry Davies 1920-C
Heat Sink (Q_{18}) 1	Wakefield Type NF205 (27¢)
Printed Circuit Boards	1/16" x 5" x 5" (2 required)
	1/16" x 4¾" x 3¼" (1 required)
Printed Circuit Tape Resist,	1/16" wide



Circuit Symbol	Inductance Nom. mH	d.c. resis. (Ohms)	Diameter (inches)	Approx. no. turns	Wire size
L1, L2, L3	93	220	1.0	3400	#36
L5	37	130	0.781	2400	#36

Note: A 240 k (minimum) 2-watt resistor is used as a coil form. The inductance of L5 will be about 37 mH with 2300 turns. 2400 turns are wound, and then turns are removed as described in text.

Fig. 3. Coil winding data.

impedance is 60k ohms. At 3 kHz, Q_1 has a voltage gain of 10 dB and Q_2 a gain of 28 dB. L_1 and C_4 form a series-resonant circuit in the emitter of Q_2 and provide maximum gain at 3 kHz. The emitter follows (Q_3) provides a low driving impedance for the Schmitt trigger (Q_4 and Q_5) and also the resistive attentuator R_{13} and R_{16}. The response of the band-pass amplifier is shown in *Fig.* 4. The rejection is 25 dB at 80 kHz and about 43 dB at 120 Hz.

The Schmitt trigger[4] (Q_4 and Q_5) is a regenerative bistable circuit, the state of which depends on the amplitude of the input voltage. To explain the circuit operation, assume that no base bias is supplied to Q_4 from R_{15}. Q_4 would then be OFF and Q_5 would be biased ON by the base current through R_{17} and R_{18}. The voltage at the emitters of both transistors is then determined mainly by R_{19} and R_{20}, and is about 5.5 V d.c. If the input level to Q_4 base rises above 5.5 V d.c. plus the base-to-emitter drop of 0.6 V d.c., or about 6.1 V d.c., Q_4 begins to conduct and regeneratively turns OFF Q_5. Q_4 *is* then ON, and the voltage at the emitters is now about 4.5 V d.c. and is less than the initial value of 5.5 V d.c.—because R_{17} is larger than R_{20}. When the input voltage is reduced to 4.5 V d.c. plus the base-to-emitter drop, or about 5.1 V d.c., Q_5 will again conduct. During the transition period, the change in Q_5 collector voltage will be about 19 volts. Potentiometer R_{14} which selects the trip point on Q_4, is adjusted to provide a symmetrical square wave at the limiter output emitter follower, Q_6.

The limiter output is coupled to the discriminator[5] input by the d.c. blocking capacitors C_9 and C_{11} L_2, and C_{12}, connected to base of Q_7, have a parallel resonance of 3520 Hz and the waveform at the base of Q_7 (which is very nearly sinusoidal) will have maximum amplitude when the incoming frequency is 3520 Hz. Q_7, which is nonconducting in the absence of a signal, is connected as an emitter-following-detector and will rectify the positive portion of Q_7 base waveform and provide a voltage across R_{26} and C_{11}. The voltage will be a maximum at 3520 Hz and will decrease as the frequency is decreased below 3520 Hz. The low side of R_{26} and C_{11} are clamped to a fixed voltage of +13 V d.c. by zener diode CR_1. For the incoming signal, the common tie point of R_{26}, R_{27}, CR_1 cathode, and so on, can be considered as a signal ground, with the output signal being coupled by R_{29} to the base of Q_9.

L_3 and L_{13}, connected to the base of Q_8, have a resonance frequency of 2420 Hz and will detect the negative portion of the incoming waveform. This will provide a voltage drop across R_{27} and C_{13} that will decrease as the frequency is increased above 2420 Hz. The outputs from the two detectors are then combined by the two summing resistors, R_{29} and R_{30}, and applied to the base of Q_9. The over-all discriminator response, for an input squarewave of 19 volts peak-to-peak, is shown in *Fig.* 5. Note the output voltage at the base of Q_9 represents a change in voltage, relative to the reference voltage of about 13 V d.c., and does not indicate an absolute voltage.

Q_9 through Q_{11} comprise a low-frequency amplifier with a voltage gain of 35 dB. The bandwidth extends from about 0.2 Hz to 1000 Hz, with the high-frequency response being deliberately limited by C_{16}, C_{19} and C_{20}. The upper cutoff frequency of 1000 Hz was chosen in order to have virtually flat response to 250 Hz. At 250 Hz the voltage gain is 34.7 dB, or a decrease of 0.3 dB. At 3000 Hz the voltage gain is 22 dB. To prevent loading on the discriminator output, the impedance looking into the base of Q_9 is typically 450 k ohms, with a "worst" case minimum value of 300 k ohms. The base bias voltage for Q_9 is derived from zener diode CR_1. The direct coupling of stages from CR_1 anode to the emitter of Q_{10} may, at a first glance, appear to be unstable with variations in the zener voltage of CR_1 and changes in temperature. This, however, is not the case because the d.c. voltage gain of stages Q_9 and Q_{10} is less than unity, and any change in the voltage at the anode of CR_1 will appear as a smaller change at the emitter of Q_{10}. The emitter-follower Q_{11}, provides a relatively low output impedance (about 150 ohms) for the following 250-Hz low-pass filter.

The 250-Hz low-pass filter[6] has three poles and is terminated by R_{41}. Although the filter has been designated as a 250-Hz filter, the actual cutoff frequency was designed to be 600 Hz—the reason being that when the 35-dB amplifier and low-pass filter are combined, the over-all response will have a cutoff frequency of about 250 Hz. The frequency response of the combined low-frequency

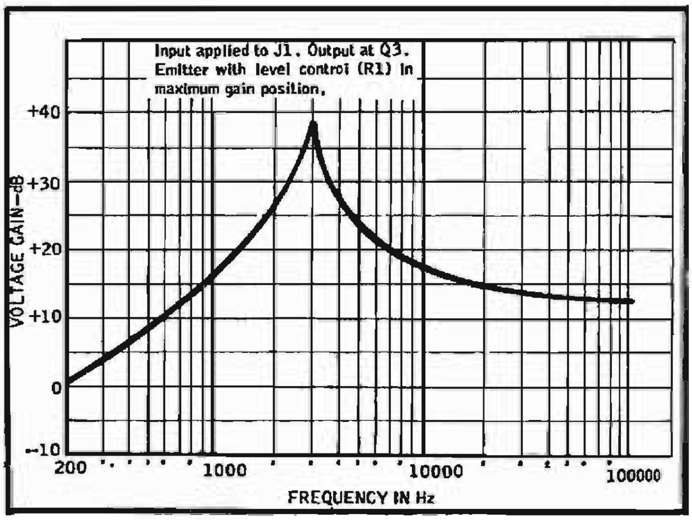


Fig. 4. Response of the band-pass filter.

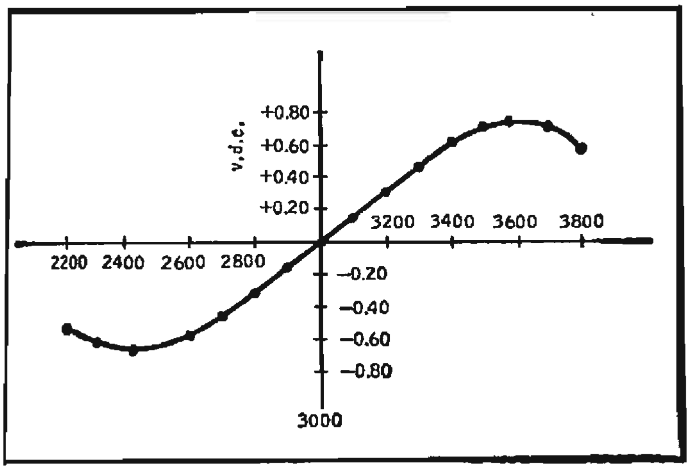


Fig. 5. Response of the discriminator circuit.

amplifier and low-pass filter is shown in Fig. 6, which represents the response from the base of Q_9 to R_{42}. The over-all gain is less than 35 dB because of the insertion loss of the low-pass filter.

The 6-Hz low-pass filter and 6-Hz high-pass filter consist of the R-C components between R_{42} and positions 1 and 2 on S_2. Each is a two-pole filter and has an ultimate slope above (or below) the cutoff frequency of 12 dB/octave. Measured data from the filters are shown in *Fig.* 7. In addition, the relative response from *Fig.* 6 has been replotted on *Fig.* 7. Thus, a glance at Fig. 7 will readily indicate the response for any one of the three positions on the flutter rate (or filters) selector, S_2.

The voltmeter or indicating circuitry consists of stages Q_{13} to Q_{16}. Q_{13} and Q_{14} provide an input resistance of about 130k ohms. C_{15}, connected to the emitter of Q_{14}, removes the d.c. voltage from the range resistors, R_{51} thru R_{55}. C_{15} (and R_{51}) could be eliminated; however, this will result in a much longer meter-settling-time, when switching to various positions on the range switch.

Q_{15} and Q_{16} comprise a conventional two-stage R-C coupled common-emitter amplifier, with an open loop voltage gain of about 25 dB. Feedback is applied from the collector of Q_{16} to the emitter Q_{15} via the full-wave bridge and microammeter. The closed loop voltage gain is about 15 dB, with the exact value determined by the setting of R_{60}, which is used to calibrate the 0.30 percent range and R_{58} the 1.0 percent range. No calibration is provided for the 3.0 percent range; however, the error should be no more than 5 percent on the full-scale reading—the tolerance on the resistors. If the percentage of flutter is high enough to warrant use of the 3.0 percent range, a 5 percent error should be of little concern. CR_{11}, in parallel with the meter, limits the maximum meter current to about 300 microamperes, or three times the full-scale value of 100 microamperes. Typically, the input sensitivity at the base of Q_{13}, is about 0.5 V rms required for full-scale deflection.

The power supply consists of a full-wave bridge rectifier followed by a series regulator. Taps are selected on power transformer T_2 to provide a nominal output voltage of 32 V rms (34 V rms measured). After rectification, the average voltage at the collector of Q_{18} will be about 40 V d.c.-with an input voltage of 115 V a.c. Q_{18} is merely an emitter follower, with the base voltage determined by the zener voltage (33 V d.c.) of CR_6. The d.c. output voltage, due to the base-to-emitter drop in Q_{18}, will be about 32.5 V d.c.

Using the 0.30 percent flutter range and 115 V a.c. as a reference, an increase in the line voltage to 125 V a.c. will cause less than a 1 percent increase (or error) in the percentage of flutter. Reducing the voltage to 105 V a.c. will cause a decrease of about 0.6 percent and 100 V rms will yield a decrease of about 2 percent; i.e., the meter will read 0.294 percent for 0.30 percent.

Packaging Concept

To make a professional looking unit, a BUD "Portacab" meter case was selected to house all components. The indicator, a Simpson 4½-inch Wide-Vue meter, was chosen for accuracy, ease of reading, and compatibility with the Portacab case. The front panel controls and input connections were positioned as functionally as possible. For example, all input-output jacks are mounted at the bottom of the front panel and below the three controls. With this arrangement, the various connecting jacks and wires will be less likely to obscure the front-panel controls.

All electrical components, except those on the front panel, L_4 and T_2 are mounted on printed-circuit boards, which are attached to a basic 5x7x2-in. aluminum chassis. The front panel and chassis are then fastened together to form an assembly which can be slid easily into the shell of the Portacab meter case.

The design objective for the internal layout was to have easy access (for test purposes) to all electrical junctions. This is achieved by mounting two printed circuit boards vertically and with the components facing outward. The third board is mounted horizontally with the components facing the open side of the chassis.

The first board contains the band-pass amplifier, Schmitt trigger, discriminator, and a portion of the low-frequency amplifier. On the second board are the last stages of the low-frequency amplifier, filters, and voltmeter circuitry. The third board contains the 3-kHz oscillator and regulated power supply. Printed circuit boards No. 1 and No. 2, and L_4 are mounted vertically (with angle brackets) on the top of the basic chassis. The third board and power transformer are mounted underneath the chassis. The interconnections between boards (and the front panel) are made with No. 20 or No. 22 stranded wire. The input lead and 3-kHz oscillator output are, however, routed with RG-174/U coaxial cable.

Photographs of the unit are shown in *Figs.* 8 and 9.

Parts Considerations

A list of the materials required to make one unit is given in the PARTS

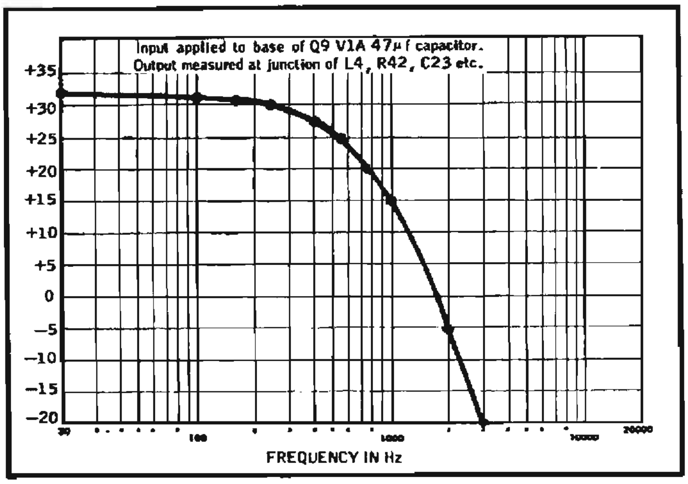


Fig. 6. Frequency response of the low-frequency amplifier and the low-pass filter.

LIST. All parts except the inductors, are off-the-shelf items, and can be purchased from Allied Radio and Newark Electronics. (Neither firm lists all the parts in their catalogs.)

L_1, L_2, L_3 and L_5 are fabricated, and L_4 can be purchased from Barry Electronics, 512 Broadway, New York, N.Y. 10012. L_4 is a two-henry inductor described by Barry as an FTR Mini-Choke, catalog No. 11-38, and is priced at 40¢. The choke, obviously a surplus part is a staple item and has been in the Barry catalogs for several years. A commercial equivalent can, of course, be used. A UTC VIC-13 variable inductor will work very well, although the price is \$7.20 and it would have to be adjusted so the inductance was 2 Hy. An alternate 2-Hy fixed inductor would be the UTC MQB-7 priced at \$14.70.

The cost of parts, for one unit, will be about \$99. Of this figure, the meter and cabinet cost about \$27, capacitors \$24, resistors and pots \$15, transistors \$11, diodes \$8 and so on.

At a first glance the cost of parts may appear high. But remember, only one bargain part, L_4, has been used. Besides, one commercially available vacuum-tube flutter meter sells for about \$500. The unit was designed for performance and not to be sold for a particular price. The cost also reflects the use of 5 percent resistors, compared to 10 percent resistors that are often used. The use of 5 percent resistors increases the cost about \$4.50. After considering the initial tolerance and the effects of temperature and aging on resistors, it is felt the additional cost can be justified easily.

All transistors, except Q_7 and Q_8; are NPN planar silicon. The 2N3710, made by Texas Instruments, was selected because of the price (49¢ each) and size. It has a plastic enclosure, a minimum h_{fe} of 90 and a maximum collector-to-emitter voltage of 30 V d.c. Because the 2N3710 is relatively new and has a different base configuration, the base connection is shown in *Fig.* 9. The 2N3053, made by RCA, was selected because of the cost *vs.* performance. Other transistors that can be used, for Q_{17} and Q_{18}, but at greater cost, are the 2N2102 or the 2N2270. An old standby, the 2N697 can also be used, but because of the type of construction, the dissipation of the 2N697 is less than the three previously described transistors.

During the early design phase it was not known exactly what the limiter peak-to-peak amplitude would be. For this reason germanium transistors were selected for Q_7 and Q_8, because of the lower base-to-emitter voltage in comparison to silicon transistors. (At the time, a delivery problem was also encountered with the silicon complement of the 2N3710, the 2N3702). As it turned out, the limiter amplitude is large, about 19 volts peak-to-peak, in comparison to the difference in base-to-emitter voltage of germanium and silicon transistors (0.2 volts *vs.* 0.6 volt). If the reader wishes to make the flutter meter an all-silicon unit, the 2N3702 may be substituted for the 2N1307, and the 2N3710 for the 2N1306, providing the printed circuit board layout is modified.

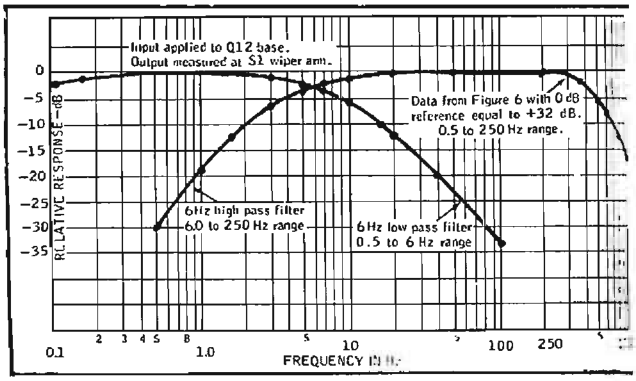


Fig. 7. Frequency response of both filters.

All capacitors in the parts list except the disc ceramics, are called out by a manufacturer's part number.

With many solid-state devices, there are vast differences in price. This is the case of zener diodes CR_4 and CR_6. Of the firms making readily available zener diodes, International Rectifier appears to have the lowest prices on the 1N964B and 1N973B.

Winding the Coils

Data for winding L_1, L_2, L_3 and L_5 are given in *Fig.* 3. A 2-watt resistor, with a minimum resistance of 240k ohms, is used as a coil form. The sides of the coil form are circular discs made from an insulating material such as Bakelite, plexiglass, or any phenolic material. These discs are fastened to the resistor using an epoxy adhesive, such as Eocobond 26. Small holes are then drilled in the sides of the coil form. The wire is then passed through the holes and soldered to the resistor leads with the solder joints close to the body of the resistor.

The coils can be made with the aid of a ¼-inch electric drill and a Variac to control the speed. The coil form is secured to the drill by putting one end of the resistor lead into and firmly against the end of the drill chuck. It is also helpful to place a small washer between the drill chuck and the body of the resistor.

In the absence of a suitable counter to indicate the number of turns, there are many ways the builder can wind the approximate number of turns. Obviously, turns can be added and the inductance or d.c. resistance measured until the values shown in the table of *Fig.* 3 are obtained. Alternately, tur can be added and the reasonance fr quency of an L-C network can b measured. That is, L_1 can be tailor with C_5 (0.03 uf) So the resonan frequency is 3 kHz; L_2 can be tailor with C_{12} (0.022 uf) so the resonan frequency is 3520 Hz; and L_3 can be tailored with C_{13} (0.47 μf) so the ne work frequency is about 2420 Hz.

Two of the frequencies just give 2420 Hz and 3520 Hz, are the desig center values for the discriminato Even if the inductance values specifi in the table of *Fig.* 3 are obtained e actly, the network resonance freque can be in error by about ±5 per ce because of the ±10 per cent toleran on the capacitors. Variations in the d criminator network frequency up and even slightly above 5 per cent w have virtually no effect on the final pe formance.

Making the Printed Circuit Boards

The printed-circuit board layouts fo all three boards are shown in *Figs.* 1 11 and 12. The board layouts are r produced full scale, so that no inte mediate steps are required to red the size. Boards 1 and 2 are 5 in. squa and board 3 is 3¾ by 4¾ in. Th measurements are also specified in t parts list and can be verified by mea urement of *Figs.* 10 through 12.

To show how the printed-circ boards are made, board No. 3 will used as an example. This is the easi board to make and contains the le number of components. After the boa has been cut to size, the copper cl side should be cleaned to remove a dirt or film that has accumulated. Clea ing is most easily accomplished wi crocus cloth. Ajax cleanser or a lar eraser. The board is then laid on t workbench or flat surface, with t copper side up. SCOTCH or maski tape is used on alternate corners t hold the board in position. The sketc of board No. 3 (*Fig.* 12) is then plac squarely on top of the printed-circu board. A center punch, with a very fi tip is used to locate all holes. Alte nately, if you do not wish to have

(*Continued on page* 5

FLUTTER METER

(*from page 26*)

series of small "pinholes" on the magazine pages, *Figs.* 10 through 12 must be retraced, and the tracing used as a template to locate the holes. The various connections of *Fig.* 12 are then made by placing 1/16 in. printed-circuit tape resist on the printed-circuit board.

The two most common etching solutions for printed circuit boards are ferric chloride and ammonium persulfate. Ferric chloride is normally bought in solution form, and one pint is more than enough to etch all three boards. Ammonium persulfate is a powder, and to make a solution, 2.5 pounds of Ammonium Persulfate are added to one gallon of water. In addition, a minute amount of mercuric chloride (a poisonous powder) is added, as a catalyst, in the amount of 26.7 milligrams of mercuric chloride to one gallon of water. Both of the etching solutions just described have their advantages and disadvantages. Rather than elaborate more, I suggest those who are undecided use the ferric chloride solution.

A plastic or glass tray can be used to hold the solution to etch the boards. The boards are then submersed in the solution, and to speed up the etching process, the tray can be heated (to about 160° F) and agitated. The time to etch a board will vary, but with heat and agitation, the time will range from about 15 to 30 minutes. After etching, the boards should be washed with water and the printed circuit tape resist removed. The boards can then be cleaned (as described previously) and appropriate holes drilled for the various components.

When soldering the components to the boards it is a good idea to heat-sink the 1N126 germanium diodes and also the resistors associated with the resistive divider in the range switch. All components, except the 2N3710, are mounted flush against the printed circuit boards. Because of the close spacing of the leads on the 2N3710, no attempt was made to have the printed-circuit board layout coincide with the wire leads on the 2N3710. The outside leads are spread apart and the 2N3710 is mounted about one-eighth of an inch above the board. Most of the other parts should fit on the boards fairly well; however, because the boards were originally laid-out using a 1:1 scale (as opposed to a 4:1 that is often used) some of the parts may not fit as well as a commercial board. The layouts should suffice, however, and are by far easier to use than point-to-point wiring. The components can be mounted on the boards with the aid of *Figs.* 13, 14, and 15.

Preliminary Printed Circuit Board Tests

After the parts are mounted on a printed circuit board, it is advisable to perform some preliminary electrical tests on each board before it is fastened to the chassis. The reason for this is that if a circuit is not functioning normally, it is easier to isolate troubles on the board level.

To test board No. 3, the output leads from the power transformer, T_1, can be soldered temporarily to the bridge rectifier, CR_1 thru CR_3. The voltage at the emitter of Q_{18} should be about 33 V d.c., indicating the regulator is functioning. Q_{17} should oscillate at 3 kHz and the output amplitude at R_{63} (or J_3) should be 0.75 V rms.

To test board No. 1, +33 V d.c. from board No. 3 can be used to supply the d.c. voltage. The d.c. biasing on the stages can be checked with the aid of the voltage readings on the schematic. The voltages were measured with no signal input. Note that the measurements at the collectors of the Schmitt trigger may be reversed, depending on the states of Q_4 and Q_5. The discriminator-detector can be checked as follows: the d.c. voltages measured *across* R_{16} and R_{17} should be close to zero, with a maximum upper limit of about 50 millivolts. This measurement will assure that neither Q_7 or Q_8 is leaky. A 3-KHz, 0.1-V rms minimum signal is then applied to the input. The d.c. voltage *across* R_{16} should increase to 1.1 V d.c. and *across* R_{17} the voltage should be about 1.6 V *d.c.*

Board No. 2 can be given a preliminary test by measuring d.c. voltages. Except for the voltages at the collector of Q_{15}, which depend upon the setting of R_{66}, most measurements should be within about ±10 per cent of the values shown on the schematic.

Build your own

Solid-State Flutter Meter

ARTHUR E. GLADFELTER

Inasmuch as the construction part of the first part of this article ran a little ahead of the figures, we cover again the complete construction of the printed-circuit boards and continue with the preliminary testing of the boards, then proceed with the calibration in this installment.

IN FOUR PARTS—PART 2

NOW THAT YOU HAVE SECURED all the parts and wound the four coils, you are ready to proceed with the assembly of the boards. *Figure* 9 shows the lead position on 2N3710 transistors. In *Figs.* 10 and 11, the holes for these small devices are shown in an obtuse triangle form, with the collector at the apex in every case, but with the base and emitter locations indicated by letters B and E respectively. Note that two pieces of wire are shown to make connections where it was not possible to provide crossings of the foil pattern—always a problem in laying out printed circuits.

Making the Printed Circuit Boards

The printed-circuit board layouts for all three boards are shown in *Figs.* 10, 11 and 12. The board layouts are reproduced full scale, so that no intermediate steps are required to reduce the size. Boards 1 and 2 are 5 in. square, and board 3 is 3¾ by 4¾ in. These measurements are also specified in the parts list and can be verified by measurement of *Figs.* 10 through 12.

To show how the printed-circuit boards are made, board No. 3 will be used as an example. This is the easiest board to make and contains the least number of components. After the board has been cut to size, the copper clad side should be cleaned to remove any dirt or film that has accumulated. Cleaning is most easily accomplished with crocus cloth. Ajax cleanser or a large eraser. The board is then laid on the workbench or flat surface, with the copper side up. SCOTCH or masking tape is used on alternate corners to hold the board in position. The sketch of board No. 3 (*Fig.* 10) is then placed squarely on top of the printed-circuit board. A center punch, with a very fine tip is used to locate all holes. Alternately, if you do not wish to have a series of small "pinholes" on the magazine pages, *Figs.* 10 through 12 must be retraced, and the tracing used as a template to locate the holes. The various connections are then made by placing 1/16 in. printed-circuit tape resist on the printed-circuit board.

The two most common etching solutions for printed circuit boards are ferric chloride and ammonium persulfate. Ferric chloride is normally bought in solution form, and one pint is more than enough to etch all three boards. Ammonium persulfate is a powder, and to make a solution, 2.5 pounds of Ammonium Persulfate are added to one gallon of water. In addition, a minute amount of mercuric chloride (a poisonous powder) is added, as a catalyst, in the amount of 26.7 milligrams of mercuric chloride to one gallon of water. Both of the etching solutions just described have their advantages and disadvantages. Rather than elaborate more, I suggest those who are undecided use the ferric chloride solution.

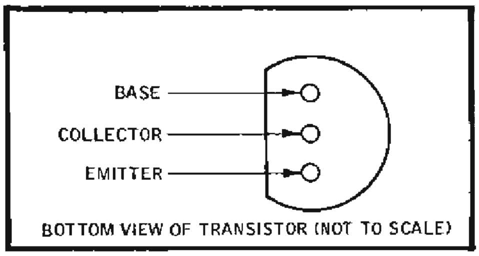
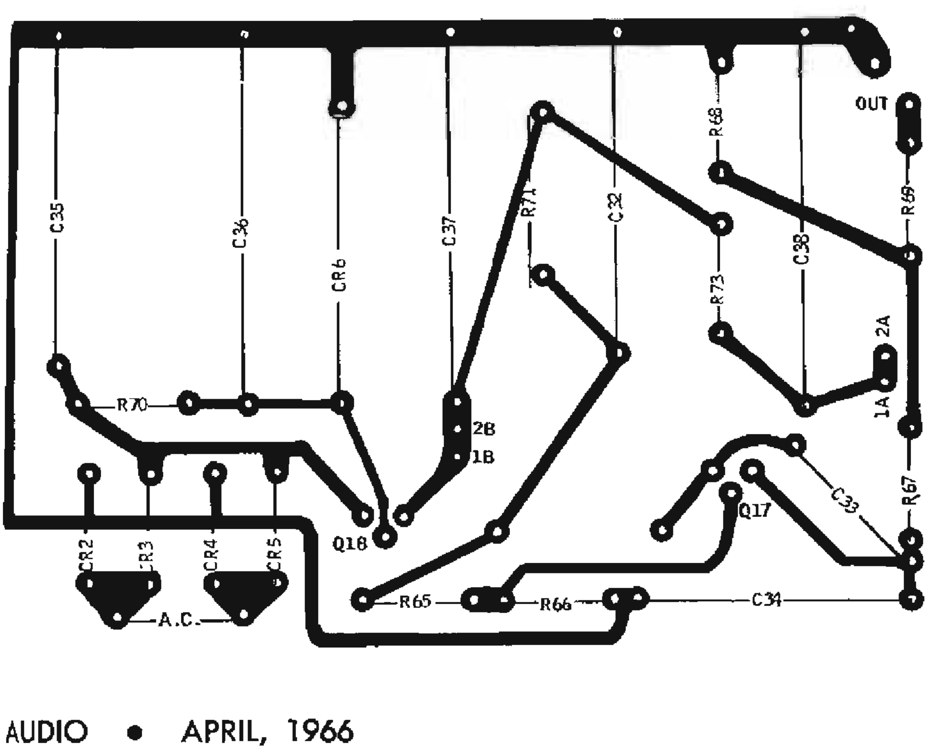


Fig. 9. Terminal arrangement of the 2N3710 transistors.

Fig. 10. Full-scale layout of board No. 3—the power supply and 3000-Hz oscillator unit. The component locations are identified by the colored lettering. The corners of the board are indicated, while the crosses at the upper right and lower left corners show the mounting-hole locations.



A plastic or glass tray can be used to hold the solution to etch the boards. The boards are then submersed in the solution, and to speed up the etching process, the tray can be heated (to about 160° F) and agitated. The time to etch a board will vary, but with heat and agitation, the time will range from about 15 to 30 minutes. After etching, the boards should be washed with water and the printed circuit tape resist removed. The boards can then be cleaned (as described previously) and appropriate holes drilled for the various components.

When soldering the components to the boards it is a good idea to heat-sink the 1N126 germanium diodes and also the resistors associated with the resistive divider in the range switch. All components, except the 2N3710, are mounted flush against the printed circuit boards. Because of the close spacing of the leads on the 2N3710, no attempt was made to have the printed-circuit board layout coincide with the wire leads on the 2N3710. The outside leads are spread apart and the 2N3710 is mounted about one-eighth of an inch above the board. Most of the other parts should fit on the boards fairly well; however, because the boards were originally laid-out using a 1:1 scale (as opposed to a 4:1 that is often used) some of the parts may not fit as well as a commercial board. The layouts should suffice, however, and are by far easier to use than point-to-point wiring.

Preliminary Printed Circuit Board Tests

After the parts are mounted on a printed circuit board, it is advisable to perform some preliminary electrical tests on each board before it is fastened to the chassis. The reason for this is that if a circuit is not functioning normally, it is easier to isolate troubles on the board level.

To test board No. 3, the output leads from the power transformer, T_1, can be soldered temporarily to the bridge rectifier, CR_1 thru CR_5. The voltage at the emitter of Q_{10} should be about 33 V d.c., indicating the regulator is functioning. Q_{11} should oscillate at 3 kHz and the output amplitude at R_{69} (or J_1) should be 0.75 V rms.

To test board No. 1, +33 V d.c. from board No. 3 can be used to supply the d.c. voltage. The d.c. biasing on the stages can be checked with the aid of the voltage readings on the schematic. The voltages were measured with no signal input. Note that the measurements at the collectors of the Schmitt trigger may be reversed, depending on the states of Q_4 and Q_5. The discriminator-detector can be checked as follows: the d.c. voltages measured *across* R_{16} and R_{17} should be close to zero, with a maximum upper limit of about 50 millivolts. This measurement will assure that neither Q_7 or Q_8 is leaky. A 3-KHz, 0.1-V rms minimum signal is then applied to the input. The d.c. voltage *across* R_{16} should increase to 1.1 V d.c. and *across* R_{17} the voltage should be about 1.6 V *d.c.*

Board No. 2 can be given a preliminary test by measuring d.c. voltages. Except for the voltages at the collector of Q_{15}, which depend upon the setting of R_{60}, most measurements should be within about ±10 per cent of the values shown on the schematic.

Calibration

After determining that all circuits are functioning normally, the unit is ready for calibration. Three potentiometers are used to calibrate the unit. R_{11} is the first pot to be adjusted. With R_1 in the maximum gain position a 3-kHz signal of about 0.1 V rms is applied to the input. Then, R_{11} is adjusted so that the square wave observed at the emitter of Q_6 is symmetrical. Alternately, if an oscilloscope is not available, R_{11} can be adjusted with a d.c. voltmeter. The philosophy behind this method of adjustment is that Q_4 (or Q_5) will be conducting 50 per cent of the of the time. With this in mind—and assuming a collector-to-emitter saturation voltage of about 0.3 v.d.c.—the calculated average voltage at the collector of Q_5 will be about 16.5 v.d.c. Using a d.c. voltmeter, R_{11} is then adjusted so the voltages at the collector of Q_5 is 16.5 v.d.c.

R_{60}, the second potentiometer to be adjusted, controls the gain of the voltmeter circuitry. It is adjusted with the RANGE switch in the 0.3 per cent position. To show the reasoning behind the setting of R_{60}, it will be necessary to take a second look at the basic definition of flutter, Eq. (1). Rearranging Eq. (1) in terms of df_o will give:

$$df_o = \frac{\sqrt{2}\,(f_{avg.})\ (f_k)}{100}$$

where: f_k = per cent Flutter

df_o = Maximum deviation from the average frequency

$f_{avg.}$ = Average frequency (3000 Hz)

With $f_{avg.}$ = 3000 Hz, f_k = 0.30 per cent, df_o corresponding to full-scale deflection is:

$$df_o = \frac{\sqrt{2}(3000)\,(0.3)}{100} = 12.73\ Hz$$

Now, 12.73 Hz is the peak deviation from the average frequency and peak-to-peak deviation on the discriminator curve will be twice the peak deviation or 2×12.73, or (12.73) 25.46 Hz f a full-scale reading of 0.30 per cent r flutter.

After knowing the peak-to-peak d viation in frequency for a given pe centage of flutter, a corresponding a plitude can then be determined fr the slope of the discriminator S cur *Fig.* 5. Assuming arbitrary frequen of 3200 Hz and 2800 Hz (on Fig. the corresponding d.c. voltages will + 0.310 V d.c. and −0.310 V d.c. spectively. The rate of change in f quency (Δf) with respect to the r of change in voltage (ΔV) will then

$$\frac{\Delta f}{\Delta V} = \frac{3200-2800}{0.310-(-0.310)}$$

$$= \frac{400\ \text{Hz}}{0.62\text{V}} = 650\ \text{Hz/volt}$$

Or, stated another way, the incom frequency will have to change 650 for a voltage change of 1.0 volt p to-peak at the discriminator output is, at this point, more convenient to c vert the amplitude directly to an reading, that is, 1.0 volt peak-to-peak

$$\frac{1}{2\sqrt{2}}\text{V rms} = 0.3535\ \text{V rms}$$

$$= 353.5\ \text{mV}$$

As a result, 650-Hz peak-to-peak viation will correspond to 353.5 rms. The voltage v correspondin 25.46 Hz or 0.30 per cent flutter be:

$$v = \left[\ 25.46\ \text{Hz}\ \right]\left[\frac{353.5\ mV\ rms}{650\ Hz}\right.$$

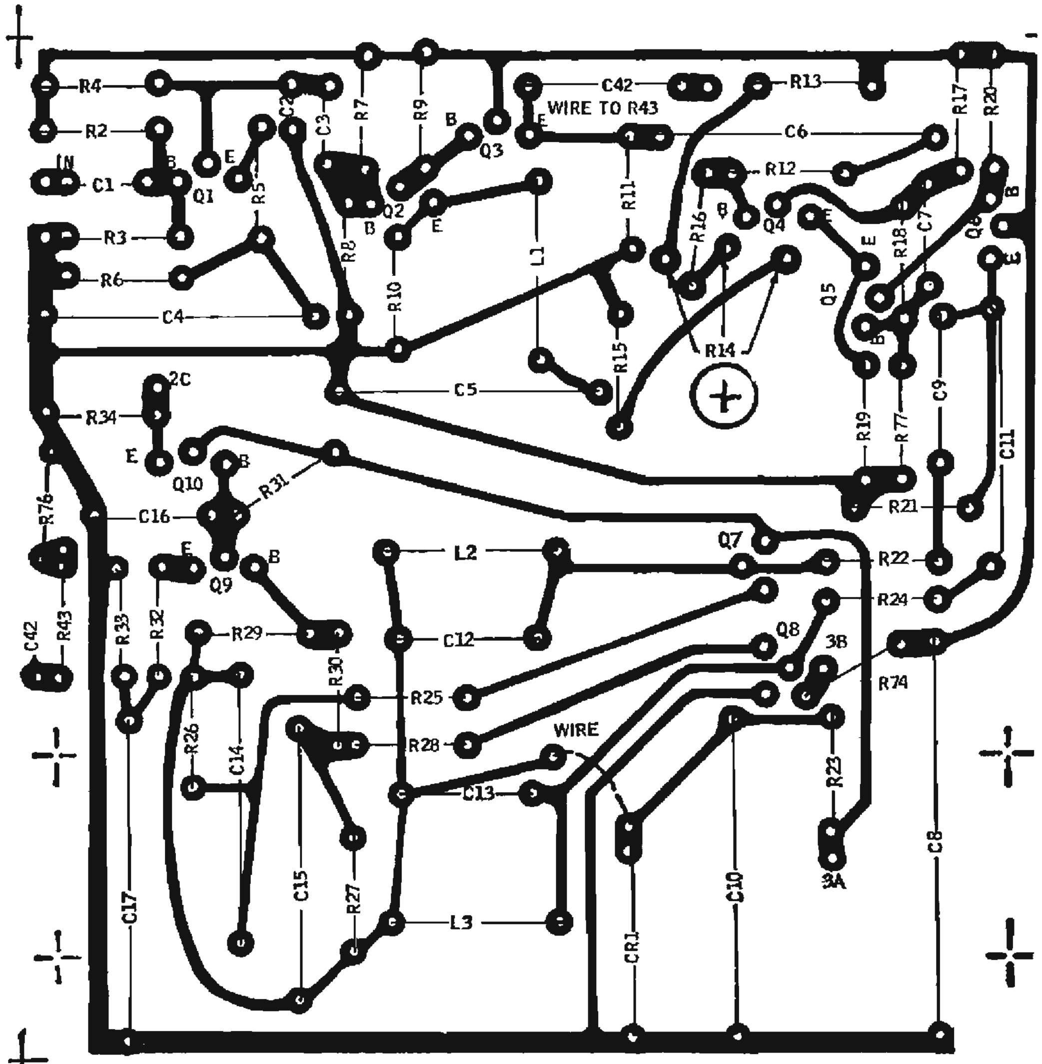
$$v = 13.8\ mV\ rms$$

This means the rms voltage at the criminator output will be 13.8 rms when the input frequency de tion represents 0.30 per cent rms flu

To set R_{60}, the input signal is moved and R_1 is adjusted to the m imum gain position. The FILTER sw is placed in the 0.5-to-250 Hz posi and the RANGE switch in the 0.30 cent position. The output signal fro sine-wave generator is then coupled the base of Q_9 via about a 10-µf ca tor. An rms meter is also connected the base of Q_9. With the frequency tween 22 and 28 Hz, the oscillator put amplitude is increased until meter indicates 13.8 mV rms. R_{60} then adjusted so the meter indica full scale, or 0.30 per cent rms flu

R_{52}, the last potentiometer to be justed, calibrates the 1.0 per cent ra R_{52} must be adjusted after R_{60}. For full-scale reading of 1.0 per cent flutter, the discriminator output vo will be 3.33 times the voltage for 0.30 per cent full-scale reading.

Fig. 11. Full-scale layout of circuit board No. 1. Crosses at sides of panel are centers of holes for mounting brackets, made of ⅜ in. wide .064 aluminum strips.



ing this, and with the same set-up as described previously, it is only necessary to turn the RANGE switch to the 1.0 per cent position and increase the oscillator output amplitude so the external meter reads $3.33 \times 13.8 = 46.1$ mV rms. R_{52} is then varied so the meter indicator reads full scale or 1.0 per cent rms flutter.

The input amplitude (from the recorder output) although not critical, should be at least 0.10 V rms. The fixed resistors, R_{13} and R_{76}, determine the meter deflation when the RANGE switch is in the LEVEL SET position. R_{13} and R_{76} have been selected so the meter will indicate about 85 per cent of full-scale deflection, when adjusting the input potentiometer R_1 for the correct LEVEL SET amplitude. A line can then be placed on the meter scale at about 85 per cent of full-scale deflection to indicate the LEVEL SET position. (This marking was omitted on the writer's unit)

Performance

The preceding calibration is all that is necessary to calibrate the flutter meter. In order to evaluate the basic design however, a transistorized flutter simulator was made. This simulator can also be used to calibrate other existing flutter meters.

The flutter simulator, *Fig.* 13, is a bistable multivibrator (flip-flop), with a nominal repetition frequency of 3 kHz. The deviation in performance from a basic flip-flop can be explained as follows: Normally the base-bias resistors (R_{80} and R_{81}) would be returned to ground, and the frequency would be relatively independent of any variation in supply voltage. In *Fig.* 13 however, the base-bias resistors are returned to ground through a 100-ohm resistor, R_{82}. A voltage from a sine-wave generator is then impressed across R_{82} via R_{83}. The varying voltage appearing across R_{82} will not affect the actual base circuit R-C time constants, but will change the ultimate or peak voltage to which the base capacitors can charge. Now, for a given voltage "trip-point" on an exponential R-C charging waveform, an increase in the peak charging voltage will cause a decrease in charging time. Thus, when a negative voltage is impressed across R_{82}, the peak charging voltage will increase and cause the frequency to increase. Similarily, a positive voltage impressed across R_{82} will cause a decrease in the peak charging voltage,

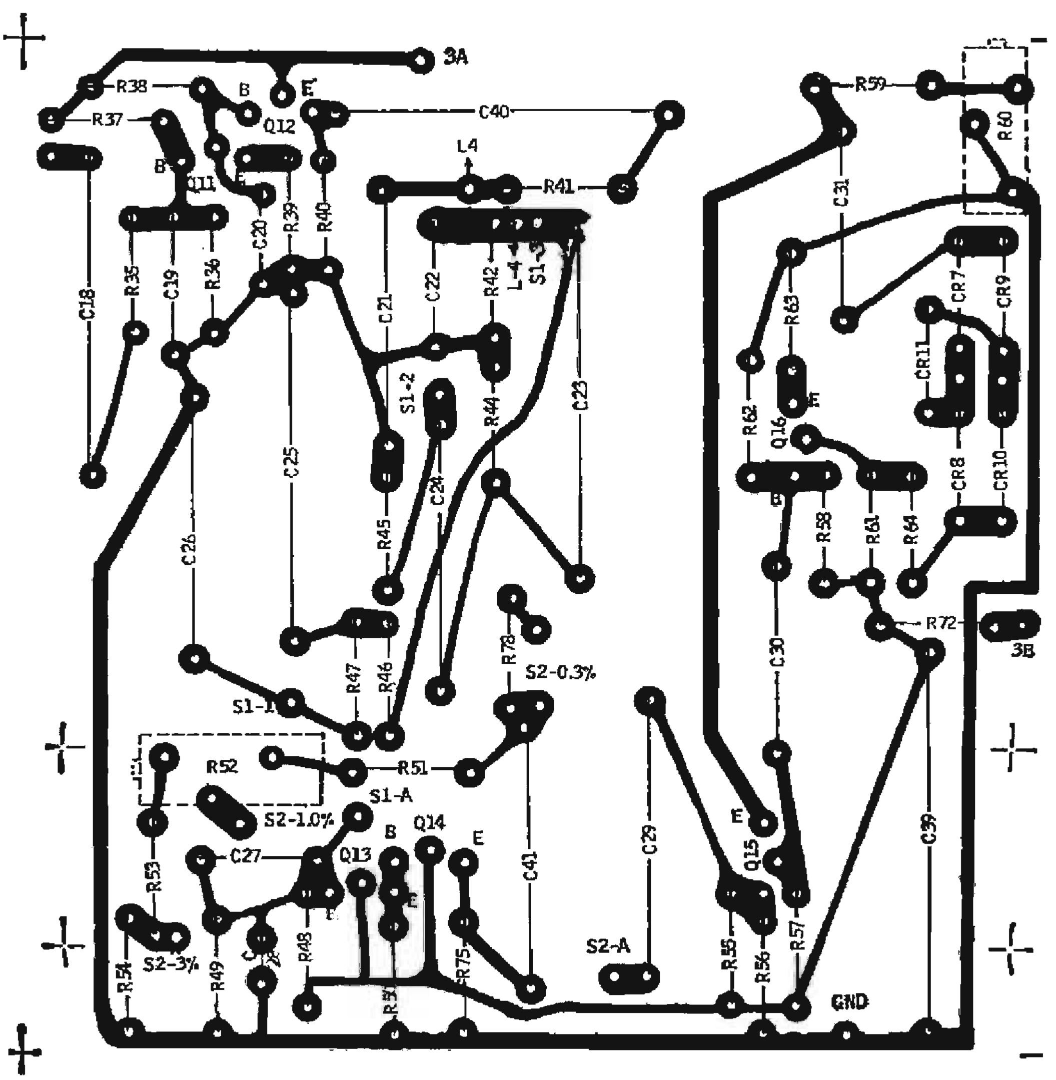


Fig. 12. Layout of board No 2. 3A and 3B indicate connections to power supply; input is to top end of C_{18}. Connections to switches are indicated as S1-1, S2-3%, etc.

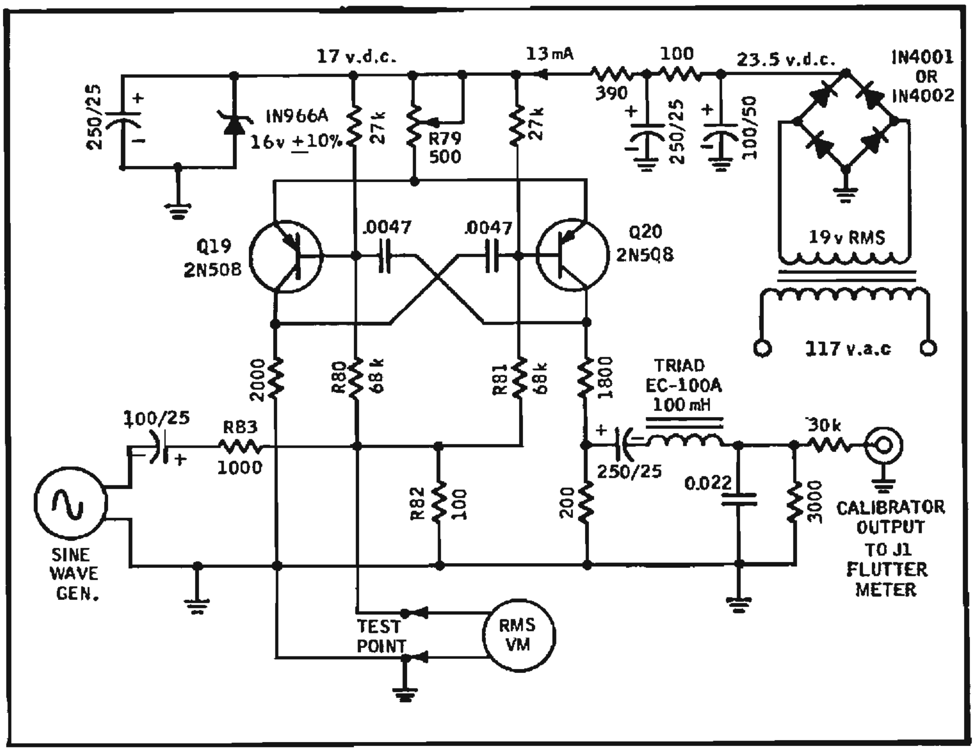


Fig. 13. Schematic of flutter simulator used to calibrate meter.

and the frequency will decrease. Consequently, the generator output amplitude will correspond to a percentage of flutter and the frequency will represent the flutter and the frequency will represent the flutter rate. The frequency of the multivibrator may be adjusted to exactly 3000 Hz with potentiometer R_{79}.

The multivibrator output is taken from the collector of Q_{20}, attentuated by the 200-ohm and 1800-ohm resistors, and applied to a two-pole 12 dB/octave low-pass filter. This filter, which has an upper half-power frequency of about 4.5 Hz, removes the odd harmonics from the multivibrator square-wave output and provides a waveform that is essentially sinusoidal at the calibrator output.

To calibrate the flutter simulator it was necessary to measure the deviation in frequency (Δf) *vs.* the change in voltage (ΔV). These results are plotted in *Fig.* 14. Using the frequency deviation corresponding to 0.30 per cent rms flutter and converting this to an rms voltage, the level across R_{82} will be about 0.067 V rms when the flutter simulator is generating 0.30 per cent rms flutter. When the rms meter indicates 0.223 V rms, the output frequency deviation represents 1.0 per cent rms flutter. Note the rms readings just given were measured *across* R_{82} and do not indicate the sine-wave generator output amplitude. This amplitude will be about 11 times that indicated by the rms meter.

The first item to be evaluated with the flutter-simulator was the over-all frequency response. If the detector charge and discharge time constants (R_{25}, R_{26}, R_{27}, R_{28}, C_{14}, and C_{15}) were not chosen correctly, the flutter meter would not have provided uniform frequency response to the higher flutter rates. Using the calibrator, the over-all frequency response is down 1.0 dB at 170 Hz and down 3 dB at 240 Hz. At the low flutter rates the circuitry is flat to well below 0.5 Hz; however, at flutter rates below about 4 Hz, because of the damping characteristics of the meter movement, the indicating meter tends to follow the flutter peaks rather than the rms values. For very low flutter rates, the reading can be approximated by using the average of the high and low meter indications. Actually, if very low flutter rates are to be observed, it is better to use a d.c. oscilloscope, pen recorder, or any other suitable indicator that has response to d.c. Any of these instruments can be connected to the oscilloscope output jack J_4 on the flutter meter. Based on the tests just described, the flutter-rate frequency response should be sufficient. In fact, uniform response up to 120 Hz is adequate, except for the most critical tests.[1]

The second test was to evaluate the flutter meter tracking error. This was done by injecting various known percentages of flutter into the flutter meter. The flutter meter actual readings and that represented by the rms meter connected to the flutter simulator were then recorded. These results are summarized as follows:

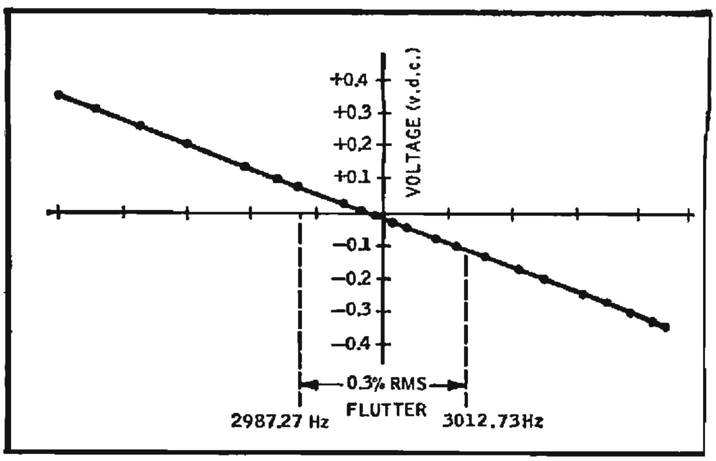
Range	Flutter meter should read	Flutter meter actual reading
0.3%	0.30%	0.30 %
0.3%	0.15%	0.14 %
0.3%	0.10%	0.093%
0.3%	0.05%	0.04 %
1.0%	1.0 %	1.0 %
1.0%	0.8 %	0.795%
1.0%	0.6 %	0.59 %
1.0%	0.4 %	0.37 %

Thus, on the 0.30 per cent range and for readings greater than about 0.05 per cent, the absolute accuracy should be within 0.01 per cent flutter. The error does increase on the lower portion of the 1.0 per cent range; that is, where the reading was 0.37 per cent for 0.40 per cent. The tracking error at the low-scale readings is due, to a large extent, by the nonlinear characteristics of the diodes in the meter-bridge-rectifier circuitry. The tracking error could be reduced by providing more open loop gain in the meter amplifier, Q_{16} and Q_{17}. Along with the greater gain would be the possibility of low-frequency instability or motorboating. For this reason, the slight tracking error was accepted.

Although the flutter meter will probably be used at room temperature, a temperature test was performed on the entire flutter meter. The meter was placed inside a test chamber and the flutter simulator was used to provide a reference of 0.25 per cent rms flutter at +75°F. With the temperature increased to +122°F, the flutter decreased to 0.23 per cent, or a change of 0.02 per cent of flutter. Similarly, reducing the temperature to +32°F, increased the flutter to 0.26 per cent, or an increase of 0.01 per cent flutter.

To be continued

Fig. 14. Calibration of simulator in frequency deviation vs. voltage.



Build your own

Solid-State Flutter Meter

ARTHUR E. GLADFELTER

In this installment the author finishes his discussion of the operating characteristics of the flutter meter. Also included, and certainly as relevant as the instrument itself, is a discourse on the standards of flutter measurement.

IN FOUR PARTS—PART 3

The flutter meter does have three peculiarities, all of which are minor and can be readily explained. The first is the turn-on settling time. After the power is switched ON, the meter will indicate full scale (and possibly vary from zero to full scale) for about the first 30 seconds. This, of course, is the various capacitors charging. The lengthy settling time is easily understood when one considers the low-frequency response of the various stages.

The second item deals with the meter zero position. With the meter ON and no input signal, the meter will indicate zero per-cent flutter; however, when a 3-kHz signal (which theoretically has zero per-cent flutter) is applied to the input, the residual flutter is about 0.013 per cent. This was traced to the interaction between the power transformer and the discriminator coils. I consider the problem minor, unless accurate readings must be made down in the 0.02 to 0.04 per cent region. If so, the residual reading can be reduced considerably by placing a sheet of mumetal near the discriminator coils. Alternately, the residual reading can be reduced to zero by mounting the power transformer externally or by using an external d.c. power supply.

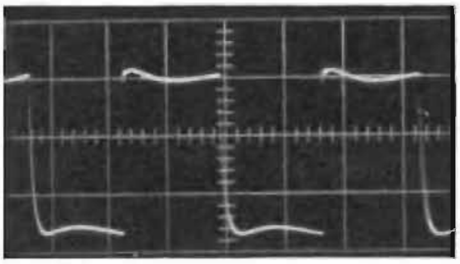


Fig. 15 (A). Multivibrator output measured at the junction of 1.8k- and 200-ohm resistors. Vertical sensitivity: 0.5 volts/cm; horizontal sweep: 100 μsec/cm. Per cent flutter: 0.25% rms; flutter rate 90 Hz (Sweep: from left to right).

The third item that some readers will invariably detect is the result of a sudden change in temperature on CR_1. Because of the positive temperature coefficient of CR_1, a sudden change in temperature, such as might be caused by a blast of air, will cause a small change in the zener voltage of CR_1. The sudden change is, however, amplified and can be readily observed on the meter. The temperature dependence of CR_1 is a minor problem, and is practically non existent when enclosed in the meter case.

The dynamic performance of the flutter simulator and flutter meter can be shown most easily with the waveforms of *Fig.* 15. For the photographs, the sine-wave generator used in conjunction with the flutter simulator, was adjusted for 0.25 per cent rms flutter and a flutter rate of 90 Hz. (A) in *Fig.* 15 shows the multivibrator 3-kHz output waveform, measured at the junction of the 1800- and 200-ohm resistors. Note at (A), *Fig.* 15, a small amount of flutter (or jitter) can be detected on the last cycle of the waveform. (B in *Fig.* 15 shows the multivibrator waveform after passing through the 4.5-kHz low-pass filter. This is the waveform of the flutter simulator output, or the signal that is applied to the flutter meter input (J_1). (C) shows the waveform at the emitter of Q_3. Note that in comparing (B) and (C), the band-pass amplifier has improved the waveform. The limiter output, measured at the emitter of Q_6, is shown at (D) in *Fig.* 15. All these figures show a nominal frequency of 3 kHz. It is virtually impossible, however,

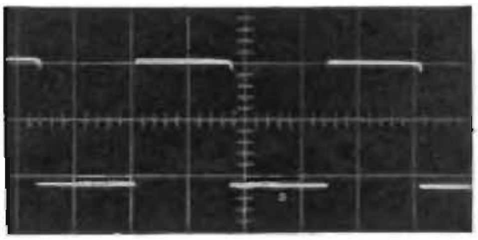


Fig. 15 (D). Limiter output at the emitter of Q-6. 10 volts/cm; 100 μsec/cm. Per cent flutter: 0.25%. Flutter rate: 90 Hz.

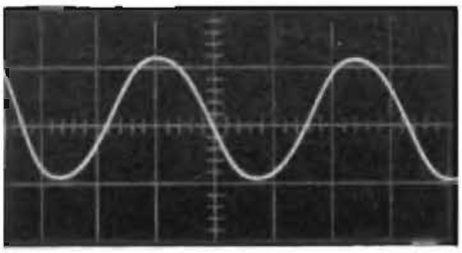


Fig. 15 (B). Calibrator output. 1.0 volt/cm; 100 μsec/cm. Per cent flutter: 0.25% Flutter rate: 90 Hz.

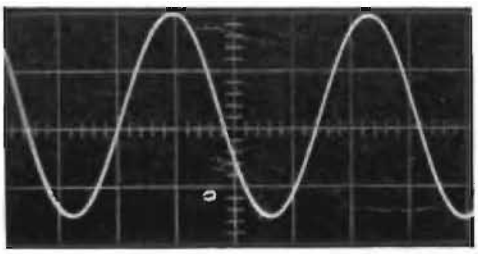


Fig. 15 (C). Emitter of Q-3 (in flutter meter). 5 volts/cm; 100 μsec/cm. Per cent flutter: 0.25%. Flutter rate: 90 Hz.

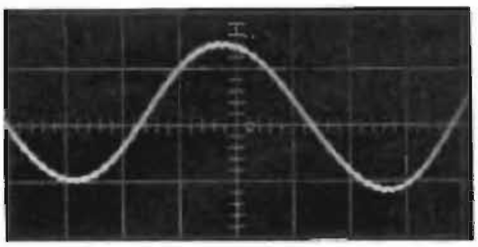


Fig. 15 (E). Oscilloscope output (J2). 1.0 volts/cm; 2.0 milliseconds/cm. Per cent flutter: 0.25%. Flutter rate: 90 Hz.

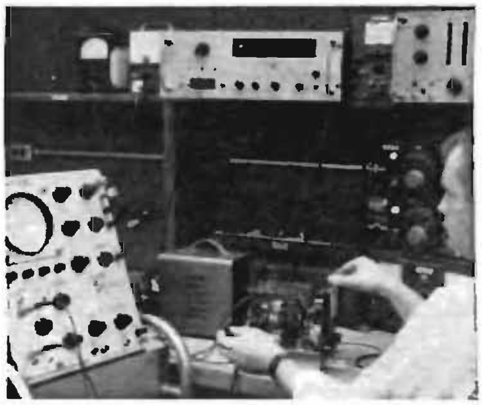


Fig. 16. R14 is being adjusted until a symmetrical waveform is observed. The voltmeter is connected to the collector of Q-5 and indicates about 16.99 volts d.c.

by looking at the waveforms, to measure the flutter with any degree of accuracy. (E) shows the waveform after passing through the discriminator and low-pass filter. This was measured at the scope output jack (J_2). The basic flutter frequency of 90 Hz has been recovered, along with the very low amplitude 3-kHz components, that are riding on top of the 90-Hz flutter rate.

Measurement of Flutter

This portion of the article describes briefly an accepted standard that does exist for the measurement of flutter. The standard, "Flutter in Sound Recording," is by the American Standards Association (ASA) Z57.1-1954.[7] The "Standard" indicates that a constant wavelength "test tape" or a test tape having relatively negligible flutter content is to be used. The test tape is played on the machine under test and the readings are observed on a flutter meter. Percentage of flutter is to be that for one operation alone, that is, playback only.

Standard flutter test tapes are available from Ampex. These are recorded on a highly refined tape transport and the flutter content is guaranteed to be less than 0.03 per cent, with a typical value of 0.015 per cent. The actual flutter content is, however, marked on the container of each tape. Test tapes are available for tape speeds of 3¾, 7½ and 15 inches/sec. Tape length is 600 feet and the price is $21.95.

Even though the ASA Standard has existed for many years, it was impossible until recently to comply with it due to the lack of a suitable test tape. This has led to the common use of an Alternate (Non-Standard) method which does not require the use of a special tape. This method allows one to record and reproduce the tape on the same machine. Thus, the flutter obtained in this manner is for two operations, and must be converted to represent flutter for one operation. The Alternate Standard states: "It must be understood that when this procedure is used, flutter components due to a given excitation in recording and reproduction add vectorially". It further states that ". . . while testing machines with its own recorded signal, . . . if the measuring equipment indicates rms flutter, and if conditions are such that equal flutter is to be expected in recording and reproduction, the most probable flutter for either operation alone is 0.707 of the measured flutter."

A report published previously[8] compares measured data from the Standard and Non-Standard method. The tests performed indicate that with the Standard method, the flutter-meter needle is much easier to read. Also, with the Non-Standard method, cancelation and addition of flutter components can lead to results which misrepresent the true flutter in the machine. The actual difference in methods is dependent upon the particular machine being tested, and is related to the frequency and relative phase of the flutter vectors.

In addition to the previously described flutter Standards, reference is oftentimes made to "Weighted" and "Unweighted" flutter. The flutter meter, as described in this article, will indicate "Unweighted" flutter when the *Filter* switch is in the 0.5- to 250-Hz position. To measure "Weighted" flutter it is necessary to insert an additional band-pass filter after the existing 250-Hz low-pass filter. The filter provides minimum attentuation at about 4 Hz, and has half-power frequencies of about 1.2 and 12 Hz. For more detailed information on the filter requirements, the reader should consult reference[9]. Because of a more restricted bandwidth, the "Weighted" flutter can never exceed the "Unweighted" flutter. For example, at 7½ inches/sec. the NAB Standard[9] allows the maximum "Unweighted" flutter to be 0.20 per cent rms and the "Weighted" flutter to be 0.07 per cent rms.

In the previous discussion of flutter measurement, it has been assumed that the tape recorder is totally free of tape-dropout. The reader should understand, however, that a flutter measurement on a tape machine with excessive dropout can give an erroneously high flutter reading, even though the actual flutter may be very low. This is because tape dropout produces a very large voltage at the discriminator output.

To be concluded

Upon completion of the instrument—and on the occasion of any subsequent servicing, it is desirable to have at hand a table showing the element voltages for each of the transistors. Table II provides all of this information in easy reference form.

POSITION	VOLTAGE (v.d.c.)
Q1 E	10.5
Q1 B	11.0
Q1 C	18.2
Q2 E	8.6
Q2 B	9.3
Q2 C	17.4
Q3 E	16.9
Q3 B	17.4
Q3 C	27.1
Q4 E	4.4
Q4 B	5.1
Q4 C	4.8
Q5 E	4.4
Q5 B	2.1
Q5 C	27.1
Q6 E	26.5
Q6 B	27.1
Q6 C	27.1
Q7 E	13.2
Q7 B	13.2
Q7 C	29.9
Q8 E	13.2
Q8 B	13.2
Q8 C	0
Q9 E	12.5
Q9 B	13.2
Q9 C	22.0
Q10 E	19.5
Q10 B	22.0
Q10 C	29.9
Q11 E	1.5
Q11 B	2.1
Q11 C	14.5
Q12 E	14.0
Q12 B	14.5
Q12 C	29.9
Q13 E	14.4
Q13 B	15.0
Q13 C	31.0
Q14 E	13.7
Q14 B	14.4
Q14 C	31.0
Q15 E	0.8
Q15 B	1.4
Q15 C	24.0
Q16 E	1.3
Q16 B	1.9
Q16 C	15.0
Q17 E	10.4
Q17 B	10.0
Q17 C	25.5
Q18 E	33.4
Q18 B	32.9
Q18 C	41.5

E=EMITTER
B=BASE
C=COLLECTOR

Note: Readings taken with no signal input and with an input line voltage of 116 volts a.c. Voltages associated with the Schmitt trigger, Q-4, Q-5, and Q-6 may vary depending on the state of the transistors.

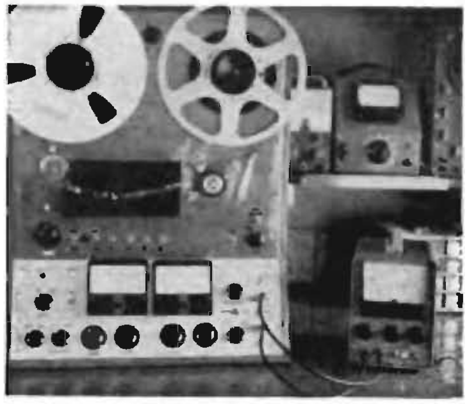


Fig. 17. The flutter and wow meter connected to a tape recorder for flutter measurements.

Build your own

Solid-State Flutter Meter

Fig. 1. The completed flutter meter with its protective case removed.

ARTHUR E. GLADFELTER

Conclusion

In this final section the author makes his acknowledgements and offers his references. Also included is a box of corrections on what has already transpired.

WHILE designing the flutter meter and also while preparing the article, I was given assistance by one particular individual, I would like to thank Edward J. Gleeson, Principal Engineer with The Bendix Corporation, York, Pa. for his help in the design of the detector, discriminator, and low-pass filter. He has also reviewed the written portion of the manuscript.

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[7]ASA-IRE Standards on Sound Recording and Reproduction, "Method of Determining Flutter content," A.S.A. Report Z57.1-1954.

[8]D. E. Morgan, "Flutter measurements by the American Standard method." *J. Audio Eng. Soc.,* 9:232, 234, 238, July 1961.

[9]J. A. Moore, "NAB reel-to-reel tape standards," *Broadcast Engineering,* September, 1965, Pages 30 and 32.

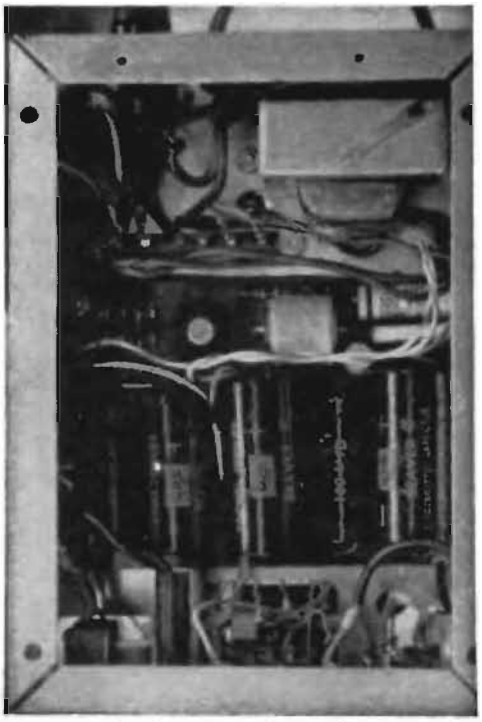

Fig. 3. A bottom view showing the parts layout of the completed flutter meter.

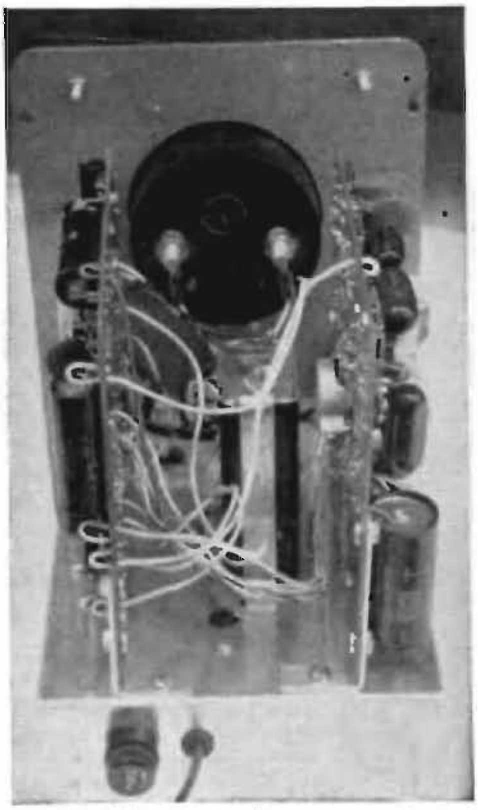

Fig. 2. (Above) A direct rear view of the flutter meter. The two circuit boards and associated wiring may be seen.

Department of (error) Amplification

In a complex story errors will sometimes creep in regardless of our watchfulness. The following summarizes corrections to this series. All save the last, are in the March issue.

Page	Column	Line	Article Reads	Correction
21	3	22	–32 deg.	+32 deg. F
22		Schematic	R72 500	R72 560
26	3	7	(0.22uf)	(0.022uf)
26	3	9	(0.47uf)	(0.047uf)
26	3	17	+5%	±5%
22 April	1	4-6	An extra line crept in here. ". . . and the frequency will represent the flutter rate." is the correction.	

The Early Flutter Spoils the Recording

LEWIS A. HARLOW

Every user of a tape recorder occasionally encounters flutter which may be in the recording itself or simply in the playback process. The suggestions given here will aid in eliminating flutter wherever it occurs.

THE EXPERTS WHO USE tape recorders in telemetry and other rarified applications have coined the word "stiction" to describe a fault which, if present at all, nullifies the best professional efforts and makes the tapes completely worthless. "Stiction" is a sticky kind of friction that causes the tape to chatter rather than to flow smoothly by the heads of the machine. In music recording and reproduction with tape, stiction at a somewhat lower frequency is better known as flutter, and with rare exception, the causes of the fast stiction and the slower flutter are the same.

NAB standards specify that flutter shall not exceed 0.2 per cent when recording and reproducing from the same equipment. Most of the good brands of recorders claim this low level of flutter (or even better) in their specifications, and can justify the claim with tests of a new machine under laboratory conditions. An audible flutter is very likely to appear at some later time in the output from this good machine, but fortunately it can usually be reduced to satisfactory inaudibility without the need for outside professional service. The corrective is mostly a matter of a change or two in your habits of tape handling and tape selection—plus a little more good housekeeping than is presently being practiced.

If audible flutter is present in playbacks in the better music room, it is most frequently heard at the start of the PLAY cycle, and often it disappears completely after the first four or five minutes of playing. Slow sustained tones from a piano recording are the most likely to be superimposed with flutter. Almost as bad an offender is the slowly played classic clarinet, probably because the rank and file clarinetist is taught to play with less vibrato than is traditional from any other solo instrument. The classic organ—if its tremulant stop is deenergized—is also very sensitive at showing up the flutter problem in your recorder, but from an organ you never know exactly what to expect in the way of sound and are likely to blame the flutter on the instrument of origin or the taste of the performer. Unless the mechanico-electronic flutter of your tape recorder is very bad, the normal vibrato of a human voice or most of the other solo orchestral instruments will conceal the flutter effectively.

But why should this flutter be worse at the beginning of a tape than later? Why does it usually disappear completely after four or five minutes into the tape? What condition is different at the start to cause the phenomenon?

The obvious difference and the important one is that the take-up reel is starting empty. The function of this reel is to accept and store tape that is fed to it at a constant speed by the capstan. Uninhibited, the take-up reel should want to turn faster than it does—but just a little faster. When empty, the reel turns rather rapidly because the tape is coming onto a core of modest circumference. As the reel fills with tape, as shown in *Fig.* 1, the circumference increases and the reel speed becomes slower and slower. There is a clutch somewhere below-deck, and the slippage of this clutch permits the speed variation.

By NAB standards, the take-up reel should accept the steadily arriving tape under a constant tension of 5.5 ounces, but it rarely does. The clutch effect, attained in some designs of recorder by motor slippage and in others by slippage of belts or spring-loaded or weight-loaded felt washers, is rarely precisely adjustable. In many recorders, the clutch tension is maintained by coil springs, adjustable by cut and try if at all.

Then there is the matter of FAST FORWARD function which this same reel must perform. This forward movement should be at least fairly fast, and there is required a bit of the clutch which can be considerably more vigorous than the ideal minimum for take-up service.

Suppose, then, that your clutch, by design or disrepair, does not yield to the theoretical demand for uniform tape tension of 5.5 ounces. On a seven-inch reel with standard 2.25-inch diameter core, *the pull at the start of take-up can be three times as strong as it becomes when the reel has filled with tape.* This is just a matter of leverage. It can contribute importantly to the stiction or flutter at the start of the RECORD and PLAY cycles.

the other tape tensions designed into the average-to-good recorder. The capstan pulls with an average tension of 23 ounces—against a drag across the heads of about six ounces. There should be enough factor of safety here so that a somewhat variable pull by the take-up reel will make no difference, but the evidence in practice is quite damning.

And there is professional corroboration that the small empty core of the take-up reel is a troublemaker. In the early days of tape recording, all of the tape manufacturers sold you tape on seven-inch reels with 1.75-inch cores. Then, suddenly and uniformly, the standard changed to 2.25-inch cores. These are compared in *Fig.* 2. Why? Certainly the tape producers would have liked to sell you the extra 150 feet of 1.0 mil tape that the older reels would hold, and certainly you would have appreciated the extra few minutes of playing time per side. Continuous playing time was then (and still is) an important sales feature of recorder and tape specification. There must have been a pretty good reason for the step backward—a reason like the stiction-flutter problem that could be eased by increasing the diameter of the reel core. Commercial recorders have always been equipped for large-core reels, and the very-small-core experiment on the household recorder had turned out to be a bad idea. No better reason has been advanced, and the increase in the standard of core size for the home recorder is still not quite enough.

What to do about it

As previously mentioned, the tape tension adjustments on your recorder (if there are any) are probably extremely inaccessible and should only be approached with professional equipment and skills. This should not be a deterrent, though, to the project of reducing flutter to an inaudible level.

Ideally, there should be no pull at all from the take-up reel. You can prove this for yourself by running the tape over the edge of the recorder and letting it fall to the floor by its own weight and collect in a pile. Try this as you record a test sampling of slow piano, live or from a disk. Now play back the sampling, again disposing of the tape over the edge of the recorder and onto the floor. If you have been bothered by flutter at the start of your piano tapes, now it will probably have disappeared. This cure—playing onto the floor—is of course impractical for a long selection, either recording or playing back, and what you will be needing is a special take-up reel with oversize core—the more oversize the better.

Your special reel with oversize core will require a little finding. It is only recently becoming available as an "empty reel" through the better electronics supply stores and catalogs. The seven-inch reel with four-inch core pictured in *Fig.* 3 was acquired with a recorded RCA-Victor tape on it. The tape, a satisfactory investment in its own right, was then transferred to a standard reel, thereby releasing the special reel for service of great value as a flutter-resistant take-up reel.

Somewhat surprisingly, you can play or record onto this big-cored reel the entire contents of a standard seven-inch reel, the 1200 feet of 1.5-mil or the 1800 feet of 1.0-mil that you would buy as a reel of tape (but not as much as you yourself could wind onto it to its absolute capacity). The 1200 or 1800 feet will load the special reel full but not to the point of spillage, though the reel would not hold this much if loaded by fast forward or rewind; both of these processes tend toward rather loose winding. You can rewind *from* the special reel safely and easily, and this of course you may wish to do to free the special reel for other service as may be needed in flutter-free recording and playback.

If your recorder happens to accommodate no larger than a five-inch reel, your shopping expedition is slightly different. Recorded music tape or a catalog is not the source for a five-incher with oversize core. The five-inch special reel shown in *Fig.* 4 has a 2.75-inch core (instead of the standard 1.75), and it was acquired with a Dubbings test tape on it. Right away the Dubbings tape was respooled onto a standard reel so the big-core special would be available for critical take-up service—like, for instance, a test tape. (And please note the importance which a manufacturer of test tapes attaches to the big-core idea).

This big-core five-incher will accept in take-up the full 600 or 900 feet of tape sold commercially on five-inch reels. Again the special reel is loaded to capacity and would be overloaded if charged by the fast forward or rewind process. The Dubbings Company tape came from Long Island City, New York, and its reel is stamped "American Molded Products Co., Chicago." You can try either source in your shopping. Of course you will not need a big-core five if your recorder will hold the even bigger cored seven previously suggested.

Big-core reels are a chore at best if

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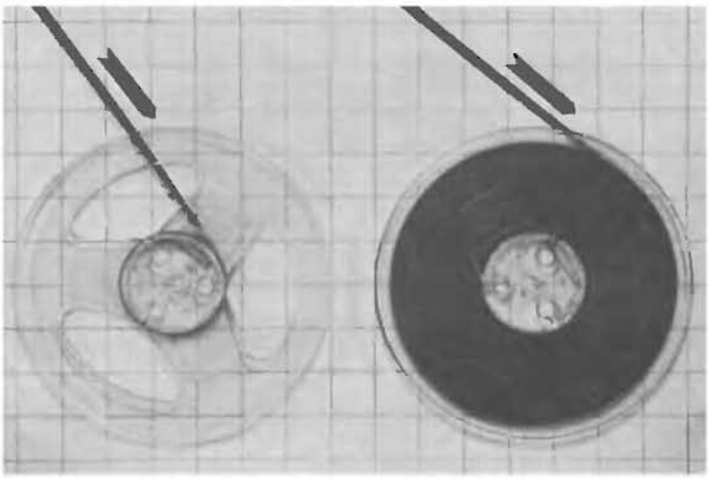


Fig. 1. If the clutch slippage of your take-up reel is poorly adjusted or sticking, the tape pull at the start of take-up (left) can be three times as strong as it becomes later in the take-up cycle.

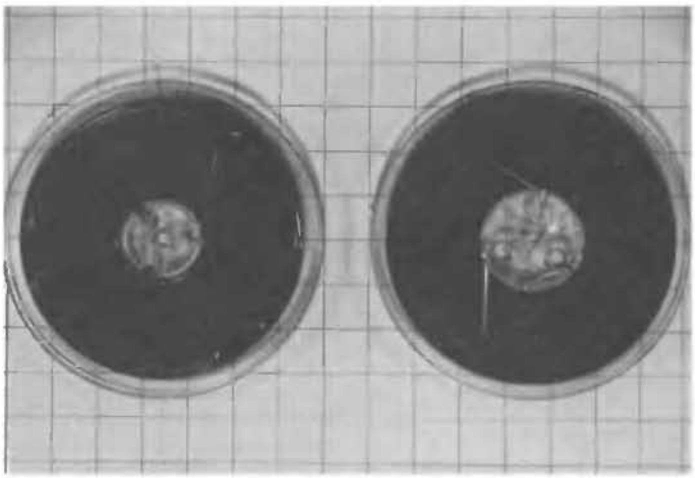


Fig. 2. About ten years ago, the industry increased reel core size from 1.75 inches (left) to 2.25 (right). The loss was playing time but the gain was minimized flutter at the start of play.

HARLOWE

(from page 20)

they must be rewound after each critical service in order to be freed for use in the next. In general you will use them for RECORD only and take the chance in PLAYBACK that your "flutterlessly" recorded tape will play to your satisfaction onto a standard-core reel. If it doesn't, you still have the good tape and can take the special measures to get good play out of it.

Recorder housekeeping

The heavy pull from a practically empty small-core take-up reel is not the only possible cause of flutter at the start of your tape making or playing. At the beginning of operation, your head pole-pieces are stone cold (or at least ambiently cold) and after four or five minutes they will have warmed up considerably from the scrubbing friction of the somewhat abrasive tape. With operating temperature attained, the accumulation of oxide and gunk acquired from casual living and neglect of maintenance may scrub off or melt off and permit the rest of the recording to sound pretty good. There are two recourses.

The first is a five-minute warm-up period *with tape running*. (A two-minute warm-up of motors and rotating rubber is always a good idea).

Or better than warming up the pole-pieces, why not clean them—and then forevermore keep them clean! Use an approved liquid cleaner, and use it often! And if your recorder has pressure pads, scrub them just as often with a stiff dry brush until all of the glaze has disappeared and they are felts again!

Tape selection

All of the "plastic" recording tapes are elastic to an extent, and this elasticity contributes to the general problem of flutter and stiction. The acetate-based tapes are elastic to the breaking point. The untensilized polyester-based tapes (Mylar is the registered trademark of the Dupont brand of polyester) are elastic to the point of permanent elongation.

Thick acetate is obviously less pliant than thin. Thick polyester (1.5 mil) is similarly less stretchable than the thinner (1.0 mil) and still less than the thinnest (0.5 mil).

All other conditions being under control, the thicker the tape, the less the flutter problem—provided that the tape being used is well "lubricated" in manufacture. Manufacturing techniques, formulas, and quality control measures vary from brand to brand, and here should be offered a word of warning about some of the nameless or bargain brands. The bargain tape you may buy may be excellently and uniformly lubricated and satisfactory in every way for high quality music reproduction—or it may be a job lot that failed to pass inspection standards that justify a well-advertised brand name. You'll have to buy it and try it to find out.

The tensilized (pre-stretched) polyester is of course the safest and surest of all buys, not because it is pre-stretched but because it is manufactured to be a premium tape in every respect—including freedom from the flutter which bad tape can exaggerate. The price is high, though, and the tensilized tape should not be needed for excellent music reproduction if all of the other safeguards against flutter are observed.

That old tape recording

Ten or fifteen years ago when tape recording was a rather new idea, things were happening that were just as worthy of being recorded as are the aural events of today. It may be that you have been around this long and that you cherish a few tapes, abominably recorded, that can never be replaced. They were recorded on an old machine, probably inadequately ventilated and these tapes may have been thoroughly cooked in the first run through. Perhaps they flutter from start to finish. Perhaps they are even squealers. A squeal is a stiction-level of flutter, caused just like any other flutter, but its frequency is high enough ((usually 2000-3000 Hz) to become a musical pitch that can be heárd as mechanical sound without the aid of electronics as the tape passes the heads in replay. Ordinary flutter is too low in frequency (about 100-200 Hz) to display this phenomenon.

This squeal may also be recorded on the tape as an electronic signal, but the better chances are that it is not. The initial recording was probably finished before the machine had completed its warm up to the "too high" level. Then the tape acquired its cooking by sitting around on a hot recorder for the damaging hour or so. Then —on the first playback—it was discovered to be a squealer.

If the squeal has *not* been recorded electronically, this tape can often be restored to a condition for fairly satisfactory listening. Run the tape at fast forward across the surface of a sponge moistened with a silicone lubricant, or—if your recorder has pressure pads, apply a touch of the lubricant to the pads and start playing, observing of course all of the safeguards you would apply to the flutterless play of a tape recorded in 1966. You may be very pleasantly surprised. Æ

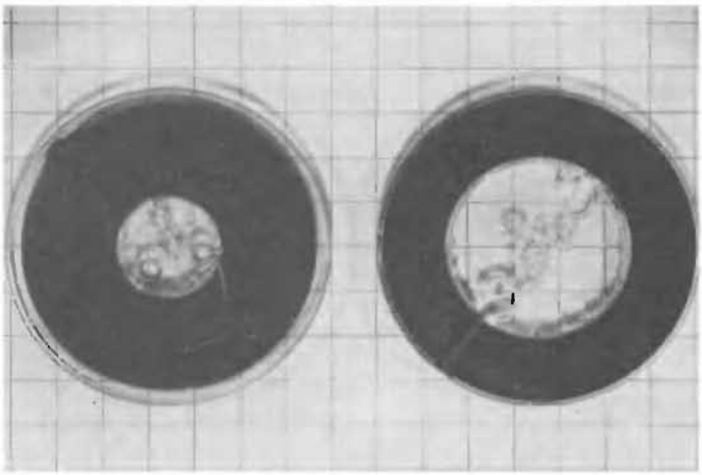


Fig. 3. Special 7-inch reel with 4-inch core (right) will correct starting flutter which is often still present in the use of standard reel (left). If handled with care, special reel will accept standard 1200 feet of 1.5 mil tape.

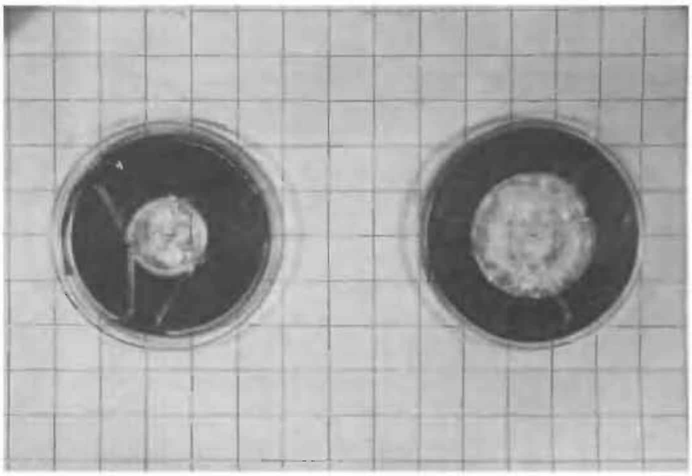


Fig. 4. If your recorder accepts no larger than 5-inch reels, the special reel (right) with 2.75 inch core will minimize starting flutter. With care, this reel will accept standard load of regular 5-inch reel (left) with 1.75 inch core.

Automotive Tape Cartridge Systems

The three tape cartridges shown together to compare sizes. Note lead pencil in foreground for reference.

What about tape cartridges? How do they work and how well do they work. A serious presentation of the characteristics of each of the three types which compete in the automotive field.

For as long as tape has been available there have been dire predictions of how this recording medium would replace the disc as a carrier of music. Well, it doesn't take a keen observer to note that this has not happened yet. Reel-to-reel tape is coexisting nicely, however, but it has not been able to prove itself either as economical or as convenient to use as a long-playing disc.

Almost as long as there have been tape recordings there has been talk of placing tape into a cartridge and thus making it palatable to the "mass-market buyer" of music.

We at AUDIO have sat back as these systems have developed partly because we have felt that none of them is sonically attractive enough to warrant the attention of the serious audio buff. It has become increasingly clear over the past twelve months that the automotive industry has succeeded where the audio sales field has failed. They have put cartridge tape players on the map. The automobile is perhaps a "natural" for a tape cartridge player. Disc players on involved gimbal mounts have not attracted the automotive accessory buyer, and car manufacturers could not market them though they have tried.

But tape in a pre-threaded carrier presents no mechanical problems to car installation. They can be made compact and d.c. operated drive mechanisms of reasonably accurate speed are not hard to design.

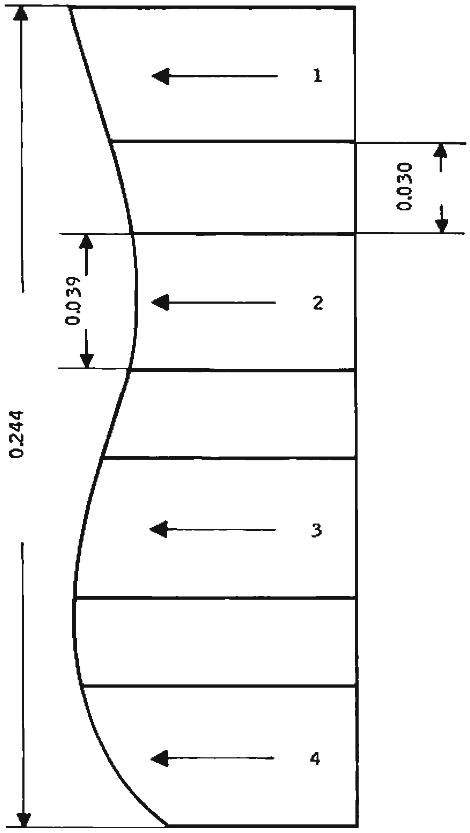
The tape cartridge did not start with the car, to be sure. There have been famous (and infamous) attempts to place cartridge systems before the public. Most of these have involved the placement of reel hubs into a carrier that could, in turn, be easily inserted into a player or recorder. The broadcast industry has long been familiar with closed-loop cartridge systems that could make spot announcements and the like. In fact, one of the present day systems is an outgrowth of a broadcast machine.

This is the Fidelipac system. For professional application there are several sizes of cartridges, each holding a maximum quantity of tape—controlling the maximum playing time. Of course, broadcast versions are full-track player/recorders of sophisticated design, wide-band response, highly accurate and repeatable playing time, and outstanding durability.

It was not much of a trick to standardize on one size of cartridge (the smallest available) and convert the players to four-track operation in stereo. These Fidelipac cartridges are re-entrant type systems. That is, the tape is on a single hub; it is caused to move past the heads by pulling from the *inside* of the hub and placing it back on the *outside* of the same reel of tape. *Figure* 1 illustrates a current Fidelipac-tape cartridge.

This sort of cartridge can be played only in one direction. While it can be played at fairly high speed—thus making it possible to have a fast-forward mode—you cannot play backwards. It just is not possible to pull from the edge of the tape reel and replace tape at the hub end. This necessitates the obvious need to place the two pairs of stereo channels in the same direction. Since we are dealing with an endless loop of tape some method of switching the head scanning from one set to another must be available. In practice this is not too difficult to achieve and has not been a problem of tape cartridges. What is a problem is finding a selection that is buried somewhere within an endless tape that has no identifiable beginning or end. *Figure* 2 shows the track configuration that is standard on these tapes.

The real boom in tape cartridge players did not come until the introduction of the Lear-Jet system. With the cooperative efforts of Lear-Jet (which makes personal jet aircraft as a side interest), Ford Motor Company (which makes automobiles), and RCA (which makes practically everything the other two do not), the Lear-Jet system took off. Basically, this is also a re-entrant cartridge similar in many respects, including size, to the Fidelipac system. It differs, however, in two important areas. First, as *Fig.* 3 indicates, it contains a built-in idler and second—and by far more important—it has eight tracks of information side-by-side on the tape. Thus it offers four pairs of stereo programs—just double what Fidelipac can do on the same tape. Of course, the tracks are squeezed closer together making head design and mounting considerations that much tougher. In *Fig.* 4 can be seen the track layout on these cartridges.



(A)

Fig. 2. Track dimensions and direction of tape travel for the Fidelipac four-track cartridge used in automobile tape playback systems.

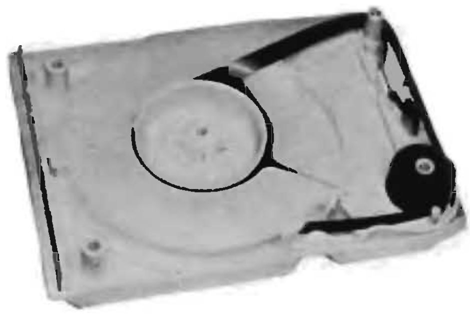


Fig. 3. The Lear-Jet type of cartridge, showing the built-in idler, which contributes to smooth operation. This is an eight-track cartridge, using conventional ¼-in. tape.

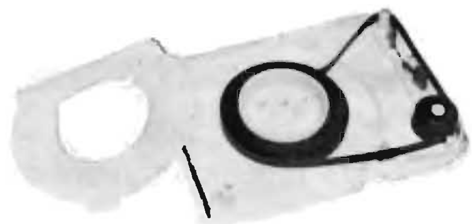


Fig. 4. Same cartridge as shown in Fig., except with tape guiding disc removed.

Fig. 1. The Fidelipac tape cartridge with cover removed.

Track Layout Compared

Figures 2 and 5 compare standard four- and eight-track configurations. Note that in stereo performance on four-track tapes, tracks one and three comprise one stereo pair while two and four are the other. With the eight-track tapes it gets a bit tricky. Tracks one and five, two and six, three and seven, and four and eight are the stereo pairs.

With such close track proximity, crosstalk can become a serious problem. But careful attention to detail can hold separation between channels to 35 dB according to Alfred E. Dusey and Robert A. Wolf of Motorola writing in the January *Journal of the Audio Engineering Society*. Low frequencies are poorer in this respect, of course, but separation can still be held to 30 dB.

Environmental Conditions

One does not often think in terms of environment for a tape player but when one considers the atmosphere that can exist in a car quite a wide range must be accommodated. "The storage temperatures encountered in automobiles in the continental United States range from −40°F to +185°F. Automotive radios (and tape cartridge systems—Ed.) must withstand these conditions, must recover, and must operate over a −20°F to +140°F range.

"Automotive electronic equipment must withstand 40 *g* impact, 5 *g* vibration, and operate well during 1 *g* vibration. In addition, the equipment must operate over humidity ranges up to 95 per cent relative, and d.c. supplies of 11 to 16 volts. It must not be affected by the interference generated by the electrical systems of the vehicle." This is excerpted from a discourse on adapting tape to the automotive environment, written by John P. King of the Ford Motor Company's Radio Engineering Department, in the AES *Journal.*

The Question of Friction

Both the Fidelipac and Lear Jet cartridges work by pulling the tape out from the center hub. This creates friction problems that are different from those encountered in reel-to-reel operation. Now there is a strong *sideways* pull across two surfaces of tape as it is withdrawn from the hub. A speck of dirt introduced on the tape could scratch the emulsion surface during this withdrawal. So it can be seen that cleanliness is vital. Cartridge end covers are supplied but we must wonder how often they will be used, particularly in the less-than-clean conditions that occur in the usual family car.

From the AES *Journal's* report on cartridge systems, "Friction within the tape loop is all-important for correct operation of the cartridge. Excessive friction can cause speed variations and ultimate jamming or seizure of the loop along its path. Insufficient friction can cause similar problems by preventing transmission of rotational energy from the center of the slack loop to its outer periphery, thus resulting in a lack of tape-up tension at the downstream side of the operating loop. With proper design of tape and cartridge this seemingly paradoxical situation can be resolved, so that the take-up function will proceed smoothly without inhibiting supply. . . .

"The action whereby the tape is continually extracted from between a neighboring inner turn and the hub is perhaps the most severe mechanical abuse to which the tape is subjected anywhere in the loop. The geometry of extraction is most difficult to analyze or optimize, but experienced cartridge designers seem to have developed a tape path imposing the least possible torture on the tape. The hub-flange combination is driven by the act of tape removal, the tape driving the hub as a belt drives a pulley. Thus, the outside surface of the hub moves at a linear velocity equal (except for slippage) to that of the tape; the flange, fastened to the hub and moving at the same rotational velocity, has linear speeds at any distance from the center of rotation depending on the radius at that point. For full (400-foot) cartridges of the type considered here, the ratio of linear velocity of the platter (at a radius of about 1¾ in.) to that of outer turns of tape is about 2 to 1. . . .

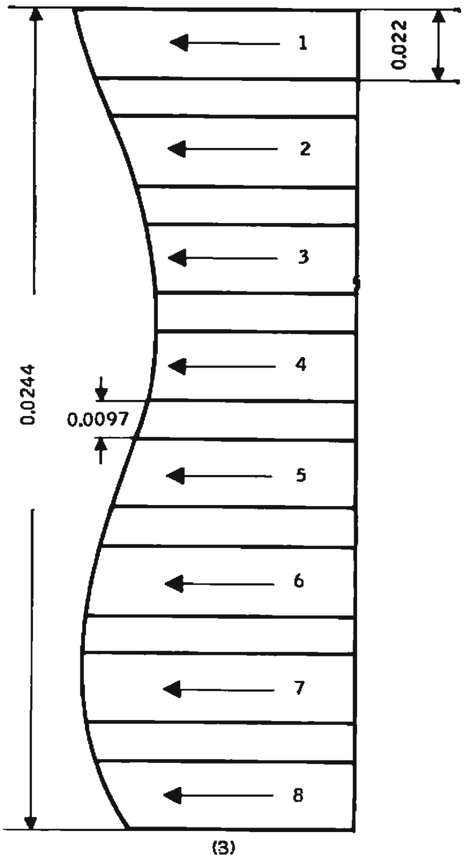


Fig. 5. Arrangement and dimensions of tracks on the Lear-Jet type of cartridge. Note that all tracks travel in he same direction, since the cartridge can not be turned over, just as in the Fidelipac type.

"The force of friction between tape edges and the face of the flange is the means by which the rotational velocity of the flange is imparted to the slack loop, where it provides most of the initial take-up force. The fact that the flange is overdriven with respect to tape velocity also results in a significant force which tends to separate the turns of the tape pack. This force, of course, is also transmitted to the tape from the flange by means of tape-edge friction.

"Any appreciable frictional drag on the tape along its path from the pressure roll to the outer periphery of the slack loop tends to inhibit the take-up function. It is also reflected into the slack loop, where it appears as a part of the back tension at the input side of the capstan.

"From these considerations it is readily recognized that the cartridge and tape function together intimately as a system, and that it would be foolish to undertake the design of one without full consideration of the other." Raymond C. Smith and Peter J. Vogelgesang of 3M.

These considerations are, of course, undertaken. However, frictional problems and the attendant flutter and wear do occur in re-entrant cartridges. Their minimization does not obviate the fact that the problem exists. It should be pointed out that while these cartridges do not appeal to the audio buff as a medium of superior performance, they are attractive to and sufficiently durable for the automobile owner seeking programmed music on the highway.

We have avoided any discussion of frequency response simply because these cartridges operate at 3¾ ips. and it has already been shown that this speed is entirely satisfactory for wide-band reproduction of music.

Philips

The Philips system, available here under the US logo, Norelco, and a number of licensed manufacturers, differs from the two just discussed in a number of ways.

First, it is a reel-to-reel cartridge. Second, it eschews the usual ¼ in. tape stock to use a tape width of 0.15 in. Third, it operates a speed of 1⅞ ips. There are other differences to be discussed later.

In a reel-to-reel cartridge you can place the hubs closer together if you eliminate the reel flanges. RCA did this in their earlier cartridge, Philips has done the same. *Fig.* 6 shows the Philips unit (they call it a "cassette" as a distinguishing term) in relationship to the others. As can be seen, it is much smaller. So much so that four cassettes will fit into the plastic carton designed to hold one Lear-Jet unit.

Reel-to-reel operation offers the obvious advantage of fast-forward and rewind modes. This leads naturally to a system that lends itself to recording operation as well as the playback of commercially-recorded tapes. (This publication does not use the redundant term—*pre-recorded.* Something is either recorded or it is not recorded, hence blank.)

Figure 7 shows the track configurations used on these cassettes. Note the fact that tracks one and two are one stereo pair, while three and four are the other. The result is that a *mono* half-track player will scan both stereo channels providing an effective mono signal. (We have not experimented with this as yet, but we still remember the early days of two-track stereo. As often as not the two channels were out of phase with each other resulting in gross distortion when they were scanned by a full-track head. Whether this problem exists at all remains to be seen.)

It becomes obvious that when you reduce the width of the tape and thus reduce the width of each track you increase the signal-to-noise ratio and you increase the tendency to crosstalk from one channel to the next. If you will recheck *Fig.* 5 you will see that the track width and distance between tracks on the cassette tapes is similar to the configuration of the eight-track cartridge tapes. There is one important difference, however. On the eight-track tape the adjacent tracks carry different information; any audible crosstalk (and 30 dB down *is* audible) will be highly objectionable, while the four-track Philips tape only serves to reduce *stereo* separation somewhat. And 30 dB of stereo separation, even the 20 dB claimed to exist at 200 Hz, is quite good.

Fig. 6. The Philips cassette with cover removed. This is a 4-track cartridge, using tape 0.15 in. wide in a reel-to-reel configuration, permitting play in either direction.

The Philips system seems to be gathering force in home-type and portable equipment while the two cartridge systems move forth in automobiles. Certainly it is our feeling that the cassette with its record facility offers features that the re-entrant cartridge does not.

Recorded Music Tapes

RCA sparked the cartridge wars with their endorsement of the Lear-Jet system (while in no way abandoning their own system which they still promote). Most of the other record manufacturers have followed suit with eight-track cartridges. All along there have been a smattering of Fidelipac cartridges, the amount grows daily. A recent announcement, in fact, states that record giant Columbia is releasing (with caution) some four-track cartridges. (They have been releasing eight-track cartridges.)

The entry of these divergent systems has created a unique packaging arrangement for the record companies. While the major companies are their

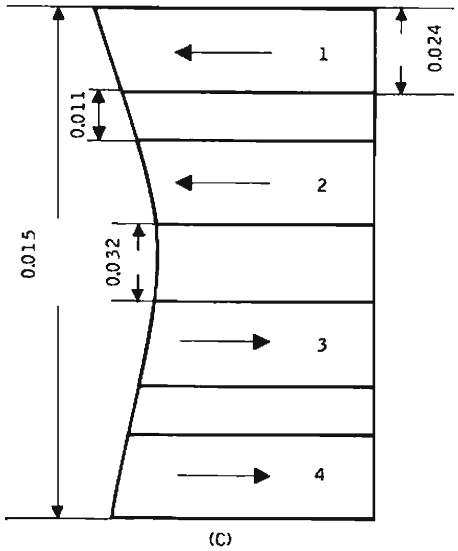


Fig. 7. Track dimensions and direction of travel in the Philips-type cassette. Note that tracks 1 and 2 constitute the right and left channels in one direction, and tracks 3 and 4 in the other. Thus a mono signal can be obtained by using a head which scans two adjacent tracks simultaneously.

own disc manufacturers and even the minor companies will do their own mastering, the tape-cartridge field has spawned a number of independent cartridge tape duplicators. Operating on tape masters supplied by the record companies they process the cartridges and are active in their actual distribution. One of the largest is Ampex, which spews out any number of tapes for any number of companies. But they are by no means alone.

The result has been a less identifiable (by sonic characteristics) product. Of course, the import of this to the consumer is one that the consumer must decide for himself. Another thing he must determine on his own is which of the cartridge systems is the right one for him. (Perhaps none of them is.) Obviously if the record companies cannot decide which system to back, how can the consumer? We should mention that compatible stereo/mono commercially recorded tapes are beginning to appear in cassette form. Mercury Records, the U.S. outlet for Philips, is in the forefront but others are joining in.

Each of the systems has its virtues and vices. Each offers sufficient playing time to eliminate that as a factor. The eight-track cartridge will offer twice that of its four-track brother (up to 400 feet or ⅔ hour × 4 or 2). This is continuous play in one direction on a machine that will automatically switch tracks.

The cassette, however, offers a maximum of 60 or 90 minutes of play dependent on which of two types you buy. The differences are in tape thickness. But this time is split because the cassette must be flipped over to play the second track(s) as with any reel-to-reel machine. In time there may be automatic reverse play and record units just as there are with conventional reel-to-reel machines but this is not in the offing now.

It might be inferred that there is no compatibility between systems and this is not far from the truth. It is possible to design Fidelipac/Lear Jet compatible players (and some have appeared.)

Certainly the engineering problems are not formidible since the cartridges are so close in size and design.

But the Philips cassette stands quite alone. It is not interchangeable with any other system and is not likely to stray from that position.

More Confusion

We have only touched on the three systems that seem to have captured the public's fancy—fickle though that is. We don't mean to imply that these are the only systems in existence. The RCA reel-to-reel cartridge using standard tape at 3¾ ips and four-tracks is still at hand. So is a 3M system including players that automatically change cartridges. Then there is an Orrtronics system which uses a cartridge much like the Fidelipac unit except that the tape travels at right angles to the hub, while it is traversing the play head. While there are legitimate engineering claims to support this system, we doubt that it will receive general acceptance. Perhaps not because it is (or isn't) better, rather because it is too late.

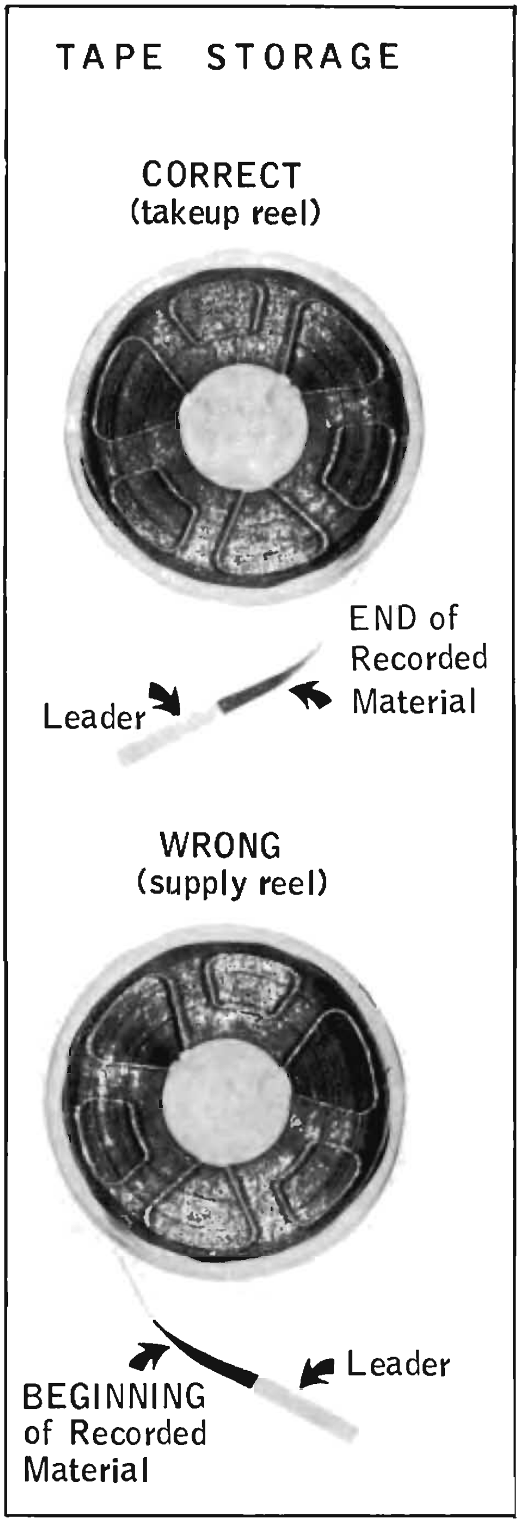
The battle lines have been drawn. Fidelipac four-track is on one side grappling with Lear-Jet's eight-track system. Philips' miniature cassette sits on the other side fighting with both of them. Who will win?

Prognostications are dangerous things. They are always better made with the advantage of hindsight, something we have not had as yet. But we can make the estimate that this battle will go on for some time to come and we do feel that it is actually drawn as indicated so that the ultimate winners will prove to be one of the two cartridge systems and the cassette system. For each offers specific advantages that the other cannot. And each can be directed at different markets. We see the cassette appealing to the audio buff that wants to do his own recording while the cartridge remains a medium for commercially recorded music. Neither, in our estimation offers a serious threat to the disc as a purveyor of quality music recording. We may eat these words someday, but the table is not yet set. Æ

The new NAB magnetic tape standards

Part 1 HERMAN BURSTEIN

The National Association of Broadcasters' reel-to-reel magnetic tape recording and reproducing standards are examined in depth. Comparisons with older NAB standards and current RIAA practice place today's NAB standards in proper perspective.



IN JUNE 1953, NAB (National Association of Broadcasters) issued *Recording and Reproducing Standards* for magnetic (and disc) recording. Over the next dozen years a gap gradually developed as these standards were outstripped by advances in a relatively new art—better tape, better heads, better electronics, better transports, in-line stereo heads, quarter-track format, etc. Many in the industry tended to follow the practices of its leaders so that a set of shared practices grew up amounting to de facto standards. However, there was not the degree of conformity that official standards command and that work to the advantage of the consumer. To illustrate, for a long time the purchaser of prerecorded tape could not be sure that it would be matched by the playback characteristic of his tape machine to yield flat frequency response. To close the gap, the NAB issued in April 1965 *Magnetic Tape Recording and Reproducing Standards, Reel-to-Reel*.

Standards can be very confusing. In this and succeeding articles, we shall try to clarify these standards, closing the information gap suffered by many tape enthusiasts by discussing the new NAB standards. We shall dwell on all points that we believe are of interest and significant to the home tape recordist. Where appropriate for a complete view of a point, we shall compare the new standards with the old, as well as with *Standards for Magnetic Tape Records* issued by RIAA (Record Industry Association of America, Inc.) in July 1965. Whether directly quoted or paraphrased, the new NAB standards are indented to separate them from our comments.

To catch our errors in interpreting the 1965 NAB standards, Mr. John G. McKnight, an Ampex Staff Engineer who participated in developing these standards, has been kind enough to review our discussion. However, responsibility for any remaining errors and for comments on the standards is solely ours.

Magnetic Tape Dimensions

> The 1965 NAB standard sets the width at 246 mils ± 2 mils.

Thus the tape may vary between 244 and 248 mils. The rated width of 246 mils agrees with the 1965 RIAA standard and with industry practice since 1959. The 1953 NAB standard permitted a broader range, 244 to 250 mils. Narrowing the range of tape width helps reduce problems of incorrect azimuth and of poor tape-to-head contact due to tape skewing or cupping as it passes through guides too wide or too narrow.

> Total thickness of magnetic tape may not exceed 2.2 mils. Standard reel diameters, hub diameters, and minimum lengths of tape to be supplied on these reels are:

Reel Diameter	*1.5 Mil Base*	*1.0 Mil Base*
3"	125 ft.	200 ft.
5	600	900
7	1200	1800
10.5	2500	3600
14	5000	7200

Tape with a ½ mil base is "not recommended" except for the 3-inch reel supplying 300 feet.

Tape Uniformity

> For constant signal level applied to the tape the signal level recorded on the tape shall be "uniform within ± 0.5 dB throughout a given reel." The test speed is 7½ ips, and the test signal is 400 Hz, recorded at a level matching the "NAB Standard Reference Level."

This level is a specified tone on a special test tape, to be discussed later in connection with signal-to-noise ratio.

> Bias is adjusted to obtain maximum recorded 400 Hz signal by playing back the tape and using a standard VU meter that reads average output.

Magnetic Tape Wind

> The tape shall be wound with the oxide-coated surface facing toward the hub of the reel. Recorded tape normally should be wound so that the start of the

program material is at the outside of the reel.

This "head out" or "tail in" winding permits the tape to be played immediately. However, for long storage and for other reasons, the NAB standard advocates "head in" winding; now the end of the program is at the outside of the reel, so that the tape must be rewound before it can be played. Tapes stored in this manner shall be clearly marked to prevent accidental playing in the reverse direction.

A footnote explains that tapes stored "head in"

> will have slightly less preprint than postprint. This is generally desirable because postprint tends to be masked by the program material and reverberation effects. Also, rewinding a tape immediately before playing tends to reduce print-through. Another advantage of rewinding before playing is that stresses are relieved and any adhesion of adjacent layers will be eliminated. A further advantage is that tape wound on the take-up reel in the play mode of operation usually is wound more smoothly than when wound at high speed. Therefore, there is less chance of damage during storage or shipment or due to temperature and humidity changes.

However, smooth winding can also be had with "head out" storage. A tape recorded and played in both directions, as is common in the home, will enter storage slowly and smoothly wound. In the case of one-way tape, if time is no problem, the tape can be slowly rewound by putting it through the playback process.

Four-Track Stereo Format

> When tape is first unwound from the supply reel and moves from left to right with the coated side facing away from the observer, the track numbers from top to bottom are 1, 2, 3, 4. Tracks 1 and 3 are used simultaneously for one direction of tape travel; tracks 2 and 4 for the other. Tracks 1 and 3 are used first. Tracks 1 and 4 carry the recording for the left channel, as viewed by an audience; tracks 2 and 3 carry the recording for the right channel. Tracks 1 and 3 (and similarly tracks 2 and 4) are recorded with in-line gaps, producing in-phase signals on the tape.

The in-phase requirement means this: If stereo playback equipment produces in-phase signals from a full-track tape, it will also produce in-phase signals from properly recorded stereo tracks. (Mr. McKnight adds: Although not so indicated in the NAB standard, the phasing specification only applies to low frequencies.")

The four-track stereo provisions agree with the RIAA standard. The latter further requires leaders at least 3 feet long at each end of a recorded tape. If these leaders are used for identification, RIAA states they are to be yellow at the beginning (start of tracks 1 and 3) and red at the end (start of tracks 2 and 4).

The NAB standard is quite clear as to track width but less clear as to width of the islands between tracks.

> The recorded tracks for four-track recordings shall be 0.043 + 0.000 —0.004 inches in width. The center-to-center distances between tracks 1 and 3, and between tracks 2 and 4, shall be 0.134 + 0.002 —0.000 inches. The four tracks shall be equally disposed across the tape with a tape width of 0.244 inches and the outer edges of tracks 1 and 4 coincident with the edges of the tape.

The pieces of this puzzle apparently can be put together as follows:

1. Track width shall be between 39 and 43 mils.
2. The distance between the top of track 1 and the bottom of track 4 shall be 244 mils, corresponding to minimum tape width.
3. The 4 tracks shall be of equal width, and the 3 islands between tracks shall be of equal width.
4. Inasmuch as the 4 tracks can occupy between 156 mils (4 × 39) and 172 mils (4 × 43), the 3 islands share what remains from a total of 244 mils—between 72 and 88 mils. Dividing this remainder by 3, island width can vary between 24 and 29⅓ mils, depending on track width.

But a piece of the puzzle seems left over: the NAB center-to-center provision. From the center of track 1 to the center of track 3 equals a span of 2 tracks and 2 islands, which may total 134-136 mils; hence 1 track plus 1 island may total 67-68 mils. However, this provision seems redundant in view of Point 4 above. According to Point 4, a maximum track width of 43 mils requires an island width of 24 mils so that 4 tracks and 3 islands will exactly fill 244 mils; thus a track and an island total 67 mils. And a minimum track width of 39 mils requires an island width of 29⅓ mils, for a total of 68⅓ mils. So the center-to-center provision appears unnecessary; to boot, it presents a trivial contradiction involving ⅓ mil.

The RIAA standard is essentially the same as the NAB one except that RIAA does not require total track span to be exactly 244 mils. Instead it sets 244 mils as a maximum. RIAA also makes the center-to-center provision, which now is necessary to define island width. Accordingly, depending on track width, island width may vary from 24 to 29 mils, and total track span may vary from 240 to 244 mils.

The differences between the NAB and RIAA track requirements are very minor and present no significant problem as to correspondence of tracks under the two standards.

It may be noted from the foregoing that the tape (246 mils) is slightly wider than the space spanned by the head gaps (244 mils). Mr. McKnight points out that "the purpose for recommending head width *less* than the tape width is to eliminate the problem of 'grooving' of the head at the edge of the tape. If the tape edge is outside of the core, no groove is produced, and the need for re-polishing the face of the head is eliminated."

Two-Track Stereo Format

Under the NAB standard a four-track stereo head is incompatible with two-track stereo recording, unless there is provision for moving the head down. Whereas the lower gap of a four-track stereo head will nominally span the region 134-177 mils from the top edge of a 244 mil tape, track 2 of a two-track stereo recording nominally begins 159 mils from the top. Thus 25/43 of the lower gap will span unrecorded space, with consequent serious deterioration of signal to noise ratio—unless the head can be moved down for two-track stereo.

On the other hand, the RIAA standard permits, though it does not require, compatibility. A specific track width is not stated; only a minimum island width of 30 mils is provided. In this case, track 2 of a two-track recording begins 137 mils from the top of a 244 mil tape, corresponding very nearly with the lower gap of a four-track stereo head, which starts 134 mils from the top.

Mr. McKnight comments: "It was felt that two-track should be considered a 'professional mastering' format, and that to degrade crosstalk by narrowing the island, in order to have reproducibility on home equipment, was not in the best interests of the broadcasting industry. The RIAA specification . . . I'm sure, is to allow two-track/four-track compatibility, especially on home-grade two-track recorders. There's nothing to say that U.S.A. two-track recorders for the *home* market shouldn't use the narrower island; NAB is for broadcasters."

Performance specifications of the new standard will be examined next month. Æ

PART 2

The New NAB Magnetic Tape Standards

HERMAN BURSTEIN

Tape speeds, and playback-recording characteristics outlined in NAB standards are examined in this installment

IN THE FOLLOWING discussion we are largely concerned with NAB reel-to-reel, magnetic tape performance specifications. These principally apply "to all high quality magnetic recording and reproducing equipment used for music and speech programs where superior performance is of primary importance." There is also a brief section applying to "special purpose limited performance systems."

Tape Speeds

Of key interest, reflecting progress of the tape art, is the new, preferred tape speed.

> "It shall be standard that the preferred tape speed be 7½ inches per second."

This contrasts with the 1953 NAB standard, which designated 15 ips as the "primary standard" and 7½ ips as the "secondary standard."

> The 1965 standard designates 15 ips as a "supplementary tape speed."

In 1953, 30 ips played the same role.

> The 1965 standard further designates 3¾ ips as a "supplementary tape speed."

No mention of 3¾ ips appeared in the 1953 standard.

> Tape speed tolerance is ±0.2%, applying to any portion of the reel of tape in use.

The RIAA standard permits a slightly greater tolerance, ±0.3%. The 1953 NAB standard said nothing on this point.

At the risk of repetition, it is necessary to anticipate our discussion of signal-to-noise ratio to help make clear why 7½ ips has become the preferred speed and 3¾ ips has risen to the status of a supplementary tape speed. The NAB S/N specifications are just as high at 7½ as at 15 ips on an unweighted basis (giving equal weight to all frequencies in the audio spectrum when measuring noise). In fact, on a weighted basis (reflecting normal hearing characteristics at low volume and, therefore, greatly deemphasizing noise in the bass region and moderately deemphasizing noise in the treble region), slightly higher S/N is specified for 7½ than for 15 ips.

> On a weighted basis, 7½ ips permits a higher S/N ratio because playback equalization "remains the same for both speeds while the tape noise increases with tape speed."

For 3¾ ips the NAB standard specifies S/N that, on a two-track or four-track basis, is overall nearly as high as for 7½ ips. In fact, for two-track operation, specified S/N is 1 dB higher for 3¾ ips than for either 7½ or 15 ips when noise is unweighted; and 1 dB higher than for 15 ips when noise is weighted. Only on a full-track basis is specified S/N significantly less for 3¾ ips than for the higher speeds.

Playback and Recording Characteristics

The tape playback head is a "velocity" device, responding to rate of change of signal; the lower the frequency, that is, rate of change, the lower the head output voltage. Thus for constant magnetic flux (equal level of recorded signals) on the tape, head output voltage essentially declines in proportion to frequency —bass loss. In recording, there are serious treble losses owing to various magnetic phenomena. Altogether, the record-playback response of a tape system in the absence of equalization is a camel hump: high in the middle and low at the bass and treble ends. Therefore the tape amplifier must provide bass boost and treble boost to restore flat response. But the amount of boost can vary in two ways:

(1) It can be provided either in recording or in playback or in a combination of the two. (2) The machine may contain the minimum amount of equalization necessary for flat response; or, to get as much signal on the tape as is consistent with tolerable distortion and thus improve the S/N ratio, the manufacturer can increase recording boost above the minimum amount required. Because equalization at any given speed can therefore vary all over the lot, it is eminently desirable to have standard recording and playback equalization to permit interchangeability of tape among machines and to achieve optimum practice (best compromise among the conflicting requirements of wide frequency response, low distortion, and high S/N ratio).

If one specifies a playback characteristic and at the same time specifies flat record-playback response, the recording characteristic is implicitly defined. Such is the practice in tape recording, because it is easier to measure the playback characteristic (boost or cut applied to a signal coming off the tape) than the recording characteristic (boost or cut in the magnetic flux actually recorded on the tape).

Next month the author will examine equalization curves.

The New NAB Magnetic Tape Standards

HERMAN BURSTEIN

Playback, record, and frequency response characteristics are examined in this installment

PART 3

THE NAB PLAYBACK characteristic is largely but not entirely an equalization curve provided by the playback amplifier. Rather, it is the combination of such a curve plus frequency irregularities (treble losses and bass losses or gains) that a particular playback head exhibits (as heads in general do). This *combination* must achieve a specified playback response with respect to constant flux on the tape. If there were an ideal head—with no response irregularities—the playback equalization and the playback characteristic would be identical.

In practice, with high quality heads, equalization provided by a playback amplifier comes within a few dB of the total playback characteristic specified by NAB.

One playback characteristic is specified for 7½ and 15 ips. A second characteristic is specified for 3¾ and 1⅞ ips.

For 7½ and 15 ips, the playback characteristic is in effect: bass boost commencing (3 dB up) at 3180 Hz, rising 6 dB per octave as frequency declines, and leveling out (3 dB below maximum boost) at 50 Hz. In microseconds, the specified turnovers are 50 and 3180 μsec.

For 3¾ and 1⅞ ips, the playback characteristic is in effect: bass boost commencing at 1770 Hz, rising 6 dB per octave as frequency declines, and leveling out at 50 Hz. The specified turnovers are 90 and 3180 μsec.

The 1953 NAB standard provided a playback characteristic only for 15 ips, and this was the same as the present characteristic for 7½ and 15 ips. The RIAA standard provides playback characteristics only for 7½ and 3¾ ips, and these are the same as the NAB ones. Inasmuch as the NAB standard does not show the playback characteristics in the manner we have described, whereas the RIAA standard does, we present here RIAA Fig. 1. Note, however, that the NAB tolerances, which we shall describe later, are somewhat different and on the whole tighter, than the RIAA tolerance of ± 2 dB.

The NAB playback characteristics, while *equivalent* to our description and to RIAA (Fig. 1) and to NAB in the 1953 standard, are *actually* presented as shown in Fig. 2. There are two reasons for the new form of presentation: (1) It helps get away from the misconception that the playback characteristic is merely a specific equalization curve in the playback amplifier without regard for response irregularities of the playback head. (2) It serves to show how the flux recorded on the

(Continued on page 38)

Fig. 1—RIAA reproducing characteristics for magnetic tape.

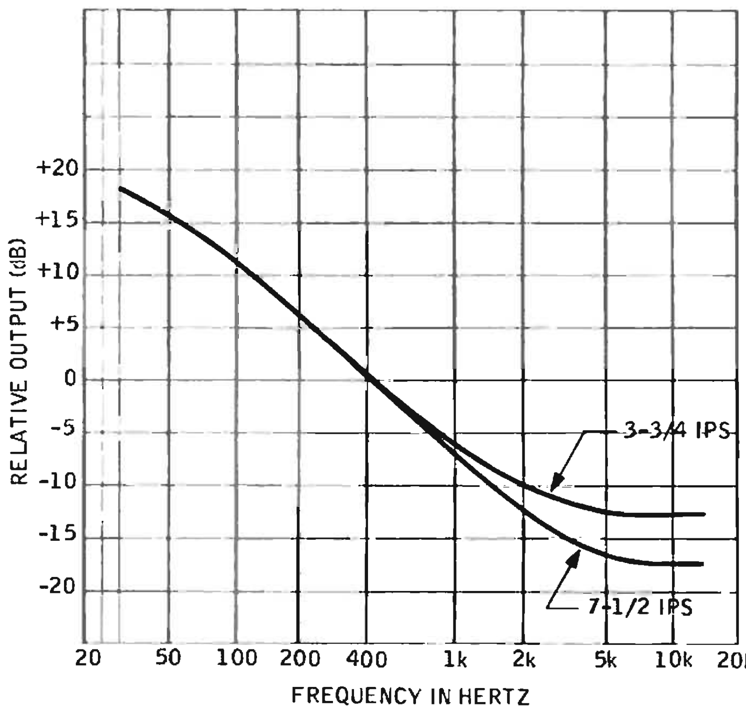
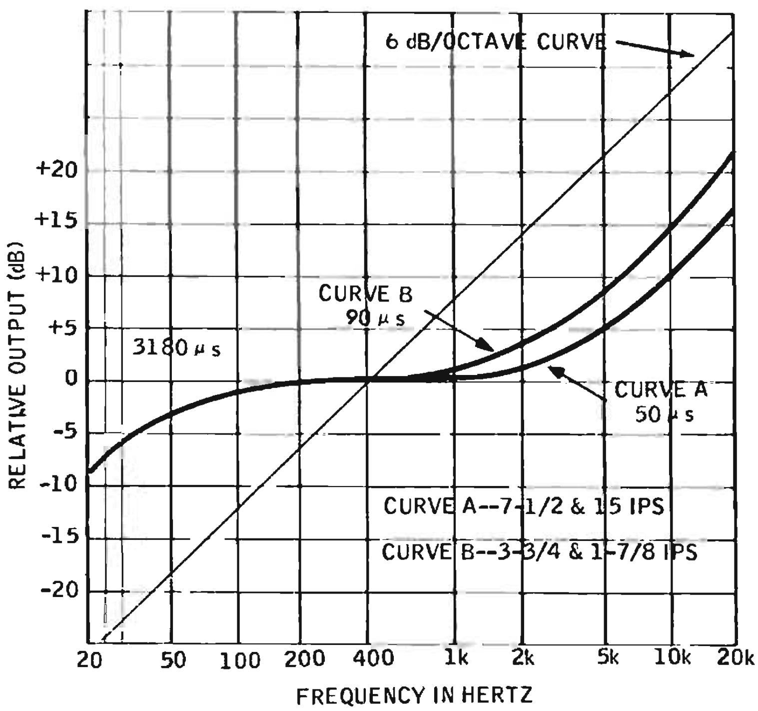


Fig. 2—NAB reproducing characteristics for magnetic tape.



The New NAB Standards

(Continued from page 38)

plicity and elegance at the time of writing the NAB Standard):

["A Recorder Flux Characteristic, which is the tape flux produced by constant input voltage to the Recorder. (A concept not used by NAB.)

["A Reproducer Flux Characteristic, which is the output voltage produced by constant flux input to the Reproducer. (Fig. 2.)

["A Relative Reproducing Characteristic, which shows the deviations (or allowable deviations) of a system from the Standard Reproducing Flux Characteristic. (Fig. 3.)

["A Relative Recording Characteristic, which shows the deviations (or allowable deviations) of a system from the Standard Recording Flux Characteristic. (Fig. 4.)"]

Frequency Response Specifications

When measuring playback response in the manner previously described (by playing the NAB Standard Test Tape for a particular speed), response shall be within the following limits referred to 400 Hz: At all speeds, +1 db between 20 and 20,000 Hz. At 7½ ips, −1 db between 100 and 10,000 Hz, and dropping to −3 db at 30 and 15,000 Hz. At 15 ips, −1 db between 100 and 15,000 Hz, and dropping below this range to −3 db at 30 Hz. At 3¾ ips, −1½ db between 100 and 7500 Hz, and dropping outside this range to −3 db at 50 Hz and −4 db at 10,000 Hz.

Recorded response, measured in the manner previously described, shall be within the following limits referred to 400 Hz: At all speeds +1 db between 20 and 20,000 Hz. At 7½ ips, −1 db between 30 and 10,000 Hz, and dropping above this range to −2 db at 15,000 Hz. At 15 ips, −1 db between 30 and 15,000 Hz. At 3¾ ips, −1½ db between 50 and 7500 Hz, dropping above this range to −4 db at 10,000 Hz.

(Continued next month)

PART 4

The New NAB Magnetic Tape Standards

HERMAN BURSTEIN

FREQUENCY RESPONSE AND SIGNAL-TO-NOISE RATIO MEASUREMENTS

LAST MONTH, WE CONCLUDED PART 3 of this series by examining frequency response specifications of the NAB standards for magnetic tape. Response specs were enumerated. The NAB response curves are repeated here to complement the following summation, which further clarifies recorded response. See Fig. 1.

Thus the distance of Curve A *above* the 0 dB line equals the *decline* of recorded flux at high frequencies. And the distance of Curve A *below* the 0 dB line equals the *boost* in recorded flux at low frequencies. This decline and this boost in recorded flux are implicitly part of the NAB standard.

Measuring Frequency Response

Because practical heads are not ideal heads, the playback characteristic of a tape machine cannot be checked simply and directly by measuring equalization of the playback amplifier. Instead, it is necessary to follow the indirect course of checking frequency response when playing an "NAB Standard Test Tape."

"The signal frequencies are recorded on these tapes in such a manner that they would supply a constant output level when reproduced on an ideal reproducing system." (The Ideal Reproducing System is one with the NAB playback characteristic, as measured by special methods described in an Annex of the NAB standards.)

It is impractical to measure recorded flux directly, except by laboratory methods. Therefore the following indirect procedure is provided by NAB.

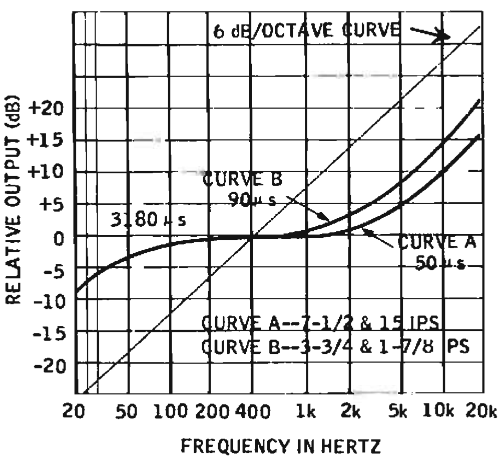
(1) Playback response is measured when playing an NAB Standard Test Tape. (2) Record-playback response is measured, using "normal operating bias" and recording frequencies at the same level as on the NAB Test Tape. (3) The difference between measurements 2 and 1 is the "recorded response."

"Recorded response" is therefore the deviation from the recording characteristic implied by Fig 1—that is, from the complement of Curve A or Curve B, depending on tape speed.

In the case of playback response, the NAB standard recommends a roll-off at the rate of at least 6dB per octave outside the frequency limits specified for each speed.

The standard further states that when measuring playback response with a half-track or quarter-track head, "a low frequency boost may be expected" inasmuch as the NAB Test Tape is full-track.

Fig. 1—NAB reproducing characteristics for magnetic tape.



Such boost is due to fringing. That is, at long recorded wavelengths (low frequencies) the head tends to respond to more of the tape than is spanned by the gap. The standard does not indicate the magnitude of boost due to fringing inasmuch as this depends upon the individual head. It only says:

"Refer to the instructions supplied with the test tape for further details."

Inasmuch as the NAB Test Tapes were not yet available when this was written, no further explanation is offered here, except to note that with a playback head of high quality, fringing boost should not exceed 2 or 3 dB around 50 Hz.

It is noteworthy that the NAB standard gives no specifications for record-playback response, except those which may be inferred by combining the separate specifications for playback response and recorded response. Instead, the standard states:

"The recording equalization of a recorder/reproducer should be adjusted for an over-all response which matches as nearly as possible the response of the reproducer from the NAB Standard Test Tape. This response is standardized, rather than the simple over-all record-reproduce response, in order to assure better interchangeability of recorded tapes."

To illustrate, assume that a tape machine exhibits a 5 dB peak at 10,000 Hz when playing the NAB Test Tape. Record equalization should be such as to produce the same 5 dB peak at 10,000 Hz on a record-playback basis. If record equalization results in the same playback response as does the NAB Test Tape, the machine's recording characteristic necessarily matches the NAB standard.

In other words, it is undesirable that an error in the playback characteristic (for example a treble peak) be compensated by a complementary error in recording (a treble dip). If there are complementary errors, the machine stands doubly isolated from machines that meet the NAB standard: It cannot play their recordings satisfactorily. Its own recordings cannot be played satisfactorily on these other machines.

Measuring signal-to-noise ratio

To measure S/N one must, of course, have a reference level—an audio signal of specific magnitude with which the noise level can be compared. For example, in measuring a power amplifier, the reference level may be the unit's maximum rated power. Or, to apply a common denominator to all amplifiers, it is frequent practice to use 10 Watts output as the reference: The amplifier is driven hard enough to produce 10 Watts, the corresponding output voltage is measured, the drive signal is removed, and the remaining noise voltage is measured. S/N, ordinarily expressed in dB, is the ratio between the measured audio and the noise voltages.

For other components, an appropriate reference level becomes problematical. What should be the reference for measuring a preamplifier's S/N on magnetic phono input? The measurement will depend on the magnitude and frequency of the input signal. Therefore it has become frequent, although not universal, practice to base S/N upon the preamp output when it receives a 5 millivolt, 1,000 Hz input signal (which roughly corresponds to the average output of the average magnetic phono pickup when playing the average high quality phono disc).

An appropriate reference level becomes still more problematical for tape recorders. The 1953 NAB standard said that it should be the output level obtained when playing back a tape containing a 400 Hz tone recorded at a level producing 2% total harmonic distortion, with the "recording system operating under normal operating conditions . . . (and) using tape that is normally available."

But what are "normal operating conditions"? Tape recorder manufacturers vary in their frequency response objectives and accordingly, in the amount of bias current employed. The result is differences in the amount of 400 Hz signal impressed on the tape for given distortion. And what is "tape that is normally available"? For a given frequency response at a given tape speed, the amount of 400 Hz signal recorded at given distortion tends to vary according to kind of tape used —conventional, high-output, low-noise, etc. And it tends to vary according to tape coating thickness. True, with the exception of high-output tape, these differences in recorded level are not very profound. They tend to be on the order of 1, 2, or perhaps 3 dB. But in an art where every improvement of 1 dB is virtually an engineering triumph, there is a need to be quite precise about the reference level.

Therefore the new NAB standard seeks to introduce a reference free from variations in operating conditions and in the tape used. In effect it seeks to postulate as the reference a specific amount of magnetic flux presented to the playback system.

"The NAB Standard Reference Level shall be that 400 Hz level which is equal to the recorded level on the NAB Primary Reference Tape." The Primary Reference Tape, made under laboratory conditions, "is a tape of the normal general-purpose type which has been selected for average characteristics of output, sensitivity and distortion. The 400 Hz recording on it was made at 7½ ips with bias adjusted for maximum output (at 400 Hz), at an output level 8 dB below that which produced 3% third harmonic distortion . . . It is . . . a practical convenient method of specification consistent with the magnetic recording and reproducing process. Since neither the tape nor the measurement conditions can be duplicated exactly in the field, all NAB Standard Test Tapes contain a 400 Hz recording at the NAB Standard Reference Level within ±0.25 dB as a means for making this level available."

In sum, the new NAB reference level for S/N measurement is a 400 Hz tone on an NAB Standard Test Tape. The same reference level is used on the seperate test tapes for 7½, 15, 3¾, and 1⅞ ips.

John G. McKnight, Ampex staff engineer who was on the NAB committee that formulated the new tape standards, comments on reference levels, as follows:

"Since, in magnetic tape recording *applications*, it is common to express levels in decilogs ("decibels"), rather than using the basic units themselves, it would be convenient to have a reference quantity for a magnetic recording decilog system. (This reference quantity would be comparable to 1 mW as a reference quantity for the dBm, or 1 V as the reference quantity for the d1V.) Such a quantity may be arbitrarily chosen; it is convenient that it be the basic SI unit (Wb/m) or a decimal multiple of it; and that it be in the order of magnitude of commonly-used flux values, namely, 150 to 320 nWb/m. Thus, a convenient quantity for a "decilogs, flux" would be 0 d1 = 100 nWb/m.

"In a practical recording and reproducing system, the levels are indicated on some sort of "volume indicator"; for example, a VU meter, a quasi-peak-reading meter, etc. The choice of flux level for the "operating level"—that is, the flux level when the meter points to its "0 d1" mark—it is an operating quantity determined by experience with a recording system. When recordings are to be interchanged, as in broadcasting applications, and with master tapes for phonograph disc manufacturing, it is very desirable that a uniform "operating level" be adhered to. Surpisingly enough, most of the standards—BS, EIA, IEC, CCIR, RIAA and SMPTE—make absolutely no mention of an operating level. Those who do consider an "operating level"—Ampex Corp., DIN, and NAB,—do not employ uniform terminology and practices.

"The Ampex Reproducer Test Tapes contain an "operating level" section in the sense defined above. The NAB "standard reference level" is identical to the NAB "standard recorded level," and is, in fact, also an "operating level" as defined above. The DIN Standards call out setting the operating level of a recorder by means of a *distortion* measurement; the "Bezugspegel" (reference level) on the DIN Test Tapes is not referred to in the other DIN Standards. On the other hand, the "Bezugspegel" is used as the operating level in German broadcasting practice.

"The SMPTE has standardized a "signal level," which is "for use in controlling magnetic sound recording levels and standardizing methods of signal-to-noise measurements. . . ." Since no description is given of operating practices, this signal level is really an arbitrary reference quantity similar to the "100 nWb/m" mentioned above; it is not an operating level."

"The clear separation of the "reference quantity" and the "operating level" is very desirable: the reference quantity is only a measurement unit, and, once chosen, needs never be changed. On the other hand, the operating level is variable because it is influenced by the tape, the equalization, the volume indicator, and other factors."

As previously remarked, the NAB Test Tapes were not yet available when this was written. Measurement methods, however, are not totally uncertain inasmuch as the new procedure can be related within a small number of dB to present measurement practices.

Assuming an NAB Standard Test Tape is available, the new procedure is as follows. (1) Play the 400 Hz tone on the test tape and measure the output of the playback system. (2) Put another tape through the recording process with bias but no audio signal applied to the tape, and measure the output of the playback system. (3) Calculate the ratio, in dB, of the first to the second measurement; this is the S/N ratio.

The NAB standard further specifies that "the response of the measuring system shall be uniform ±0.3 dB from 30 to 15,000 Hz. Response at 20,000 Hz shall be 3 dB below the 400 Hz value, falling at the rate of at least 12 dB per octave above 20,000 Hz."

The reason for rapid cutoff of high frequency response of the measuring system is to exclude inaudible noise frequencies from measurement, particularly the bias frequency.

"The indicating meter (for measuring output) shall have the dynamics of the Standard Volume Indicator (ASA Standard C16.5-1961). The measuring system shall have a full-wave rectified average measurement law."

How does the foregoing relate to current practices? Some parties, including manufacturers of high quality tape recorders, specify the reference as the output level of a 400 Hz tone recorded at 3% total distortion or third harmonic distortion. (Inasmuch as third harmonic is the principal distortion component for tapes, total harmonic and third harmonic distortion are approximately the same for present purposes.) Others employ the Ampex test tapes containing a 700 Hz tone at "operating level." This signal is recorded at a level producing about 1% harmonic distortion on the tape, and about 6 to 8 dB below the level producing 3% harmonic distortion. We have already noted that the 400 Hz reference signal on the NAB Primary Reference Tape is to be recorded 6-8 dB below the level producing 3% harmonic distortion on the tape (with "normal" bias, and excluding amplifier distortion). Therefore the Ampex "operating level" and the NAB Standard Reference Level are about the same. If a person sets his own reference level by recording a 400 Hz tone at 3% harmonic distortion, his reference is *about* 6-8 dB higher than the Ampex and NAB references; the actual distance depends on how nearly the tape used by the individual resembles the magnetic characteristics of the tape selected by NAB for making its Primary Reference Tape.

As an approximation, we may say that a person employing 3% harmonic distortion as the reference, and using conventional tape (1½-mil thickness and not a special-purpose type such as high-output, low-noise, etc.), should reduce his S/N measurement about 8 dB to conform to the new NAB standard or to a measurement based on the tone at "operating level" on an Ampex test tape.

(Continued next month)

The New NAB Magnetic Tape Standards PART 5

HERMAN BURSTEIN

MAGNETIC TAPE MEASUREMENTS AND SIGNAL-TO-NOISE SPECIFICATIONS

MR. MCKNIGHT OF AMPEX COMMENTS THAT when the NAB Standard Test Tape appears (reel-to-reel samples are presently being evaluated), its "reference level will probably be either the same as the Ampex Operating Level or else 2 dB lower than that level." Therefore the person now using the Ampex test tape for reference obtains S/N measurement that, as events will decide, is the same as or 2 dB higher than the measurement obtained by using an NAB Test Tape.

Assuming the NAB Test Tape is available, the new procedure for measuring S/N raises several questions.

1. Under the old NAB procedure, a signal is recorded on the tape, and tape output is measured in playback; the tape is again put through the recording process but with no signal input, and tape output is measured again. This procedure takes the erase capability of the tape machine into account; unerased signal is measured as noise. Under the new NAB procedure for measuring S/N, erasing capability is not taken into account. Nor does any other section of the new standard refer to erasing capability. (Mr. McKnight observes that "... erasing capability is a function of the recorded wavelength, and also the bias current used in the original recording, the tape type, etc. Therefore a single-frequency measurement of erasure is of doubtful value. I know of no satisfactory standard procedure for evaluating erasure, though it is possible that one could be written." He notes, too, that professionals ordinarily use a bulk eraser.)

2. In measuring noise of a tape "recorded with bias but with no signal," the NAB standard does not specify what kind of tape should be used. Presumably, one should be using "conventional" 1½-mil tape. But if not, significant differences in measured noise may occur, depending upon the oxide thickness and formulation. The NAB standard partly recognizes this problem by stating: S/N measurements are "figures of merit for comparisons of systems noise." That is, they are intended not to indicate absolute S/N performance but to enable one tape system to be compared with another in the matter of S/N. On the other hand, it is possible that *relative* differences among tape machines may vary according to tape used. For example, measured S/N may be the same for Machines A and B when conventional tape is used, but may be 2 dB better for Machine A, which has quieter electronics, when low-noise tape is used.

3. The NAB standard does not state how much bias should be used in making the S/N measurement. Presumably, one would be using the bias normally provided by the manufacturer or user of the machine.

4. Some machines are underbiased in order to achieve extended treble response at speeds below 15 ips. But underbiasing increases distortion and decreases the recorded level. This is tantamount to reduction of the machine's S/N capability. Yet such a reduction is not reflected in the NAB standard for measuring S/N, inasmuch as the standard does not require the machine to record an audio signal. However, *in another part of the standard,* the machine is required to achieve a certain level of performance with respect to distortion, so that a machine manufacturer cannot cheat excessively by underbiasing.

"The over-all record reproduce system total harmonic distortion ... shall be less than 3% rms for a 400-Hz sine-wave signal recorded to achieve a reproduce level 6 dB above the NAB Standard Reference Level."

But is this requirement enough? First, the NAB Primary Reference Tape does not reach 3% harmonic distortion until a level 8 dB above the NAB Standard Reference Level, while the machine under test is permitted to (almost) reach 3% distortion at a level about 2 dB lower. Second, the NAB standard does not specify the kind of tape to be used in measuring distortion; high-output tape enables a tape machine to reduce distortion at a given output level. All in all, if a machine is unprobably be specified under certain conditions). Also, with such a 'Reference Level,' tape manufacturers and/or recorder/reproducer manufacturers could — if they would — show a specification for the flux level at which a certain amount of distortion is expected. Such data is given in the sales literature for German blank tape."

Signal-to-noise specifications

When measuring noise on an unweighted basis in accordance with the NAB procedure previously described (using the Standard Reference Level on the NAB Test Tape, and using a measuring system with a specified type of meter and response declining in a specified manner above 15,000 Hz), NAB specifies minimum S/N ratios for the following tape speeds and track formats:

Tape Speed	*Full-Track*	*2-Track*	*4-Track*
15 ips	50 dB	45 db	—
7½	50	45	45 dB
3¾	46	46	45

The NAB standard also provides for measuring noise on a weighted basis "to give a more useful indication of the subjective signal-to-noise ratio than the unweighted measurement." The weighting consists of a great deal of bass cut and a moderate amount of treble cut [Fig. 1], which includes a suggested weighting network. This frequency response is intended to be "similar to that of the ear at low volume levels."

The weighting network [such as the one in Fig. 1] is inserted in the system employed to measure noise output level. Because the network produces some attenuation at all frequencies, it is necessary to increase the gain of the playback system to avoid understating noise and therefore overstating S/N. This increase in playback gain is made on the basis of a special 1,000 Hz tone on the NAB Standard Test Tape, recorded at a level producing output equal to the 400 Hz Standard Reference Level. After output of the 1,000 Hz tone is measured without the weighting network, playback gain is adjusted so that this tone produces the same measured output with the weighting network inserted. On a weighted basis, the NAB standard specifies minimum S/N ratios as follows:

Tape Speed	*Full-Track*	*2-Track*	*4-Track*
15 ips	58 dB	53 dB	—
7½	60	55	52 dB
3¾	57	54	52

At 15 ips, full-track, the new NAB standard specifies 50 dB unweighted S/N, based on a 400 Hz reference level 8 dB below 3% harmonic distortion (on the NAB Primary Reference Tape). This translates into 58 dB S/N based on 3% harmonic distortion. In contrast, the 1953 NAB specification was 55 dB S/N, based on a recording level that produces 2% distortion at 400 Hz. Inasmuch as the difference between the recording levels that produce 2% and 3% harmonic distortion is about 3 dB, the 1953 specification also translates into about 58 dB S/N based on 3% harmonic distortion.

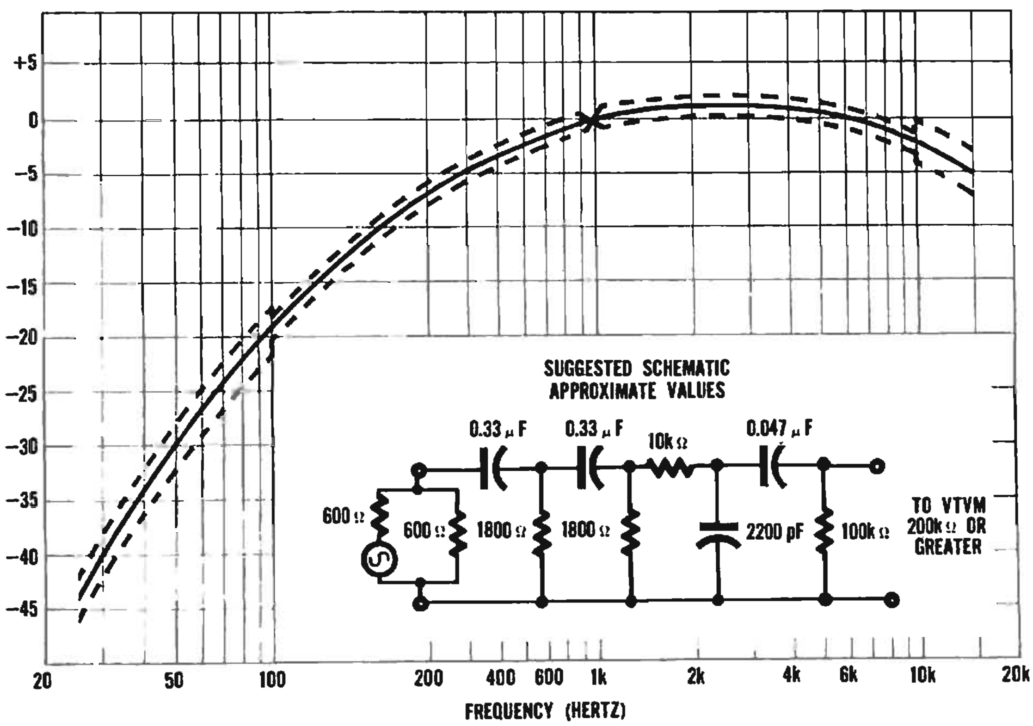
Thus the new standard *seemingly* reflects no advance in the state of the art with respect to S/N. However, we know that frequency response, noise, and distortion are inter-related so that a possible improvement in one or two respects can be foregone in order to achieve an improvement in another respect. With S/N and distortion specifications held at about the 1953 level, the 1965 NAB standard represents an improvement confined out of choice to frequency response. (More specifically, slightly better frequency response is now specified at 7½ ips than formerly at 15 ips. On a record-playback basis, the 1953 standard permitted response to be 5 dB down at 50 and 15,000 Hz at 15 ips; the 1965 standard permits record-playback response to be 5 dB down at *30* and 15,000 Hz at *7½ ips*.)

Mr. McKnight of Ampex comments: "The question of S/N versus speed is a good deal more complicated than the NAB standard admits. . . . The NAB figures should never have been called signal-to-noise ratio. They should have been called either 'noise level' or 'reference-signal-to-noise ratio.'

"The measurements here completely ignore the loss of high-frequency 'power response' at lower speeds. It certainly is *not* true that, because the NAB noise specifications are (nearly) the same at all three speeds, the true dynamic range at the three speeds is equivalent. This reduction of high-frequency dynamic range with reduced speed shows up in the NAB Test Tapes; the frequency run is at Standard Reference Level at 15 ips, but (in order to avoid tape saturation at high frequencies) the frequency run is 10 dB lower at 7½ ips and 15 dB lower at 3¾ ips. The price paid for slower tape speed is bound to be poorer performance. In the U.S.A., equalizations are such as to make the poorer performance show up in the high-frequency power response (not specified) rather than in the noise (specified)."

TO BE CONCLUDED

Fig. 1—NAB weighting curve for weighted noise measurements.



The New NAB Magnetic Tape Standards

PART 6 (CONCLUSION)

HERMAN BURSTEIN

Recording level, flutter specifications, and crosstalk are examined in this final installment

Recorded program level

FOR TAPE MACHINES USING A VU meter as the recording level indicator, it is frequent practice to adjust the meter to produce a meter reading of 0 VU in accordance with the reference level on a test tape, usually the 700 Hz tone at "operating level" on an Ampex test tape. The meter is set so that its pointer swings to 0 VU when a 700 Hz signal is recorded at a level producing the same output in playback as does the Ampex test tone.

With the appearance of the NAB Standard Test Tape (not available at this writing), it would probably be the practice to equate 0 VU on the meter with the Standard Reference Level.

> "Recorded program material shall produce the same reference deflection on a Standard Volume Indicator (ASA Standard C16.5-1961) as that produced by a 400 Hz sine wave signal recorded at the NAB Standard Reference Level."

This signifies that the pointer of a VU meter shall swing to 0 VU at the Standard Reference Level.

Flutter specifications

NAB uses the term flutter to include wow.

> "The unweighted flutter content when reproducing an essentially flutter-free recording of 3,000 Hz at any portion of the reel of tape in use shall not exceed the following:"

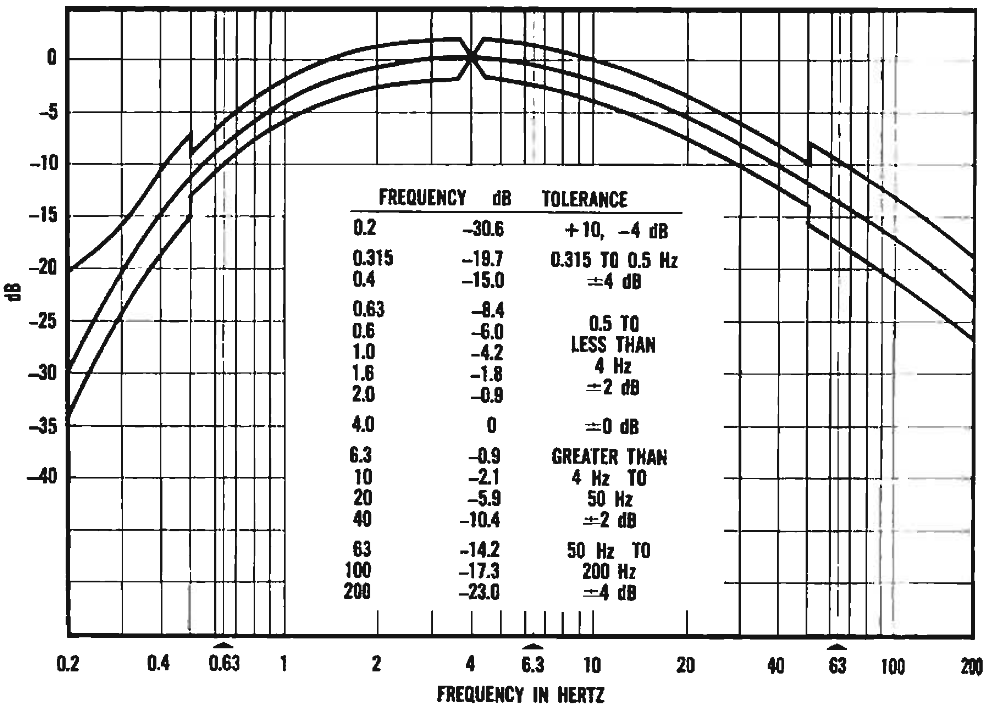
Tape Speed (ips)	*Flutter (rms)*
15	0.15%
7½	0.20
3¾	0.25

To achieve uniformity among flutter measuring systems and procedures, which is rather a problem, the NAB standard carefully spells out the requirements for measurement.

> "Unweighted flutter content shall be measured over the frequency range of 0.5 Hz to 200 Hz. The response of the measuring system shall be 3 dB down at 0.5 Hz and 200 Hz, and falling at a rate of at least 6-dB-per-octave below and above these frequencies, respectively. At low frequencies, where the meter pointer follows the wave form, the maximum deflection shall indicate the rms value. The indicating meter shall have the dynamics of the Standard Volume Indicator (ASA C16.5-1961), a full-wave rectified average measurement law, and shall be calibrated to read the rms value of a sinusoidal frequency variation. ... The meter shall be read for random periods throughout the length of the tape, noting the average of the peak readings, but excluding random peaks which do not recur more than three times in any 10-second period."

The NAB standard also provides for weighted flutter measurements that correspond to the characteristics of the average ear. The weighting curve is shown in Fig. 1. A specific network is not suggested by NAB but is left to the manufacturer of the measuring system, or else to the user.

Fig. 1—NAB weighting curve for weighted flutter measurements.



FREQUENCY	dB	TOLERANCE
0.2	−30.6	+10, −4 dB
0.315	−19.7	0.315 TO 0.5 Hz
0.4	−15.0	±4 dB
0.63	−8.4	
0.6	−6.0	0.5 TO
1.0	−4.2	LESS THAN
1.6	−1.8	4 Hz
2.0	−0.9	±2 dB
4.0	0	±0 dB
6.3	−0.9	GREATER THAN
10	−2.1	4 Hz TO
20	−5.9	50 Hz
40	−10.4	±2 dB
63	−14.2	50 Hz TO
100	−17.3	200 Hz
200	−23.0	±4 dB

On a weighted basis, rms flutter is not to exceed 0.05% at 15 ips, 0.07% at 7½ ips, and 0.10% at 3¾ ips.

The 1953 NAB standard merely specified that "the instantaneous peak flutter and wow shall not exceed 0.2% . . . when recording and reproducing on the same equipment." There was no specification of the measuring system and procedure, and no provision for a weighted measurement. On an unweighted basis, at 15 ips, the 1953 and 1965 standards are on a par inasmuch as 0.15% rms is nearly equivalent to 0.2% peak.

Crosstalk

Crosstalk concerns signal leakage between adjacent tape tracks. To measure crosstalk it is first necessary to adjust recording and playback gain so that equal signals recorded on adjacent tape tracks result in equal output levels. A signal is then recorded on one track, and the output levels of that track and the adjacent track are measured. The crosstalk ratio, for the adjacent track, is the ratio between the two output measurements, expressed in dB.

The NAB standard specifies that the crosstalk ratio for the adjacent track shall be at least 60 dB between 200 and 10,000 Hz. Measurement must be made with a tuned voltmeter to eliminate the effect of noise. In making the test, bias is not to be applied to the adjacent (unrecorded) track. The signals are to be recorded at the same level as the 400 Hz tone in the frequency response portion of the NAB Standard Test Tape. This tone is at Standard Reference Level in the 15 ips Test Tape; at a level 10 dB lower in the 7½ ips Test Tape; and at a level 15 dB lower in the 3¾ and 1⅞ ips test tapes.

The above specifications apply to two-track and four-track mono systems and to four-track stereo systems.

Why isn't a crosstalk test specified for two-track stereo? The answer is that such a test is encompassed in the stereo channel separation test described in the next section. In the case of four-track stereo, the adjacent track does not correspond to the "other" channel (in a given direction of tape travel, either tracks 1 and 3 are recorded, or else 2 and 4). But in the case of two-track stereo, the tracks for the two channels are adjacent. Therefore, in measuring channel separation, one necessarily includes crosstalk between tracks.

Stereophonic channel separation

Channel separation deals with leakage between channels of a stereo system. Largely or entirely, it involves leakage between the channel amplifiers and between the two sections of a stereo head. To measure channel separation it is first necessary to adjust recording and playback gain so that equal signals simultaneously recorded through both channels and their corresponding tracks will produce equal output levels. A signal is fed into one channel, and the output levels of that channel and of the "other" channel are measured. The channel separation ratio for the "other" channel is the ratio between the two output measurements, expressed in dB.

The NAB standard specifies that the channel separation test be made with bias applied to both tracks. Between 100 and 10,000 Hz, stereo channel separation should measure at least 40 dB.

Although not explicitly stated, the level at which the test should be made is the same as used for crosstalk measurement.

Special-Purpose, Limited-Performance Systems

In a special, relatively brief section, the NAB standard sets less demanding specifications for "lightweight portable magnetic recorders . . . where adequate voice intelligibility and interchangeability of recorded tapes are of primary importance . . . (but which are) not suitable for maximum fidelity recording of speech or music." No mention is made here of 15 ips. On the other hand, provision is made for 1⅞ ips. Inasmuch as we are largely concerned with high-fidelity performance, we shall deal very briefly with this section.

Speed tolerance is ±2%, compared with ±0.2% for high-quality systems. Unweighted flutter content may not exceed 0.5% rms, compared with a maximum of 0.15% to 0.25%, depending on tape speed, for high-quality systems. Playback response specifications are the same for all three limited purpose speeds: response may be +2 dB at any frequency, −2 dB relative to 400 Hz at 200 and 3,000 Hz, and −5 dB at 100 and 5,000 Hz. Recorded response may be +2 dB at all frequencies, −2 dB between 200 and 5,000 Hz, and −5 dB at 100 Hz.

Thus on a record-playback basis, a limited purpose system may be down as much as 10 dB at 100 Hz and 7 dB at 5,000 Hz. While it "is not intended to restrict the frequency range of voice-recording systems which have the inherent capability of wide-range recording, without distortion, it is, however, often considered desirable to limit the extreme low-frequency response for improved speech intelligibility. . . . Attenuation above 5,000 Hz is recommended (in recording) to reduce the chance of high-frequency tape overload at the lower tape speeds."

On an unweighted basis, S/N is specified for all three speeds as 46 dB full-track, 43 dB two-track, and 40 dB four-track; S/N specifications are about 3 to 4 dB higher for a high-quality system.

Standard test tapes

The NAB standard provides for test tapes at speeds of 15, 7½, 3¾, and 1⅞ ips. All are to be full-track recordings and to contain five parts: (1) 60-second azimuth tone of 15,000 Hz at 15 and 7½ ips, 10,000 Hz at 3¾ ips, and 5,000 Hz at 1⅞ ips; (2) 20-second 400 Hz tone at the NAB Standard Reference Level; (3) Another 20-second 400 Hz tone at the same level at 15 ips, 10 dB lower at 7½ ips, and 15 dB lower at 3¾ and 1⅞ ips; (4) A series of signals for testing frequency response. These signals are at the Standard Reference Level at 15 ips, 10 dB lower at 7½ ips, and 15 dB lower at 3¾ and 1⅞ ips. All speeds include the following frequencies: 30, 50, 75, 100, 250, 500, 750, 1,000, 2,500, and 5,000 Hz. In addition there are frequencies of 7,500 and 10,000 Hz at 3¾ ips; and of 7,500, 10,000, 12,000, and 15,000 Hz at 7½ and 15 ips. (5) 60-second 1,000 Hz signal at a level equal to the Standard Reference Level.

Though not examined here, the 1965 NAB standard further contains annexes on methods of tape speed measurement, on the playback characteristics at various speeds, and on a Primary Calibrated Reproducing System for the purpose of calibrating Standard Test Tapes. Æ

Why Look-A-Like Recorders Can Be $$$ Apart

HERMAN BURSTEIN

Tape recorder buying checkpoints are examined by the author to determine what accounts for price differences

It is puzzling to many an audio buff that we can go into hi-fi dealer's store and find tape machines with great similarity in all respects but price, which can differ by a factor of 2:1 or more. If two machines are about the same size, look alike, and perform basically the same functions, what can account for such a vast price difference? In general terms the answer lies in the following:

1. Variety of features, conveniences, and functions.
2. How well a machine performs.
3. Reliability — how long a machine continues to perform well.

Before getting down to specifics, it should be stated that we are not arguing a case for either expensive or inexpensive tape machines. We are just trying to explain the differences between them. At the same time we hope to provide worthwhile clues to finding that particular machine most commensurate with your needs and budget.

Features

Instead of attempting a complete list of features offered by one tape machine or another, we will just provide enough examples to make clear the point that price must go up with both the number and nature of features offered.

Most home machines are put through their paces of "start," "stop," "normal forward," "fast forward," and "fast reverse" (and perhaps "pause") by levers, knobs, or pushbuttons. For greater ease and speed of operation, some machines supply the force to actuate the mechanism: pushbuttons responding to a light touch control solenoids that perform the actual work. Convenient but costly.

There is a major trend to the automatic reversing machine, which eliminates the need to stop the transport at the end of a reel, lift the reels off the machine, reverse it, and restart the transport. Instead, using one of a variety of sensing techniques (metallic foil, period of silence, etc.), the machine stops automatically at the end of the tape and almost immediately proceeds to operate in the reverse direction. Such a machine must cost appreciably more than its non-reversing counterpart in order to incorporate a sensing direction, and facilities to reverse motor direction and switch from one set of tracks in the forward direction to another set in the reverse direction.

Because it has separate amplifiers for its two channels, a stereo machine has the essential *capability* for sound on sound: to record on one channel while playing the other. But the normal mode of operation is to record on *both* channels or play on *both* channels. To simultaneously record on one and play on the other calls for special switching, including temporary insertion of a dummy load on the bias oscillator to prevent significant changes in bias and erase current for the channel which is recording. Otherwise, such changes might appreciably affect distortion and treble response, and they might injure the erase head (owing to excess oscillator current). Further, there must be provision for mixing the newly-recorded input signal and the previously recorded signal.

Number of speeds adds to cost. This is not merely a matter of slight extra mechanical complexity in the transport. With each change in speed there are accompanying changes in recording and playback equalization; also, depending on how hard the manufacturer strives for good performance, there may be changes in bias current, recording level, and recording level indication. Moreover, as machines introduce ever-lower speeds, playback heads with ever-narrower gaps are needed to maintain good treble response. A high order of technology, skill, and precision go into such heads.

A highly desirable feature is the ability to monitor the tape; that is, play the tape as it is being recorded to compare the incoming and recorded signals. This requires separate record and playback heads, separate record and playback amplifiers for each channel, and a switch to alternate the machine's output between the incoming and playback signals—all at no small added cost.

Many more features that add to cost might be named, such as remote control, voice-actuated operation, synchronization with slides, mixing facilities with separate gain controls for each input signal, pause button (to stop the transport without disengaging the record or playback electronics), etc. But by now the point should be clear.

Performance quality

The basic criteria of high fidelity performance are low noise, low distortion, and faithful frequency response (wide and smooth). These electro-acoustic criteria apply with full force to the tape recorder. In addition, there is a fourth, mechanical criterion: good tape motion, which essentially denotes speed that is accurate and constant. Two machines identical in functions and features may differ appreciably in the extent to which they measure up to these criteria, and therefore will tend to differ in cost.

Noise. Of the four criteria, tape recorder noise seems to get least attention although it is just as important as the others. Attention inclines to be diverted to frequency response and tape motion, which are no longer a problem in home machines; at least not at speeds of 7½ and 3¾ ips. But noise remains very much a problem, and a significant element of tape machine cost is the effort to keep noise at a level low enough to be considered "high fidelity." Preamplifiers, power amplifiers, and tuners attain signal-to-noise ratios of 55 dB and substantially more. But for a tape recorder to do so (with reference to 400 Hz recorded at a level producing 3% harmonic distortion on the tape) is still a relative rarity, achieved through extra-careful design and layout and through components of extra-high quality.

An important component of noise is produced in recording as the result of distortion in the bias waveform. To reduce waveform distortion to a minimum, the tape machine manufacturer must be willing to go to extra expense in such matters as oscillator circuitry, quality of the oscillator transformer (a toroidal unit is considered best by some), and quality (stability, precision, and overload characteristics) of the capacitors and resistors used.

Counted in with noise are cross-talk between stereo channels and inadequate signal separation between adjacent tape tracks. To hold cross-talk to a minimum requires that a stereo head be suitably constructed to provide good isolation between its two sections, and that the tape amplifiers for the two channels be properly laid out to prevent the signal in one amplifier from leaking to the other. Good adjacent track separation requires that the tape heads be precisely constructed and positioned.

Before leaving the subject of noise, it is important to note indications that the Dolby Noise Reduction System (discussed in the March and April 1967 issues of AUDIO) may be making its way into home tape machines. Briefly, the system divides the audio range into several bands, emphasizes low-level signals in each band prior to recording, and de-emphasizes these low-level signals and, simultaneously, noise in playback. Unless the original Dolby system is greatly simplified, incorporating it into a tape machine would appear to add an item of considerable cost.

Frequency Response. Where frequency response is concerned, the chief problem seems to be that of preserving treble frequencies to the upper limit of the audible range for adults—about 15,000 Hz. As stated before, this depends in good part upon making the gap of the *playback* head (not the record head) sufficiently narrow; the lower the speed, the narrower must be the gap to avoid an abrupt treble drop. It is costly to construct a head with a gap that is very narrow and at the same time very straight (for well-defined response) and deep (to withstand wear).

As tape speed is reduced, there are intensified treble losses in recording owing to magnetic phenomena. In part these losses can be overcome by reduced bias. But a reduction in bias means an increase in distortion unless recording level is lowered. A satisfactory lowering of recording level can take place only if noise is kept low. And keeping noise low, as we have already seen, is expensive. In sum, good treble performance at low speeds is not achieved through design and components of the garden variety.

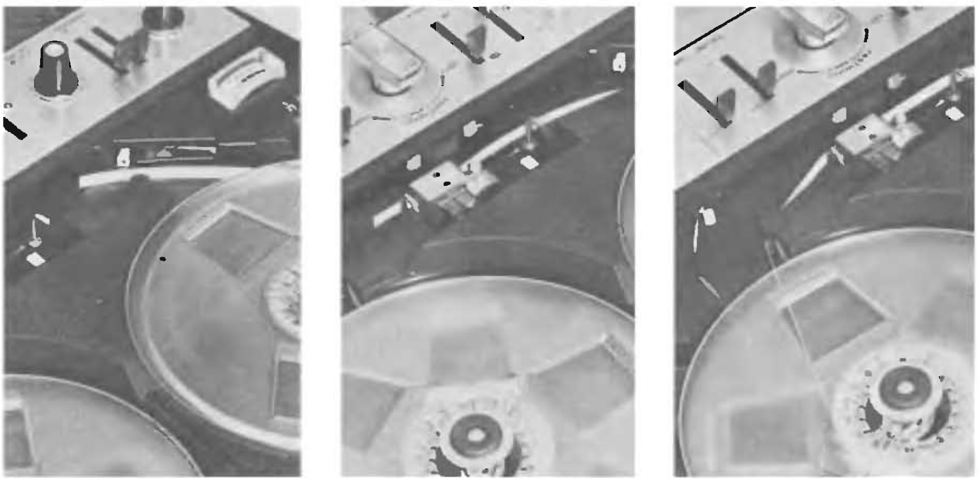
In reproducing bass, the playback head tends to exhibit "bumps" and a hump in response owing to the tendency of the head as a whole, and not merely its gap, to respond to long magnetic wavelengths (low frequencies) on the tape. These irregularities can be minimized through optimum angle of approach of the tape to the head, through optimum "wrap" of the tape about the head, and through suitable adjustment of playback equalization circuitry.

For the rest, flat response depends upon expertly designed record and playback equalization, and upon the use of precision components that fulfill this design. It is not sufficient that record-playing response be flat. In addition, playback response must be flat with respect to a standard test tape—in other words, playback response must conform to a specified industry characteristic to assure flat response when playing tapes recorded on other machines (commercial tapes or otherwise).

To the extent that he seeks to satisfy the various requirements for wide, smooth frequency response, the tape machine manufacturer faces ascending costs.

Distortion. Not only should the record and playback amplifiers of a tape recorder have inherently low distortion but they should be able to accommodate a wide range of input signal levels without overloading; that is, there should be sufficient "headroom" for large incoming signals even though these are eventually reduced to appropriate size before reaching the record head.

Fig. 1—Automation has come to reel-to-reel tape recorders, including automatic threading. Ampex, Bell & Howell, and Sony Superscope are among manufacturers offering this luxury. Below, is a Bell and Howell Autoload® recorder which, on pressing a lever, activates a blower that lifts tape from the supply reel, carries it under the tape lifter, up toward the takeup reel, and drawn around the reel hub, whereupon the blower shuts off automatically. (Stop-action, strobe-light photography was used here.)



Distortion further depends on the quality of the tape heads, particularly the recording head, which if improperly constructed may overload before the tape does at low frequencies.

Thirdly, distortion depends on the steadiness of tape motion. Flutter (rapid variations in tape speed occurring from about 20 to several thousand times per second) modulates the audio signal, resulting in spurious frequencies: distortion. The audible effect is grainy, gritty, or even coarse sound in place of "silky smoothness." If poor design of the tape path, rough heads, or excessive pressure-pad force causes the tape to rub across the head in the manner of a bow against a violin string, the audio signal is correspondingly modulated and distorted.

Finally, and perhaps most important, distortion depends on the recording level. If the tape machine designer has gone to the effort and expense of reducing noise about as far as the state of the art permits, he can afford to let the user record at a moderate level—most of the time causing less than 1% harmonic distortion and seldom or never causing more than 3% distortion. But if corners have been cut to save money and the machine is therefore relatively noisy, the designer may have the user record at appreciably higher levels—often causing more than 3% distortion—to achieve a non-objectionable signal to noise ratio. How does the designer get the user to record at one level or another? Simply by the way he calibrates the recording level indicator. For example, if he intends the user to operate at a high recording level to cover up noise, the indicator is adjusted to say "halt" (the magic eye will close or the VU meter will swing to 0 VU) only at a high recording level.

Parenthetically, this brings us to the much disputed question of VU meters versus magic eyes in home tape recorders. I have long argued that for home use the "magic eye," though less expensive, is at least as good as and probably preferable to the VU meter if you needn't conform to professional specifications. The "magic eye" gives a more definite indication of excessive recording level. As it is an electronic device, it responds quickly to sound peaks. VU-type meters have gained popularity for two reasons: they look "professional," and of course, they are easier to integrate into transistor circuits than are "magic eyes," which are really tubes.

It is ironical to contemplate that a cheap, non-standard "VU meter" is probably more costly than a magic eye. If it is *truly* a VU meter, meeting A.S.A. specifications as to frequency response, rapidity of response, and ability to withstand overloads, the cost becomes much higher, even though the home user gains little, if anything.

Separate Record and Playback Heads. Earlier we pointed out that separate record and playback heads (and with these, separate record and playback amplifiers) are a desirable though costly feature that permits monitoring the signal on the tape as it is being recorded. Probably a more basic reason for separate heads is that they permit the best overall performance in terms of frequency response, distortion, and noise. Although a single head for use in both recording and playback can give good performance, it is nevertheless a product of compromise. Separate heads permit each to be designed for the specific, intended purpose. For recording, a head should have a relatively wide gap and low impedance. For playback, it should have a much narrower gap and high impedance (to maximize signal output and therefore the signal-to-noise ratio at a stated distortion level).

Motion. We have already mentioned the need for minimum wow and flutter to obtain maximum transparency and clarity in recording. This requires a well-balanced motor of high quality, careful design of the tape path, accurate machining of rotating parts, and mechanical devices to filter out deviations from constant speed (such as a heavy flywheel). Good motion further denotes accurate speed, constant speed from beginning to end of a reel, and facile transition from one operating mode to another (for example, from "normal forward" to "fast reverse") without spilling or breaking tape. Good motion increases cost.

Reliability

Though a tape machine initially meets all its specifications, it still must stand the test of time. Reasonably long, uninterrupted periods of correct operation are obviously important. This too can be built into a machine, but again as additional cost.

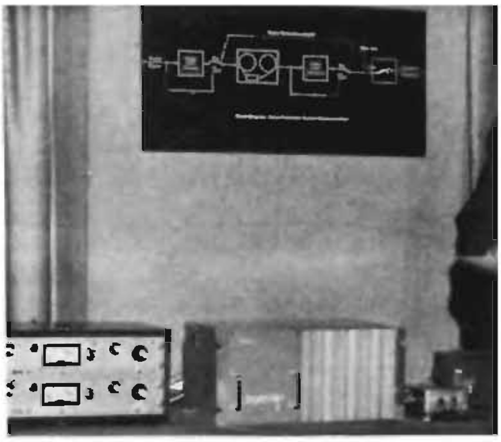
To give a simplified idea of how this works, assume that a tape machine consists of 200 parts, that malfunction of any part will disable the machine, and that each part has 0.001 chance of going bad within a year. Then, based on probability theory, the machine has about 18% chance of breaking down within a year. The manufacturer may feel this is good enough in view of the low price he is charging for his machine. Another manufacturer, however, may seek to offer much greater reliability, though at a higher price, by using more expensive parts (such as resistors of higher wattage rating and capacitors of higher voltage rating) having only 0.001 chance of failure within a year.

Reliability is not only a matter of good parts, but also of good design and proper assembly. The extra engineering hours that a manufacturer may devote to designing, testing, and redesigning a machine to make it failure-resistant must be reflected in higher cost. To prolong head life, for example, he may abandon pressure pads and switch to a more expensive system of tape guides and tape tension to maintain intimate tape-to-head contact (essential for good treble response). When a machine must be extra rugged to withstand all day, day-in, day-out use, he may go to three motors instead of one. To make sure that his product meets specifications and *keeps meeting them,* he may institute an extra-tight system of quality control over incoming parts and outgoing final product. A conscientious job of testing a complex tape recorder and putting it through its many paces takes time and money.

So when you wonder why tape recorder "A" cost more than tape recorder "B," though they appear to be identical, consider the foregoing factors. Æ

Audio Noise Reduction: Some Practical Aspects

RAY M. DOLBY



Part I: Operating principles and details of a professional noise-reduction system

A NEW GENERAL-PURPOSE, professional audio noise-reduction system (Dolby Laboratories model A301) was discussed last year in AUDIO Magazine.[1] The design philosophy of the system was subsequently presented in a technical paper.[2] Intended as a continuation of this discussion from a more practical point of view, the present article will examine the device itself and consider various operational aspects, especially in relation to magnetic tape recording.

The purpose of the noise-reduction system is to reduce noises—hiss, print-through, hum, as examples—that normally arise in the tape recording and playback process.

Beyond a certain point in the care taken in designing, maintaining, and operating recording equipment, any gains in signal-to-noise ratio unfortunately become increasingly difficult. Such gains may be difficult not only because of limitations imposed by the physics of the matter, but because of expense or general impracticability. A 10 dB reduction in tape hiss, for example, can in theory be obtained by the use of tracks ten times as wide. Or any desired reduction of print-through can be had at the inconvenience and expense of interleaving the layers of the recording tape with a suitable shielding material.

Clearly it is a matter of great practical significance to have a method of side-stepping these difficulties, which is the intention of the A301 system to be described.

Before going into the details of operation, it is necessary to appreciate how the system is connected into the recording chain. Referring to Fig. 1, the signal is first treated by the recording-processor half of the noise-reduction system before being recorded on tape. During playback, the signal is fed through the other half of the system, where it is restored to normal; at the same time, hiss is reduced by 10 to 15 dB (depending on the frequency), and hum, rumble, and print-through are reduced by 10 dB.

In general terms, the noise-reduction system must operate around or enclose the noise-producing element of the chain. Therefore, a requirement for use of the system is that the signal must be available for pre-processing before being fed into the noisy element and for post-processing after it emerges.

It should be appreciated that the system is not capable of separating the noise from the signal in the normal sense. All it does is pre- and post-process the signal in such a way that the signal effectively becomes less susceptible to the addition of noises. Pre-emphasis and de-emphasis and, in a somewhat different way, frequency modulation and demodulation are further instances of complementary pre- and post-processing systems which improve noise immunity.

In professional audio there are many ways in which a noise-reduction facility can be utilized, but a notable application is in the making of master tapes for high-quality phonograph records. When the original signal is put on tape in processed form it is protected from the usual sources of noise encountered in recording, dubbing, storage, and final playback. In this connection, hiss and hum are the most common noises, but print-through is undoubtedly the most serious flaw when it does occur.

Principles of operation

The A301 system may be thought of from two points of view. First, it is a compression-and-expansion system operating in four frequency bands. Second, it is an automatic, signal-operated equalizer which continuously controls the recording and playback equalization characteristics in such a way as to improve the overall signal-to-noise ratio. From both views, the A301 is a three-dimensional signal processing system with an overall gain of unity, but with intermediate transmission properties which are functions of amplitude, frequency, and time.

Viewing the device as a compressor-expander, the main feature which distinguishes the system from previous ones along similar lines is that the signal as a whole does not pass through any variable-gain elements. Referring to Fig. 2, high-level signals pass straight through the direct path of the system (amplifiers only). Thus, they are not altered in any way whatever. By this means, the usual distortions and tracking troubles of compressor-expanders are avoided.

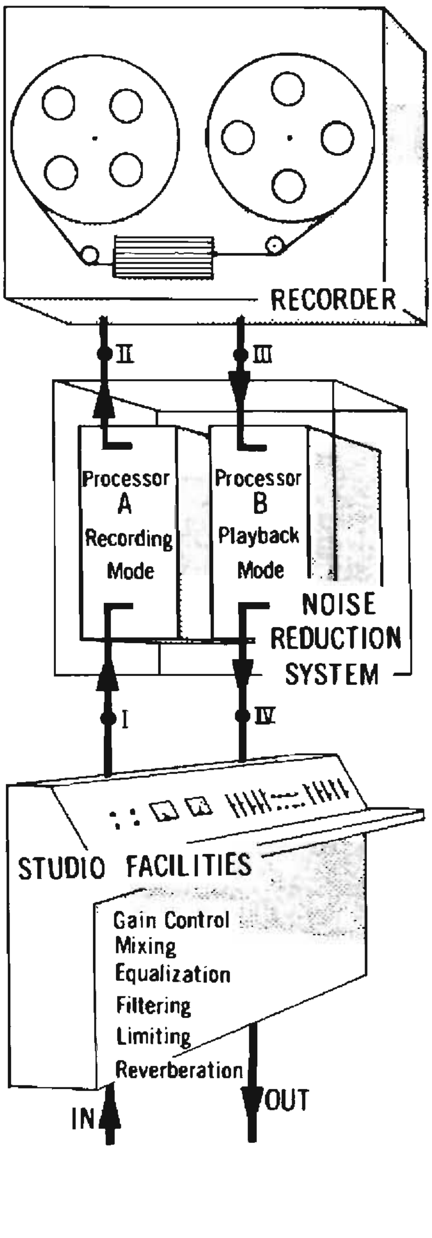
Low-level signals, which are of relevance for noise-reduction purposes, are handled in a side chain (differential network) comprising four band-splitting filters and low-level compressors. Whenever the signal amplitude is low in any band, the output from the compressor is large in comparison with the same component in the main path. The addition of the differential component to the straight-through component thereby results in a boosted output signal. The situation at high levels is that the differential component is compressed substantially; being small in comparison with the main signal, its contribution is negligible.

A complementary operation is performed during playback, with the differential component in this case being subtracted from the main signal. Since the gain of the playback unit is decreased at low levels, the desired noise-reduction effect is achieved.

An important aspect of the system is that identical differential networks are used in both the recording and playback modes. Inspection of Fig. 2 shows that, basically, an extra component is added and then it is subtracted; what is left *must* be the original signal. Insofar as correct restoration of the signal is concerned, the networks can have almost any characteristics whatever, with the proviso that they are the same.

A feature of the process, it should be noted, is that no pilot signals are used in controlling the playback operation. In effect, the signal is its own pilot. The playback processor contains full information on the principle by which the recording unit operated; this information, together with the processed signal itself, is sufficient for the playback half to recreate the original signal.

Fig. 1—Use of the Dolby noise-reduction system in one channel of the audio chain. The signal is processed before recording and after playback; noise is reduced and processing operations cancel out, restoring the signal to normal.

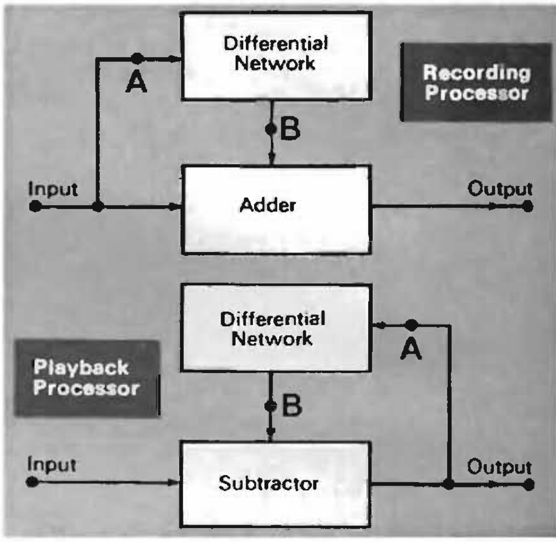


The transfer characteristics of the two processor units are shown in Fig. 3. When the differential component, Fig. 3C, is added on a decibel (dB) basis to the input signal, the recording characteristic in Fig. 3A results. It can be seen that at very low levels the input signal is amplified, whereas at high levels the transfer characteristic essentially rejoins the input signal line. The inverse (playback) characteristic, shown in Fig. 3B, is formed by subtracting the differential component from the input signal. The result is reduced gain at low levels (noise reduction) and nominally unchanged gain at high levels.

A noise-reduction system with transfer characteristics as described above, but with only one full-frequency compressor band, would have good characteristics with regard to distortion, tracking ability, and so on, but it would suffer from poor noise-reduction properties. Full noise reduction would be obtained only at low signal levels, while at high levels the noise would have its usual value. Moreover, "swishing" and "breathing" would be produced under dynamic conditions, a familiar behavior of limiter and compressor circuits in general.

There is a fairly widespread misconception of the reason for noise-modulation effects; they are usually attributed to excessive recovery time in the limiter or compressor control circuitry. In fact, such behavior is evident even when extremely short recovery times are used. A steady-state phenomenon, the effect arises because of the inability of the signal—which usually occupies the mid-frequency portion of the spectrum—to mask low- and high-frequency noises adequately.

Fortunately, the masking effect makes it difficult or impossible for the ear to perceive noise in the *same* frequency range as the signal. By exploiting this naturally occurring noise reduction and suitably fitting compression and expansion noise re-



duction to it, it is feasible to provide for all normally encountered eventualities of signal and noise and to produce an overall reduction of *perceivable* noise.

The process of joining real noise reduction to apparent noise reduction must take into account the diminishing efficiency of the masking effect with increasing separation of the noise and signal frequencies. To this end, it is necessary to handle the audio spectrum in several independent frequency bands. Figure 2 shows the arrangement used in the A301 system, in which four bands are employed.

The bands are divided as follows: band 1, 80 Hz low-pass; band 2, 80 Hz-3 kHz band-pass; band 3, 3 kHz high-pass; band 4, 9 kHz high-pass. Conventional 12 dB/octave filters are used for bands 1, 3, and 4, while band 2 is designed to have a frequency and phase response which is complementary to that of the other bands.

In the recording mode, the outputs of all the bands are combined with the signal in the main path in proportions which result in a uniform low-level 10-dB boost up to about 5 kHz. Above 5 kHz, the boost rises smoothly to 15 dB at 15 kHz and then levels out. The amount of noise reduction follows the same pattern, since the frequency response in the playback processor is complementary to that in the recording processor.

For fairly low-level program material, the full amount of noise reduction is obtained in all bands. But with increasing level, the noise re-

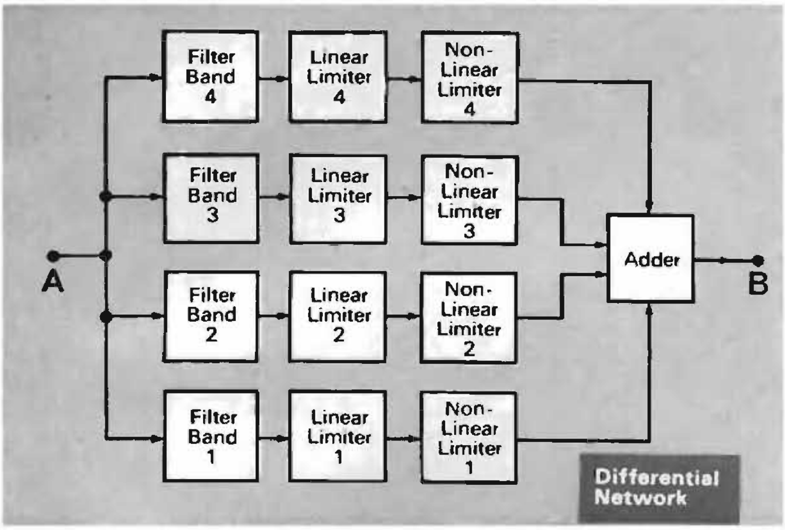


Fig. 2—Basic block diagram of Dolby system. During recording, a differential network adds a low-level signal to the straight-through signal. In playback, the low-level component is subtracted. The differential network (right) consists of four band-splitting filters and low-level compressors. Terminals A and B show how the network is connected.

duction in band 2 decreases progressively, whereby the masking effect then assumes control of mid-frequency noise perceptibility. Noise reduction under signal conditions therefore arises most of the time from low-level pre-emphasis, followed by complementary de-emphasis, due to the actions of bands 1, 3, and 4.

Because the bands do not have sharply defined boundaries, they produce useful noise reduction outside their nominal pass-bands, a fact which has been taken into account in establishing the frequency divisions. Thus, when band 2 is paralyzed, band 1 provides noise reduction up to about 120 Hz, with band 3 being effective down to about 1.8 kHz. With signals containing fairly high-level high-frequency components, band 3 is also blocked, in which case band 4 provides noise reduction down to about 5 kHz. Band 4 is rarely blocked, except by signals such as loud cymbal crashes.

All of the bands work together, in varying degrees of momentary noise reduction, in their respective frequency ranges. The overall result is a noise level which is less (or appears to be less) than the original noise level and, equally important, is constant (or appears to be constant).

Referring to Fig. 2, a further feature of the system which should be noted is the non-linear limiter following the compressor (linear limiter) in each of the four bands. In practice, the non-linear limiter circuits are simply symmetrically biased diode clippers. Without the clippers, a tone-burst applied to the input of the system would normally cause the output to overshoot by 10 to 15 dB during the attack time (that is, the time taken for the control-signal circuitry and compressors to respond). But the amplitude of the differential component is so small in comparison with the signal in the main path, it is possible to bias the diodes in such a manner that any overshoots are confined to 2 dB with peak level inputs. Such clipping may seem to be a very dubious procedure. In actuality, however, the limiters operate linearly except with the most percussive program material. When they do operate, the clippers are inaudible because of the masking effect of the high-level transient components present in the main path.

The attack time of the system is variable in the range from about 1 to 100 ms, automatically adjusting itself to the size of the amplitude transition. For small transitions it is an advantage to use long attack times in order to minimize the generation of modulation products, but for large transitions it is clearly best to minimize the duration of the overshoot.

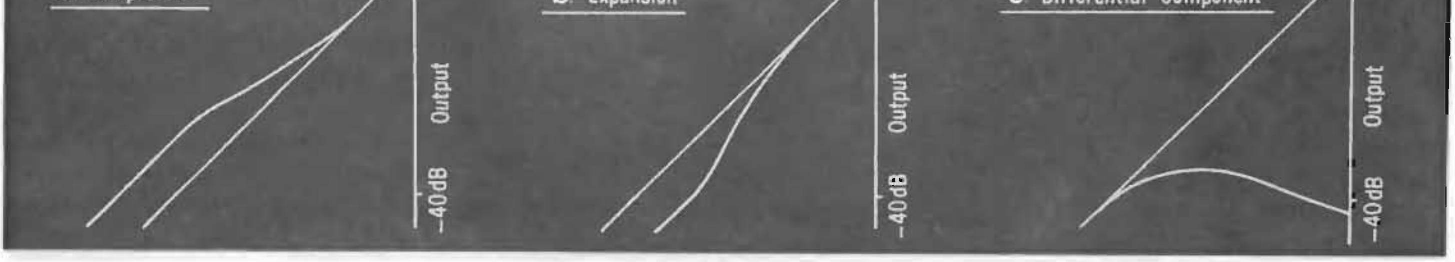
Non-linear control signal integration circuitry is similarly used to provide optimum smoothing of the rectified control signal, while minimizing the recovery time of the noise-reduction action following cessation of high-amplitude signals. Low-frequency distortion is thereby held to a fraction of a per cent at 30 Hz, and the recovery time (less than 100 ms) is short enough that no "swishing" or "breathing" effects are perceptible under even the most difficult program situations, such as "clap sticks" in a dead studio.

Block diagram

Turning to the block diagram in Fig. 4, one of the two signal processors in an A301 unit is shown, together with the power supply. Each processor consists of an amplifier module, a control module, and two compressor modules (each of which contains two compressors).

The signal enters the unit through the bridging input transformer, T403 (or T404). It is fed to potentiometer RV101, which is adjusted to give a standard operating level

Fig. 3—Transfer characteristic curves. Compression curve (a) is produced by adding the differential component (c) to the straight-through signal. The expansion characteristic (b) is formed by subtracting the differential component (c) from the straight-through signal according to the negative-feedback configuration shown in Fig. 2.



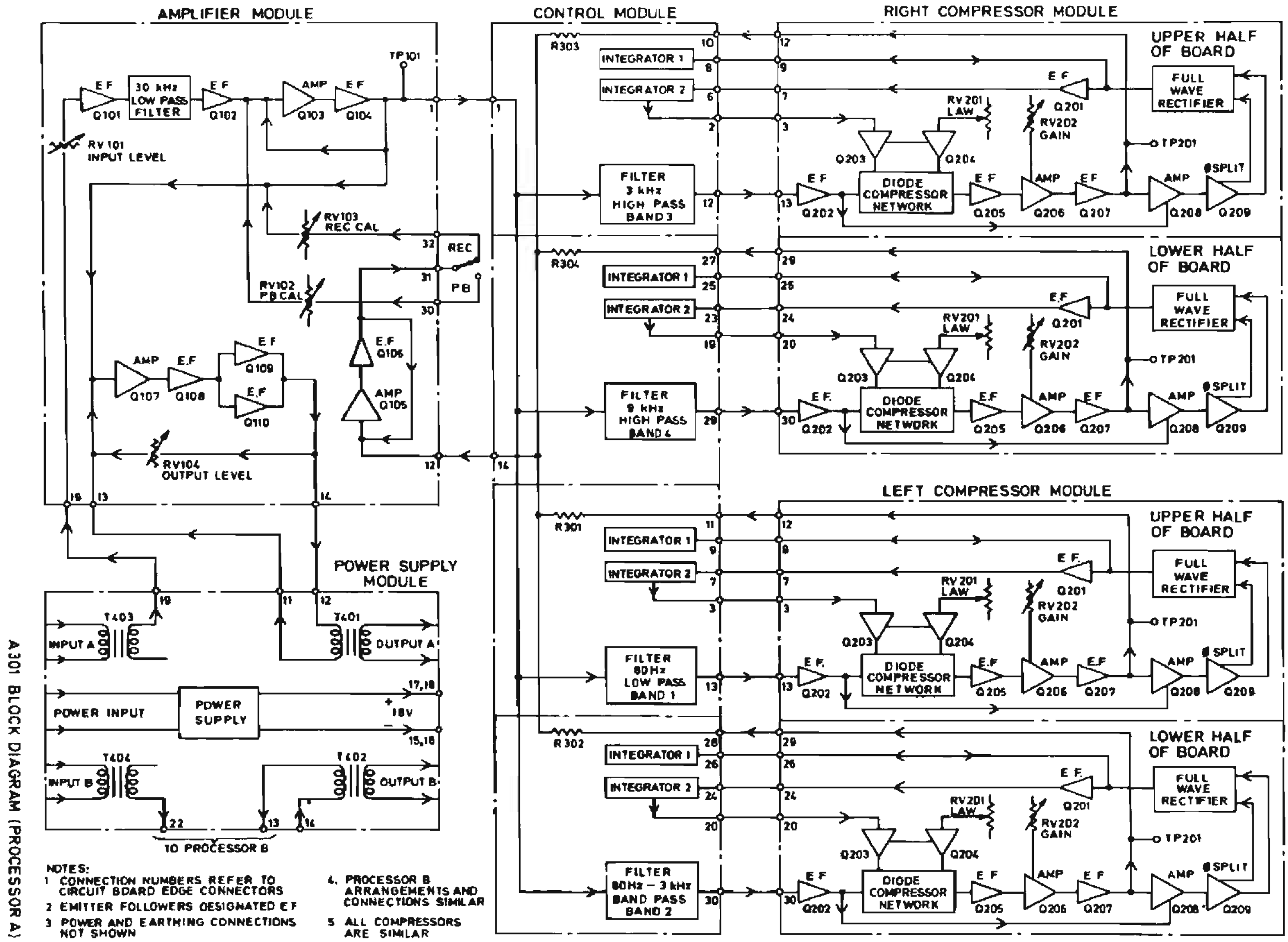


Fig. 4—The audio- noise reduction-system consists of two identical processors, one each for left and right channels. A Processor (amplifier, control, and two compressor modules) is shown above.

within the system. After passing through the 30-kHz low-pass filter, which removes any tape recorder bias or other undesired high-frequency signals in the input, the signal is fed to the filter driver-amplifier, Q103 and Q104. The output from the amplifier is passed to the filters in the control module and also to the output amplifier, Q107-Q110.

The output amplifier itself is fairly standard, having an output impedance of 600 ohms and a clipping level a little over +18 dBm (that is, 14 dB above 0 VU on the normal +4 dBm standard). As with the input, the system output is left floating to minimize line noises.

The control module coordinates and controls the operation of the four compressors. All functions which differ from band to band are contained in this module, an arrangement allowing identical compressors to be used for all bands.

Following band-splitting, the signal is distributed to the four compressors, being fed in through the emitter follower Q202, which in turn drives the diode compressor network, a combination of two germanium and two silicon diodes.

The compressor circuit takes advantage of the fact that a diode's dynamic resistance can be controlled by the direct current flowing through it; transistors Q203 and Q204 produce a control current which determines the impedance of the diodes. The diodes form part of an attenuator network, a balanced configuration being employed to cancel the d.c. component. Because of the low signal amplitudes handled in relation to the curvature of the diode characteristics, distortion produced in the compressor is negligible.

The compression threshold in all bands is 40 dB below peak operating level, defined according to the European convention of taking the nominal 2% distortion point on magnetic tape as peak operating level. In VU terms, the threshold is 36 dB below 0 VU.

The output of the compressor is amplified by Q205 and Q206, passed through the diode clipper circuit (between Q206 and Q207), and returned to the control module through the emitter follower, Q207.

The output is also amplified further by the control-signal amplifier, Q208, and passed to the phase splitter, Q209, and full-wave rectifier circuit. The fed-forward signal from Q202 to Q208 should be noted. By suitably combining this signal with the output of the compressor, the resultant control signal produces the down-turning characteristic shown in Fig. 3C.

After being pre-integrated by the fast time-constant integrator 1 in the control module, the d.c. control signal is fed through emitter follower Q201 and back again to the control module for further integration. Integrator 2 is an RC circuit with a back-

(Continued on page 55)

NOISE-REDUCTION SYSTEM

(Continued from page 22)

to-back combination of germanium and silicon diodes in parallel with the resistor. This configuration provides the desired non-linear integration properties. Finally, the smoothed control signal is returned to the compressor and is used to control the current in the compressor diodes by means of Q203 and Q204.

The four compressor output signals returning to the control module are combined by the precision resistors R301-R304 and are fed back to the amplifier module. After being amplified by the feedback amplifier, Q105 and Q106, the resultant noise-reduction signal is fed to a switch which determines whether the processor operates in the recording or playback mode. For recording, the noise-reduction signal joins the main signal additively between Q104 and Q107; during playback, the noise-reduction signal is combined subtractively at Q102-Q103.

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NEXT MONTH: Dr. Dolby discusses applications of the noise-reduction system.

Fig. 1 (left) — Audio Noise Reduction System A301. Fig. 2 (below) — Power-supply module removed.

Audio Noise Reduction

PART 2 (Conclusion)

RAY M. DOLBY

More aspects of the Dolby professional noise reduction system and practical applications

Construction

THE A301 audio-noise-reduction system described last month is powered by an 18-volt regulated supply (the power supply module may be seen in position on the left in Fig. 1 and separately in Fig. 2). Having a total noise content of about 100 μV peak-to-peak and an extremely low output impedance, the unit powers all circuits with a minimum of further decoupling. The power supply module also contains the input and output transformers for the system.

To obtain the full benefits of modular construction, the system has been designed to use a minimum number of different modules. The eight circuit modules in the A301 system comprise only three different types: amplifier, control, and compressor. This simplifies troubleshooting significantly.

Fiberglass printed circuits are used in the modules, component numbers being silk-screened with epoxy paint. All resistors are 5%, 2%, 1%, and ½% high-stability carbon film types. Electrolytic capacitors are used where necessary, but band-splitting filter capacitors, time-constant capacitors, and critical coupling capacitors are of the polyester type, while tantalum capacitors are used in situations where high values and reliably low leakage currents are required. Tolerances are tightly controlled, with more than 200 components being bridged, matched, or specified to better than 1%.

The 99 transistors used in the system are of the high-gain silicon planar type, enabling large amounts of negative feedback to be used in the circuitry. The circuits are thus highly stable with regard to transistor spreads and ambient temperature variations. Wherever possible, the circuitry has been designed to be temperature independent. Where this is not possible or practical, temperature compensation is used (of the 163 germanium and silicon diodes in the system, 44 are used for temperature compensation).

System alignment

Because of the stability of its circuitry, the system does not require routine alignment. Nevertheless, an alignment procedure is useful for troubleshooting purposes or for giving the system a thorough check. The two signal processors are adjusted independently, following a standard procedure which ensures that all modules of a given type are fully interchangeable.

During alignment, a module is removed and a special alignment card, providing several switch-selected calibration and test functions, is inserted in its place; the module is then plugged into a connector on the alignment card.

Apart from the power supply voltage adjustment, the A301 has six different types of controls on it: input level, output level, record calibration, playback calibration, compressor gain, and compressor law. The first four controls are on the amplifier modules, the last two on the compressor modules. The control modules do not require adjustment.

Each amplifier module is adjusted by feeding a calibration tone into the system from an oscillator. The input level control accommodates the various signal standards encountered, giving standardized operating conditions within the system (100 mV at the amplifier test point). The output level control, being adjusted in a complementary way, yields an overall processor gain of unity. After the input and output level controls have been set, the record calibration and playback calibration controls are used to standardize the gain of the noise reduction amplifier Q105 and Q106. This procedure gives the required 10 dB of noise reduction at 1 kHz, which also automatically sets the 15 kHz value at 15 dB.

The alignment card is then shifted to a compressor module position and switched to the compressor gain mode, which bypasses the filters and inserts a precision 50 dB attenuator between the filter driver amplifier output and the compressor inputs. As the resulting level is below the compressor threshold, the compressor gain can be accurately set (10 mV at the compressor output). Then the alignment card is switched to the compressor law mode, which raises the input to the compressor by 20 dB. The law control is then adjusted to give a compression of 6 dB (that is, 50 mV at the compressor output).

The above compressor procedures are carried out on the two compressor modules (two compressors each), and finally the whole routine is repeated for the second processor.

Because of the international exchange of master tapes, it is necessary that the above alignment procedure should take into account the various signal standards used throughout the world. This is accomplished by means of the two main 15 i.p.s. alignment tapes in use, Ampex-NAB and DIN.[1] Unfortunately, these tapes employ different reference levels—the DIN level being approximately 4 dB higher—and it is necessary to allow for this fact when adjusting the input and output level controls of the A301 system.

For standardization purposes the system is adjusted with a calibration tone equal to the signal level produced by a DIN alignment tape (that is, recorded flux of 32 mM/mm at 1 kHz), each processor then being set for 100 mV at the amplifier module's test point, as explained previously. Thus, the Ampex-NAB tape, the most widely used reference in the U.S., should produce a voltage 4 dB below 100 mV at the test point; during adjustment of the system, it is therefore necessary to use an oscillator signal which is 4 dB *higher* than

Fig. 3—Changeover system. Added to the back of Model A301, the changeover facility enables one unit to be used for recording and non-simultaneous playback on two tracks. Both remote control and manual control are provided.

the level produced by the Ampex-NAB tape. This procedure correctly ties the two alignment tape standards together and also compensates for the many different line levels in use. The most common practice in U.S. studios is that the Ampex-NAB level corresponds to a 0 VU signal level of +4 dBm, 600 ohms, or 1.23 volts rms. The level which should be fed into the A301 system for alignment purposes is thus +8 dBm or 1.95 volts.

[1] J. G. McKnight, "Absolute Flux and Frequency Response Characteristics in Magnetic Recordings: Measurements, Definitions, and Standardization," *J.A.E.S.*, *15*, 254 (1967).

Although noise-reduction tapes will be recorded and reproduced under standardized level conditions on an international basis by use of the above procedures, it is still necessary to remember that there are several equalization curves in use throughout the world (NAB, CCIR, DIN, and others[1]), and that these must be accommodated in the usual way.

Operational aspects

Since the A301 comprises two independent signal processors — each of which may be connected either in the recording mode or playback mode—one unit may be utilized for recording on two tracks, playback on two tracks, or recording and simultaneous playback on one track. The operating modes of the processors are normally determined by two toggle switches on the back of the unit.

Whenever weight, space, or economy are considerations, it is possible to exploit the dual mode facility. In many recording situations, particularly multi-track, it is not customary to monitor line-out during the "take" itself. Also, for remote sessions on two tracks, it is convenient to be able to get along with only one noise reduction unit. In such cases, line-in is monitored for setting balances, while line-out (the raw noise-reduction tape playback) can be checked with regard to dropouts and other tape or recorder defects. After listening awhile, it is possible to become somewhat accustomed to the bright, breathy, bigger-than-life sound of processed signals, thereby allowing at least some judgment of the artistic merits of the recording during the "take" itself. Of course, during a proper playback, the noise-reduction system is changed over to the playback mode.

While a switch is provided for determining the operation mode, it is necessary to remember that the unit must also be shifted appropriately to the input or output circuit of the tape recorder. In a simple setup a signal switchbox can be built or the XLR cables can be physically replugged.

For more sophisticated installations, a remote changeover system, (shown in Fig. 3) is used. This facility, added to the back of the unit, includes a 24-volt power supply and provides for relay-controlled changeover of the processor mode and all signal connections. When connected to the record relay circuit of the tape recorder, the changeover system operates automatically.

The way in which the noise-reduction system is actually used in the studio chain will now be considered in more detail. While the arrangement is very simple (see Fig. 1 of Part I), it is necessary to observe the operational rules which are implicit in the figure.

Basically, the signal should not be operated upon by the normal studio facilities while it is in the processed condition (points II and III, between processors and recorder). But the signal may be manipulated as usual either before or after the system, at points I or IV. The tape recorder should have level frequency response and, as discussed previously, the absolute sensitivity of the recorder should be adjusted with a test tape.

If these requirements are not met, the output from the noise reduction system may contain low-level errors in frequency response and signal dynamics. In fact, however, the allowable tolerances are reasonable. A gain-error of 4 dB, for example, may produce barely discernible changes in high-frequency response on a direct A-B comparison.

An attempt should be made to keep overall gain-errors within 2 or 3 dB, which is normally not very difficult. But in a complex studio using many different kinds of equipment, even this fairly liberal requirement is met only with some degree of care. For example, a termination resistor left off a source having a true 600-ohm impedance will cause a voltage rise of 6 dB (which will result in a brighter sound than normal during playback). Nevertheless, such mistakes can usually be avoided. And if a studio uses standardized practices throughout, such as consistently terminating lines on either the sending end or the receiving end (as opposed to mixing the two methods), then use of the noise-reduction system is simply a matter of inserting the unit in the lines to and from the recorder.

It should be appreciated that the requirement for tape recorder standardization does not mean that the level actually recorded on the tape must be standard, only that the A301 compression law should always correspond to standard magnetic conditions on the tape. After the noise-reduction system and recorder have been adjusted, the recording engineer is free to set the output of the mixing console to any level required for full modulation of the particular type of tape used. Note that compensation for tape sensitivity (for example, for high-output tape) should be accomplished in the above way, not by modifying the record and playback gain settings on the tape recorder, as is usually done.

While the foregoing standardized operating conditions are recommended, occasions sometimes arise in which, for one reason or another, it is necessary to use non-standard level conditions. In such cases a level-setting tone should be recorded on the beginning of the tape, the equivalent level used being indicated —i.e., whether the tone corresponds to the Ampex-NAB level or to the DIN level (4 dB higher).

Regarding the question of tape itself, engineers often have strong feelings about the best kind to use. Normal, low-noise, and high-output are the main types considered for conventional recordings. With noise reduction, however, the type of tape selected is not so important since hiss and print-through are reduced by the system, and the distortion situation can be alleviated simply by recording at a slightly lower level. Nevertheless, most engineers feel that low-noise tape is optimum with the system. The print-through tendency is corrected by the system, a very low noise level is obtained, and the overload properties are good, especially at high frequencies; a good balance between the various limiting factors is thereby obtained, resulting in excellent performance overall.

Regardless of the tape used, it sometimes takes a while to become accustomed to the latitude which a 10 dB increase in dynamic range provides. When first using the noise-reduction system, many recording engineers instinctively continue trying to record at as high a level as possible. This is a practice to be discouraged, since two or three dB of the increased signal-to-noise ratio can well be sacrificed for a reduction in distortion.

Regarding the editing of noise-reduction tapes, most editors prefer working with the signal restored to normal. However, for certain passages the facility of monitoring the processed signal is used (the play-back-noise-reduction unit being switched out); in this way a kind of magnified view of the recording is obtained, whereby doubtful noises (such as the honking of a distant horn or musicians whispering) can be identified quickly. Moreover, if a splice sounds acceptable—in terms of differences in hall acoustics, for example—while the signal is in the processed condition, then the splice is sure to pass inspection after the signal has been restored to normal.

Under ideal conditions the original tape is edited and is used in the final tape-to-disc transfer, an operating procedure which is still often used for classical music. However, where tapes are bought or sold abroad, copies are usually involved. In such cases one-to-one copies of processed tapes are made for distribution on two ordinary recorders. Whether using an original or a copy, a noise reduction unit is used in the final tape-to-disc transfer.

Since the A301 system is a complementary one, it follows that it cannot be used to reduce the noise on old normally recorded material. If a normal tape is treated by a playback processor, the noise will be significantly reduced. But this will be at the expense of introducing errors in low-level signal dynamics and a loss of high-frequency response. However, if it is necessary to use old material, the system can successfully minimize any further degradation during re-recording; the noise level of the copy will then be substantially the same as that of the original (theoretically, 0.4 dB higher).

So far, the discussion has concerned use of the system in a standard way, whereby the rules are followed and one can be sure that the output signal will always be identical in all respects to the input signal. Situations arise, however, in which there is a strong temptation to use the system in a non-standard way for the purpose of reducing the total number of units required. Unless an engineer has some understanding of the system and is of an experimental bent, such practices should be avoided. In principle, it is not permissible to operate on the signal without first restoring it to normal; the "control signal," which is contained in the processed signal itself, will be altered in the process. Nevertheless, some non-standard procedures, when used cautiously, often produce satisfactory results. For example, moderate amounts of equalization and filtering usually may be introduced in the processed signal without detrimental effects. Limiting can also be done, provided the gain control of the limiter is set to yield an overall gain of unity at all levels below the limiting threshold.

In the mixing-down of three-, four-, and eight-track recordings, a matter of even greater significance is that processed signals can often be mixed together with results which are indistinguishable, after de-processing, from those of the standard case in which the signals are de-processed before mixing. In such a procedure it is sometimes necessary to readjust the resultant level slightly (to the playback unit), to take into account the degree of correlation of the signals being mixed. In general, an experimental approach is required; as long as the rules are being broken, the only rule of any consequence is that *if it sounds right, it is right.*

It should be appreciated that the installation of a noise reduction system by no means marks the end of noise problems in a studio. On the contrary, the removal of tape as the limiting factor often uncovers a whole new layer of deficiencies which the meticulous engineer will wish to correct: microphone, pre-amplifier, and mixer noise; air-conditioning, fluorescent lights, creaking chairs. In addition, if the above factors are well under control, there will be further surprises. The soft puffing noises of the felt dampers on pianos will be revealed, and many other instruments (and their players) will be found to be unmistakable generators of wide-band noise. Æ

BEHIND THE SCENES

BERT WHYTE

The Cassette Tape Format: Pros & Cons

At the recent Consumer Electronics Show (an industry trade show) in New York it was well nigh impossible to walk into a manufacturer's exhibit that did not feature some form of cassette recorder. The proliferation of new units, especially from the Japanese companies, was of amazing proportions. It was very obvious that the cassette format had "arrived." There were tiny battery-operated portable mono and even stereo cassette recorders; recorders combined with AM and FM radios; recorders built into elaborate "entertainment" consoles; integrated stereo cassette recorders with amplifiers and speakers. There were stereo cassette playback decks, even stereo cassette changers from Norelco and Aiwa which automatically played back up to six cassettes. Every manufacturer I talked with outlined his extensive and elaborate marketing plans for maximum exploitation of the cassette concept.

While the 8-track cartridge is generally regarded as a playback medium, the cassette has been touted for its recording capabilities as much as for its function for playback of pre-recorded music. In fact, I would say that at the Show, the emphasis was on the "hardware" and the recording aspects of the cassette. The attitude of many people I talked with was that the "plus" of the recording facility in the cassette system had a great attraction for the consumer, even if his primary interest was in pre-recorded cassettes. It was also felt that this was the reason why the cassette system would eventually prevail over the 8-track cartridge. This of course, remains to be seen. With the accelerated pace of developments in magnetic recording technology over the past few years, prognostication is a very precarious pursuit.

Admitting the values of the recording function, it is also apparent from the ever-mounting sales figures that the pre-recorded stereo cassette is a resounding success. Let's take a look at this plastic phenomenon and discuss some of its positive and negative qualities.

Advantages and disadvantages

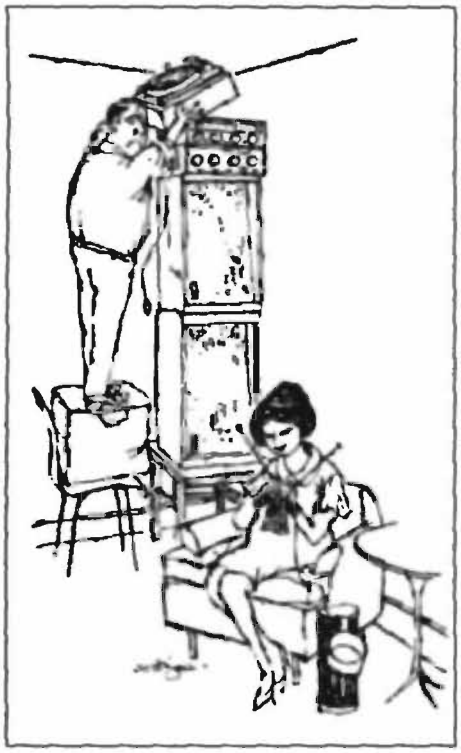
The stereo cassette is indisputably small, light, easy to handle and easy to store. It is rugged and, unless grossly abused, should last almost indefinitely. Its tiny size stimulates miniaturization of drive mechanisms and electronics. The narrow 150-mil tape can be had in an ultra-thin configuration which will afford 120 minutes of playing time at its 1⅞-ips speed. The cassette permits the use of fast-forward and rewind, which allows a certain degree of program selectivity when combined with a footage counter. The two pairs of stereo tracks are well separated so there is no crosstalk problem. There is crosstalk between the tracks in each stereo pair, but since the tracks both run in the same direction, the crosstalk isn't audible. Under laboratory conditions the frequency range of a stereo cassette can be 40 to 12,000 Hz with a signal-to-noise ratio (S/N) of 45 to 48 dB (unweighted). Under optimum playback conditions in the home, regular production-run cassettes afford 60 to 9000 Hz with S/N ratios of 40 to 45 dB (unweighted). It should be noted here that this S/N ratio is better than the levels of many cassette playback units!

Needless to say, like any other developing technology, the cassette has various problems. Some of these problems are in the cassette itself, others are in the playback units. For example, in the standard cassette there is a mu-metal shield to reduce stray hum fields. In practice it has been found that with certain drive mechanisms the shielding is inadequate and, try as you might, you just can't ground out the hum. In new cassettes which should be on the market very soon, the mu-metal shield has been extended in a "wrap-around" style; with a very slight additional cost to the pre-recorded cassette manufacturer, this has reduced hum significantly. There are also some new playback units which use synchronous motors. Unfortunately, the narrow width of cassette tape contributes to playback noise on a ratio of almost 2 to 1 compared to standard ¼-in. tape. The cassette tape has a 0.5-mil base and a 0.2-mil coating of magnetic oxide. Standard ¼-in. tape has a 0.4-mil coating. This difference in oxide thickness produces about a 5 dB loss in the cassette tape because of the very short wavelengths at the 1⅞-ips speed. The cassette tape must have the thin base and coating to ensure a supple tape which permits good head "wrap" and contact to maintain good frequency response. It is also a matter of sheer space and playing time, an important factor in the tiny confines of the cassette. One bright spot is the production of a new oxide which is reported to gain back about 3 to 4 dB of the loss attributed to the thin coating on cassette tapes. As far as the "Cronar" chromium oxide tape is concerned, it is not available at present. In any case, it is too thick, is somewhat abrasive, and could cause accelerated wear in the type of heads furnished with cassette playback units. Further, it would require the bias frequency to be almost doubled. In the considered opinion of some engineers, there has been and there is such rapid progress in the development of iron oxides that they doubt chromium oxide would be much of an advantage

. . . . at least not in the area of home audio.

Another thing that plagues the duplicators of pre-recorded stereo cassettes is the variation in the width of their duplicating tape. The standard calls for 150 mils with a tolerance of plus zero and minus 2 thousandths. If the tape is slightly narrower, it cuts into the edge track (which is the left channel) and may cause a loss of 3 to 4 dB in signal. There is an appreciable number of pre-recorded stereo cassettes which are down in level on the left channel. It is usually not your machine which is out of kilter, but this tape variation . . . so you will have to use your balance control. You will also encounter dropouts in pre-recorded cassettes. Much of this is caused by fingerprint residues on the thin coating, but this is expected to be eliminated by the use of automatic machinery instead of the hand assembly of cassettes.

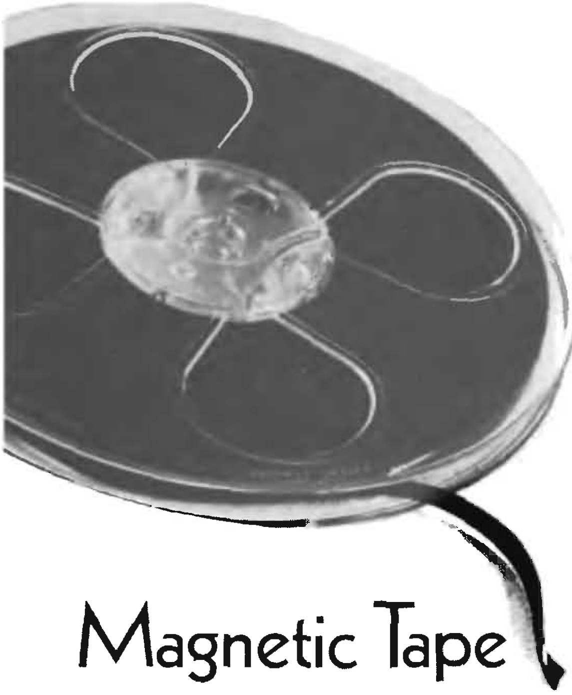
The wow-and-flutter content of most pre-recorded stereo cassettes is usually quite a bit less than the wow-and-flutter of the various playback units. This is one of the penalties of the 1⅞-ips speed and is likely to remain a problem until the development of an inexpensive servo-control drive. It should be noted that there are two new cassette recorders made by 3M/Wollensak and Harman-Kardon which use full-sized drive components, large flywheel, etc., and are in general built to "professional" standards, and for which excellent figures are quoted for wow and flutter. I saw and heard both units at the Consumer Electronics Show and they seem to hold much promise. I hope to obtain sample units and give them the "lived with for awhile" treatment.



There seem to be two schools of thought regarding the duplication of stereo cassette tapes. One method is the Ampex-style use of multiple-slave tape machines. The other is what is known as the "common mandrel" system. The mandrel is the equivalent of a large-diameter, massively heavy axle on which are mounted many large take-up reels which pull the tape over a recording head . . . one to each reel. Since the reels are all driven by the same mandrel or "axle," it is claimed there is very little difference in wow-and-flutter content between the tapes on each reel, and a low overall figure for wow and flutter. The Dubbings Co. of Copiague, New York uses the common mandrel system; I am indebted to Chief Engineer Trevor Campbell for a most fascinating and instructive tour of his plant. Mr. Campbell says that, under optimum conditions, wow and flutter are about equal between the multiple-slave tape-machine system and the common-mandrel system, but he feels the latter system is easier to maintain to specifications. He also feels that scrape flutter is easier to control, especially since he is using dubbing ratios of 16 to one and is currently experimenting with ratios of 24 to one. This drops the scrape-flutter frequency much further into the bass range, with subsequently less effect on transient response. Mr. Campbell uses the Ferrite heads which wear very little and thus avoids "lipping" and tape skew which could cause track misalignment.

At the present time, any evaluation of the pre-recorded stereo cassette must be considered in the light of several different kinds of listeners. For the mass market—those with integrated or "compact" systems with small limited-range speakers, who audition mainly "pop" type material, at relatively low playback levels in small apartments or homes—the stereo cassette affords them a quality of sound equal if not superior to what can be obtained from a disc system in the same price range. With the added advantages of longevity and non-"scratchability" of their recorded material and the recording capability of their cassette unit, the appeal of the cassette system is readily apparent and the burgeoning sales not surprising. For the serious audio buff, especially those oriented to classical music, the stereo cassette leaves much to be desired. For one thing, the number of classical cassettes is comparatively limited. Ampex, for example, issues but one classical cassette to every 15 or 20 pop productions. It is true that D.G.G. and a few other companies are trying to build up fairly extensive catalogs of classical cassettes, but this still makes for scant classical representation. Far more serious than the lack of material is the quality of sound to be found on classical stereo cassettes. To put it bluntly, it is not remotely competitive with the high-quality stereo disc or 7½-ips stereo tape, and only marginally and occasionally equal to 3¾-ips material. The restriction of frequency response, especially in the bass range, the dropouts and the all-too-frequent distortions of various type, and above all, the almost traumatic hiss caused by the poor S/N ratio, simply negates its virtues of size and handling. When "pop" cassettes are played over a high-quality system the results are somewhat better, but the hiss still is too high. I must admit that the present cassette playback units add their increments of noise, and some of the newer units may help the overall noise picture. I will also admit I heard some remarkably quiet stereo cassettes of generally good quality under laboratory conditions, and presumably this kind of quality will be forthcoming in the not-too-distant future. As things stand now, with the currently available quality of sound on stereo cassettes, the audio buff will most likely use them for background music.

Over the past months we have discussed various formats of slow-speed tape. The quality picture with all of them is not particularly bright, at least as far as the devotee of high-quality sound is concerned. A friend of mine, one of the most respected engineers in the tape field, who chooses to remain anonymous, sums up the slow speed tape situation thusly: . . . there are many avenues of approach to upgrade the slow-speed tape where it can eventually begin to compare with top-quality disc and tape. But all of the effort expanded in doing this will prove that reaching this goal will be very expensive and that, except for certain convenience factors, it will not be as good as a 7½-ips tape. The factors needed to improve 7½-ips tape are far easier of accomplishment and far less costly. The extra tape required on a 7½-ips tape is cheap and is getting cheaper all the time." A radical view in this age? Unquestionably, and only time will tell. Æ



Magnetic Tape

AL FANNING

How to choose the right type from among all-purpose, low-noise, low-print-through, high-output tapes

MAGNETIC AUDIO TAPE is one of the most ingenious and resourceful partners a hobbyist could have, if he understands how to choose and use it. This thin ribbon of plastic—shiny on one side and dull on the other—is an endlessly versatile tool. But there is much to bewilder a tape recordist—imperfect tape, different thicknesses, many trade names, and so on.

Good-quality tape can't be made without unremitting care in manufacture and fanatically intense inspection of the finished tape, because the process requires extreme precision in a number of important respects. Obviously, the maker must have skill and, literally, devotion to quality if he is to put out tape that is uniformly good. So your chances of getting good tape are a lot better if you buy "brand" tape from the top names in the field. The "white box" tape sold at bargain prices carries no such initial presumption of quality. One box of it may be pretty good, and the next wildly off the beam. Some white-box tape is instrument tape that has been *rejected* by manufacturers (You wouldn't want a dropout when a computer is calculating *your* income tax would you?).

What do you watch out for? Whatever the origin of tape, it may have any of the faults discussed in the following. Some of the most important arise from the fact that the strength of the signal varies with the thickness of the coating of magnetic particles that forms the active side of the tape. To understand this and some of the other faults, let's review quickly how tape is made.

The process is a continuous one in which the plastic backing, at this point two to three feet wide, moves slowly through a machine that flows the liquid coating onto it. Next the broad band of plastic goes through an oven that hardens the coating. Then it goes to a slitting machine which cuts it, in one operation, into 40 or 50 ribbons which are (for open-reel sound tape) one-quarter inch wide.

The preparation of the coating to go onto magnetic tape is an extremely demanding process. The coating consists of billions of particles of iron-oxide powder, the "magnetic" part of magnetic tape, mixed into liquid glue, the "binder," which holds them onto the plastic backing. The oxide particles are needle shaped and each one is only a few millionths of an inch long. When the binder hardens, the particles are, ideally, laid out along the tape in a perfectly even "carpet" that has the same magnetic properties at every point on the tape.

The coating must be flowed onto the backing with a thickness uniform within a few millionths of an inch. And the backing must have a mirror-smooth surface on that side.

How deficiencies originate

What happens if the coating *does* change in thickness a tiny bit, getting a little thinner, say, at some point? The signal level will suddenly drop a few decibels, a very disconcerting fault found in poor tape. Elimination of such changes depends, in the final event, on 100% inspection. A lot of tape doesn't get it, as already suggested. The reason for this sensitivity to coating thickness is simple: the thicker the coating, the more iron there is on the tape and the stronger the magnetism produced at any point by a given signal in the recording head.

Now let's suppose that the iron is not evenly dispersed in the coating, or that the iron particles themselves are not unifrom in size. Then the tape will have a high noise level: it will be very hissy, and this is probably the most common fault of poor tapes. We can see that the particles together produce a kind of "grain" in the magnetic coating. The finer and more even this grain, the lower the noise.

If the tape backing is microscopically rough, the coating will have tiny variations in thickness that are heard as noise. So use of a low-grade backing pushes up the noise level.

The binder itself has to be just right, very tough and smooth when it hardens but *not* brittle or stiff, because the tape must be flexible enough to conform to the heads in recording and playback. If the binder lacks the necessary toughness or is brittle, the coating may tend to flake off after a time, or the oxide may rub off in large amounts in the form of a gummy dust. A flaky or powdery tape should be removed from the machine the instant it shows its character. You can detect extremely good binder quality by running a finger nail over the coating side of the tape. But this may not show up a more moderate coating weakness that would be unacceptable on your machine. Deposits around the heads or guides are often the first signs that a tape is shedding its coating.

If the binder is very rough or sticky, you have some other troubles: you may get extreme head wear, or a squeaky tape, or high flutter, (from uneven motion), or poor high-frequency response, or a lot of noise. This shows how important the adhesive chemist is in tape manufacturing. He has to come up with a glue that has just the right properties when hard. Top-quality tape today has nearly always been given another operation to reduce binder roughness: the coating side is lightly polished to make it very smooth, so it passes over the heads with a minimum of friction. At slightly higher cost, you can get tape that is "lubricated" with a silicone to make it slide even better. Some manufacturers put a lubricant in the coating as a routine matter. Low friction at the heads is of great importance because, among other things, it reduces "modulation noise," the mushy garbling sound that seems to be in back of the music in wide-range reproduction. Modulation noise is really a very fast flutter that results from the tape's sticking a little on the head, then jumping forward, then sticking, etc. The less it sticks, the lower this noise. And that adds up to a lot of trouble from a rough or sticky binder.

Now let's suppose that the mixing process has not been complete, so that there are little clumps or nodules of the powder on the surface of the tape. When one of these passes over the head, it tends to push the tape away from the head. We lose high frequencies, or maybe the whole signal, for a fraction of time. We have, in other words, a "drop-out," a kind of hole in the sound. If the loss is very short, a few milliseconds only, it may not be noticeable in sound recording (in digital instrumentation recording any drop-out is serious because the loss of just one of the microscopic dots of magnetism will change the story on the tape). A longer drop-out of course, is very disturbing.

A more subtle fault found in much poor tape, and one depending on a combination of factors in manufacture, causes high-frequency deficiencies. The oxide characteristics, the shape and size of the particles, and a number of other things affect the high-frequency response of tape. Since getting super highs in tape recording is a challenge, you are badly licked if the tape itself falls off considerably in this part of the spectrum. It is worthwhile making a trial recording on any tape you are not sure of if you are going to use it for a job in which the ultimate in fidelity is important.

As a final entry in this litany of tape sins, consider what happens if the slitting machine does not cut the tape ribbons absolutely straight, as is sometimes the case. The edge of the tape may be "curly," which tends to produce uneven motion with attendant flutter, wow, increased modulation noise. The tape may skew back and forth over the heads, causing gross variations in signal; or it may repeatedly rub against the flange of the supply or take-up reel, again a cause of flutter or wow.

Types of magnetic tape

So much for the sins of tape. Let's talk about which *good* tape the savvy recordist chooses for any given recording job. First there is the division into the two types of backing material, acetate, polyester, and polyvinyl chloride.

Acetate is a little cheaper and is satisfactory for all ordinary recording jobs, including those in which you are aiming at top fidelity. Its inferiority to polyester shows up mostly when it comes to life in storage. Extreme changes in humidity and temperature may degrade the acetate backing somewhat, making it brittle or causing the coating to peel. Polyester is more resistant to such changes.

The polyester is also considerably stronger than acetate at a given thickness. Thus the 1-mil polyester is, roughly, about as strong as 1½-mil acetate, and this makes 1-mil polyester very attractive for all-around amateur recording. On the other hand, a lot of professional recording is done on acetate, partly because of one fault of polyester: its tendency to stretch before breaking. If a tape breaks without stretching, it can usually be put together with little or no loss of the recording, but a stretched section of tape is ruined.

For a little more money still, though, you can get "tensilized polyester," which has both high strength and a reduced tendency to stretch (it's pre-stretched). To sum up, if your machine runs at fairly low tension or is so arranged that it very seldom breaks tape, the straight polyester is a good all-around tape for recordings you want to keep indefinitely. If you have occasional tape breakage or stretching, use tensilized tape for irreplaceable recordings.

The third type of tape base is polyvinyl chloride—PVC—which is available from a few manufacturers. This material is claimed to exhibit a strength similar to polyester, and is extremely supple, thus providing excellent head contact. It is usually pre-stressed, and is not affected by humidity.

Concerning the three backing thicknesses, 1½, 1, and ½-mil: the thinnest, the ½-mil, should be used only if you can't do without its increased playing time on a particular job. It has a greater tendency to break or stretch than thicker tape. It also exhibits increased "print-through," which is the transfer of the signal from one layer of tape to another in storage. The unhappy outcome is stronger pre- and post-echoes.

There are tapes with specialized characteristics, including "high output," "low print-through," and "low noise." But select them with care because you may lose a desirable characteristic to gain another one. For example, the high-output tape may be handy if you are recording from some source that puts out a very low signal, forcing you to run the gain on the tape recorder very high, a possible cause of increased noise. But note that higher-output tape tends to have high print-through. So if print-through has been troubling you, high-output tape is likely to make it worse.

You can reduce print-through by recording at a lower level, by using a thicker tape, or by using a low-print tape at some loss in sensitivity to signal. The low-print tape may be indicated if you must record fairly high and if you have plenty of low-noise signal so that the sensitivity of the tape is of secondary importance. Low-noise tape is especially useful with slow-speed recorders, where signal-to-noise ratio and high-frequency response is poorer than at higher tape speeds. This tape reduces background noise and lifts the high end. But a problem arises here: recorder bias requirements are different from bias current needs of other tape oxides. Thus a recorder bias adjustment may be needed to match low-noise tape. This will allow a recordist to take full advantage of the tape's attributes. This is not an acute problem, though, because there is generally sufficient bias margin to produce desirable results with magnetic tape having special characteristics. Performance will be further enhanced, however, when bias current is right on the button for the type of tape used. This squeezes out that last drop of higher fidelity.

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MAGNETIC TAPE

(Continued from page 48)

Which tape to choose

To conclude, the best tape for you depends much upon a variety of factors, including recorder fidelity; tape-handling ability, recording signal strength and quality, recording time without changing reels, degree of fidelity required for an application, tape recording permanence, and recorder bias.

Experimentation with a few tapes might help you to decide which tapes are best for what purposes. One way to do this is to purchase one reel of each type and splice lengths of each type together. Then record the same selection on each length, in turn, and play back the results. You may decide that low-noise magnetic tape will serve best for 75% of your recording needs. Perhaps the other 25% will be made up of magnetic tape chosen from among a few other types, say, long-play tape for recordings that promise to consume an inordinate amount of time (conferences, party games, complete broadcasts, etc.). Maybe "white box" tape will be fine for letting friends and relatives hear what they sound like, low-print tape when you intend to store recordings for a long time, and so on.

The judgment will have to be made by *you*, however, because no one knows your recorder or your needs better than you do. Æ

Layman's Guide to Tape Recorder Specifications

AUDIO ENTHUSIASTS pay fairly fancy prices for good home tape recorders; tags on high-quality machines are often $300 and upward. Yet their specifications are not quite as fancy when compared with those for other audio equipment such as tuners and amplifiers. A high-priced tape machine may claim S/N (signal-to-noise ratio) of only 50 dB, while tuners and amplifiers vaunt S/N of 60 dB, 70 dB, and even higher. A tape machine may claim relatively flat response over the range of 50-15,000 Hz, while other equipment will claim virtually flat response from 20 to 30,000 Hz, 50,000 Hz, or higher.

Understandably, in judging the quality of a tape recorder by its specifications—presented by the manufacturer or in an equipment profile—the audiophile will want to know what can be considered excellent performance in light of today's state of the art. And he will want to know what is good, and what is only fair. It is the purpose here to help him acquire such knowledge, guided by (1) performance achieved by today's top-flight home machines, and (2) performance standards set by NAB (National Association of Broadcasters) for professional machines.

Much of what we have to say concerns three principal criteria of performance: frequency response, S/N, and distortion. We shall also talk about such things as motion, crosstalk, channel separation, tape speeds, head gaps, various tape recorder features, and the like.

Interdependence of Frequency Response, S/N, and Distortion

We must remind readers how interdependent the three parameters noted above are in the case of a tape recorder. An improvement in one aspect of performance necessarily entails a sacrifice in one or both of the other two aspects. For example, a significant extension of treble response—say from 15,000 to 20,000 Hz at 7.5 ips (inches per second) —could be achieved if one is willing to let distortion rise appreciably and/or let S/N drop appreciably. A substantial improvement in S/N could be obtained by letting distortion and/or frequency response go to pot. And so forth.

Why is this so? Perhaps we can most easily explain by illustration. Suppose that at 7.5 ips, using conventional tape, a machine correctly specifies that frequency response is down no more than 2 dB at 15,000 Hz; that S/N is 55 dB at peak recording level; and that harmonic distortion is 3% at peak recording level. Without changing the machine or tape or state of the art, what could we do to extend treble response to, say, 22,000 Hz for Fido's benefit? One thing that might be done is to decrease bias current a little. The effect of reducing bias current through the record head is to reduce "bias erase"—the tendency of the head to erase the very frequencies it is trying to record. Since bias erase becomes more severe as frequency rises, a reduction in bias current results in improved treble response.

So far, so good. But something else happens when bias current is cut. Distortion rises, and quite sharply. Hence the gain in treble response comes at the expense of more distortion. On the other hand, might it be possible to avoid the rise in distortion? Yes. How? Simply by recording at a lower level.

But with less signal recorded on the tape, we get lower S/N in playback. Hence we have traded better treble response for a deterioration in S/N. Or we could take a compromise position by trading the treble gain for moderate sacrifices in both distortion and S/N.

With a little imagination, the reader can figure out the consequence of trying to achieve an improvement in S/N or distortion.

S/N Specifications

S/N deals with the ratio, in dB, between the desired audio signal and noise due to the record and playback amplifiers, hum picked up by the playback head, and tape noise.

S/N and distortion specifications tend to be the ones most puzzling, not only to amateur audiophiles but to professionals as well. This is because the meaning of these two specifications is tied to a precise definition of peak recording level (also known as maximum permissible recording level), and unfortunately this definition is not always clearly given.

For a number of years it has been the practice of high-quality machines for home use (and even professional use) to define peak recording level as that which at 400 Hz produces 3% harmonic distortion on (conventional) tape. (To make an important digression: VU meters in machines of professional quality are generally set to read 0 VU either at a recording level that produces 1% distortion or, pretty much the same thing, at a level about 8 dB below that which produces 3% distortion. The reason is to allow for the mechanical lag of the pointer, which cannot follow sharp transients and therefore tends to under-indicate peak recording level. By keeping the VU pointer at or below 0 VU, the recordist therefore has a margin of safety of about 8 dB.) Experience has shown that a recording level which does not exceed 3% harmonic distortion at 400 Hz, where audio energy is apt to be most intense, results in quite acceptable reproduction. The forthcoming NAB test tapes contain an "NAB Standard Reference Level," which is a 400 Hz tone recorded at a level 8 dB below that producing 3% third harmonic distortion at 7.5 ips.

Therefore let us be satisfied to consider peak recording level that which at 400 Hz produces 3% harmonic distortion on conventional tape at 7.5 ips. Accordingly, S/N is the ratio in playback between a 400-Hz signal recorded on a machine at peak recording level and all the noise produced by that machine in the absence of such a 400 Hz signal.

Using conventional tape, the NAB standard calls for S/N to be at least 53 dB at both 7.5 and 3.75 ips for quarter-track operation (and about the same for half-track). However, there are some home machines that under these conditions can achieve S/N as high as 56 dB, and perhaps a bit more. Therefore excellent, good, and fair S/N performance might be defined as shown in Fig. 1.

The peak recording level is sometimes defined as that which produces 1% harmonic distortion. Then the specified S/N ratio should be adjusted upward by about 8 dB. Thus if the specification states that S/N is 44 dB below the 1% distortion level, we can state that S/N is about 52 dB with reference to the recording level that produces 3% distortion. The reference level may be given as 0 VU. Properly—but not always—0 VU should correspond to a recording level that produces about 1% harmonic distortion and is about 8 dB below that which produces 3% harmonic distortion. So again we can add 8 dB to a specification based on 0 VU. WARNING: Be sure

Parameter		Excellent	Good	Fair	Unit
S/N		54-56	50-53	45-50	dB
Frequency response	7½	30-16,000	50-13,000	50-10,000	Hz
	3¾	50-12,000	50-10,000	50-8000	Hz
	1⅞	50-8000	50-7000	50-5000	Hz
Uniformity of playback response		±1	±2	±3	dB
Uniformity of record/playback response		±2	±3	±5	
Wow and flutter	7½	0.1	0.2	0.25	%
	3¾	0.2	0.25	0.3	%
	1⅞	0.3	0.4	0.5	%
Absolute speed error		0.3	0.5	1.0	%
Crosstalk		60	54	48	dB
Channel Separation		40	35	30	dB

Fig. 1—How to interpret Tape Recorder Specifications.

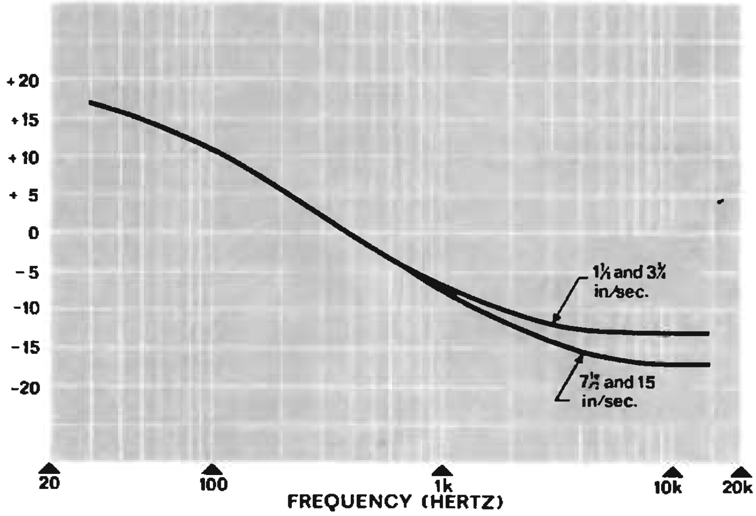


Fig. 2—Standard reproducing characteristics for magnetic tape records. Reference frequency: 400 Hz.

that the distortion level is 1% when the VU meter is driven to 0 VU while recording a steady 400 Hz tone. To be sure, consult the tape machine's specifications or its manufacturer or an equipment profile of the machine.

On occasion one may find that peak recording level corresponds to 5% harmonic distortion at 400 Hz. Then one should subtract about 6 to 8 dB from the specified S/N. Ror example, a rating of 58 dB would be reduced to about 50-52 dB.

One more word of caution: Make sure that the S/N specification pertains to the entire record-playback process, including the heads. At times it has been found that a specification was related only to the tape electronics, enabling a manufacturer to claim a juicy 60 dB.

Distortion

Much of the preceding section bears on the distortion specification as well as on the S/N one. We have seen that distortion is really a matter of definition rather than of design in the case of a tape recorder. For the purpose of calculating S/N, the accepted practice of high-quality home tape recorders is to use as a reference the recording level which produces 3% harmonic distortion on the tape at 400 Hz. The practice might just as well have developed of rating S/N on the basis of 2% or 1% distortion; but this would not mean that the tape machine produces any less distortion. The distortion level is due to the amount of signal that one tries to get onto the tape, not to the quality of the tape machine.

Tape recorder specifications rarely mention IM (intermodulation distortion). The fact is that when a recorder is producing only about 1% THD (total harmonic distortion), IM is substantially higher; a typical figure might be 6% or 8% IM. When a machine is producing 3% THD, measured IM might rise to a figure like 12% or 15% or more. Because such figures are distressing to the eye, they are omitted. But the results appear considerably less distressing to the ear. For one thing, we are talking about IM at or close to peak recording levels, which are only occasional. For another, some part of the measured IM products may be outside the range of hearing.

Frequency Response

If one were to record and play a tape with a machine containing no frequency equalization, there would be considerable bass and treble loss in playback. For reasons having to do with maximization of S/N and minimization of distortion, it has been found desirable to correct for the bass and treble losses by supplying bass boost largely in playback, and treble boost laregly in recording. Accordingly, there are standard playback characteristics for each tape speed; the characteristic shows how much bass boost shall be supplied in playback. No standard characteristic is specified for recording. Instead, broadly stated, the standards provide that recording equalization (mainly treble boost) shall be such that, when taken in conjunction with playback equalization, the result shall be flat response (within specified tolerance limits). In other words, the record-playback process shall be such that the magnitude of each audio frequency relative to 1000 Hz (or relative to any other reference frequency) shall be the same at the output of the process as at the input.

The playback characteristic of a tape machine should be as shown in Fig. 2. For 15 and 7.5 ips, the standard response curve consists of about 36 dB of bass boost, commencing (3 dB up) at 3180 Hz and leveling off (3 dB below maximum) at 50 Hz. For 3.75 and 1.875 ips, there is about 31 dB of bass boost, this time commencing at 1770 Hz and again leveling off at 50 Hz.

It must be clearly understood that this playback curve is not simply that of an equalization circuit in the tape playback amplifier. Therefore it is not sufficient to measure the playback equalization of the tape amplifier to determine whether playback response is correct. Instead, the standard response curve is the *combined* result of the equalization circuit plus the deviations of the actual playback head from those of an "ideal" head. Compared with an ideal head (which displays response steadily rising 6 dB per octave for constant magnetic flux through the head), actual heads tend to display irregularities in the low-bass and high-treble frequencies. Typically these consist of moderate treble loss (due to gap width, and coil and cable capacitances), and exaggerated bass (due to the entire head and not merely its gap reacting to the flux emanating from the tape at long wavelengths, i.e. low frequencies). The playback amplifier is required to compensate for such aberrations, resulting in the *total* response curves of Fig. 2.

It is beyond the capability of most people to measure the total playback response of a tape machine, including that of the playback head. Therefore the means of checking playback response is to play a test tape and measure response at the output of the tape amplifier. (At the time of writing, the NAB test tapes are not yet available. However, other test tapes, such as those of Ampex, serve in the interim.) If the machine has perfectly accurate playback response, the output of its tape amplifier will be flat. In other words, all recorded frequencies on the test tape will have the same measured amplitude at the amplifier output.

The criteria of playback response are

the width of the frequency range covered and the uniformity (flatness) of response over this range. In broad terms, the NAB standards suggest that high-quality performance requires a range of about 30 to 15,000 Hz for 15 and 7.5 ips; of about 50 to 10,000 Hz for 3.75 ips; and of about 100 to 5000 Hz for 1.875 ips. And for the bulk of each indicated range, these standards suggest a uniformity of about ±1 dB for 15 and 7.5 ips; of about +1, −1.5 dB for 3.75 ips; and of ±2 dB for 1.875 ips. Reduced to tabular form, these criteria also appear in Fig. 1.

The indications of Fig. 1 must be read with understanding and probably also with some grains of salt. For one thing, the suggested ranges presume that there is a compromise among the conflicting requirements of broad frequency response, low distortion, and high S/N. For another, the reader should not be too literal. For example, if the playback response of a machine at 7.5 ips is ±1 dB to 15,500 Hz rather than to 16,000 Hz, this too is probably excellent. Also, the reader should allow for some extra dropoff at the extremes of the audio range. To illustrate, if a machine maintains ±1 dB response between 50 and 12,000 Hz at 7.5 ips, but drops off to −2 dB at 30 and 16,000 Hz, this again is probably excellent.

We turn now to the question of record-playback response, and here we have a problem, namely the extent to which we should allow the NAB standard for professional machines to affect our viewpoint concerning machines for home use. The intention of NAB is that, regardless how inaccurate the playback response of a particular machine may be, its recording characteristic shall be accurate, enabling tapes recorded on this machine to yield flat response when played on *other* machines with accurate playback response. In other words, inaccuracies in playback response shall *not* be compensated by opposite inaccuracies in recording equalization so as to yield flat record-playback response.

This is appropriate for professional machines, which turn out recordings to be played in large part on other machines. But the recordings of home machines are played largely or entirely on the same machine. Therefore, in the case of home machines, we *are* concerned with the flatness of record-playback response, quite apart from the accuracy of playback response (important only in playing tapes made on other machines, such as pre-recorded tapes). Accordingly we shall be guided by NAB standards only in the following sense: permissible deviations from accurate recording will be added, more or less, to permissible deviations from accurate playback to arrive at a concept of accuracy on a record-playback basis.

Excellent, good, and fair *range* of record-playback response may be considered the same as for playback range of response, already discussed, and are also shown in Fig. 1. As before, understanding and salt are needed. For example, if response at 7.5 ips is down 2 dB at 13,000 Hz and 3 dB at 16,000 Hz, that is still excellent.

Head Gaps

Because a narrow head gap plays an important part in achieving extended treble response, specifications often state the gap width. Thus a typical figure for a playback head or a record-playback head is .00005″ — 50 micro-inches.

The recordist should realize that a narrow gap is crucial only for playback. For recording, a relatively wide gap is more desirable. If the tape machine has separate record and playback heads, an appropriate value for the record head is more nearly 500 than 50 micro-inches.

The question then is how narrow the gap need be to sustain treble response. A close answer may be obtained from the equation $G = 900{,}000S/2f$, where G is gap in micro-inches, S is tape speed in inches per second, and f is frequency in Hz. To illustrate, assume it is attempted to achieve substantially flat response to 10,000 Hz at 1.875 ips. The indicated gap width is $G\ (900{,}000 \times 1.875)/(2 \times 10{,}000) = 1{,}687{,}500/20{,}000 = 84$ micro-inches.

Since in actual tape machines one finds playback gaps as narrow as 50 micro-inches, and perhaps narrower, it is unlikely that gap width of this order is a limitation on treble response—assuming the gap has very straight edges as well as being very narrow. Much more likely as the cause of the limitation are the severe magnetic losses that occur in recording.

Tape Speeds

Earlier discussion has suggested the quality of performance which can be achieved at 7.5 ips, which is now the NAB standard speed, and at 3.75 ips. More explicitly, in the present state of the art—encompassing tape heads, tape electronics, tape, knowledge of optimum equalization techniques, and mechanical design— high-fidelity performance is readily achievable at 7.5 ips, and within grasp at 3.75 ips. In some top-flight home machines, performance at 3.75 ips is little if at all distinguishable to the ear from that at 7.5 ips, although instruments can measure differences. Hence it is desirable that a home machine offer tape speeds of at least 7.5 and 3.75 ips.

It doesn't defy memory to recall a time when tape recorders of good reputation sought to extend response only to about 8000 Hz at 7.5 ips. Today this kind of response is attainable at 1.875 ips, along with reasonably good performance in other respects (S/N, distortion, motion, and so on). Accordingly, 1.875 ips may now be considered a "serious," and therefore desirable, speed for such things as background music, speech, and other applications where requirements are less stringent than for high fidelity reproduction but nonetheless fairly demanding. The art advances.

Format

Quarter-track operation, either stereo or mono, is the usual format for home machines. That is, provision is made for recording four tracks, each nominally 43 mils wide, with three islands of 24 mils each between tracks. With tape moving from left to right and the oxide facing away from the operator, the tracks are numbered 1 to 4 from top to bottom of the tape.

For stereo, tracks 1 and 3 are used in one direction of tape travel, respectively carrying the left and right channels; tracks 4 and 2, respectively carryinging the left and right channels, are used in the other direction. For mono, the operating sequence is 1-4-3-2. In mono, the machine should be capable of recording each track individually; to make this point clear, the machine should not be erasing track 3 while recording track 1, and vice versa; nor should it be erasing track 2 while recording track 4, and vice versa. By the same token, the machine should be capable of playing each track individually.

Motion

The NAB standards require that flutter and wow, on an unweighted, rms basis, shall not exceed 0.2% at 7.5 ips, 0.25% at 3.75 ips, and 0.5% at 1.875 ips. Wow consists of a few (up to about 10) deviations per second from average tape speed; flutter consists of more frequent deviations. The flutter and wow percentage shows the relationship between the amount of speed deviation and the average speed.

Taking into account not only the NAB requirements but also the present capability of high quality home machines, these criteria are also tabulated in Fig. 1.

NAB requires that professional machines have a speed error of no more than 0.2% (plus or minus) throughout the reel. This means that in an hour of recorded material, there shall be a gain or loss of no more than 7.2 seconds, which is a rather rigorous requirement. The figures for speed accuracy are also shown in Fig. 1.

As an important operating convenience, particularly where time is money, professional machines provide for very fast wind and rewind of tapes, with a travel rate of about 1200 feet in 30 seconds being typical. In home machines, however, this is apt to take more like 60 to 90 seconds, and sometimes as much as two minutes. To us, a fast wind specification of about 60 to 90 seconds for 1200 feet appears more desirable than one appreciably under 60 seconds. The reason is that home machines seldom wind the tape as smoothly as professional machines do, and excessively fast winding may cause tape stresses that deform the tape and distort the sound.

Crosstalk

This refers to signal leakage between adjacent tape tracks. NAB specifies that between 200 and 10,000 Hz the difference between the original signal on one track and the leakage signal on the adjacent track shall be at least 60 dB. This may also be considered the standard of excellence for home machines. Inasmuch as S/N, although excellent, is likely to be a smaller figure—say 55 dB—then crosstalk is apt to be masked by other noise. Excellent, good, and fair performance criteria are tabulated in Fig. 1.

Channel Separation

This refers to signal leakage between channels of a stereo system—between the left channel and right channel sections of the tape heads, or between the left channel and right channel sections of the tape electronics. NAB specifies that between 100 and 10,000 Hz there shall be channel separation of at least 40 dB; this may be considered the standard of excellence for home machines as well, also shown in Fig. 1.

Why is the requirement for channel separation considerably less stringent than for crosstalk? The reason is that the question of channel separation arises only for stereo signals, which are ordinarily related to each other. Leakage of signal from one channel to the other is therefore unlikely to be audible, and if audible it is unlikely to be disturbing. But in the case of crosstalk, where adjacent tracks contain unrelated signals (see the earlier section on Format), a small amount of signal leakage can be audible and disturbing.

Record-Level Indicator

Inasmuch as the present subject ties in with distortion and S/N, the reader is referred back to the earlier sections on the latter two subjects for necessary background.

If the indicator is a magic eye tube, it has virtually instantaneous and full response to sharp transients. It truly indicates the level of signal peaks, which cause the most distortion. Accordingly, the magic eye should be adjusted so that it barely closes when recording a steady 400-Hz tone at a level that produces 3% harmonic distortion on the tape. Should the eye not close until a higher recording level is reached, excessive distortion may appear on the tape. Should the eye close before the 3% distortion level is reached, the recorded signal level may be of insufficient magnitude, resulting in poor S/N.

If the indicator is a VU meter (or a meter with reasonably similar characteristics), being a mechanical rather than electronic device it will not respond fully to signal peaks; it will tend to under-indicate their magnitude, inducing the operator to record at too high a level, with consequent excessive distortion. Therefore it is customary to provide "headroom" by having the meter indicate peak recording level—that is, 0 VU—well before a steady 400-Hz signal produces 3% THD on the tape. The general procedure is to allow about 8-dB headroom. One way is to adjust the meter to indicate 0 VU when recording a 400-Hz signal at a level 8 dB below the level which produces 3% harmonic distortion on the tape. Another, and approximately equivalent, way is to adjust the meter to indicate 0 VU when recording a 400-Hz signal at a level which produces 1% harmonic distortion on the tape. Hence the specifications should indicate that one of these two methods of adjusting the VU meter has been employed.

(A number of persons believe that 8 dB is often not enough headroom. Some program material may contain peaks as much as 15 and even 20 dB above the level indicated by the VU meter. Therefore, an electronic eye or an oscilloscope may be desired as an adjunct to or substitute for the VU meter as a record-level indicator. Another alternative is a peak-reading meter, rather than one which indicates average level as does the VU meter.)

Other Things to Consider

Thus far in the main we have been discussing the quality of performance of a tape machine as revealed by its specifications. The specifications may have a good deal further to say which is of interest to the audiophile. However, the additional information is likely to bear not on measures of performance but on *features* which provide operating convenience and flexibility. Following are a number of the features that may be specified. (The order of presentation is of no significance.)

Sound-on-Sound. This provides for synchronized, sequential recording of two or more sound sources on a single track. For example, one person can become a trio, quartet, and so on by: recording his voice on track 1; playing track 1, recording this signal on track 3, and at the same time adding his voice afresh to track 3; playing track 3, recording the track 3 signal on track 1, and adding his voice afresh to track 1; and so on as long as desired.

Echo Effect. This permits the recorded sound to be repeated on the same track at small intervals and at successively diminishing magnitudes, thus achieving an effect akin to an echo.

Reverse Operation. This saves the operator the inconvenience of having to interchange the tape reels on the hubs when desiring to operate the tape in "opposite direction" — for example when desiring to record tracks 4 and 2 after recording tracks 1 and 3. Instead, the machine permits the direction of tape travel to be reversed. A number of machines feature automatic reverse; some only in playback, and others both in recording and playback; some only in one direction of tape travel, and some in both directions.

Tape Lifter. Some machines permit the tape to remain in contact with the heads when the tape is being rapidly wound or rewound, thereby accelerating head wear. The tape lifter spaces the tape away from the heads during rapid wind.

Pause Control. Generally, if the tape machine is in the recording mode and if the transport is stopped, the machine is automatically taken out of recording mode, thereby avoiding accidental erasure when operation is resumed. However, a pause control may be offered that enables the user at his option to stop the transport yet keep the machine in recording mode.

Tape Counter. To facilitate finding a passage in a reel of tape, most home machines provide a digital counter with a reset knob or button.

Push-Button Controls. Some machines use relatively "primitive" means of putting the transport through its modes of operation — by knobs or a "joystick." Others offer push-button controls. These may still be mechanical devices, although fancier in appearance. Or they may control solenoids, enabling the user to operate the transport easily and quickly with a light touch.

Inputs and Outputs. The tape recorder should have at least two inputs, one for high-level sources such as a tuner or audio preamplifier; and the other for low-level sources such as a microphone (and sometimes for a magnetic phonograph pickup, including proper equalization). If the tape machine contains its own power amplifier and speaker, it should still provide an output jack for feeding an external amplifier. For minimum distortion, the output signal should be taken at a point prior to the output transformer (if any) of the internal amplifier. It is further desirable that the machine provide a second output jack for an external speaker; in this case the signal would be taken from a point following the internal amplifier.

Returning to the subject of a low-level input for microphones, most home machines provide only for accepting the signal from a high-impedance microphone, which puts out a relatively high signal. Some, however, provide for accepting the signal from low-impedance microphones, which deliver less signal but permit a long run of cable between the microphone and tape recorder without appreciable loss of highs. (If the tape recorder will accept only a high-impedance microphone and the recordist wishes to use a low-impedance microphone, it is necessary for him to insert a step-up transformer between the microphone and the recorder.)

Input & Output Sensitivity. High-level sources such as FM tuners, TV, audio preamps, and so on, ordinarily deliver at least 0.5 V. Therefore the figure for input sensitivity of the tape recorder on high-level input should be no more than 0.5 V; in other words, no more than 0.5 V should be required to drive the tape recorder to peak recording level as shown by the record-level indicator. To allow some margin of safety, a number of machines have input sensitivities ranging down to about 0.1 V.

Similarly, the tape recorder should have adequate sensitivity on low-level input. The sensitivity figure should be no more than about 10 millivolts, while a sensitivity of about 2 or 3 millivolts is preferable.

Inasmuch as most high-fidelity amplifiers or preamps can be driven to the desired level by a signal of about 0.1 to 0.5 V, the tape recorder should be able to deliver a correspondingly strong output signal. Allowing for a margin of safety, the tape recorder should be able to deliver peak signals of about 1 V to following audio equipment.

Automatic Equalization Change. Equalization requirements vary with tape speed both in recording and playback. Most home machines automatically provide the necessary change in equalization as the user selects the desired tape speed. However, in some top-flight recorders this change is not performed automatically, and it is necessary for the operator to keep his wits about him to make sure that he does not use the wrong equalization in recording. (Making a mistake in playback is much less consequential, because no *permanent* harm is done by forgetting to set equalization correctly.)

A-B Switch. A machine having separate record and playback heads and thereby permitting simultaneous recording and playback will usually (but not always) incorporate an A-B switch. When recording, this permits the playback signal to be compared with the incoming signal, so the operator can check recording quality for such things as faithful frequency response, low distortion, and high S/N. By flipping the A-B switch between the positions representing the input and playback signals, he obtains an almost instantaneous and valid comparison. The comparison can be made by means of earphones, or by one's audio system if properly connected to the tape recorder. (The A-B switch is not strictly necessary, because audio preamps generally contain such an A-B switch.)

Bias Indication. Correct setting of bias current through the record head is critical in order to achieve the optimum combination of extended treble response, low distortion, and high S/N. This setting is performed by the manufacturer at the factory, or else by a competent technician or machine operator. To make sure that bias is at the proper value, some of the best machines enable bias to be read on the machine's VU meter at the turn of a switch.

Adjustment Facilities. In order to achieve optimum performance, the tape recorder should permit a variety of fine adjustments, including bias magnitude, bias frequency, input gain, output gain, record-level indication, erase current, azimuth, and perhaps others (such as equalization). To the extent that the tape machine contains controls facilitating such adjustments, one can get the maximum performance designed into the machine. (Of course these adjustment controls add to the cost of the machine and to the complexity of maintaining it.)

Editing Facilities. If the user wishes to edit tape in fine detail—for example by making splices at a given syllable or musical note—it is important that the transport provide easy access to the portion of the tape that is directly in front of the heads. It is also necessary that the operator be able to move the tape by hand past the playback head and thereby find exactly the sound he is looking for. Some machines pay careful attention to such editing facilities.

Number of Motors. It is well settled by now that a tape transport with a single motor rather than three (one for driving the capstan and the other two for driving the supply and takeup reels) can deliver mechanical performance of excellent quality. However, for greater durability, greater operating convenience, and faster and smoother rapid winding, three motors tend to be superior. These virtues are generally of more importance to the professional user than to the home recordist. There is some gain with a one-motor machine in terms of a lighter, more compact, and less expensive machine.

The better home machines usually employ a synchronous motor to drive the capstan; this helps to achieve accurate speed and keep speed constant throughout the reel. Æ

How Tape Recorder Bias Controls Fidelity

ANDREW H. PERSOON*

How to adjust bias for best performance from your tape recorder.

MAGNETIC RECORDING TAPE provides a superior method for the permanent recording of information, but it is limited by the natural phenomena of magnetic properties. Fortunately, the shortcomings created by these magnetic phenomena can be compensated for by electronic measures. This article will make no attempt to explore the mathematical or theoretical realms of magnetic recording but will present a simplified explanation of high-frequency bias, its requirements and limitations, and methods of adjustments.

Every magnetic medium exhibits a non-linear characteristic because the magnetization, resulting from an exposure to a magnetic field (such as that produced by the recording head), is not directly proportional to the strength of the field. This non-linear characteristic, if not corrected, would result in severe distortion of the audible recorded information. The use of a high-frequency bias current; applied through the recording head, is the standard method of compensating for the non-linearities in the transfer of electro-magnetic signals onto magnetic recording tape.

The high-frequency bias signal is usually generated by an oscillator circuit in the recorder electronic system and is added to the signals generated by the microphone or supplied by the recorder input circuits. The bias is a high frequency, usually 30 to 100 kHz, which is above the range of hearing. Therefore, during playback of only the bias signal, one could not hear or identify any tones which would indicate its presence. By adding the bias signal to the audio signal, a resultant signal is produced (Fig. 1). In most recorders, the two signals are simply combined without any form of modulation. The resulting signal is what the record head inductively converts from electrical signals into magnetic fields which influence the magnetic tape.

As previously stated, every magnetic medium exhibits a non-linear characteristic. This non-linearity is best illustrated by the Transfer Characteristic Curve which is mathematically derived from a family of hysteresis loops (Fig. 2). The hysteresis loops and transfer curve indicate the degree of tape magnetization which results from an exposure to a magnetic field such as that produced by the recording head. The transfer curve also indicates that the non-linearities exist only at the extremely low signal level (center portion of the curve) and at the very high signal levels (saturation areas) which are at the extreme ends of the curve. The remainder of the curve is relatively straight and allows linear and proportional transfer of magnetic signals.

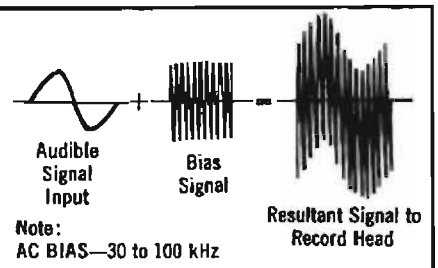
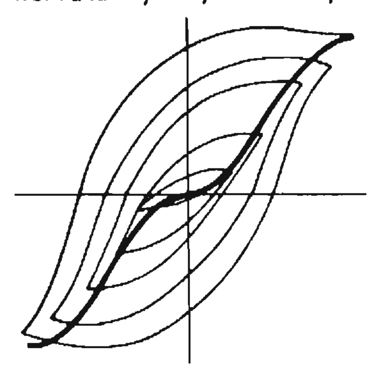


Fig. 1. Production of resultant signal.

Fig. 2. Transfer curve derived from a family of hysteresis loops.



The transfer curve shown in Fig. 3 illustrates the resulting tape magnetization from a magnetic signal generated by the recording head. The curve is typical of those for recording tape and no attempt is made to show non-linearities and signal losses created by either the record head or recorder electronic systems.

As the magnetizing force increases (greater record-head output in terms of magnetic-flux-field intensity) the resulting tape magnetization also starts to increase. Notice that the vertical segments of the transfer curve are relatively straight. It is within these straight segments of the curve that undistorted recording take place. The straight segments indicate that a linear and proportional relationship exists between a given input and the resulting output. This relationship may change for different types of tape because of differences in the magnetic properties exhibited by various oxide coatings.

The straight portions of the curve continue until saturation in either the positive or negative direction occurs. At the saturation points, no effective additional tape magnetization will occur even if the magnetizing force continues to increase. Recording into the saturation levels may produce distortion and tape noise, and may reduce frequency response.

To visualize the recording process, the transfer curve illustrates the resultant signal waveform (sum of bias and input signals), and its transfer

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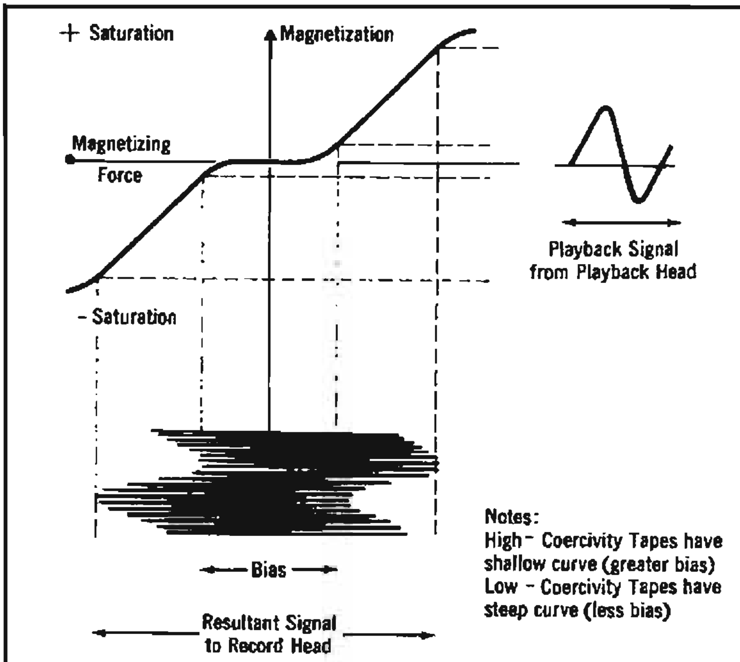

Fig. 3. Recording tape transfer curve.

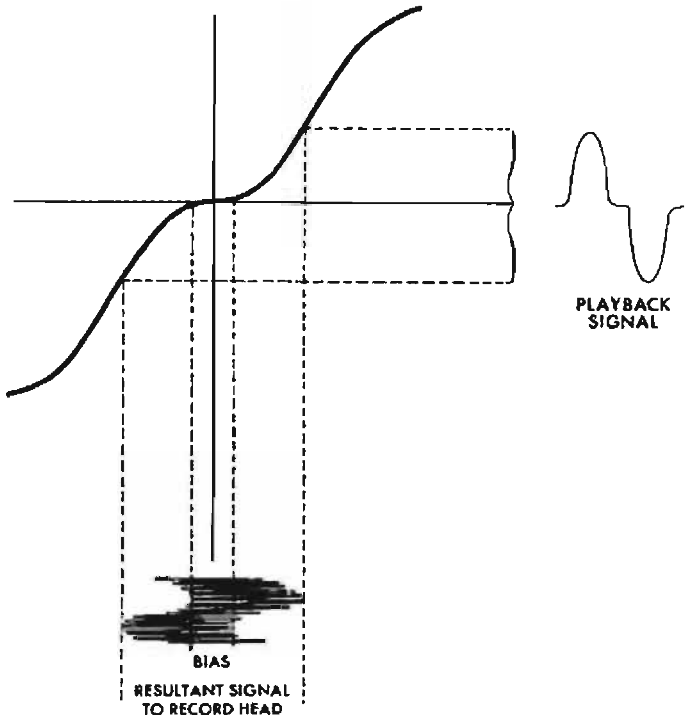

Fig. 4. Under-bias condition.

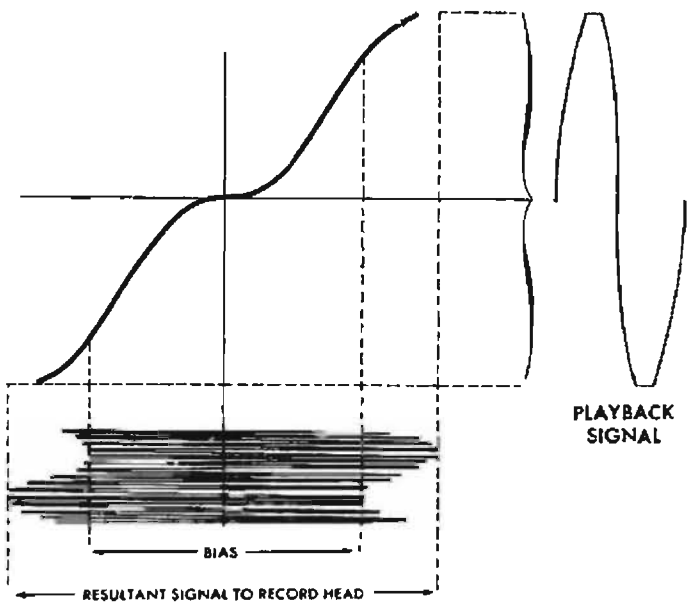

Fig. 5. Over-bias condition.

across the curve to form the recorded signal waveform. (Fig. 3) Observe that the signal with bias essentially bridges the "zero-point" and the low-signal-response portion. The bias position across the curve allows the signal-changing portions of the input waveform to fall onto the linear segments of the curve.

The shift of the input waveform across the transfer curve to form the recorded-signal waveform shows that the non-linear segment is essentially removed by the bias signal, and the recorded signal is relatively distortion free. Also, it can be visualized that either a low- or high-bias condition will drive the signal onto the non-linear segments of the curve and will cause distortion.

With a low-bias condition (Fig. 4), the low-level input signals fall onto the "zero-point" region and either may be severely distorted or not even be reproduced at all. In a high-bias condition (Fig. 3), the high-frequency response will decrease. The high frequencies will distort sooner or go into saturation because of a phenomenon called "self-erasure." Also, the signal-to-noise ratio may be reduced causing undesirable tape noise.

The transfer curve is typical of most magnetic recording tapes but each particular type of tape will exhibit a different slope, and a different "zero-point" region, as well as different saturation peaks. The differences of the curve shapes are created by the individual magnetic properties exhibited by each tape type. As the shape of the curve changes so do the bias requirements.

A low-coercivity tape has very steep linear segments and will require less bias current. On the other hand, a high-coercivity tape has relatively shallow linear segments which require a greater bias current. Because of the differences in tape magnetic properties, the slope of the curve changes and the bias level required to eliminate distortion will change accordingly.

To evaluate the changes of bias requirements involved with different types of tape, the wavelength response of the tape must be considered. Bias current is required to eliminate distortion but is also directly involved with frequency response and output. In terms of

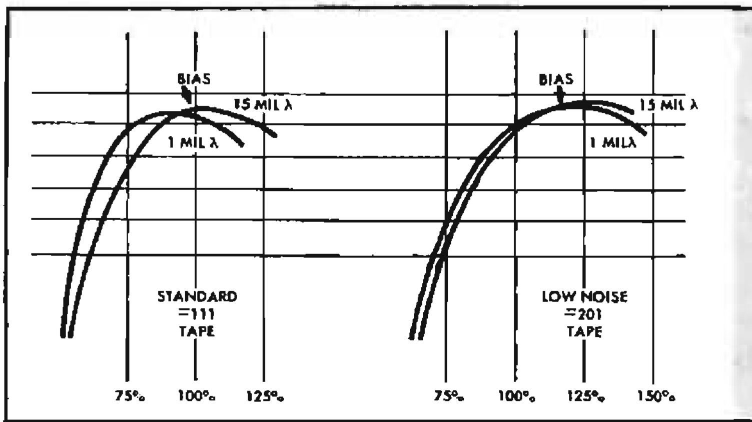

Fig. 6. Bias curve.

response and output, the bias requirement is related to tape construction such as: type and thickness of coating; quality of oxide dispersion forming the coating; and smoothness of the coating surface.

As a general rule, high-frequency response can be improved by using a tape with a high-coercivity oxide, relatively thin coating depth, and a smooth (specially prepared) coating surface. These improvements of high frequencies may have an opposite effect for the low frequencies to the extent that they may not be reproduced with the same efficiency. This situation becomes apparent in the bias curves (Fig. 6).

Because of the slight differences that may occur in the reproduction of the different frequencies, the actual bias setting is of a selective nature. The ideal situation would be one where the bias setting is at the point of peak output for all frequencies. Note that a bias setting is easily accomplished for the low-noise tape as shown in the bias curves (Fig. 6). For this particular tape both the high-frequency (1-mil wavelength) and the low-frequency (15-mil wavelength) output peaks coincide with each other allowing the bias setting to be at the overall output peak.

In the case of the standard tape, the high- and low-frequency (short- and long-wavelength) output peaks do not coincide at maximum output. The bias setting could be at either output peak or at the mid point. In normal recorder adjustment however, the bias setting most often used is at the output peak of the longer wavelengths. This setting is justified because the greatest percentage of recorded information is in the low- or mid-range portion of the frequency spectrum. To compensate for any unbalance in response output, the equalization settings of the recorder are adjusted until the overall output frequency response is flat.

Typical Bias-Adjustment Procedure

The bias settings shown in the illustrations indicate only a relative bias-level comparison between two different types of tape. The percentage-value relationship will generally hold true for most recorders. Specific information on bias adjustment or settings is impossible to enter into here because of the large variety of recorders in use. Most recorders have their own individual requirements and specifications for bias current (or voltage) adjustments. If a bias-level adjustment is attempted, care should be taken to assure correct settings. The recommendations of the recorder manufacturer must be followed precisely.

As mentioned in the preceding paragraphs, the low-midrange-frequency (longer-wavelength) output peak is generally used to obtain the most desirable bias setting. The normal adjustment frequency (for 7½-ips tape speed) is 500 to 1000 Hz. This audio signal is available from an audio, function, or signal generator which most electronic repair facilities have available.

The following adjustment of recorder bias is typical of many machines now in use. For stereo recorders, the adjustment procedure must be followed for both channels. Before attempting any adjustment, be sure that the machine is operating properly, the record and playback heads are clean and in good condition, and thoroughly review the manufacturers service manual. The bias-adjustment range, location, and function of controls, and the operation and scale of the output meters (VU meters) must be understood. Since the bias setting is determined by the type of recording tape, establish the basic type of tape you use most often. Prepare the machine for normal recording at 7½ ips with a low-signal-level input (approximately 20 dB below tape saturation).

Set GAIN, RECORD VOLUME, or LEVEL adjustments low to avoid the possibility of recording in the saturation levels. Adjust the signal generator (1000-Hz, signal source) for a low-voltage output and connect to the recorder input terminals. If the recorder is a three-head type, listen to the recorded signal while recording the 1000 Hz. Slowly increase the bias and observe any increase of output as indicated on the VU meters. An increase of intensity of the playback signal should also be heard. Continue to adjust the bias, starting at low output, until the maximum output signal is observed. Continue to increase bias until the output begins to drop, indicating an overbias condition (Fig. 6), and return the bias setting to the point of maximum output.

If the recorder is a 2-head type, the set-up procedure is similar except that a series of short recordings is made, each with a change in bias, and then played back afterward. A simple method is to voice-identify the recording segment and bias setting and record the 1000-Hz signal for 10 seconds, readjust the bias and record another segment. Repeat this procedure over the entire range of bias control. Then play back all the recorded segments noting which one has the greatest fidelity and intensity, and set the bias accordingly.

The recommended bias setting for most recorders is where maximum output is indicated for the 1000 Hz signal. This setting coincides with the low-frequency (long-wavelength) output peaks as shown in the response-curve illustrations.

After the correct bias adjustment is obtained, a corresponding equalization-control adjustment may be required in some cases to compensate for differences in overall frequency response. Usually a simple listening test of recorded material will determine if the overall response is correct. Æ

The Cross-Field Technique

FRITJOF BRODTKORB*

Description of a biasing technique which is claimed to improve performance and frequency response at low tape speeds

Obtainable bandwidth in magnetic tape recording is limited by tape speed. When the wavelength of the signal recorded on the tape approaches the width of the recording zone, the frequency response is severely reduced.

To move the critical frequency upwards, the recording zone must be made narrower. This can be accomplished either by reducing the bias and using thinner tape or by introducing the cross-field technique. The aspects of the two alternatives are discussed here, and it is claimed that the cross-field technique on long-play tape gives 6-dB lower tape noise than the conventional technique on triple-play tape.

As lower tape speeds become more common for sound reproduction from magnetic tape, the difficulties in maintaining a satisfactory dynamic range are accentuated. The limitations are imposed partly by the increased preemphasis required for obtaining a given frequency range at a satisfactory signal/noise ratio.

Recent developments in these fields, based on the work done at Tandbergs Radiofabrikk A/S.

The tape recording technique developed rapidly during and shortly after the second world war when plastic tape with a coating of magnetic material was introduced, opening the possibility for application of high-frequency biasing.

The magnetizing process is inherently nonlinear, of course. If the tape is magnetized by the signal alone, the resulting signal played back from the tape will be severely distorted. In the early days of tape recording, steel wire was used as the magnetic medium. It was then found that the distortion could be reduced by exposing the wire to a permanent magnetic field superimposed on the signal field during recording. This resulted, however, in a strong background noise from the wire during playback. A great step forward was taken when the permanent magnetic field was replaced by an alternating field of high frequency, as the plastic tape coated with a thin magnetic layer became available. When such a tape is exposed to a magnetic field simultaneously excited by the signal and the high-frequency bias, a virtually distortionless reproduction at a very low noise level is obtained. With this technique the magnetic tape advanced to a leading position amongst media for recording and reproduction of sound.

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In most recent years, efforts have been aimed at the development of better tapes and recording techniques which will make it possible to record at the lowest possible tape speed while maintaining an adequate frequency range.

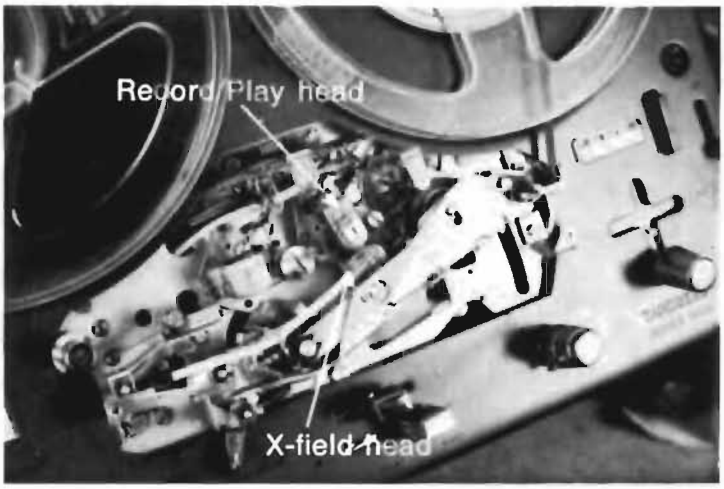
Methods to Extend Frequency Range

Before these improvements are described, it is best to take a closer look at the recording process itself.

An important limitation is imposed by self-erasing, which occurs in the recording zone as the tape wavelength of the signal to be recorded has diminished to the same order of magnitude as the width of the recording zone. More precisely, the magnetic field will be completely or partly cancelled when the extension of the recording zone along the trailing edge of the recording gap becomes greater than or equal to half the wavelength of the signal to be recorded. At a tape speed of 7½ ips, the critical wavelength corresponds to fairly high frequencies which are of no significance for reproduction of speech and music. At 3¾ ips and 1⅞ ips, however, the cancelling effect occurs within the useful frequency range when conventional recording technique is used. It is therefore desirable to make the recording zone as narrow as possible (shorter extension along the tape) in order to be able to record at short wavelengths.

Several methods have been applied to accomplish this: (1) the design of the recording head has been improved, resulting in a narrowing of the recording zone. (2) It has been found that the frequency range can be extended by using tape with a thinner magnetic coating—the so called triple-play tape. This is, unfortunately, accompanied by a reduction of the maximum available playback signal amplitude and, consequently, a relative increase in tape noise. (3) a relatively new recording technique, the *Cross-field* technique, has been developed, giving a similar extension of the frequency range for normal coating thickness (long-play tape) without any increase of tape noise.

Reduction of Bias. For thick magnetic coating, the recording zone will contract if the bias amplitude is reduced. If the signal amplitude is maintained, this will cause distortion from the deepest parts of the coating, where the bias becomes insufficient. This effect is prevailing for the lower and medium frequencies, which are the more important ones. In order to avoid this distortion, the signal amplitude must be correspondingly decreased so as to have the same depth of penetration for the signal and bias fields. This will, in turn, reduce the available playback signal amplitude. It is thus obvious that recording at reduced bias utilizes the magnetic coating poorly. It leads to less available signal relative to the tape noise, and has the further drawback that inhomogeneities in the coating are



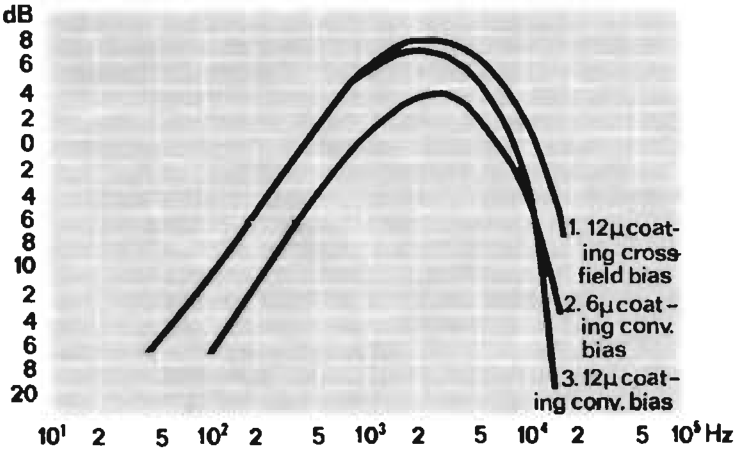


Fig. 1. Tape curves at 3¾ ips tape speed.

accentuated in the form of signal drop-outs.

Generally, it can be stated that the available recorded signal amplitude depends on the bulk of magnetic material being excited. In the lower- and middle-frequency ranges, the signal increases proportionally to the coating thickness for all tapes in current use. The noise level is mainly determined by the surface structure of the tape and is, therefore, practically constant when the thickness of the tape is varied.

A possible method for increasing the bulk of material being magnetized would be to increase the width of the track from quarter-track to half-track. This will increase the signal amplitude by 6 dB. The noise will increase only by 3 dB because of its random frequency and phase relationships. The net gain in signal-to-noise ratio is, therefore, 3 dB for a doubling of the track width.

Reduced Coating Thickness. In order to benefit from the possibilities associated with reduced bias and, thereby, narrower recording zone, thinner tapes with reduced coating thickness have been produced. These tapes will give a weaker signal in the lower and middle frequency range. In contrast, the higher frequencies will have larger amplitudes than those obtained with thick coating because of the narrowing of the recording zone. Consequently, the frequency range is extended upwards, accompanied by a general decrease in signal level. The wider frequency range is thus obtained at the sacrifice of S/N.

Figure 1 shows how the signal amplitude varies as a function of frequency for tapes with thick and thin coating. Curve 3 represents the thick tape with a coating of 12 microns (long-play tape), whereas curve 2 represents the thin triple-play tape with 6 microns coating thickness. Both tapes have been optimally biased; that is, maximum available signal amplitude occurs at 600 Hz for 3¾ ips. The curves have been plotted using conventional recording technique at a tape speed of 3¾ ips, and show the tape characteristic for constant signal recording current when played back through a flat-response amplifier. This enables the relative response for the two categories of tape to be read directly in decibels.

As shown, curves 2 and 3 intersect at 10,000 Hz. At frequencies below the crossover, the thinner coating gives a loss of 6 dB as compared to the thick coating. If the tape is run at half the speed, the crossover frequency is also reduced by the same amount to 5000 Hz. By switching to thin tape coating, the amplitudes of the higher frequencies are not reduced, and it will appear that a 6-dB gain at higher frequencies has been achieved as compared to the lower frequencies. This, however, is gained at the sacrifice of S/N in the most important frequency range where a corresponding attenuation occurs, an unfortunate occurrence because the S/N already represents a serious restriction on good sound reproduction.

The ideal solution would be to extend the frequency range at a given tape speed without deteriorating the S/N ratio. As explained in the following, this is possible by application of the cross-field-bias recording technique.

The Cross-field Technique

The cross-field technique is characterized by an extra head that introduces the bias field, close to the record head gap, in opposition to the signal field. The resulting field is more perpendicular to the tape surface, and the recording zone becomes narrower. The self-erasing effect has now moved towards considerably higher frequencies, and the recording field at middle and lower frequencies penetrates through the thicker magnetic coating. An improvement of frequency response has thus been obtained without any compromise in the signal-to-noise ratio.

In our tape recorder laboratory, extensive measurements have been made to determine the amount of signal gain obtained by using cross-field bias instead of conventional bias. The improvement can be found from Fig. 1, where curve 1 shows the resulting frequency response for cross-field bias. Comparing with curve 3 representing the same conditions for conventional bias, we find that the two techniques at a tape speed of 3¾ ips give equal amplitudes up to 1000 Hz, where the curves diverge and show a difference of 5 dB in favour of the cross-field technique at 10,000 Hz. It can thus be stated that the cross-field technique gives the same signal amplitude at lower and middle frequencies, as compared with conventional recording, and a significant signal

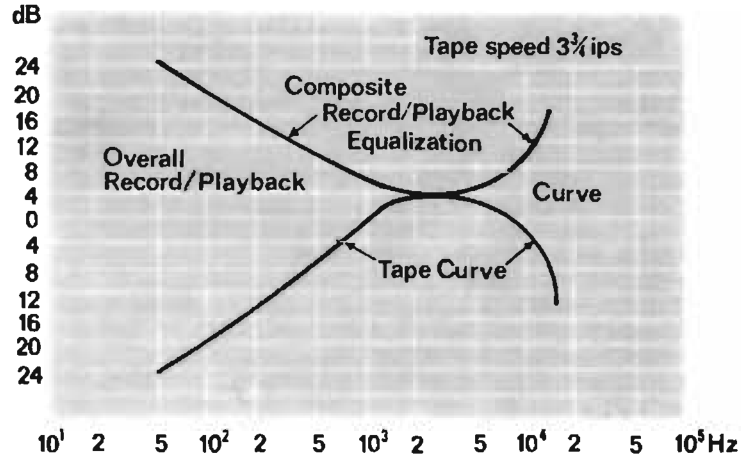


Fig. 2. Construction of Overall Record/Playback Curve.

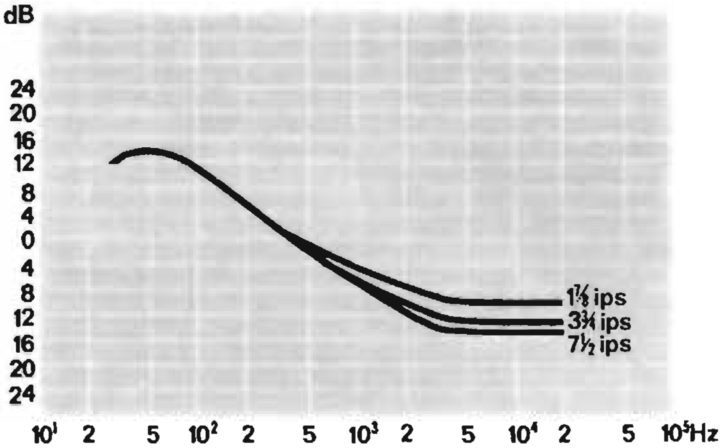


Fig. 3. Playback equalization, IEC–standard

improvement at higher frequencies.

If we compare the curve for cross-field recording on long-play tape (curve 1) with conventional recording on triple-play tape (curve 2), we find the curves to be virtually parallel to one another, with the latter 6 dB down. This means that with cross-field bias, the frequency range for long-play tape with 12-micron coating is the same as that obtained with triple-play tape (6-micron coating) using conventional technique. The gain in S/N, however, is directly expressed by the distance between the two curves; that is, 6 dB.

To recapitulate, there are presently two ways to extend frequency range at low tape speeds. One way is to make the recording zone narrower by reducing the bias. This implies the use of thinner tape, leading to a subsequent decrease of the signal level and a corresponding incease of the relative noise level. The other possibility is to contract the recording zone by means of cross-field bias, whereby the thick tape can be used and the low noise level maintained.

Practical Cross-field Technique

Frequency Corrections During Record and Playback. Before discussing design guidelines for cross-field biasing, it is necessary to review how frequency characteristics of a tape recorder arise. Figure 1 shows that the frequency response of the head and the tape alone is far from being flat. The amplitude drops off radically at the upper and lower extremes of the frequency range.

At lower frequencies, the amplitude rolls of at a slope of 6 dB/octave because recording has been done with a constant magnetic field. At the upper end, the signal drop is caused by the previously mentioned wavelength losses, together with head and tape losses. In order to compensate for this, the gains of record and playback amplifiers are increased at both ends of the frequency range. See Fig. 2.

In the lower frequency range, to the left of the tape curve peak, the playback amplifier gain is increased by 6 dB/octave down, compensating for the negative slope of the tape curve shown in Fig. 2. The location of the peak depends on the tape speed. Therefore, the break frequency for the playback amplifier is set individually for each tape speed, as shown in Fig. 3. The 3-dB points for the frequency curves at the different tape speeds are determined by time constants specified in the international IEC standard, as follows:

7½ ips— 70 μs
(corresponding to 3 dB at 2260 Hz)
3¾ ips— 90 μs
(corresponding to 3 dB at 1770 Hz)
1⅞ ips—120 μs
(corresponding to 3 dB at 1330 Hz)

The dropoff at the upper end of the tape curve is caused by recording losses which can hardly be compensated for during playback because it would result in a severe increase of tape noise. In this frequency range, therefore, the signal level is raised before recording, thereby increasing the distance between signal and noise. This is the so-called pre-emphasis, which is not restricted by international standards. It is up to the manufacturer to freely develop and improve the recording technique; he can choose recording process and pre-emphasis as desired. The only requirement is that the signal be reproduced correctly, using the standard playback curve.

Dynamic Range. If a reasonable quality is to be maintained, the tolerable amount of pre-emphasis at high frequencies with respect to the medium-frequency amplitudes is limited. The consequence of pre-emphasis is that the tape recorder will require a reduced signal level in the pre-emphasized range in order to avoid tape saturation. A less pronounced pre-emphasis gives a wider safety margin against overload phenomena. For this reason, the required pre-emphasis for obtaining the specified frequency characteristics of a tape recorder should always be stated. A flat frequency response up to 10 kHz, attained by 10-dB pre-emphasis, gives a far better dynamic range than if 20-dB pre-emphasis were used for the same achievement.

Generally, the frequency distribution of music and speech shows that the amplitudes diminish with increasing frequencies, and it is fair to presume signal levels 10 dB down at 10,000 Hz as compared to 1000 Hz. It will therefore be tolerable to increase the signal 10 dB at 10,000 Hz with little risk of tape saturation. This is supported by the fact that FM broadcast programs are submitted to such a correction before transmission. The purpose in this case is to raise the signal out of the background noise.

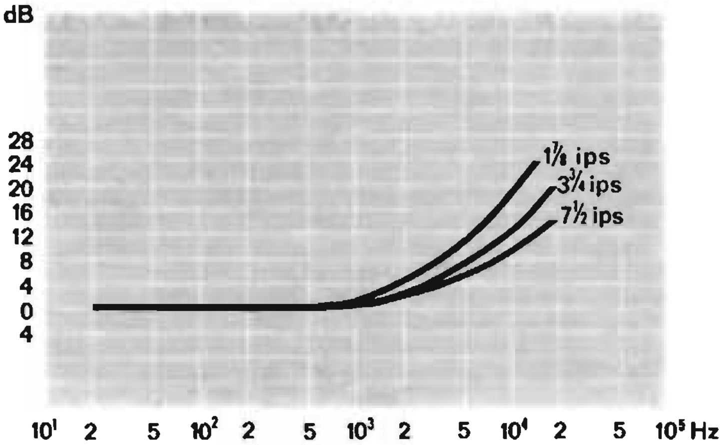


Fig. 4. Record pre-emphasis for cross-field bias.

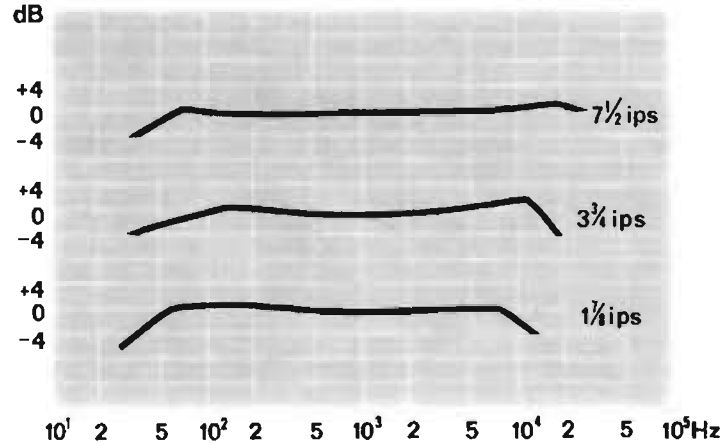


Fig. 5. Overall Record/Playback Curves for cross-field bias.

A limit of 10-dB pre-emphasis at 10,000 Hz is no longer an exaggerated quality requirement. The more modern types of music have, in fact, that much sound energy within the higher frequency range. Therefore, the pre-emphasis represents a risk of tape saturation unless the overall recording level is decreased, which will again lead to a less favorable S/N.

Improvements at Various Tape Speeds. If in spite of the above comments a maximum limit of 10-dB pre-emphasis is taken as a reference, it is interesting to compare what can be achieved at tape speeds of 7½, 3¾ and 1⅞ ips by application of cross-field biasing, with the results obtained by using thinner tape coating. We have already found that the tape curves for the two cases are virtually parallel to one another. Therefore, the pre-emhpasis curves will be equal. Thick magnetic coating gives the highest signal level and, consequently, the best signal/noise ratio.

As previously mentioned, recording losses associated with the width of the recording zone will not occur at 7½ ips. This applies for cross-field as well as for conventional techniques in the relevant frequency range. At this speed, however, other frequency-dependent losses necessitate a pre-emphasis of 8 dB. This means that the established 10-dB pre-emphasis limit is nearly reached. Because the self-erasing problem within the desired frequency range at 7½ ips does not arise, the cross-field will not alter the overall situation at this tape speed. The use of thinner tape will, however, give an increase of the tape noise by 6 dB without any advantages in return.

If the tape speed is reduced to 3¾ ips, the wavelength-depending losses begin to appear. Conventional technique and thick tape require a pre-emphasis of 20 dB at 10,000 Hz. With cross-field technique, these losses can be reduced to a magnitude where only 12 dB pre-emphasis at 10,000 Hz is required. It can thus be stated that owing to the new technique, one has succeeded in keeping the wavelength losses at a level low enough to obtain a frequency response at 3¾ ips that is approximately equal to the one at 7½ ips for conventional technique. This has been achieved without exaggerating pre-emphasis. The tape noise will, however, increase by 2 dB because of the higher playback amplification required for a given signal level in the range from 2000 Hz and upwards at 3¾ ips. See Fig. 3.

At 1⅞ ips a still greater profit is gained by the new technique. A pre-emphasis that compensates for recording losses up to 10,000 Hz will, with cross-field bias, have to be 18 dB, which is nearly the same as needed for 3¾ ips by conventional biasing. Again, 2 dB more noise will have to be accepted because of increased playback amplifier gain from 1300 Hz and upwards. In other words, this will give 4 dB more relative noise than is the case at 7½ ips.

Conclusion

The chart below is a summary of playback data for cross-field recording on thick tape and conventional recording on thin tape. Standard IEC playback curves are assumed. Cross-field recording on long-play tape at 7½ ips is taken as the reference for tape noise.

Tape speed ips	Pre-emphasis at 10 kHz	Tape noise level long-play tape cross-field bias	Tape noise level triple-play tape conventional bias
7½	8 dB	0 dB	+ 6 dB
3¾	12 dB	+ 2 dB	+ 8 dB
1⅞	18 dB	+ 4 dB	+ 10 dB

Figure 5 shows the resulting frequency curves obtained with the pre-emphasis and tape noise given in the accompanying table of playback data when the cross-field technique is used for thick tapes. The greatest advantage obtained by using cross-field recording on thick tape instead of conventional recording on thin tape is a reduction of tape noise. From the table it can be found that cross-field is 6 dB better in this respect at all three tape speeds.

It may be of some interest to know the relative increase of the tape noise when the tape speed is reduced from 7½ ips to 1⅞ ips. Taking the tabulated data for cross-field as a reference, we find that the tape noise will increase by 4 dB for cross-field, as well as for conventional technique, due to the higher playback gain required at low tape speed. Furthermore the tape speed reduction requires a 10 dB higher pre-emphasis in order to maintain the frequency response up to 10,000 Hz. Hence the overload safety margin in the upper frequency range is correspondingly reduced. The 8-dB pre-emphasis at

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THE CROSS-FIELD TECHNIQUE

(Continued from page 24)

7½ ips will allow a program of the previously mentioned standard spectral-sound-energy distribution to be recorded at maximum level in the lower and middle frequency range, with no risk of saturation at high frequencies.

A pre-emphasis of 18 dB is required at the reduced speed and, under the same conditions as above, the recording level at lower and middle frequencies must be reduced by 10 dB in order to avoid saturation at high frequencies. The tape noise will then increase by a corresponding amount.

The increase of tape noise at low tape speed consists of one fixed amount caused by the augmented playback amplification and another amount that varies from 0 to 10 dB, depending on the energy distribution of the program. The latter noise contribution is the same for cross-field recording on thick tape as for conventional recording on thin tape because the pre-emphasis is the same in the two cases. If the recorder has an instrument that indicates the maximum tolerable signal amplitude at any frequency, one will automatically set the record level according to the loudest tones. If the sound energy is concentrated at high frequencies, one will reduce the record level, and the relative tape noise will increase.

It can be stated that, for cross-field, the tape speed reduction from 7½ ips to 1⅞ ips is accompanied by a possible tape-noise increase from a minimum of +4 dB up to 4 + 10 = 14 dB, depending on sound energy distribution. The corresponding figures when one switches from cross-field recording at 7½ ips to conventional recording on thin tape at 1⅞ ips are: +10 dB up to 10 + 10 = 20 dB.

These viewpoints clearly show how important it is to consider frequency range, pre-emphasis, and tape noise as a whole when judging the quality of a tape recorder. Furthermore, these quality requirements must be considered in relation to the particular program to be reproduced. Æ

Tape Transport Maintenance

Part I: Tape Head Cleaning and Adjustment

H. W. HELLER

IN A RECENT service-shop review of repairs, we found that thirty per cent of the tape recorders brought in needed little more than routine cleaning and adjustment. A few notes from the workshop bench on general maintenance could assist one to avoid such "repairs."

Head cleaning is the sort of standard operation that is often taken for granted. However, perfunctory head cleaning can be worse than no effort at all. Also, attempting to scour and polish tape recorder heads without the right cleaning fluids may be an invitation to premature wear. Tape oxide is abrasive, especially when mixed with dust in a cement whose binding agent has been the very solvent employed to clean the heads.

First action is to clean any hard scale away, using a pointed wood scraper. There are special softwood tools for the job, but cocktail sticks, manicure picks and other wooden implements can be brought into use. Oxide can build up into thick deposits in the wedge angles of some heads, where tape guide plates are fixed to the head block. Clear such deposits first and blow away the scrapings, then tackle the head facing with a swabstick soaked in surgical spirit, cotton wool or linen tapes and pads moistened with methylated spirit, or one of the several brand-name preparations.

The important point to remember when spirit is used: clean away the residue. And never thread up the tape again until the cleaned surface is dry. It takes only a minute or two.

The exception here is the tape-head cleaning preparation that contains silicone. This is intended to clean and lubricate. The cleaning action is completed with a wipe over and, as the carrier fluid dries away, a fine layer of lubricant is left to coat the heads, guides and running surfaces. Watch that point, "running surfaces." There are some solvents intended for head cleaning *only*, which must never be allowed to attack the rubber of a pressure roller. Read the instructions on the cleaning-fluid package if there is any doubt. And if the package has no instructions, don't buy it!

Tape guides need their share of treatment, too, especially in the hard-to-see angles between the flanges and barrel. Constant tape friction can wear flats on the barrel face, and the problem of tape wear increases, as does the friction caused by a greater area of rubbing surface. If the wear has not been constant, the tape will pull toward the point of greatest pressure, the thin end of the wedge, and the double faults of mistracking and uneven tape contact will aggravate matters. If it is possible to turn the guide to present an unworn surface to the tape, so much the better, but not if doing so is going to alter the height of the guide in relation to the run of the tape through the head channel.

A few manufacturers let the guide height be the datum against which the head adjustments are made. Never alter these guides — unless you positively enjoy the tedious business of resetting the heads.

Fig. 1—When a tape deck gets into this condition it is a wonder that any high frequencies at all get through to the tape, or from the tape back to the amplifier.

Test Tape

Without a correctly recorded test tape, head setting can be a long process of "cut-and-try." The easiest kind of test tape to use is the full-track, white-noise type with Track 3 erased. Using this on either two- or four-track machines, the first adjustment is for Play head alignment, setting the head so that maximum output is obtained on the top track. Two reasons here: with this tape, the top track is fully recorded; and second, the angle of displacement may be the same as on lower tracks, but the physical movement needed to bring the head into line is greater, so a more delicate setting can be made.

A little thought about what is happening will help us understand why we make these adjustments. The frequency response of a tape recorder depends on a number of factors, important among them the width of the gap in the playback head. This must be as narrow as it can be engineered. It should be narrower than the wavelength of the highest frequency to be reproduced, which is why we get better top response when we re-play at a faster speed; the wavelength of any given note is then longer. Double the speed and twice as much tape passes the head in the same period of time.

Tilting the head so that the gap is out of true is effectively the same as widening the gap. So we find that azimuth adjustment has the most effect at the high end of the frequency spectrum. Wide-range white noise contains all the audio frequencies in equal proportion, so it comes out as a hiss with an underlying roar. Thus its use for head alignment, when a change in the hiss output is easily heard, and as easily measured, while the low-note roar is constant for reference.

The use of a test tape with track 3 erased with equipment that gives a perfectly "clean" lane up to the edge of the adjacent recorded track spacing, enables us to check four-track machines very simply. With the machine switched to play Track 3, and the head height altered up or down, there is a definite increase in white noise output once the head moves from the correct setting. As a bonus, by inverting the tape, now presenting the two quarter tracks with a similar white-noise signal, one can judge gain of the two channels of a stereo tape recorder or efficiency of the head of a mono machine.

All very well, I hear you say: I am not going to invest in a white-noise test tape that I may use once in a while, and then briefly. So the answer for some might be to make some form of test tape ourselves. First we need to ensure that the machine is recording properly, with a modulation indication

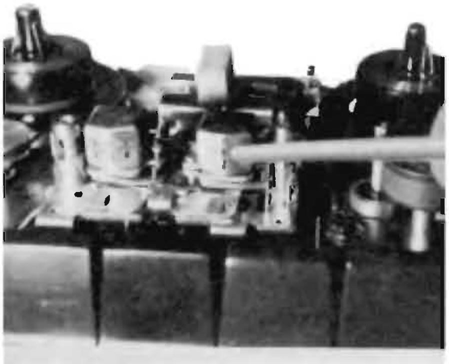

Fig. 2—Open-plan assembly such as this makes servicing easy. Pressure pads are mounted on a flap which comes up to meet the rear of the tape; falls horizontal to allow easy access when the mechanism is neutralized.

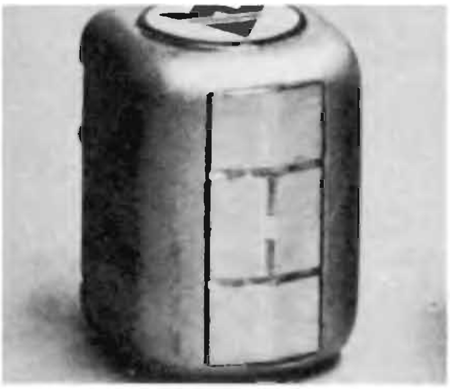

Fig. 3—A stereo head in close-up shows the gap formation. Note the head contouring and the grooves near tape edge location of the stereo head. These grooves act as a suction device when tape is moving past, but can trap oxide if not kept scrupulously clean.

up to 0 dB with an input equal to the specified level. Experience will tell us what to expect. Tape recorders differ widely; no exact rules can be laid down.

If we are sure the machine is punching modulation onto the tape, we can take it further. There is no need at this stage to investigate distortion. We simply record a signal with as much high-frequency content as possible, and, if we can, with some sustained passages of constant level. Radio stations are helpful to us sometimes, and television test signals can be poached with good effect. Do not be afraid to overload; the more signal you can get on the tape for this test the better.

Having modulated, replay this rudimentary test tape, and then rock the replay head for maximum top note output. The correct azimuth setting should be obvious. If it is not, look for head wear (a flattened portion of the facing, usually easy to see when a bright light is directed sideways at the head). Make sure that the pressure pads, if used, sit cleanly against the polished back of the tape, and that they are soft. Where pressure pads are not used, pins will often be employed to guide the tape past a contoured head. Make sure these have not been worn into flutter-producing grooves.

Head height is a lot easier to judge than might be thought. Visual inspection of record and replay heads (or combination heads) with the tape stretched across them in the playing position will show the upper edge of the top gap which should be brought to the tape edge for either two- or four-track operation. A two-track head is often set to overscan the tape slightly. The track "height" takes up 2.5 mm and there should be a safety lane of 1.8 mm between the lower edge of the two-track recorded signal and the inner edge of the lower track (that is, the track obtained when the tape is inverted). Best rough test is to modulate heavily on a new tape, then invert the tape and replay. Listen for breakthrough, which will indicate that the head setting is too low. Take care with this test, as some makers use a 3-mm track and a safety lane of 10 thou' or less.

Patience is the keyword. This is even more necessary for setting up four-track heads. Safety lanes are as small as 0.75 mm and only a minute amount of misalignment is needed to produce cross-tracking. The easiest test is to record Track 3 on a clean tape, again heavily modulating, then invert the tape and listen for cross-tracking on both Track 1 and Track 3. Breakthrough on the upper track indicates the head may be low; breakthrough on Track 3, the head was high when the first recording was made.

Emphasis on a clean tape brings us to the problem of the erase head. Here, the tolerances are a lot less exacting. The gap may overlap a millimeter or so. The gap itself is longer than that of the recording head, as well as being wider, so that the complete track can be erased, allowing for a little tape wander. For these reasons, the azimuth alignment is not nearly so critical, and visual alignment is generally sufficient. But exact height is important in four-track operation, and the method is to record a constant signal on Tracks 1 and 3, again using clean tape Then invert the tape and erase Track 3 for a spell, re-invert the tape and replay the previously recorded tracks. Misalignment of the erase head

Fig. 4—Do not overlook the upper bearing of the flywheel capstan when cleaning around the deck. Accumulated oxide at this point can quickly lead to head wear and erratic running.

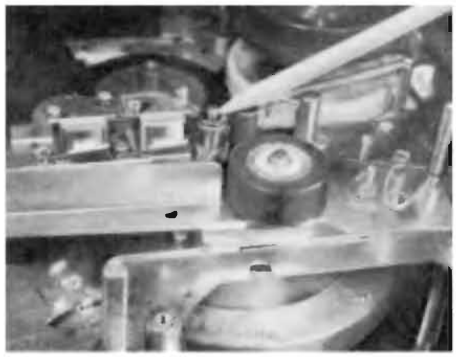

Fig. 5—Tape guide flange height can be reset by a threaded nut, but spacing is determined by the guide barrel proportions.

will cause a loss on one or the other track. A weakened replay on the upper track indicates the erase head was low, and vice versa.

Remember that a tape recorded half-track will have to be bulk erased or erased on a half-track tape recorder before it can be used quarter-track. This is because the quarter-track erase head will only cover a little more than the newly recorded track, and some of the previous half-track signal will occupy the safety lanes and break through on four-track replay.

Head and guide cleaning, setting and alignment are of primary importance. No use making a perfect job of repairing and adjusting the mechanism, or getting the highest of hi-fi from the amplifiers unless the heads are in line and doing their jobs properly. Having done this, we can check the drive system (next month), and ensure that the tape is running true. Æ

Tape Transport Maintenance

Part Two—Motor Drive and Speed Selection

H. W. HELLYER

THE HEART OF THE TAPE RECORDER drive system is the motor. Unless this part of the machine runs true, maintenance work done on the rest of the mechanism is wasted. Luckily for us, the motor is the part that needs the least attention.

Several sorts of motors are used in tape recorders. Battery models, and those designed to run on a.c. or battery power have a d.c. motor (direct current) with a rotating commutator and carbon brushes for contacts. The obvious points of wear are the brush ends, where they are continually scraped by the segments of the commutator. An occasional cleaning operation here, plus a touch of grease at the bearings will prolong life. Cleaning with a soft cloth and a drop of alcohol is usually sufficient. It is a mistake to scrape or file a commutator, however encrusted with carbon it may be. The action depends on sequential switching of a relatively heavy current, and any surface discrepancy causes arcing. The fault is cumulative and soon results in a grooved commutator plus eccentric brushes. On some of the heavier-duty motors it is possible to remove the commutator—or complete armature—and skim it down to a new surface with the aid of a lathe, but small motors seldom respond happily to this treatment. Luckily, again, small motors are cheap; it is often less expensive to replace them than to battle on with ineffective repairs.

Speed of rotation of the d.c. motor depends on the supply voltage, which gives us a means of regulation. On some of the newer machines, not necessarily designed for battery operation, this feature is used to advantage. Tandberg and Sony, for example, often employ servo-controlled d.c. motors, making quite ambitious machines independent of line-supply voltage or frequency. But servo control is a separate subject for which we shall have to find space at a later date.

Simpler methods of regulation are purely mechanical. Governors, not dissimilar in principle from the old-fashioned spinning balls of the steam engine, may be found on some of the small motors of battery portables. These work by centrifugal force. As the motor exceeds a limiting speed of rotation, contacts are opened or closed, according to design, and current is reduced by insertion of a series resistor, a diode, or a transistor regulator in the motor supply. The control falls out again as the motor speed comes down to the required figure.

This method alone would be useless, the motor continually varying between the regulator limits. For long-term control, the regulator sets the basic speed and for short-term stability we have to rely on the rest of the design, the damping effect of idlers or belts and the inertia of the flywheel.

When setting regulators or governors of the centrifugal type, always do so with the full load applied, and check overall speed with full and empty spools of tape at each side to ensure that back tensions are not affecting the torque.

Best check of d.c. motor regularity is measurement of the motor current. First the supply voltage should be checked, then minimum load current and finally the maximum current, usually in the rewind mode. This test can often give a clue to mechanical faults, where the unloaded motor draws normal current but the measurement shoots up when one or another drive function is applied.

Most of our maintenance work will be concerned with a.c. motors. The commutator type is ruled out for tape recorders (though not completely unknown). One reason is the cheaper design, with the induction motor, which is subject to some control of speed by mains frequency. Several different types are used.

Induction motors have an air gap between the moving rotor and the fixed stator. When an alternating current is applied to the stator winding, an electro-magnetic flux is set up in the rotor windings by induction. But it is the interaction of the fields of the two windings that starts the motor turning, and some method has to be found to achieve this interaction—apart from giving the motor a physical shove!

The movement of the fields to achieve starting and to keep the motor going is done by "shading" the poles of the windings, or by using an external power-factor capacitor, which provides the initial phase shift. There are other big differences between the types of motors employing these techniques.

The simple induction motor with shaded poles, easily recognized by the protruding copper shading rings in the lamination structure, will always have some "slip." The magnetic flux is changing at mains frequency (60 Hz) and the rotor is constantly trying to catch up, never quite succeeding. The overall speed is thus tied to the regular mains frequency and is independent of voltage (within wide limits), but the design is vulnerable to load changes. More load causes more slip, more magnetic flux lines are cut, a greater current flows and the torque increases. This makes such motors very useful for reeling, where the rising speed-torque characteristic is an advantage, but reduces their efficiency as capstan motors, especially in single-motor designs where the torque varies with the proportions of tape spooled.

Outside the stability limits, a reduction of supply voltage lowers the full-load torque—the full rotational force that is available at full-load speed. So we can control speed roughly by varying the supply voltage, and this is done in some three-motor machines using cheap capstan motors by inserting a heavy-duty resistor.

These resistors, switched out or changed in total value for different functions, are a vulnerable trouble-point. Look for strained wires at the terminations, for heat-spots and cracked ceramics.

The principal advantage of the shaded-pole induction motor for capstan drive is that its wow-and-flutter figure can be kept low, even though this may be outweighed by other factors of mechanical design. One disadvantage is a tendency to vibration. The motor mountings should be checked for resilience, and cable connections must always be looped to avoid strain. Design of spindle ends and bearings has to be good to keep this vibration effect down, which leads us to the obvious maintenance job of checking spindle truth and cleaning bearings.

Motor bearings, of these and the next type to be discussed, will generally be of the cup or flat-sphere variety. The alignment of the spindle in the lower bearing can be adjusted by gentle tapping (using a wooden mallet, or even a screwdriver handle, not a hammer)

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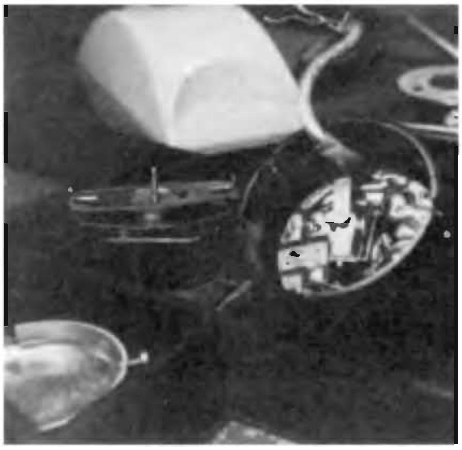

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Fig. 1—Typical synchronous motor, viewed from the lower bearing. Start capacitor is adjacent.

Fig. 2—Typical external-rotor type of capstan motor.

Fig. 3—View from below of an outer-rotor capstan motor. The fan-shaped base of the rotor can be seen.

Fig. 4—Speed-change arrangements may include change of start capacitor and some complicated switching, as shown.

Fig. 5—Battery portable motors may require the touch of a watchmaker, including the use of a jeweler's loupe and tweezers.

Fig. 6—A typical d.c. motor for small tape recorders, with suppressor circuit mounted internally on a small printed board and the regulator adjustment accessible through a hole in the shielding.

Fig. 7—Idler engagement for speed change depends on bracket adjustment and springs, as indicated here.

Fig. 8—A touch of oil at the spindle of the speed-change bracket and a clean ramp for easy sliding can make all the difference to smooth running.

Fig. 9—Motor pulley with idlers and left spool carrier shown. Note noise-reducing sponge ring inside hollow "cup" pulley.

while the motor is running. But often, the alignment depends on the upper bearing, which may be allowed some float. Nylon thrust bearings or phosphor bronze spring clamps are widely employed. The trick is to dismantle, disturbing the spindle as little as possible, clean the bearings thoroughly, lubricate lightly, then reassemble with spacers between armature and stator. These need not be exact, so long as they are nearly tight when the motor is clamped. At least three should be used spaced equidistantly, and their thickness should be equal. Then the motor is carefully clamped up, spun by hand as the spacers are withdrawn, tightened to finger-tight degree, then powered. After a few final taps and radial tightening (the way you tighten automobile wheel nuts), the motor should run true.

Mention was made a few paragraphs ago of the "poles" of a motor. These are magnetic strong-points, by the configuration of the windings. Speed of the motor depends on the frequency and the number of poles. A two-pole motor rotates at 60 x 60 revolutions per minute; that is 3600. When alternating voltage is applied the pole faces change polarity with a frequency equal to the supply.

A four-pole motor does not run twice as fast. On the contrary, the field only rotates half a revolution for each mains cycle, so speed is 1800 rpm. But the action is less jerky, and the motor is more powerful and smoother. There is a lower external magnetic field, the slower speed makes capstan drive easier to design (less speed), reduction and a larger rotor can be used, which makes for quieter and smoother running, with less vibration.

The other type of induction motor we have been hinting about is the synchronous motor, whose rotor, instead of slipping, travels at the same rate as the magnetic field.

Because the synchronous motor cannot slip, it tends to "hunt" (vary about

(Continued on page 107)

the speed of the rotating magnetic field). Instantaneous speed stability is not especially good, and so there is a chance of flutter. But this can be accommodated in the design. Drives, tensions, belts, idlers, flywheels, and so on, can be balanced to reduce the effect, and take advantage of the ability of the synchronous motor to maintain long-term speed regularity.

One type of synchronous motor is known as the hysteresis motor. The outer covering of the rotor is especially hard, magnetically—it has high remanence. (Soft material loses its magnetism easily.) This enables the rotor, as it approaches the pole to "search out" the strong field of flux and lock more exactly. Speed changing is made more accurate—even though electrical switching of poles (doubling poles to halve speed) is possible with ordinary induction motors. Exact halving or doubling is possible with a synchronous motor. This is quite apart from the consideration that an asynchronous motor with its starting capacitor would need additional capacitance for speed changing.

Capacitor start is also used with salient pole motors, which are synchronous. With these, flat spots are milled in the rotor, and laminations are of soft iron, conducting magnetic lines of force easily. This simple device gives the rotating field an easier magnetic path on one side of the rotor than on the other, and the motor tries to keep in step.

All this is academic. We cannot do much about the design, but it is as well to know what the parts of our tape recorder are doing. Start capacitors can give trouble if they are leaky, will prevent starting if open circuit, or if their value has changed drastically—a fault that may lead one fruitlessly testing up a blind alley.

Speed changing, as mentioned before, can be effected by pole switching. But this needs banks of good-quality contacts, and is very often difficult to arrange without the use of relays. Easiest method of speed changing is the provision of a stepped pulley on the motor spindle with which the intermediate drive can engage, to get the required change or changes in speed ratio. For this reason, we find pole switching confined nowadays to three-motor designs with direct drive.

These direct drive motors combine the simplicity which is their virtue with a flywheel action needed to iron out the last trace of flutter. This is done by turning the motor inside out; windings are arranged around a central axial stator and the rotor is the outside shell. Fan action is obtained by slotting the lower periphery of the motor.

Fans of the "normal" variety are prolific troublemakers. Blades work loose at the root, or are displaced and set up irregular action. The method of mounting fan-boss bushes with pressed folds of aluminum causes premature loosening. Small ticks, rattles, or at the worst, fouling that stops the motor, can often be traced to the fan. View across the sweep of the blades as they run, using a bright light, to see whether blades are eccentric.

Let us now consider methods of mechanical speed changing.

Many idler systems use ramps, with the idler pulley bracket riding up as the ramp slides, pressure being maintained by some form of spring. Two weak points are the ramp spring and the return spring that helps the idler into engagement with motor pulley and flywheel. A fault in the former will prevent true engagement on the pulley step, and weakness of the latter allows the idler to slip. In the worst case, wow will ensue. Watch for scuffed pulleys, or indented idlers that have been left in engagement with the mechanism at standstill. With three-step pulleys, a common fault is fouling of the idler against the next largest step. Distorted spindles or brackets, worn thrust washers, grit, and old grease can cause incorrect speed-idler action. Careful visual inspection often reveals such faults.

Belt-throw systems for changing speed may consist of a fork and a tongue on the motor pulley. Again, different diameters are used, but this time the steps are grooved. Look for deposited rubber in the groove, causing a "knock." Look also for eccentric pulleys that cause a belt to "wander" as it runs. Belt "tick" or "slap" can be due to either a slack belt or an eccentric pulley, or a combination of the two.

Danger with the pulley-and-fork system is a nicked belt when the pulley is displaced. A simple fault like a slackened set screw can lead to snapped belts and fouled-up decks. Look out for the tell-tale specks of rubber with belt and idler drive systems; they can be the first signs of wear. The fork that feeds the belt to its correct pulley step can get bent too easily, especially as decks are moved about, and belt spill results. Final check of position must be done with the belt running. Generally, belts are moved as the machine runs: idlers usually slide to position with the mechanism at standstill, and the interlock with the START control prevents changing on the move—which can be a drawback for some creative taping. Æ

Tape Transport Maintenance

Part III: Drive Systems

H. W. HELLYER

DESIGN PHILOSOPHY dictates which sort of tape drive shall be used—direct motor, belt, or idler. Each has certain advantages and some drawbacks. For speed exactitude, and precise control, directly driven capstan and reel carriers will be chosen; the obvious drawback is its high cost.

The choice between belts and idlers is not so simple. Quite often we find the maker has compromised, using both types of drive in the same machine, performing different functions.

Belts, or bands, may be rubber, composition or fabricised rubber, and can be round, triangular, or rectangular in cross-section. The most important criterion is evenness of cross-section, with elasticity following a close second.

Round-section belts will more often be used with a large-diameter motor pulley. It is more difficult to maintain constant diameter of a round belt, and under tension the belt 'flattens' at contact points, so angle-section belts are used with smaller-diameter motor pulleys having shaped flanges in which the belt rides snugly. The other alternative is a flat belt on a wide running surface, and fabricised flat belts which accept variations of tension for function change (play/fast wind/reverse) are favored by some designers. Where flat belts are used, the general tendency is to employ a slipping-belt drive for take-up rather than a slipping-clutch or gravity take-up, and vertically mounted machines may therefore have tensioned flat belts.

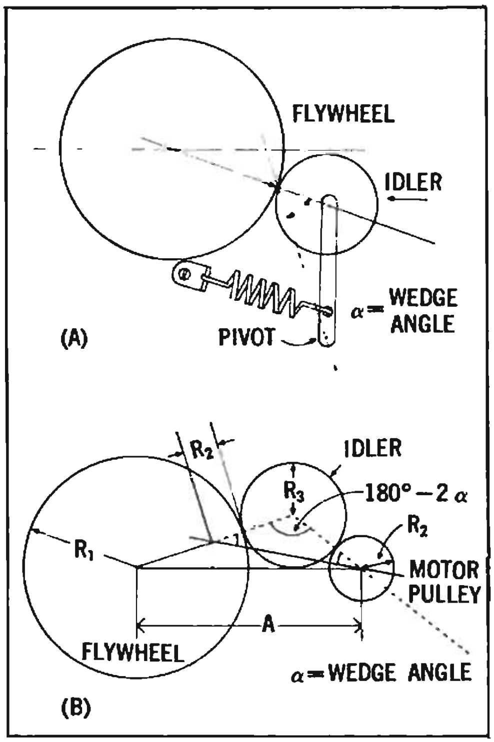


Fig. 1—(A) Wedge angle is important for correct transmission of torque. (B) Distances and wedge angles can be calculated when diameters of motor pulley and flywheel are known.

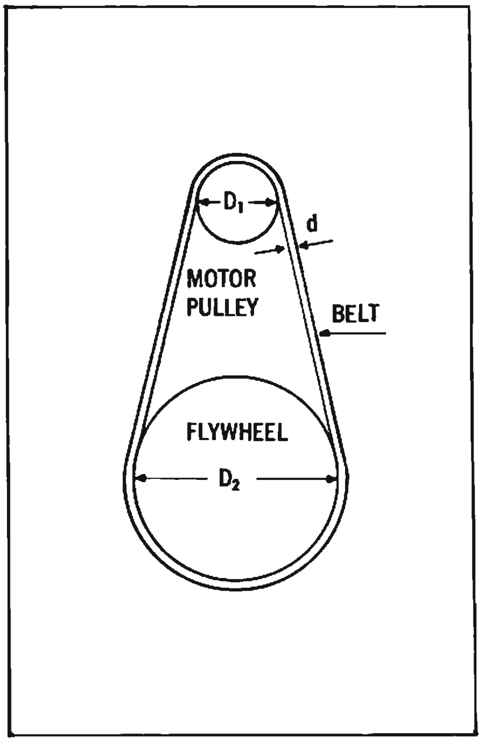


Fig. 2—Speed of a flywheel is not in inverse ratio to the diameters of motor pulley and flywheel, but has to take into account the thickness of the belt.

Idlers, jockey pulleys, puck wheels, or intermediate wheels, called variously (though not strictly correctly) by all these names, are used to couple the drive from the motor to the flywheel and to the reel carriers. Many different systems are employed, each with its own peculiarity.

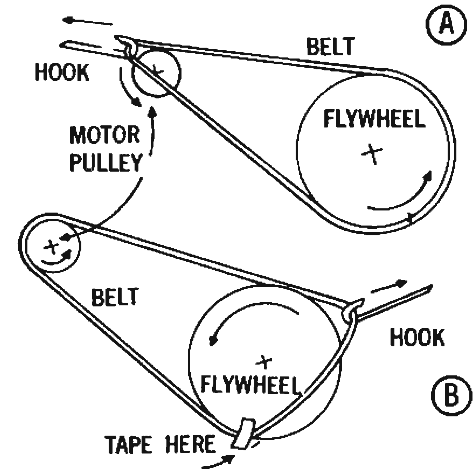
Most idlers are rubber-tired, and the type of rubber used has to be chosen carefully. As well as having good elastic properties, it must have a good coefficient of friction, the minimum of self-heating, and high abrasion resistance—properties which do not always go hand-in-hand. Because maximum tension must be achieved to reduce slip, pressure on the idler will be high. This tends to stress bearings, and makes the initial contact of idler and drive (or driven) surface vulnerable to irregular forces. For optimum transmission of drive power, systems are designed to be self-adjusting, the idler being held in contact with the driving surface by spring pressure.

Idler wheels lend themselves to designs which need reversal of driving direction, such as for the rewind operation, and to speed changes which can be effected by a stepped driving pulley. They are useful when the spacing between drive wheel and driven surface has to be varied, or is large.

Because the friction wheel is spring-loaded to make contact, some retardation is inevitable, and this has to be allowed for in design. The wedge angle for optimum conditions has been determined empirically to lie between 35 and 40 deg. for optimum drive with least slip. Figure 1 (A) shows a simple spring-loaded drive wheel, and in Figure 1 (B) we can see the design principles for an intermediate wheel. The wedge angle of 35 deg. is determined by experience, and wheel diameters should have specific relationships.

The diameter of an idler wheel should, in theory, make no difference to the speed of the driven surface. This is dependent on the ratio of the motor to flywheel diameters, in the case of direct drive. Interposing an idler reverses the direction of rotation, but should have no effect on the speed. In practice, small changes in

Fig. 3—(A) A simpler method of loading a new belt where surfaces are accessible. The hook is used to maintain tension. (B) Where other parts of the mechanism overlay the pulleys, a better method is to put the belt over the smaller diameter, secure at the rotational entry point, and use the hook to feed the belt over the larger diameter while turning gently.



diameter upset the tensioning, the wedge angle, and the calculated amount of slippage and can cause a slowing down or irregularity.

Belt Faults

Many of the factors governing regular running with idler-wheel drive also apply to belt-drive systems. Small variations in belt thickness can have a large effect on tape speed. The thickness of flat belts is more easily controlled than the diameter of round ones. Importance of exact belt diameter or thickness is illustrated with reference to Fig. 2.

If the motor pulley were geared to the driven pulley, the speed reduction at the capstan would be in inverse proportion to the diameters, i.e. $n_1/n_2 = D_2\,D_1$. Taking a practical example, a belt-driven flywheel 95.5 mm in diameter is required to rotate at 300 r.p.m. The available motor rotates at 2900 r.p.m. If a thickness of belt is 1 mm this must be added to each diameter in the above equation, giving

$$\frac{n_1}{n_2} = \frac{D_2+d}{D_1+d}$$

thus

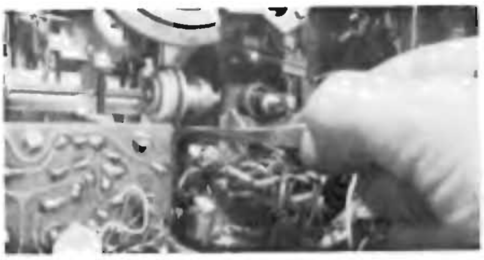
$$D_1 = \frac{n_2(D_2+d)}{n_1} = \frac{300\times 96.5}{2900} - 1 = 9\text{mm.}$$

So the effective drive at the motor has a diameter of 10mm.

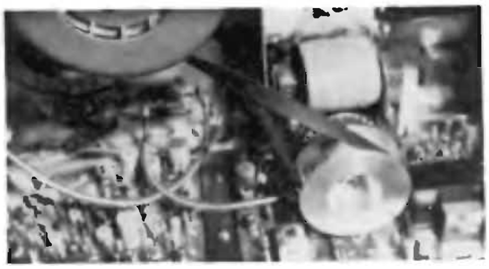
Suppose there is a variation of 10 per cent in belt thickness over its length. The resulting speed variation will be 0.1 × 100/10 or 1 per cent. The running frequency of the belt is only a few cycles and speed variations of such a low frequency can hardly be equalized by the centrifugal mass, so wow will be heard from pitch variations.

A

B



C



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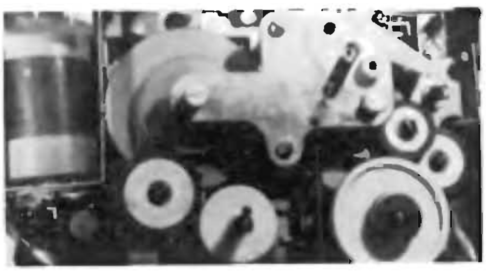
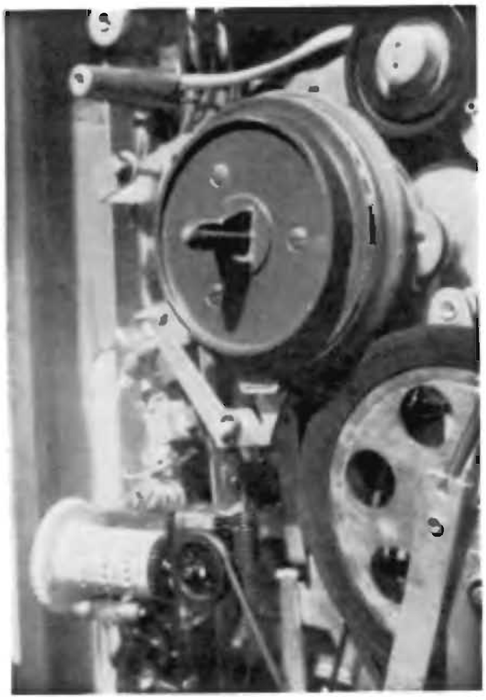
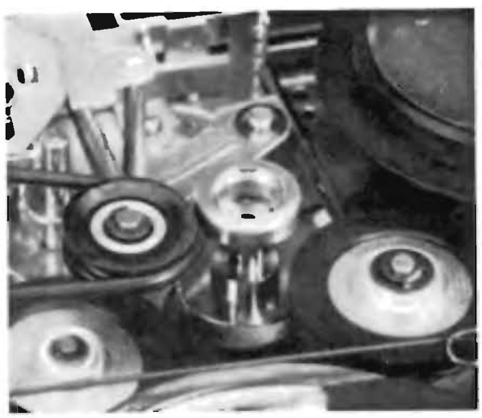


E

F



G



(A) Combination belt-and-pulley drive is used on many machines, as in this example—Sony TC-250.

(B) Small drive belts such as this one in the Uher 4000-L can be handled with a pair of tweezers, but extreme care is needed, and sharp edges of tools should be protected with adhesive tape.

(C) Flat wide belts may suffer from impregnation; small particles of metal from pulleys become embedded, imparting a surface polish. This example is a Sony CV-2000 video tape recorder.

(D) This assembly from a West German pocket portable uses a number of plastic pulleys with thin rubber tires. Check for binding bearings and out-of-line spindles.

(E) Japanese Standard portable has edge-drive from motor to flywheel and rim-drive from flywheel to take-up.

(F) Examples of the wide, thin take-up idler on a loose bracket which is liable to misalignment, and the smaller soft-tired auxiliary wheel at right. Philips 3548.

(G) Ramped idler for speed change.

If a flat belt is used, closer limits can be gained, and by hardening, then grinding the flat surfaces, deviations of less than 0.5 per cent can be achieved. From the foregoing it also appears that the larger the diameter of the motor pulley the less the effect of these deviations will be, but this has practical limits. This would require a slow-running motor, or a centrifugal mass (flywheel) of large diameter, or the addition of an intermediate drive wheel. We are back where we began.

Speed changing with belt drive can present special problems. The usual method is to feed the belt from one diameter to another on a common motor pulley. A fork drive may be used, or a tongue on the pulley.

One advantage of belt drive on simpler machines is the ability to filter out minor variations inherent in the system. Motor hum vibrations at 60 Hz can be reduced considerably by using a belt of sufficient elasticity to form an absorption circuit. This is very useful when a motor with an outer rotating cage (small diameter pulley) is employed in conjuction with a flywheel. The small diameter is dictated by the high rotational speed. Flywheel rotation should be as high as is practicable, and the short-term speed variations can be formidable unless belt elasticity is carefully chosen to filter them out. It follows that any hardening of the belt or loss of elasticity due to temperature and humidity changes will also affect drive regularity.

Neglect is the prime culprit—as has already been emphasized in this series of articles. Belts that are left in one position tend to harden and form to a shape, with flattened portions at the contact sections. The cure is a regular use of the machine, but if the condition has already set in, some remedy may be possible by heating and running the belt.

Best method of rejuvenating a formed rubber belt is to let it stand for a while in hot water, dry it, powder with French chalk or talcum, clean off surplus talcum to prevent it mixing with lubricants, and "run the belt in" for a reasonable period.

Mention of lubricants brings us to the prime enemy of belts—and indeed of any rubber or composition drive surfaces, including idler-wheel tires. This is oil or grease. Over-enthusiastic lubrication has caused more erratic tape drive than any other single fault. Oil tends to spin off rotating bearings in a fine film; grease can have a capillary action when heated, creeping up to idler surfaces. Always oil sparingly, and wipe any surplus from spindles before fitting belts or idler wheels. Diluted alcohol is the best cleaning agent, but even this has to be used with care on some composition materials, which can soften by its solvent action.

Refitting drive belts can face us with some problems. One or two tricks of the trade can make the job easier. The first thing to remember is sequence.

Unless you know the machine so well that dismantling deck parts becomes habitual, keep a note of the order in which parts are removed. Lay the parts in a logical sequence, ready for re-assembly.

Make drawings where necessary, and always include some sort of datum: 'deck front', 'head-plate', 'motor mounting', and so on. If it helps, number the operations on your guide drawing. For some of us, one map is worth a wealth of words.

When fitting a belt over two drive surfaces, with the belt under tension, some difficulty may be encountered in getting the belt to sit snugly on the larger diameter surface. Best method of attacking this problem is to loop the belt over the small surface, hold in place by stretching slightly, then feed it around the larger diameter slowly, turning in the direction it wants to go.

It is wise, when rotating flywheels or motors, to move these in their 'natural' direction of rotation.

Where a narrow flange is used, and the new belt is reluctant to stay in place, even with the tension and rotation method of fitting, one or two pieces of adhesive tape pressed lightly over the belt at the entry point will give a starter. After this, the wheel can be rotated and an outward tension maintained as it is fed into place. One useful aid for this job is that family heirloom, the button-hook.

If you can't find a button-hook, which is hardly likely today, fashion a stiff piece of wire into a nearly-closed loop and hook it over the belt. Use this, as in Fig. 3, to hold the belt away from the entry point of the flywheel, just enough to maintain tension while the wheel is rotated.

Although the use of pliers on rubber or composition belts is to be deprecated, it is occasionally necessary to employ a pair of tweezers to lift an otherwise inaccessible belt. Use the type with a flat blade and trust the clamping action to do the work. If this is not sufficient, you'll simply have to resort to the hook. But always avoid anything sharp. Protect the blade tips with a collar of rubber sleeving, or even a layer of cellophane tape.

After using any sort of adhesive tape on or near a belt, clean both the belt and any surface it contacts with alcohol. Before switching on, rotate the assembly by hand, ensuring that the movement of the motor pulley produces the correct result. After a tedious job of assembly, nothing is more annoying than a spilled belt.

Where flat belts run over tensioning pulleys, it is essential that these shall have their spindles at right angles to the belt run. Fitting tensioned belts is often easier when the tension is partly applied, so it helps to remove the strong spring and fit an auxiliary spring to give temporary weak pressure where needed—or, *in extremis*, to wedge the pulley bracket in place.

Belts that rely on forks or tongues to reposition them for speed changing should not be shifted unless they are turning. The prime cause of belt spill with some machines is speed changing when at rest.

Conversely, idler wheels that are ramped to step on different pulley diameters for speed changing should only be moved axially when in the neutral position. There are exceptions, but these are usually soft-tired idlers with lightly-sprung guide brackets. In these cases, attention to the spring pressure is needed: some slippage may be required, over-drive can be as bad as under-tensioning.

Prime cause of poor pulley drive is a binding bearing caused by an influx of dirt or a hardening of lubrication. Dismantle, taking care to note washer sequence, and clean thoroughly. Again, resort to the thumbnail sketch

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Tape Transport Maintenance

(Continued from page 48)

if several washers are used. Lay the washers in order of removal, ready for refitting. These may be of dissimilar material, to suit bearing hardness. Reassembly in the same order is vital to smooth operation.

Clean out the bearing barrel of the pulley and lubricate very lightly after reassembly. Clean the periphery of the idler with alcohol, looking particularly for embedded particles. Aluminum flywheels are particularly prone to shed surface flakes, so clean not only the pulley but the surface it drives.

Brass surfaces can polish rubber tires, producing slippage because of the hardened driving edge. If the pulley is not too far gone (hardened by constant temperature changes and polishing), treatment as for hardened belts may do the job. Scrape the edge of the pulley gently with a sharp blade, avoiding nicks and notches, to renew the surface, then treat again with alcohol before refitting.

Pulleys, idlers, puckwheels, and intermediate wheels that have been left in engagement for any length of time will present a formed surface, and cause a 'knock'. Listen as the machine runs; gauge the rotational speed from the frequency of the knock, and identify the offending idler. If it is in the early stage of deformation, heat treatment and cleaning may improve matters, but when in doubt, fit a new idler. For the cost of a few cents, future worries may be saved. Æ

Tape Transport Maintenance

Part IV: Clutch Systems

H. W. HELLYER

TALK OF BELTS and idlers leads us naturally to the subject of clutches. The function of the clutch is to transmit torque from one rotating surface to another with variation of the connecting friction. Tape decks usually have a clutch device for taking up the tape after it has passed through the sound channel and the constant-speed drive of the capstan. The variation is needed because the speed of rotation of an empty reel at the start of a recording will be considerably more than the speed of the same reel nearly full of wound-on tape.

So much is obvious, but the designer of the mechanism needs to know what the ratios of speeds will be, how much variation of friction is going to be required and what torque is available to the driven surface. He needs to know also—although with some designs it seems that the point has not been considered—what effect on the driving source the varying load will have. For us, who have to maintain the contraption which has resulted from a distortion of his dreamchild by the cost-paring guys of the production team, it helps to know how the clutch systems have evolved.

Professional tape-transport mechanisms and the earlier home recorders, as well as the more expensive versions we can buy today, are generally three-motor machines. We have taken a brief look at motors in the earlier parts of this series of articles, and we saw there that induction motors, because of their relative inefficiency, can be loaded to run more slowly, or can be

Fig. 1—Variations in torque between beginning and end of reel. Upper reel needs 3:1 variation. Lower reel holds less tape and has a smaller torque ratio. Torque compensation at right is an example of the use of a movable guide which swivels to alter wrap angle.

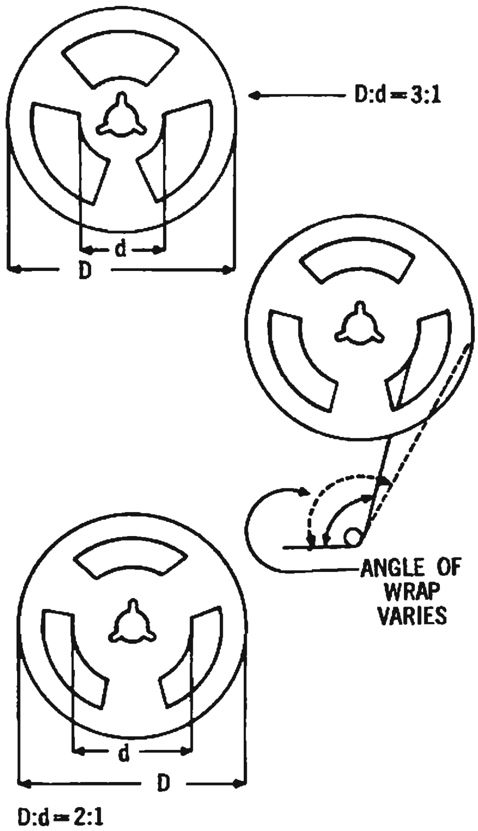
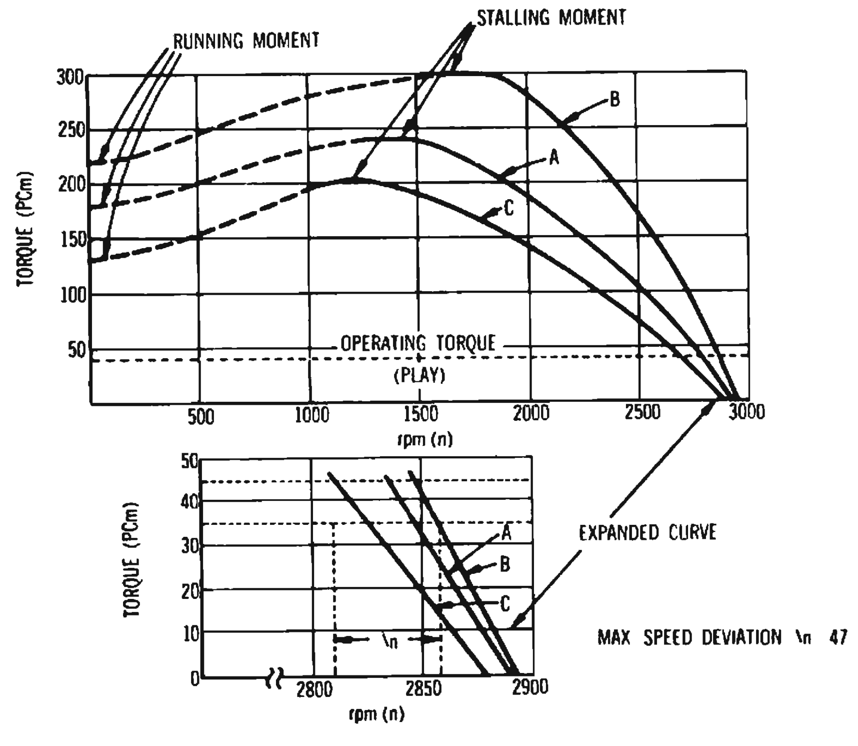


Fig. 2—Curves showing relation between torque, loading, and voltage. Curve (A) is for rated voltage, while (B) is for 10% high and (C) is for 10% low. Lower figure shows expanded curve at operating torque allowing for variation of loading, showing maximum speed variation.



powered to have less torque. Both methods are used to obtain a measure of clutch action. But the drawback with three-motor designs is that they tend to have a fairly small range of torque variations, and the tape is wound on too tightly as the amount of spooled tape increases. Ways of getting over this are to increase the size of the hub of the reel, making the ratio of diameters of full to empty reel a smaller figure, or to run the tape over a swivel guide whose tensioning action is determined by the wrap angle of the tape, as illustrated in Fig. 1.

In this respect, although they cannot achieve the fast winds of a fully powered motor, the clutched take-up mechanisms have the advantage over direct-drive machines, for the tension of the tape can be kept fairly constant throughout the whole of the spooling operation.

Some idea of the extent to which changes in voltage and of load can affect torque of a motor may be gleaned from Fig. 2. These are curves for an asynchronous motor, and are typically those that would be found for a single motor with a belt drive to weight-dependent clutches. The three curves show the connection between torque and speed at nominal voltage (a), and at voltages 10 per cent above (b) and below (c) nominal. The maximum torque of the motor has to be designed to ensure fast rewind at the lower voltage.

Operating torque for normal drive, that is during recording and replay, is small, and can be denoted by the lower dotted line. We see that the difference between the running moment and the stalling moment of the motor for each voltage is quite clearly affected by the applied voltage, and if we expand the end of the curve, as in the lower part of Fig. 2, we can determine the maximum speed variation, allowing for load variations of 10 per cent. In this case we get a figure of around 50 rpm, which is very good for the average domestic machine. It would only be achieved by careful attention to bearing friction of the rotating parts—a procedure which would also reduce the wow-and-flutter figure considerably.

In our investigations, when poor "clutch" action of a direct-drive machine is the fault, we should also look for fouled bearings, or bearings that have lost their lubrication. Not only of the motor itself—where fast winding may be perfect but take-up poor, or reeling-off retarded when running in the stalled condition, as some feed motors must—but also of the idlers or belt jockeys.

A digression is called for at this point. Poor take-up, jerky action, even a halted motor on an under-driven three-motor machine can have th most elementary of causes—dirt. Tap trapped in the flanges of closely ma chined guides, a sliver of bad splicing in a vital part of the head channel, hardened and/or overtight pressure pads: these are faults so obvious as to be mentioned with an apology to the reader. My experience as a tape recorder engineer shows that far too many faults are caused by the neglect of routine maintenance. And not *all* my customers are mechanical ignoramuses!

After the direct drive, two main types are clutch are found. Weight-dependent clutches form the largest proportion in 'table' machines, designed for horizontal operation. Modifications of them will be found in vertical machines where the inward pressure, which now takes the place of gravitational force, is effected by clamping the reel firmly to the reel carrier. Faults here are always apparent. If in no other way, they will always show themselves as a spool rattle or uneven tape stacking during fast winding.

Perhaps the only particular point that needs stressing is the tendency of spring-loaded locking devices to jam in many types of portable machine. These can be very tricky to mend, especially as so many of them depend on exact spacer washer dimensions. The circlips that provide the locking clamp to the reel carrier spindle are small, and necessarily weak. Care is needed when removing and replacing these.

Although the method of drive is not important to our fault-chasing, in practice we find that troubles in clutches often derive from the method of drive. The transfer of torque of a weight-dependent clutch is generally by a felt ring or disc. Many tape recorders have larger discs on the right side. Take-up torque is greater than the reverse torque of a feed spool. The basic turntables may be similar, but care is

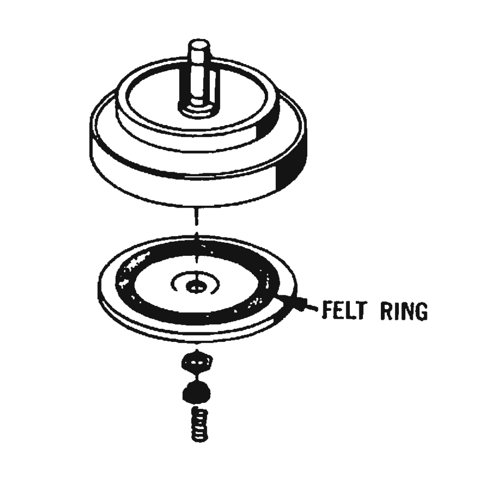


Fig. 3—Typical weight-dependent clutch, with torque transmitted to the upper portion from the driven lower portion by compression of the felt ring, which varies with the weight of tape spooled.

Fig. 4—Older type of Philips clutch assembly which relied upon the dropping action of the auxiliary spindle A allowing the upper turntable to contact the plastic pads B for fast winding. For slipping action, a plastic disc is mounted on the spindle and locked by the shaped block.

needed, as in some machines there are subtle, not always evident, differences between the spools. If you take them off, mix them up, and then refit them, you are in for a pack of trouble and some baffling symptoms.

As the amount of tape which is loaded increases, the weight compresses the felt and more torque is transmitted from the driven lower part of the clutch to the upper. Figure 3 illustrates the basic type of weight-dependent clutch, and although styles may differ, the general design is as shown, with a lower drum rotated by a belt or an idler, its bearing on the solid base of the deck or a fixed plate, rotating on a common spindle that is also fixed. The upper part has the reel-carrying spindle and a felt ring or disc beneath it, often fixed to it or as

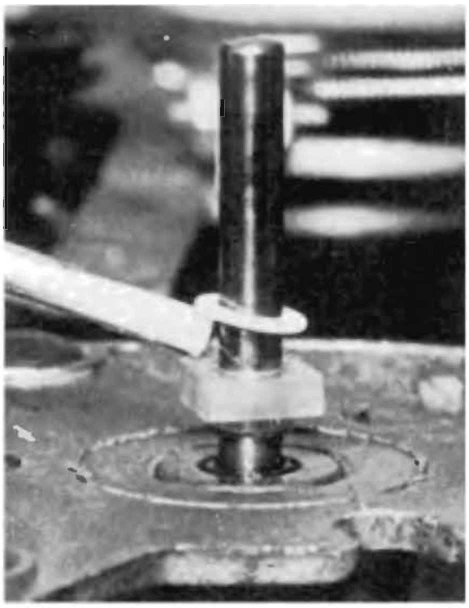


Fig. 5—Close-up of spindle and block described in Fig. 4, with Neoprene washer used for adjustment indicated by screwdriver blade.

shown, to the lower disc.

Variations include the use of a hollow spindle for the bottom portion with a linking spindle between upper and lower members. The linking spindle itself carries a disc or other device which is then pressed hard against the driving drum, locking upper and lower sections together for fast winding. See Figs. 4 and 5.

Another type has a spindle running directly through the common bearing, which is hollow, and fast winding is achieved by this spindle being pulled down from below by the lever system interlocked with the function selector.

Many of the cheaper machines have even simpler types of clutch, with only one rotating member. This is usually driven by a belt coupled to the flywheel, in turn driven by a ramped idler wheel or another belt. The reduction, and the use of a fairly slack belt gives a degree of slip and should, in theory, provide an even wind throughout the length of the tape. It will be appreciated—especially by the owner of this type of machine—that variations in external conditions will easily upset the torque, never very constant at the best of times. We have already seen what wide variations of load and voltage have to be accommodated. It is not unusual for a power supply to vary, or for internal conditions to give the same symptoms. So where a 'slack-clutch' design shows signs of poor take-up, make the motor supply your first check, even if the motor may seem to have nothing to do with the fault.

Complicated types of clutch assembly may have grown up during design by one pressure after another being put upon the designer. One such complex style of assembly arises from a basic horizontal design having been adapted to vertical operation, and the fast-winding facility of the original having to be maintained without any modification to other parts of the machinery.

Important features for this purpose in a Japanese unit I examined are the conical springs and 'starfish' washers. These are used to vary the compression. (There is an even worse French design with washers that can be rotated for different pressures, and a German version that uses felt pads in 56 different combinations to vary the coupling pressure. See Fig. 6.) The take-up clutch consists of two plastic rollers with felt between them; these are sprung apart by the conical spring, and the lower roller is idler-driven. The legs of the star spring can be bent for a compromise tension between take-up and fast winding.

Fig. 6—Position of felt or cork pads in cutouts of clear plastic lower section gives varying friction for torque compensation. Note shaping of reel carrier underside, giving a small degree of safety slip.

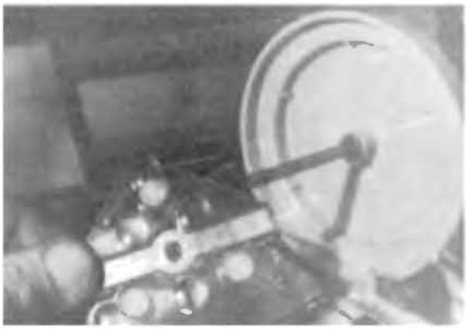


Fig. 7—Sony clutches are efficient, with simple fast-wind action by the heavy-tired idler seen at the right locking motor pulley drive directly to both upper and lower sections of clutch.

Sony has used what is perhaps the best variation on this theme, a form of weight-dependent clutch with the lower section firmly driven, and then the two portions clamped together by the insertion of a free-running idler in the system, as in Fig. 7. For fast winding, this 'clamp' simply locks the two portions of the clutch. There is no slip, no complicated adjustment, and no variation of basic drive.

A clutch system that was originally designed for horizontal operation, but has lately taken the conversion to vertical working with aplomb, is that employed by Tandberg and illustrated in Fig. 8. It has a driven lower section, and in this case the two sides have similar drive arrangements but differing adjustments to allow for the different torques. A crossed belt, engages the motor pulley on two sides—which helps reduce several of the belt-drive variations we talked about last time, and doubles a flutter frequency, throwing it well above the bothersome region. In this design, one small drawback is that turning the left spool with the motor off will also cause the right to move, making threading-up a bit tricky. Provision of a 'free' position of the famous Tandberg joystick on latest versions has now eliminated even this small fault.

The device works by the movement of a plastic 'cartridge' with a lug against which two arms press to lift or lower it when the different functions are selected. But during PLAY the tension is really provided by the compression spring. This allows the felt to be compressed sufficiently to give the right amount of friction. The felt is, of course, between the two sections of the clutch.

Small adjustments may be needed, especially of the lefthand assembly to reduce back tension which causes flutter. Most often, these adjustments are needed when going from horizontal to vertical operation. The Tandberg is not the sort of machine to be tossed around—not even the suitcase version. The left arrangement is a bit different from the right, and reference should always be made to the service manual. The auxiliary brake on the left reel-carrier assembly complicates matters.

Adjustment is mainly by the setting of the small screw in the end section of the clutch lever, accessible from the side of the machine when it is removed from the cabinet—no need for deck removal. The adjustment is for a compromise tension, avoiding excessive back tension, but not allowing the tape to spill when stopped from FAST FORWARD.

When making these adjustments, the compromise setting must often be borne in mind. Never adjust merely for 'good take-up' or 'effective fast-winding' until the complete sequence of operations has been run through. Tape spillage when the machine is stopped from running at any speed and in any direction is a fault that always gives a clue to clutch maladjustment. (Equally, brakes may be at fault: very often, brake and clutch operations are inter-dependent, as we shall see later).

Slipping-belt types of clutch, the main drive system as in Fig. 9, are usually constructed on the 'one-piece' system, with the different speeds of rotation effected by a change in belt tensioning. A jockey pulley, sometimes more than one, is sprung in against the flat belt by the lever system. The belt itself was part of last month's subject and need not occupy us here. But the jockey pulley or rider on which it rides is very much in our mind. It must be vertical to the belt run, and free to rotate, if so designed. The lever must be free to swivel and any springs used should be checked for correct tensioning. As this kind of action is constantly under changing drive tensions, special care must be taken when checking springs.

Most slipping-belt drives result from a build-up of foreign matter on pulley faces, especially the fast-moving motor pulley. Don't trust your eye: a fine polish may mask what is really a mechanically 'sticky' surface. Clean off all pulley surfaces with alcohol, run them over with cleaning ribbon a few times and ensure they are dry before refitting the belt. More will be said about flat belts and their vagaries when we deal with braking systems.

Safety clutches bother some people. The idea of these devices, usually a form of multi-disc clutch with complex adjustment, is to prevent the pull from exceeding a calculated maximum—to obviate tape damage. Braking forces can be formidable, and thin, double, or triple play tape is vulnerable to such stresses. The safety clutch may consist of a number of coupled discs, usually thin plastic, sprayed with flock or other felt-like finish. Spring tension, not always obvious in application, holds the discs together, and in contact with the clutched surface, and adjustment is for the maximum stress permitted for the tape for which the machine is designed. Then, if the pull exceeds this, because of fault conditions, fierce braking, or other troubles, the safety clutch slips, allowing the spool to decelerate more gently than with a direct clutch action. Like all safety devices, the added complication gives the service man additional problems when it goes wrong and some of these devices are the most difficult to set up.

Adjustment is often a tedious business, and will depend on being able to measure the pull of the spooled tape at all weights throughout the spooling operation. To determine the slipping moment of a clutch with a constant braking moment, a spring balance can be used, and a measured pull at a measured distance from the center of the reel (with the pull at right angles) is taken. The moment is force (poonds) x distance between center and point of application (centimetres). The pond is a force equivalent to the pull of a gram weight. For reels up to 7 inches in diameter, readings of 150 to 300 would be normal, and 8½ inch reels would give readings of 200-400, using this scale.

For purely weight-dependent clutches we need a different procedure. Place a reel of minimum diameter with only one or two turns of tape on it on the right turntable. Draw off about 50 cm from the reel and loop it to the hook of a 100-g spring balance. Switch to PLAY and allow the tape to be taken up slowly. Pull as measured should not be less than 20 p for the average domestic machine. Repeat the test with a full reel. Look for a tolerance of from 20-100 p. Check especially for variations of tension during a revolution of the turntable, which would indicate eccentric or worn bearings or spindle trouble.

Use of the correct tape has been stressed already. Oversize tape will tend to retard spooling, undersize tape will wander and cause irregular torque. Tape that is too thin causes greater weight per diameter spooled and affects torque when weight-dependent clutches are used. Æ

Fig. 8—Slack clutch drive system with slipping moment via belt from flywheel to lower section of simple weight-dependent clutch.

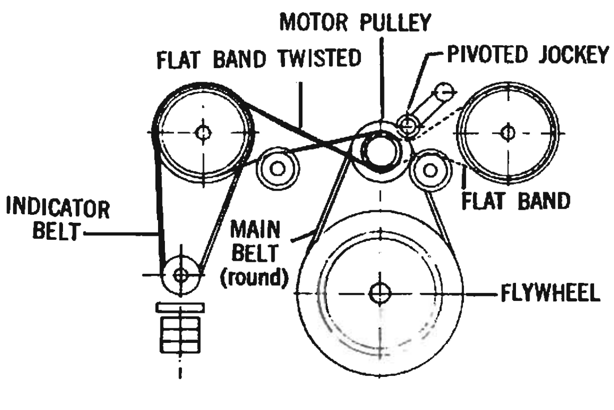
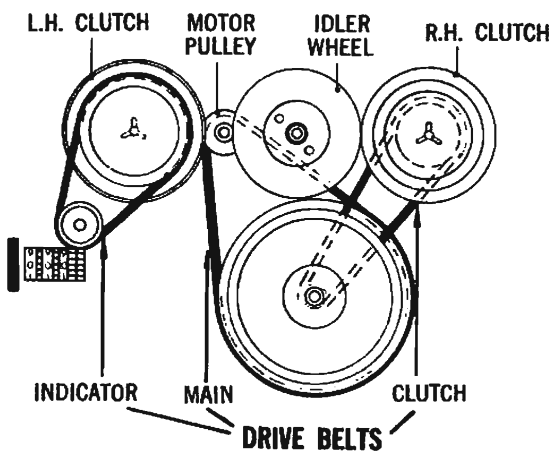


Fig. 9—Clutch action obtained by tensioning of flat-band type of drive belt, using free-rotating jockey pulley on pivoted lever.



Tape Transport Maintenance

H. W. HELLYER

PART 5/Heads, Guides, and Pressure Pads

Dust is the great enemy of tape recorder mechanisms. Audio enthusiasts care for their disks as a mother cares for her child. They wrap them in paper sleeves each time they are stored, preen them with dust collectors each time they are played, and meticulously remove all the goo they can find—and some they may only imagine—from the precious stylus point. But too many tape recorder users are content with a perfunctory wipe over the recording and replay head faces when high-frequency losses become too painfully obvious. Tapes may be stored in boxes, but no special care is taken to exclude dust or the creeping effect of humidity, and this is transferred to the head at first play.

Perhaps the reason for this neglect is that spacing-effect losses are not always so obvious to the listener in their early stages as are the pops and crackles evinced by a dirty disk.

Losses of this sort begin at the high-frequency end of the sound spectrum, and this is where other system losses have their effect, and where the hearing of the listener first begins to fail. Dirt on the heads is the primary cause of spacing-effect losses.

Loss of tape contact on the pole pieces of a head most seriously affects the response of that head. The effect on a replay head is worse when the wavelengths of the 'magnets' on the tape are short, (higher frequencies), partly because such short wavelengths have a weaker external field. Movement away from the pole faces causes dropouts.

Dropouts can be measured pretty accurately. Just lately, a good deal of work has been done into their annoyance value, and for a long time a simple formula has been used, expressing the severity of dropouts, thus

$$S=\frac{l.d.h.}{t.s.}$$

where S is an arbitrary expression of severity, l=length of impurity, d=depth, h=height, t=width of the track and s=tape speed. From this we can see that dropouts are worse at lower speeds and narrower tracks and depend directly on the volume of the impurity. Unfortunately, this expression could be challenged by any practicing engineer, because the depth of an impurity has such a drastic effect on recording.

The strength of a magnetic field varies as the square of the distance from the source of the flux. So a soiled head that causes a slight loss during replay will cause a much more severe deterioration in signal level and response when a recording made under these faulty conditions is played back with the same head. The circumstances for recording are similar insofar as dropouts have the same result—a loss of signal. But while the spacing of the tape from the head, caused by the impurity, will reduce signal strength, so will it also reduce the magnitude of the applied h.f. bias field. Reduction of bias, among other things, causes a lower signal amplitude to be recorded. So the trouble is cumulative, and the tiny scrap of dirt we have ignored impairs playback of a pre-recorded tape by a factor of X but reduces replay of our new recordings by a factor of X^4, i.e. the square of the distance times the square of the distance. This is why the complaint is often heard when a tape recorder owner brings his machine to the mechanic: "Playback is O.K. but recordings are weak." What he really means is that he has not noticed the slight degrading of replay and has perhaps unconsciously compensated for it with his controls, but the drop in recording strength is very noticeable.

We can be more precise than this and measure the spacing loss for a given distance (or size of impurity). Call the attenuation A and the distance of tape from head d, as in Fig. 2, and plot a curve whose vertical axis is the attenuation in decibels and the horizontal axis the spacing in proportion to the tape magnet wavelength, and we get something like Fig. 3. The wavelength depends on tape speed (the higher the tape speed, the longer the wavelength) and the frequency being recorded (the higher the frequency, the shorter the wavelength). Again, we note that the greatest effect is to the higher frequencies, and at the lower speeds.

The tape flux attenuation is given by the formula

$$A=e^{2\pi\cdot\frac{d}{\lambda}}$$

where e is the base of natural logarithms, 2.718, 2π is 6.28 and λ the tape wavelength. So when the spacing equals the tape wavelength $d/\lambda=1$ and $A=20$ log $e^{2\pi}$ or 40π log e which is 54.5 dB.

So again we can get down to actual figures and say that at 15 kHz, our highest frequency to be recorded, with a tape speed of 3¾ in/sec, the wavelength

Fig. 1—"Loss of tape contact seriously affects the response." This Grundig deck collected so much dirt it was a wonder it played at all!

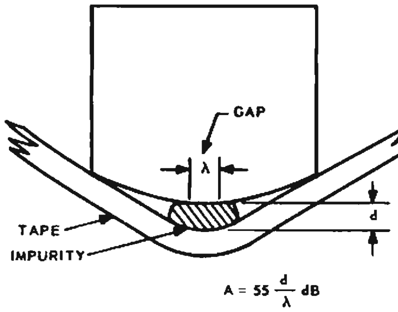


Fig. 2—Spacing loss is 55dB per wavelength.

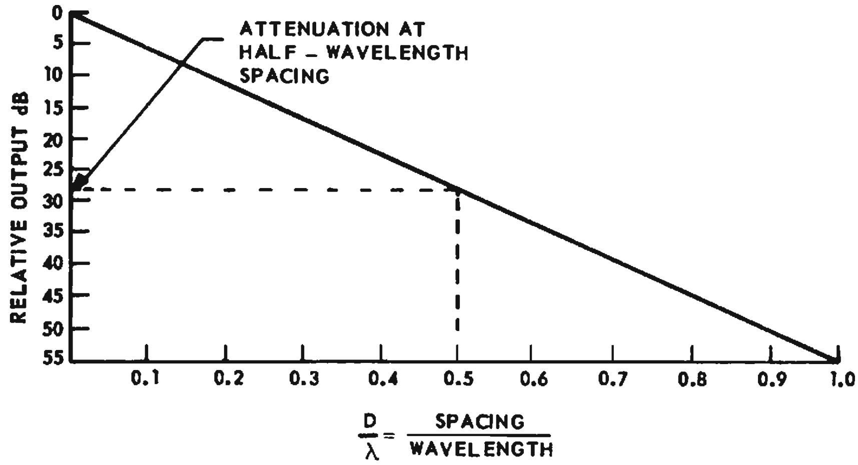


Fig. 3—Showing relationship between attenuation and wavelength.

will be 3.75/15,000 or 0.00025 in. So an impurity of only a quarter of a thou' will produce a 54.5-dB attenuation in replay. That's a voltage ratio of approximately 560:1. The loss after recording does not bear thinking about!

The foregoing dip into basic mathematics was quite deliberate. A shock treatment, if you like, to show that head cleanliness can never be taken for granted. A regular and habitual routine of cleaning and degaussing should be part of any tape recordist's schedule. There are plenty of preparations and cleaning kits available; no need to resort to worn-out toothbrushes and household detergents. The fluids are mild, non-toxic, will not damage plastics, are non-flammable, and should evaporate in reasonable time. Some cleaners carry a small amount of lubricant which is deposited while the carrier is dispelled. In other kits, the cleaner and tape lubricant are separate. One kit, at least, has a pylon arrangement which can be mounted on some decks to provide an extra running surface for the tape, with the lubricant dropped in the pylon to infiltrate the outer felt and provide a slow, regular application to the tape. When it is necessary to run old tapes through your previous sound channel, this preliminary cleaning can be an invaluable aid.

Other head cleaners may be too mild for cleaning of a heavy oxide deposit from a neglected head assembly. By the look of some tape recorders that arrive in the author's workshop, many owners operate in a coal-hole and have never heard of head cleaning. These have to be tackled with a stiff-bristled nylon brush and denatured alcohol, methylated or surgical spirit. Stubborn deposits are better removed by several ventures than a harder, protracted scrubbing.

When cleaning worn tape heads, always rubs along the line of the tape travel, not up and down. There will have been a tendency for oxide to have collected in any horizontal grooves made by wear, and cleaning this way helps to remove the deposit. Some heads have horizontal slots in the facing, aiding the tape contact by a suction effect. Oxide can build up in these. The nylon brush is the answer, *not* a poke with a screwdriver blade.

Final cleaning and polishing can be done with cotton wool on the end of a manicure stick, or, better still, the slightly pliable cottonwool sticks supplied by cleaning makers.

There are two philosophies about guides. The tape must move in a true horizontal plane, without lateral or vertical wandering, so there must be a degree of back tension. One system requires the trapping of the tape against the head with a fairly tight pressure pad, a roller, or a band. All three types are illustrated in Fig. 3. The alignment is often by a U-shaped piece, perhaps part of the head assembly, or by a closely machined central guide post, keeping the tape in true horizontal traverse.

With this first system, a mumetal shield on a sprung plate may be used, and this has been utilized as an auxiliary guide in more than one design. Great care has to be taken that this shield slides properly and completely into place and that any felt pad on its inner face is kept soft and free from impurity. Quite often, this shield is the only guiding the tape gets, apart from vertical machined pins without flanges. It can be demonstrated that a larger-diameter guide is much more efficient as regards tape wear, regularity of control (freedom from flutter), and lateral guiding effect, yet deck designers continue to fit ridiculously small pins, saving a couple of cents and eschewing the loss in quality.

The second system has critically machined guides whose flanges route the tape precisely across the head gap. Pressure pads, where these are used, only perform the function of keeping the tape in intimate contact with the head facing. The head shape assumes more importance. Many high-quality decks use no pads, but rely on the wrap of the tape around the head, and the contoured head for which much advantage has been claimed—and challenged—is very much in evidence. One incidental advantage to the maintenance man is that contoured heads are often easier to clean, and keep clean.

Machined guides present their special problems. Tape width is standardized at 0.246 in. ±0.002 in. Although usually described as 'quarter-inch' tape, the domestic variety is nothing of the kind. However, some earlier standards specified ¼-in. with a tolerance of −6 thou. So some tape is still around with a slightly oversize width, and this can jam in a closely machined guide. Any tape with poor edges, or that has been damaged, should be scrapped. Cut and splice ruthlessly. Tape is not all that expensive: and if there is valued prerecorded material on it, the practice should be to dub it off before the tape falls to bits, using a machine with more-tolerant guide dimensions. Even the less-precise

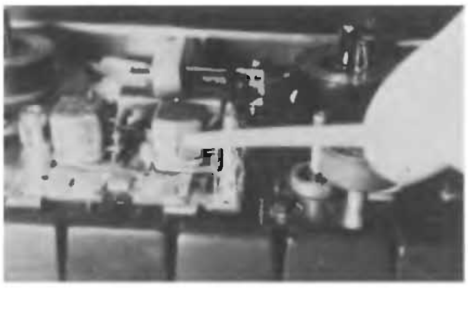


Fig. 4—Sanyo deck with contoured heads.

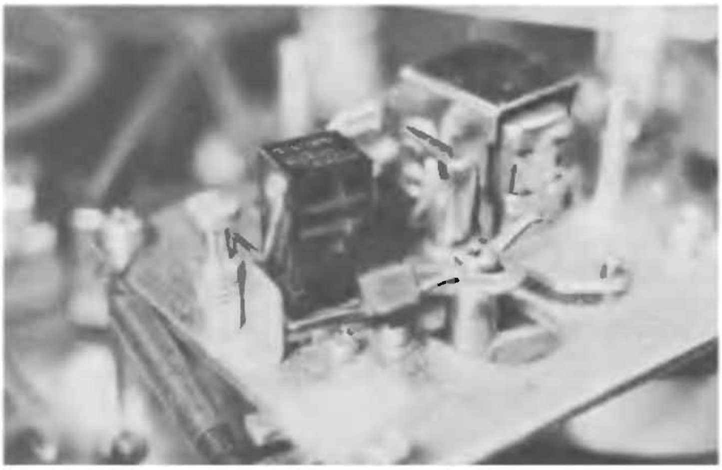


Fig. 5—Sony TC260 showing pressure pad held off by the forked arm from its contact with the erase head. In this Sony design, the Record/Replay head relies on the tape wrap for good contact.

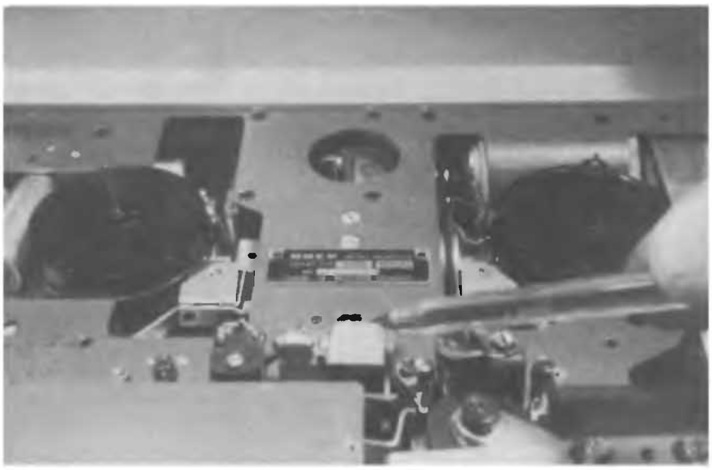


Fig. 6—Uher 4000 allows very little access for cleaning and is better served by an impregnated cleaning band.

Fig. 7—Accessibility is half the battle! The front trim of this Grundig deck lifts away completely to allow free access to the heads.

Fig. 8—The Ultimate! In this Uher Varicord design, the complete head assembly can be removed and interchanged for different track formation. Accurate machining of mountings is the secret.

guide formations of medium-class machines, with their 'play' between tape edge and flange of between 2 and 8 thou. can trap a poor tape; and the build-up of dust and oxide will rapidly diminish the tolerance.

The trouble is that some tolerance is needed, to allow for variations in tape and prevent any tendency to edge curl. But as the track width of a quarter-track tape is only forty thou', any tendency of the tape to wander, aggravated by too generous a tolerance, causes amplitude variations. Tape trapping by narrow-flanged guides or by rough guide barrels will cause flutter during record and play functions and will retard fast winding of many low-torque machines. This is one of the first areas of investigation when the fault symptoms are that the tape slows near the end of a wound reel.

Guides are often adjustable, see Fig. 9. Spring-loaded, with locknut, or simply screwed into the deck, many guides can be set to regulate the datum line for tape travel, and where this is so, setting the guides should be the first job, before any adjustment to the head positioning is made. On a strange deck, always inspect the assembly and look for some fixed datum. It may be the securing of the erase head, a popular method, or the mounting of the plate on which the heads themselves are pivoted: see Fig. 10. Setting the height of the guides according to the maker's instructions is very often an early stage in maintenance and should not be neglected. All the later adjustments that preserve full frequency response, reduce noise, avoid cross-tracking, and generally improve recording will depend on this setting.

Occasionally, we run into trouble with tape edges rubbing on spool flanges and the inevitable temptation is to alter the guide position to rectify the error. Before doing this, check the spools themselves and their carriers. The guides seldom go out of adjustment, and if they do, are more likely to exhibit their fault by being loose on their mountings. For adjustment where no information is available, first set the tape run to the fixed datum, then adjust the guides for a level run between spools, noting particularly any tendency of the tape to rub flanges or the top cover of the deck when the spools are turned by hand. After this, set the level of the erase head so that the upper edge of the top track can just be seen above the edge of the tape, stretched tightly against the face of the head. Finally, set the record and replay (or combination) heads so that the upper edge of the tape just cuts the visible upper line of the

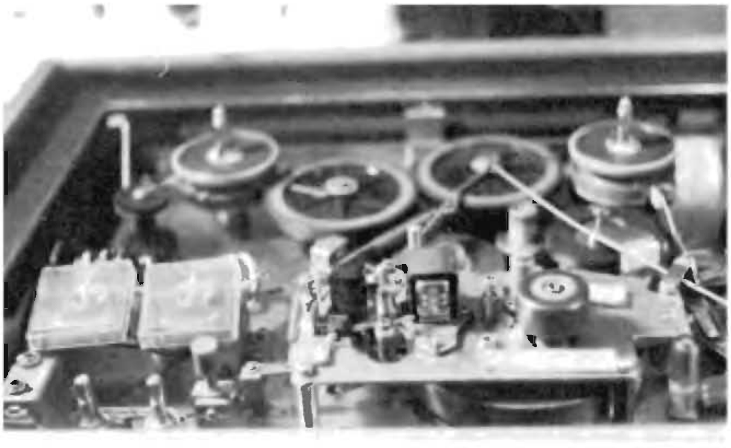


Fig. 9—Showing various adjustable guides on another Sony deck.

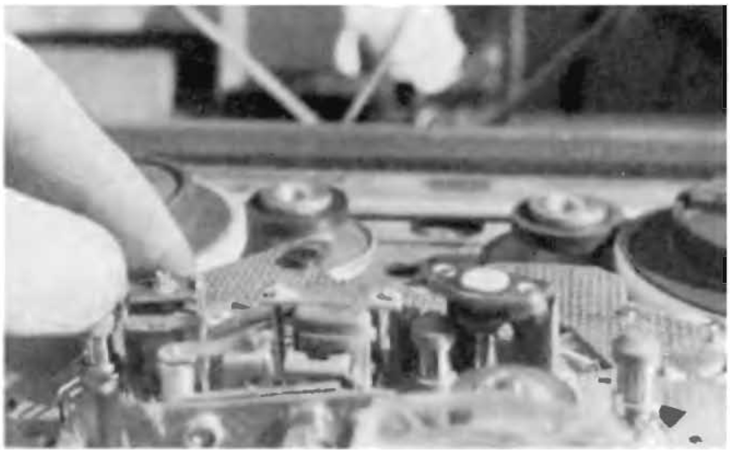


Fig. 10—Telefunken deck showing guides before, during and after tape path through channel.

polepieces. In other words, when in doubt, allow the erase head to overlap a little. Signal-strength tests will prove the final setting, and the normal rocking action of azimuth alignment completes the adjustments.

Plain brass guides, pins, and auto-stop feelers tend to wear badly if dirty tape is used. Quite often, a groove will be made in the rounded surface, and this is a prevalent cause of tape trapping, flutter, and retarded fast winding. View the cleaned surfaces on which the tape bears under a bright light, when these grooves and 'flatted' surfaces will be revealed.

Mobile guides such as free rollers, auxiliary brake pins, or tape-end-stop feelers are much in evidence on better-class machines. Because they continually present a renewed surface to the tape, they wear less readily, but a tape deck with these devices should be inspected carefully for any tendency to bind. Free-running guides should spin freely. Other types, on sprung arms, should swing readily as the tension on the tape is altered.

Watch for compensating springs that have seized, or arms that have bent, throwing the guide axis out of true vertical. Lubricate spindles very lightly, after the usual cleaning. Spring-tension arms have been mentioned in the previous sections on clutches and brakes. From our point of view this month, they can be treated as auxiliary guides and should receive the same careful treatment.

Watch out for the pin or spring guide between the heads, a form of auxiliary adjustment on many models with a sharply sprung pressure-pad plate. This device tends to wear badly when the head-gate assembly becomes cluttered with abrasive dust.

Pressure pads and the plates on which they are mounted have already been mentioned, but the adjustment of these and of the hold-off arms that limit their throw should be rechecked after other adjustments and repairs have been done. Pads must be soft, may need cleaning and perhaps resurfacing by making a small skim with a razor blade. Whatever the action that has been done, it may require a resetting of the inward tension, and this is a final check on the maintenance work around the head channel. Æ

Next:—*Brakes*

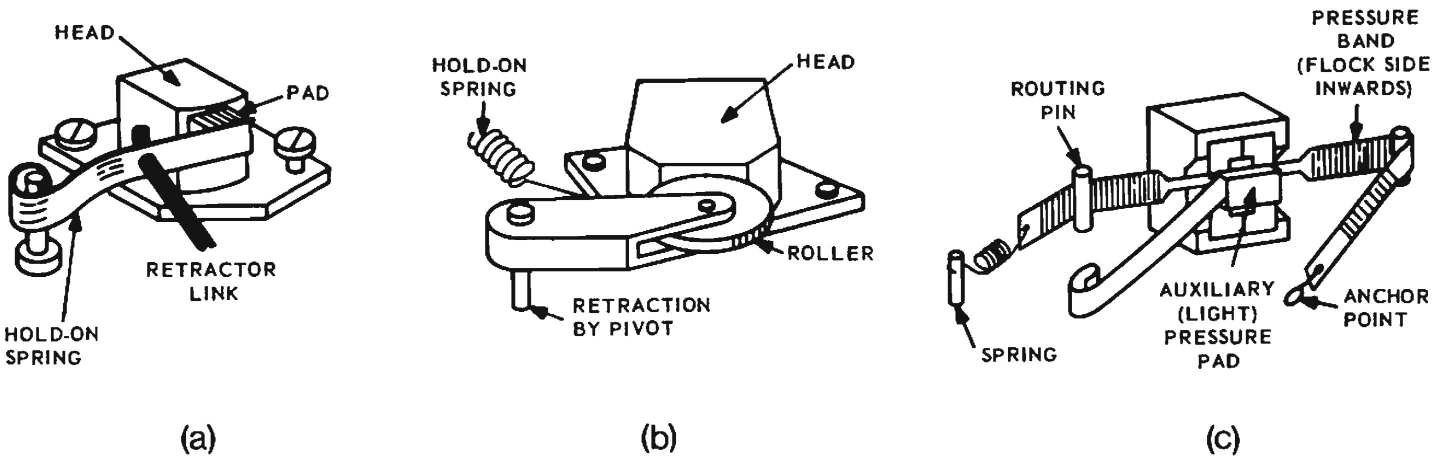
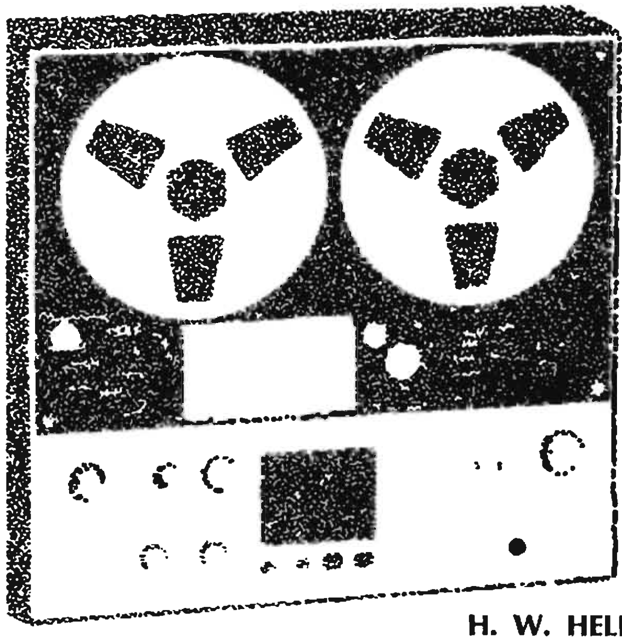


Fig. 11—Three methods of maintaining good contact of tape to head (a) pressure pad on spring arm, (b) pressure roller, and (c) felted band which a gives greater degree of wrap.

Tape Transport Maintenance



Part 6—Brakes.

H. W. HELLYER

BRAKES ARE MEANT to stop things. Although their malfunction is obvious when—in a tape recorder—they do not, more often the troubles caused by erratic brakes are those of binding, retardation, or incorrect timing. Diagnosis depends very much on a knowledge of the type of braking mechanism that is employed, how it should work, when it should work, and with what force. There are many different methods.

Basically, we are concerned with two different processes. First, and most obvious, we have to stop the spools in the shortest possible time without spillage. Secondly, and not so obvious, the correct tape tension must be supplied during playing, so that the tape enters the head channel without slap or flutter and leaves it without snatch. This operational braking is quite often the most difficult function to achieve.

The simplest type of brake is the peripheral pad—usually made of felt—which is spring-loaded to engage with the edge of the spool carrier. Levers hold it off during RECORD, PLAY, or FAST WIND. But the simplicity can be deceptive. When the STOP key is operated, the timing of the brake action is important and quite elementary braking systems can still have offset levers or brackets which enable the feed-side brakes to engage fractionally before those retarding the take-up spool. During servicing, care must be taken to preserve these small differences in brake application. Quite often they will depend on the bending of a bracket, or the setting of a screw.

Testing such simple brake systems consists of an operational run, with sudden 'stop' action applied during any function. It is a common mistake to set brakes for 'normal' braking, ignoring the crash-stop conditions from fast wind in either direction, and with full or empty spools, to which the machine may be subjected as soon as it has left your hands.

Such simple systems, too, may depend for their effective friction on the condition of a felt pad. Constant use, variations of heat and cold, perhaps the throw-off from capstan or idler bearing of minute particles of lubricant, will all cause a skin of 'brake spoil'. The effect is insidious: the brake engages quite firmly and appears correct when inspected with the mechanism at a standstill. But at the important moment when the brake begins to apply friction, the tendency of the spools to skid can cause either spillage of tape or excessive retardation—drag—with the resultant stretch of tape and the danger of breakage.

Peripheral brakes should first be checked for soft felt pads, clean cork or composition pads or shoes, and free rubbers. The last remark may cause some puzzlement: free rubbers? But the type of brake that is a small rubber wheel mounted on a spring arm which is disengaged to contact the edge of the spool carrier or brake drum when 'Stop' is selected is similar in principle to, though rather different in operation from the ordinary pad brake. The rubber wheel engages the running surface, turns briefly, then, as the spring pressure tightens, locks and grips. Usual trouble with this type of brake is binding of the wheel on its own spindle or bearing, with a consequent fierce application that is originally ineffective then *too* hard. The outcome is usually stretched or broken tape.

The cure, of course, is cleaning of the wheel mounting, easing of the spindle bracket, where this is pivoted, checking of the spring, cleaning and softening of the rubber (a bit of extra softening can help here—the wheel drives nothing) and resetting of the stops to ensure engagement at the right time.

Similar remarks apply to the simple pad brakes. Above all—keep 'em clean. Grit, dirt, oil, rubber parings, and other foreign matter spell death to brake pads, just as they do with your auto.

Where there may be some doubt about brake application, always err slightly toward the feed spool. During Play, this should come on slightly in advance, to keep the tape in correct tension through the head channel. Too much in advance, of course, means a retarded tape. The usual fault is the opposite, and a tell-tale spillage loop before the tape enters the first guide. No adjustment rule can be given, mechanisms differ in detail so much. Too often, the mechanic is left to decide for himself how much he shall bend, twist or screw the vital parts.

Servo-brake mechanisms require more than a simple 'off-on' adjustment. They work by applying a pressure to the braked drum which varies as the torque: the faster the drum is initially turning, the less the braking effect. A curve drawn to illustrate braking effect will show a pronounced difference in slope as the contact angle is increased, giving greater 'wrap.' The effect of greater wrap is not a 'tighter' brake but one that gives its retarding effect with a more rapidly increasing application.

The design problem is not one of working out the braking force, but of calculating the variation in that force between a full and empty spool rotating in one or the other direction. The servo brake tends to wedge itself on in the winding direction, i.e. the supply direction, and the contact angle, in the case of a band, or the wedge angle, in the case of a pad, has to be determined with some care. The outward force can be set, and the inward force calculated for differing loading conditions, and by reference to tables of coefficients of friction the angles can be worked out.

But like all carefully calculated plans of mice and men, external influences will make them go agley. The external influence in the case of tape recorders is inevitably dust, dirt, excessive heat, and a great growth of unwanted friction.

At first, the variation from 'as new' conditions is not noticed. By the time alarm bells ring in the mind, and the slowing spools are not as regular as they used to be, it is often too late. The cure may be to change the brake bands, reset the brakes (relaxed springs can by now have made this necessary), or change the pads, which may be of cork, felt or some rubberized composition, but should never be replaced by something different.

Shoe brakes of different sorts are used in servo mechanisms and also in straight brakes. Often, the shoe will be shaped to give the needed servo action, and care must be taken to get the shape right when replacement becomes necessary.

Wear is the big enemy, always accelerated by dust, dirt, and heat. Polished brake bands, pads, and linings are frequent causes of spillage, brake snatch, or uneven application. It often takes longer to clean such surfaces than to replace the material. As the designer has such a ticklish job in working out what materials to use, do him the honor of using the same substance he has chosen, and not a bit of the lining from your old hunting cap. Cork is a common material, with felt running it a close second. Rubber is employed at times, usually when metal rims are to be retarded, and plastic or fabric bands, some of them made from specially treated and tensilized materials, form the basis of simple servo brakes.

In the domestic machine, application is generally direct—though there are notable exceptions that use solenoid-operated brakes. But professional machines have larger spools, and are more often required to change direction, stop and start, or retard their motion from any functional operation. Auxiliary braking systems are thus employed. Stop braking takes two phases: the first is a rapid delay action, retarding the fast-moving spool, changing to a more gentle brake application as the spool decelerates. Relays with delayed-action circuits, and forms of braking magnets are often used. Quick-acting brakes which bring the spools immediately to a halt if the tape breaks, or after it runs through, are also part of the studio machine's make-up. Many of these special brakes are now being incorporated in so-called 'domestic' machines.

One of the special devices is the servo control brake. This is a form of electronic control which depends either on the rate of revolutions of the machinery compared with a standard-frequency reference source, or on a compared recorded track referred to the motor speed. Early forms were transistorized developments of the simple regulator—and not always as effective in practice as their designers may

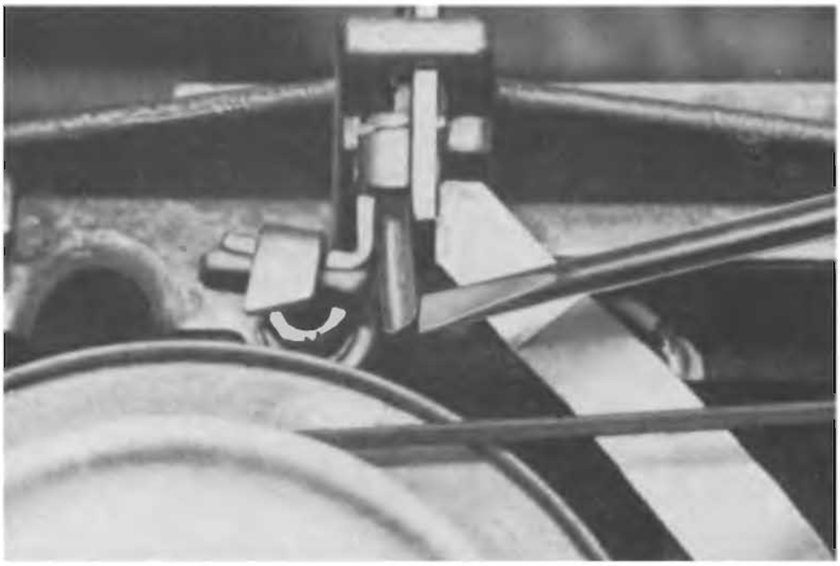


Fig. 1—Differential brake action provided by flexible tongue.

Fig. 2—Simple type of servo brake using fabric band, sometimes treated with graphite.

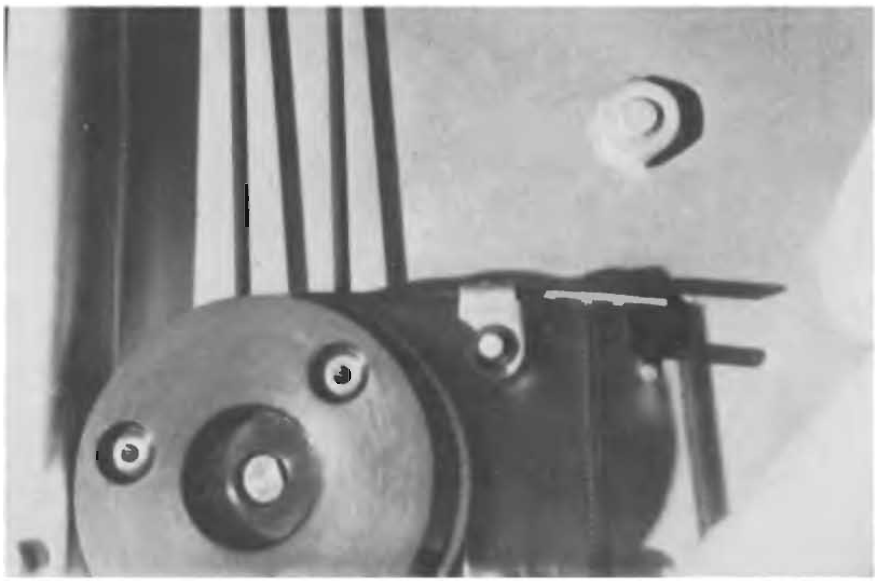


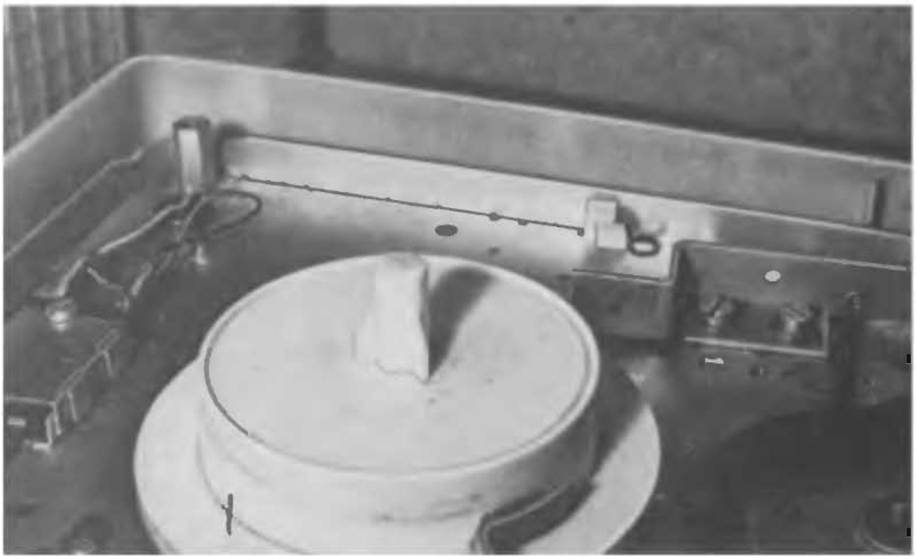
Fig. 3—The servo brake on the Revox G 36 is a steel band and the drum is rubber-lined.

have wished. It seems likely that forms of servo speed control linked with servo-assisted braking will become more common, and a section of this series of articles will be devoted to the subject when space is available.

In general, maintenance of braking systems is a matter for common-sense and knowledgeable inspection. Most frequent problem is wear and tear, broken cork or composition pads, polished linings, loose linkages, and bent brackets. It should never be forgotten that mechanical systems are interdependent. Brakes depend on clutches and drive systems: the one will show evidence of defects if the other is malfunctioning. An example is the rewind action which spills tape when stopped with a nearly full respooled tape. Premature braking may be suspected, but the trouble could be a combination of weak clutch action and 'lazy' brakes.

Watch out for the 'double-action' brake, where the two brake brackets are linked by a common rod or lever, the frictional moment being provided by the direction of spool rotation. Some peculiar effects can be obtained by wrongly adjusting the linkages, and tests should always be made with full and empty spools, both sides, before assuring oneself all is well.

Watch out also for the compensated operational brake, where the angle of the tape over a rider pin is used to give the wedge of a brake a slight pressure against a supply spool carrier. The idea is to provide constant tape tension independent of the amount of tape on the spool. Like all good ideas, it is fine when it works, the very devil to adjust when it does not. Usual adjustment is at the pivot point, and should be made for maximum action with a near-empty spool. Æ



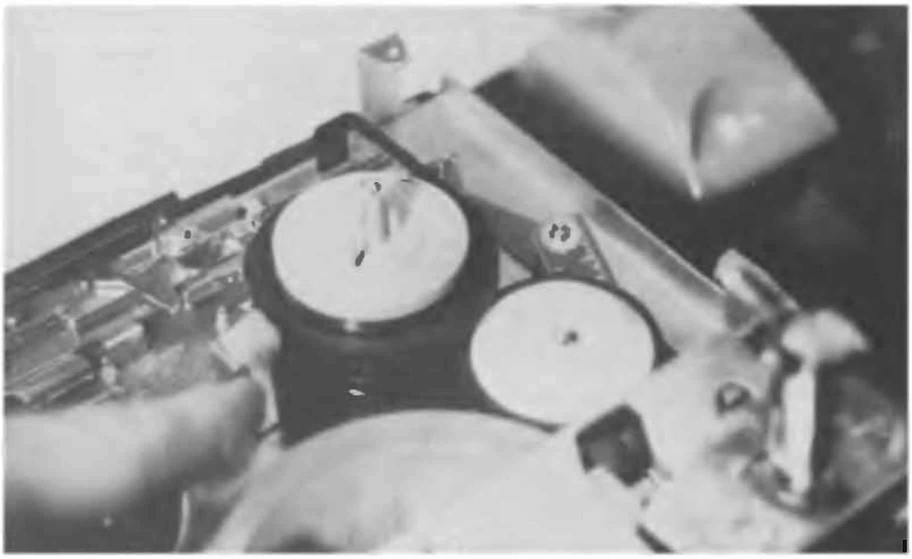


Fig. 4—(Top) Edge-contact brake as used by cheaper recorders.

Fig. 5—Auxiliary brake used by Sony at the take-up spool for the adjustment of tensioning.

Tape Transport Maintenance

Advent Model 200

PART 7: Cassette Mechanisms

H. W. Hellyer

A MAN would have to be blind and deaf to be unaware of the growth of cassette- and cartridge-loading tape recorders over the past couple of years. They will not—as some of the more hysterical reviewers have implied—completely oust the reel-to-reel machine. Each has its own virtue: each its particular application. Cassettes have come to stay, however, and the number of tape decks using a variation of the patented Philips system has swelled alarmingly (or satisfyingly depending on which side of the fence you sit).

The real audio-buff tends to look down at packaged tape, whether cassette or cartridge. He is making a mistake. Whatever system you choose, the standard of modern tape and the improvement of modern mechanics, allows you to enjoy the best of good music at home, on the move, anywhere. As Bert Whyte found (AUDIO, April 1970) for lower level listening "at apartment house level" and for sheer convenience, cassettes could not be matched. He noted the excellence of the tape in the DGG cassettes that had just come to his notice, and commented on the hiss which impaired the dynamic range when played at high levels through a high fidelity system.

Our Bert, still tingling from his train-catching tramp through the snow with the Stellavox, may have overlooked the fact that the DGG tapes were recorded to CCIR characteristics, and he does not mention what equipment was used for his replay experiments.

He is right, though, to say "Until this bug-a-boo of hiss is conquered, cassettes will never realize their potential." Right, too, to point out that progress was and is being made. Recent researches by Ed Canby have given us a clue to the future of the cassette. With Dolbyized recordings and super-quality tape, we shall get to the standard it is now possible to reach with disc and with reel-to-reel recorders of the better quality at 1⅞in./sec. Maybe sooner than you think.

Now is as good a time as any to take a more intimate look at the cassette-loading mechanism, with its special maintenance problems. Cartridges we shall have to leave for later—there are still too many ways of doing the same thing where multi-track operation is concerned. But the cassette mechanisms are all based on an original Philips' design, however much succeeding manufacturers try to trick it up. This design found its way to the market in 1963, called the EL 3300, in Europe. The U.S. invasion followed three years later reversing the usual trend.

Cassette Recorder Design

Design of a portable tape recorder is not just a matter of whittling down a larger machine. Nor is it simply achieved by arranging a battery supply and fitting a handle! There are special circuits for drive control and some ingenious ways of compromising and combining RECORD, PLAY and INDICATOR circuits. In addition to this, modern techniques of electret microphone construction permit the building of a really compact portable with no trailing leads.

Power supplies present few problems. With motors capable of achieving sufficient torque even from as little as 3 volts, and overall consumption during RECORD or PLAY limited to a few milli-amps, battery-stowage space need not be great. The present tendency is to add a simple transformer and rectifier circuit, with rudimentary smoothing, to allow operation from a. c. supply. But the price for this added convenience must often be paid in an increase of noise level during recording.

Some authorities disclaim this as a drawback. "Most recording is done in the field," they argue, "so the a.c. facility can be used on replay." But the crunch comes when one has to dub off those precious cassettes onto open reels for convenient and economical storage. Then, the use of batteries is expensive and the use of an a.c. supply can still raise the noise level more than may be tolerated. Perhaps the solution lies in the rechargeable cell—slowly gaining ground among enthusiasts.

Normal method of power-supply switching is a leaf spring actuated by the insertion of the a.c. adapter. In some cases, extra power is available by this action, and in more than one design there are three alternative power-supply circuits, adapted for use with internal cells, from an automobile battery and from house current.

But I exceed my brief as much as the exuberant Bert did and must come back to basics. And nothing could be more basic than the motor supply. Without it the recorder is just so much trash. And with it, unregulated, the recorder may be little better. As batteries run down, so the motor slows, falters and finally stops, and this is not a regular decline. It is considered better for the supply to drive the motor at full available power as long as can be, then stop. This is the way power supplies are tailored nowadays. The regulator ensures a supply until the batteries are no longer capable of sustaining the load, and then the regulator cuts in and the motor cuts out. We shall see later how this can be done.

Simple motor-control circuits start with the Zener diode, preventing the applied voltage from rising above a pre-determined amount, then combined with an adequate overload capability for long life. In the most refined designs they have sensors and differential amplifiers, monitoring the speed and regularity of the tape transport. One or two examples we shall discuss have performance specifications well up to professional rating.

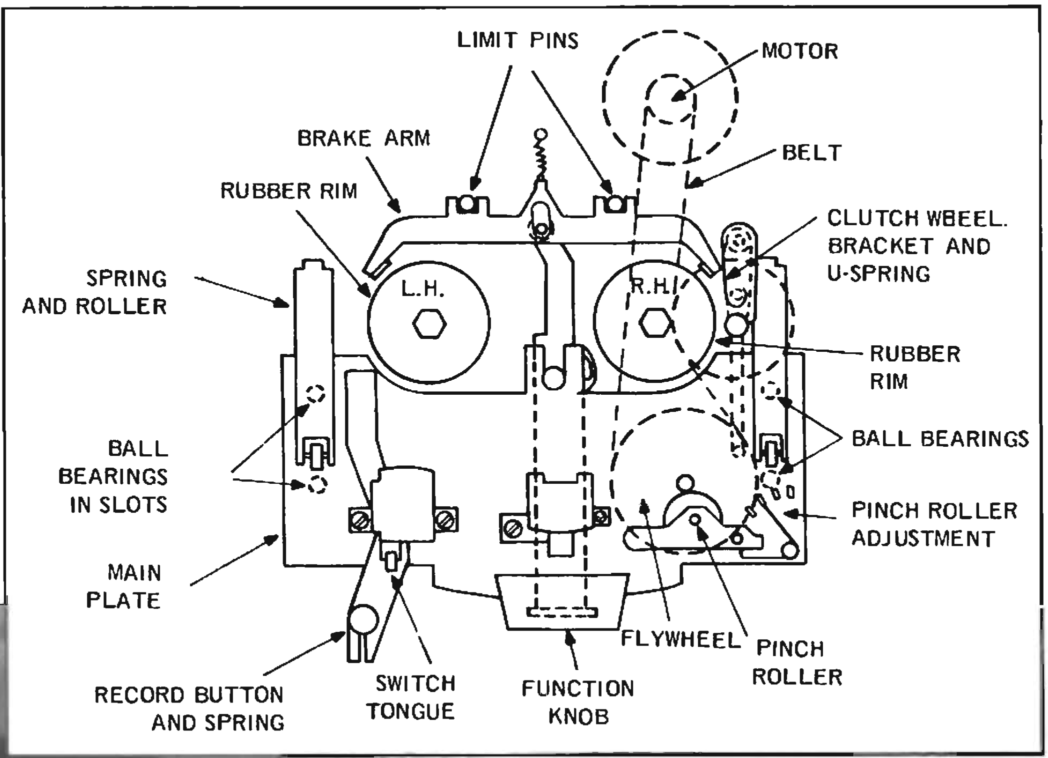


Fig. 1—Showing schematic layout of major components of Philips-type cassette recorder.

Fig. 2—With cassette partly ejected and the tape run through to its leader, the pressure pad and spring and the cutaway portions of the cassette are visible.

For most cassette tape recorders, a deck layout such as that of Fig. 1 is quite usual. Small turntables with sprung, splined tops engage the hubs of the cassette and locating pins minimize the amount of lateral play. As the run of the tape across the front of the cassette (oxide coating outwards) cannot be altered, it is necessary to advance the heads and pressure roller every time we select PLAY or RECORD. (See Fig. 2, where a partially ejected cassette has been run back to its transparent leader to demonstrate the apertures and recesses in the front of the cassette.) The darker shading of this traveling plate in Fig. 1 makes this easier to understand but does not show the various clearance slots and abutments that are needed. These can be studied better in Fig. 3, where an alternative view of the stripped deck is given.

The first thing evident is the need for a shaped cutaway for the capstan spindle and elongated holes for the cassette locating pins. These are shown in greater detail in Fig. 4, where we also become aware of the simple spring adjustment for pinch-roller pressure. The outer arm of this spring is fitted to the end of the roller arm. The nearer the roller the captured end of the spring is placed, the greater the inward pressure.

Also plainly visible in both these shots and indicated in Fig. 1 are the very important springs and rollers which guide the head plate forward correctly when RECORD or PLAY are selected. The plate runs on ball bearings, which click into detents at the limits of travel. Again, simplicity is the aim, and the designers earn a salute for the effectiveness of their baby. Once or twice, I have had trouble with the sliding plate, but the cause has generally been a bit of dirt gumming up the spring rollers and allowing them to override. The solution is obvious!

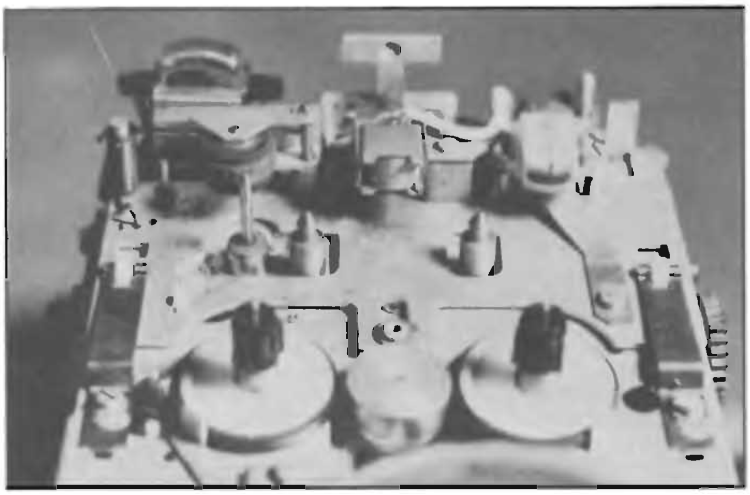


Fig. 3—Showing upper portion of Philips-design deck, looking forward toward the head and function buttons.

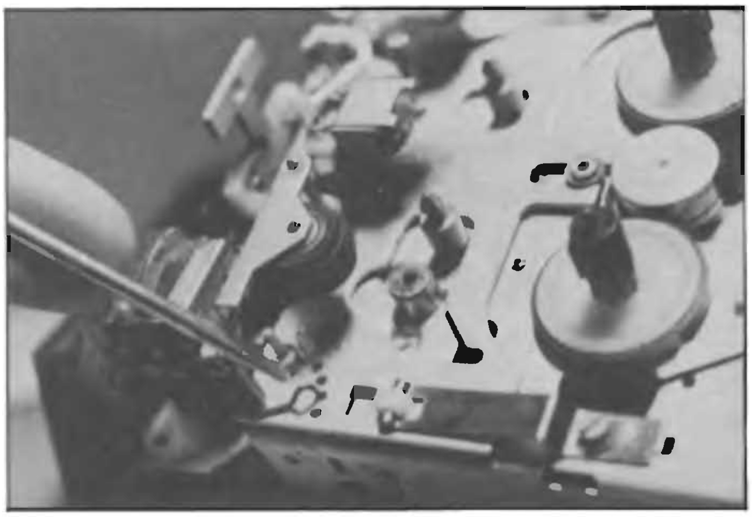


Fig. 4—Pinch roller adjustment is indicated by the screwdriver. The flywheel and capstan spindle have been removed.

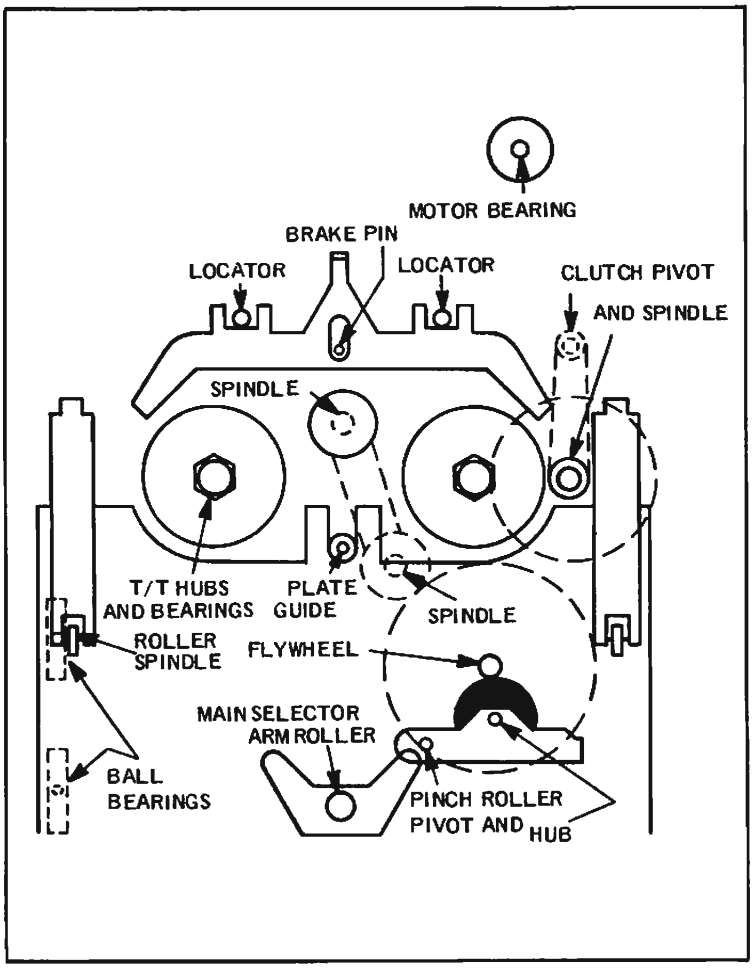


Fig. 5—Showing lubrication chart. Apply light oil to rotating parts and medium grease to sliding parts.

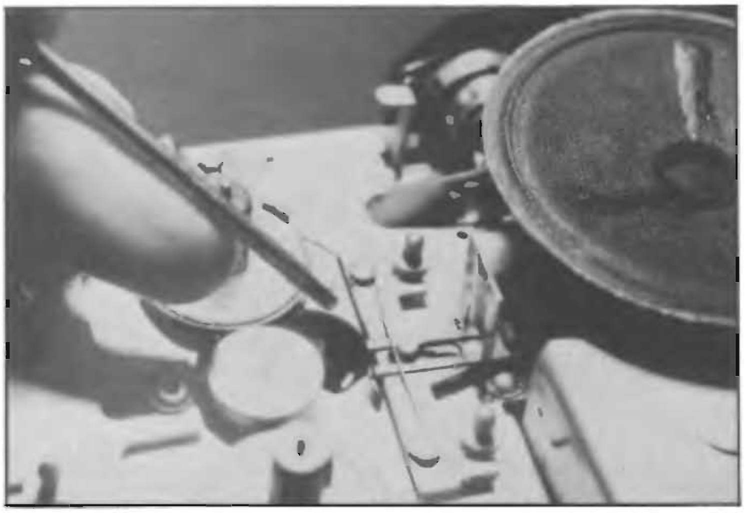


Fig. 6—The transverse spring, indicated by the screwdriver, aids engagement of the brakes, which are lugs on the bracket.

Fig. 7—Showing the underside of the deck, with main-drive belt and lower flywheel bearing removed.

Lubrication

Well, maybe not so obvious, for the most frequent failures I encounter are caused by over-indulgence in lubricants. Cleanliness comes first. My old Nanny used to tell me it was next to Godliness—on some tape decks it is next to impossible! On the Philips design only a steady hand and a fine brush may be needed. Lubrication is limited to a drop of light oil on rotating parts and a smear of medium grease on sliding parts. To make life easier, I include a rough chart (Fig. 5) showing the main lubricating points on this deck. Please note—the motor should never be oiled. Any attempt to do so will result in noisy operation since the brushes will pit. When motors get sluggish, they can sometimes by completely-stripped and cleaned and the bearings lightly lubricated. Whether or not you do so depends on your time/economy scale. Some of these motors defy reassembly anyway. A drop of oil on the top bearing is enough.

Reverting to the deck assembly, we see that the head plate is pushed forward by a roller operated by the function selector arm. This is held by a nylon circlip. In several other places in this deck we find such securing devices and they have one drawback-they dislike being disturbed. If one is servicing any great number of these cassette machines, it is wise to lay in a stock of spare circlips and washers.

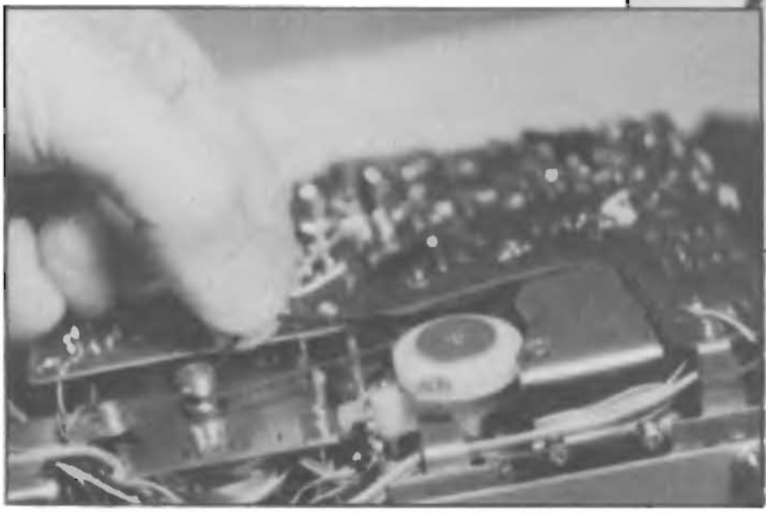
The brakes are mere lugs on the brake bracket, engaging the rubber rims of the turntables. The bracket is held off by a spring attached to the rear part of the spring blade which holds the cassette into place, illustrated in Fig. 1 but removed in Fig. 6 to show more clearly the transverse spring that aids the engagement of the brakes. Possibly because of the enclosed nature of a cassette and a consequent lack of freedom of the supply "spool" to run on, very little trouble is encountered with brakes, and the most frequent service problem I have had with the variants of these mechanisms has been a fouling of a pin in the forked slide.

Clutch Mechanisms

Not so fault-free are the clutch mechanisms, however. The main-drive belt (another vulnerable item) couples motor pulley and flywheel and runs against one section of the clutch assembly. With a motor of 3,000 rpm, this flywheel turns at 6 cycles per second to give the rated speed of 1⅞ in./sec. tape travel. Looking at Fig. 7 which has both belt and lower flywheel bearing removed, we can see the three main parts of the clutch design. Nearer the camera is the clutch wheel, driven by the belt. This is mounted on a bracket with a hook spring applying regulated inward pressure, and the two sections of the clutch have a felt disc between for transmitting the necessary torque, but allowing slippage. The upper section drives the spindle that engages the rubber rim of the take-up turntable. This can be seen in Fig. 3. Adjustment to the required 120-150 grams is made by bending the spring. Not so evident is the need for alignment of motor pulley, clutch wheel, and flywheel for correct transmission of torque. The flywheel bottom bearings, simple pads as shown in Fig. 7, permit some adjustment, and quite often the bracket fixing holes are slightly elongated. The motor seating, especially after replacement with an alternative type, may have to be packed up so that the lateral and the vertical alignments, of the motor pulley are accurate. There is a small amount of adjustment, the pulley being secured by a screw to the shaft, as shown in Fig. 8.

Fig. 9—Fast-wind action depends on two pulleys and the exact setting of the spring indicated by the screwdriver.

Fig. 8—Belt changing is easy, but the alignment of the motor pulley, flywheel and clutch grooves is important.



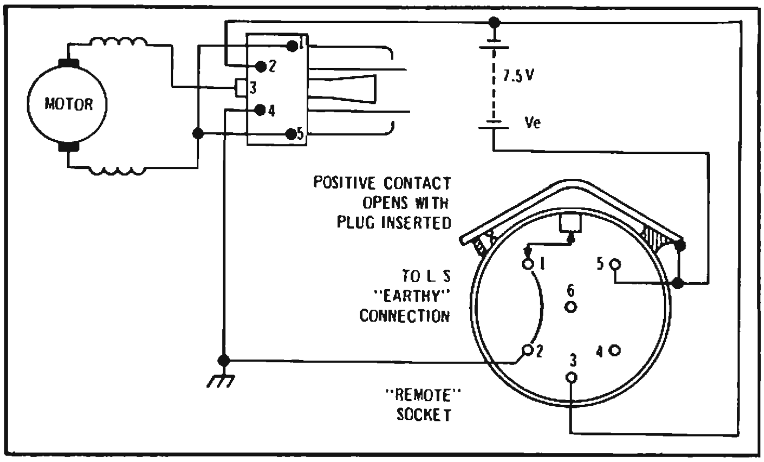


Fig. 10A—Supply and switching circuit of early Philips models.

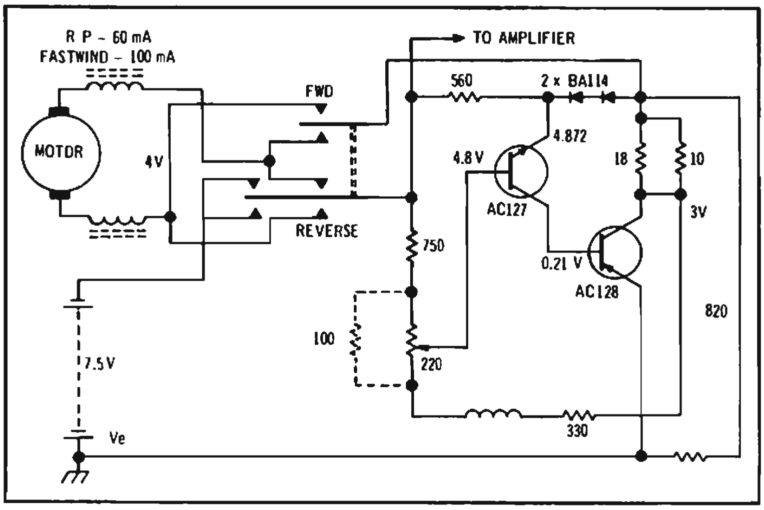


Fig. 10B—Supply, switching, and regulator circuit in later Philips-design cassette recorders.

Other parts of the clutch mechanism that need attention are the fast winding pulleys. A look at Figs. 3 and 4 shows how this is done above the deck, with a fairly large (greater circumference, higher speed) driving pulley between the two turntables. Below the deck, all is not so simple, and to aid explanation I have further stripped the basic Philips model, removing belt, flywheel and clutch wheel with its bracket (Fig. 9).

The secondary roller bearing above the deck, i.e., the drive pulley mentioned before, is driven from below by a small belt in the groove of the drive wheel nearest the camera in Fig. 9. This drive wheel engages the flywheel via its own rubber rim, and a fairly fast, direct drive is ensured, with enough of a safety slippage given by the belt coupling. Reversal is simply made by changing the polarity of the supply to the d.c. motor. The sneaky part of this assembly is that rod spring indicated by my screwdriver blade and rather obtrusive finger in Fig. 9. It should just touch the mount bushing in the FAST WINDING mode but be clear of the limiting bracket (part of the case-mounting assembly). During PLAY, however, the spring may touch this bracket but has to be clear of the mount bushing.

This whole operation is bound up with the movement of the main operating arm and the coupling to the leaf switch that can be seen in Fig. 9. There are several alternative versions of the switching, and care must be taken when replacing the switch that not only the right type is obtained, but also that the motor and supply circuits match up. With so many variants, it is easy to waste your money.

As a guide, no more than that, I have included a couple of the relevant drawings. These are neither schematic nor purely representative: They have been dreamed up in the past to put a couple of apprentices on the right road and may come in helpful for any readers who own cassette machines and are hesitant to tackle their maintenance. Figure 10a shows the original type of switch fitted in the Philips decks (the EL3300) with its wiring to motor and supply, and also the wiring of the 6-pin switched socket which is important for remote control and a.c. current operation. In Fig. 10b the later type of switch is shown and with it the simple regulator panel used to hold the motor to its correct voltage during RECORD and PLAY. More elegant regulators have been used by other firms, but some, such as Sony, eschew them altogether, preferring to use a more powerful motor and more strictly controlled mechanical design. ●

Next:—*Regulators*

TAPE RECORDER MAINTENANCE PART 8: Regulators

H. W. Hellyer

AT THE END of my last contribution (Jan., 1971) I gave a hint of things to come, while referring to methods of motor control. This applied primarily to portable cassette tape recorders, and one small circuit was given, that used by Philips in their EL3302A and similar models. Researching the subject for this continuation, and also for a lecture project I had to do at the request of my old friend Donald Aldous, (see London Letter, Page 10, AUDIO, Sept., 1970) I became convinced that the subject was worthy of a much deeper treatment than was possible as one article in this series. Our Editor (whom St. Cecilia preserve) is in apparent agreement, so I propose to go back to the beginnings and discuss the principles of regulation before leading on to some of the practices.

Reasons for this digression? They are twofold. In the first place, at the aforesaid lecture, when I took along a whole bunch of portable machines to describe and discuss with the lively South Devon Tape Recording Club, of which Aldous is the very active president, I was inundated with questions about the more sophisticated methods of servo control and voltage regulation that some of these tape recorders employ. In the second place, it was evident from some of the questions there and in my correspondence with readers of a number of audio magazines in this country (Great Britain) that understanding of the circuitry is incomplete. Power supplies are too often taken for granted. Either it goes or it doesn't.

Trouble is when it doesn't go, finding the reason is not always easy. It helps to have a knowledge of what a circuit should do before we can ferret out the causes for the stoppage.

Regulation—what do we mean? I would define this as "keeping the supply voltage constant even though the current drawn by the load varies." You may wish to apply a more elegant definition, or tie me down scientifically, but basically, this is what regulator circuits of tape recorders are designed to do. The bite comes with that last word "varies." The ordinary tape recorder has three functions: RECORD, PLAY, and FAST WIND. The first two may require much the same current, but usually the fast winding process demands more power, and the strain on the supply is more evident. Some machines need varying amounts of current for forward and reverse, even for differences between the amount of tape spooled. Simple regulation is not going to help them much when demands vary widely.

Alternative methods have motors which regulate themselves, and the supply line to the main machine can be separately and simply regulated. There are some interesting circuits in this group and we shall take a look at them later. Others employ what is now known as "servo control," where a sensing circuit picks up a mechanically generated pulse as the motor rotates, compares this with an electronically generated reference, and from the error reading feeds back to the motor a controlling change in supply voltage. Again, we have some intriguing variations.

The simplest form of regulation is the large capacitor across the battery supply. The function of this fellow is simply to bypass audio signals. It is a device to be found quite often in cheap radios as well as the more sophisticated gear we are talking about, and it does precisely the same job. In more technical terms, it reduces the apparent source impedance.

The source impedance of an ordinary layer-type battery could be a couple of ohms, and this will get bigger as the battery ages. At 100 Hz, a 1,000 μF capacitor will have an impedance only

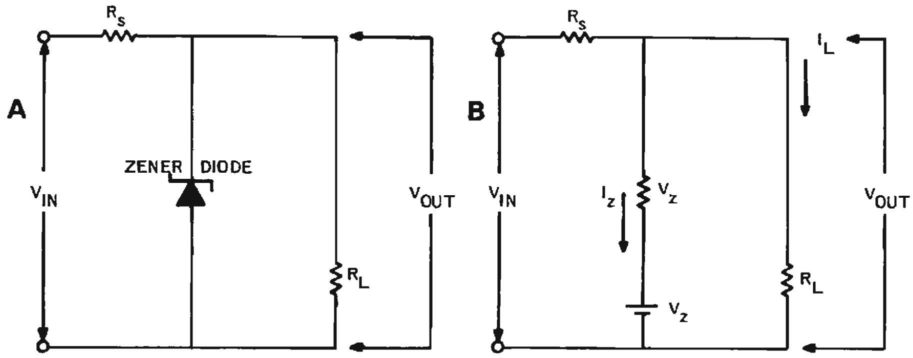


Fig. 1—Zener diode regulation, showing **A** the actual and **B** the effective circuit of the zener-controlled power supply.

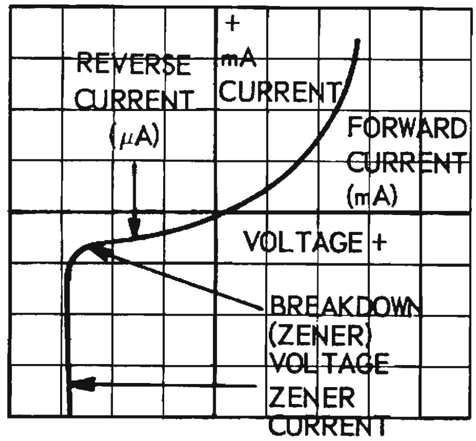


Fig. 2—Characteristic curve of a typical zener diode. Values are not marked on the axes of the graph, as these are different for various grades and specific values would be misleading. Note that reverse current is in microamps.

slightly greater than 1.5 ohms. This is in shunt with the battery impedance to alternating currents, so less voltage drop across the battery results. But if current variations are going to be greater, or, as with the portable tape recorders we are discussing, currents are direct, as used to drive the motor, then the regulating capacitor is only an added refinement.

Next simplest method of applying regulation will be found in devices where the original supply is from the a.c. electricity line. The idea is to "bleed" the power pack with a resistor across it drawing a fairly high current. Then, the current demands of the equipment, though still varying, present a proportionately less demand than when the bleeder and the power supply voltage remains more constant. But this is a brute force method. Neither of the foregoing methods needs more than our passing consideration and certainly should not require an explanatory diagram.

The zener diode across the supply line may seem little more than an elaboration on the bleeder idea. And, in fact, if we use it straightforwardly, as in Fig. 1a, it will have the same drawback—the power pack has to supply maximum current all the time. When our aim is conservancy—who wants to carry a mulepack of batteries?—we cannot use this. But if the machine is operating from an a.c. supply, and we are not too bothered about our bills, it could be used this way to act as a voltage clamp. It is more useful as a linear regulator, and we shall see it used in several of the circuits that follow, so let's take a look at the zener diode and see what it does and why.

If you forward bias a semiconductor diode, it behaves as a short-circuit—well, almost—and current flows. Reverse the applied voltage and only a tiny leakage current remains. This leakage current can be independent of the applied reverse voltage over quite a wide range, but as this voltage increases there comes a point when, zing! the current jumps from a few microamps to many milliamps—enough, indeed, to murder the diode.

This critical voltage is the breakdown voltage, V_{BR}. When C. Zener discovered it in 1934, it was just a curiosity. The "zener effect" had a few novelty descriptions, but had to wait until 1953 before K. G. McKay and K. B. McAffee published "Electron multiplication in silicon and germanium" in the *Physics Review*. An alternative theory for avalanche breakdown was proposed, and even if I understood it well enough, I would not bore you with the still arguable matter of broken covalent bonds and carrier velocities.

The important thing to cling to is that every diode will exhibit the zener effect, but not every diode will behave as we want our "zener diode" to behave, i.e., under control. Whereas, the zener diode acts as a perfectly normal diode within its "non-zener" area. They are generally designed to handle quite a hefty current for their size and, when not at breakdown, have a small leakage. When choosing zener diodes we have to look for voltage and current characteristics and to remember that the wattage rating is not voltage times current but a value slightly lower. This is because temperature comes into the calculations also. Heat sinks will be found associated with many zener diodes. Quoted specifications are for 25 degrees C. (77 degrees F.) unless otherwise stated. Voltage ranges are from around 3 volts to 150 volts or so and the power ratings cover a very wide range, up to 75 watts and more for special applications. Each diode has a correct zener voltage rating and tolerances may be 5, 10, or 15 percent. So the preferred range of values will be used, and a nominal 9 volt zener is actually 9.1 volts $\pm 5\%$, or whatever the tolerance of that particular one happens to be. Maximum zener current is equal to the quoted wattage divided by the zener voltage.

One phenomenon of the zener device is a negative temperature coefficient exhibited by those with a low breakdown voltage as against a positive temperature coefficient for those with a higher breakdown voltage. This can be used to good effect by the inclusion of two zeners in a circuit where their characteristics can balance out to give a zero temperature coefficient, at some specific current. A further speciality is the reduction of capacitance of the zener diode as the reverse voltage increases—a factor which comes handy in some television circuits.

A drawback that affects us in audio work is the rather high noise character-

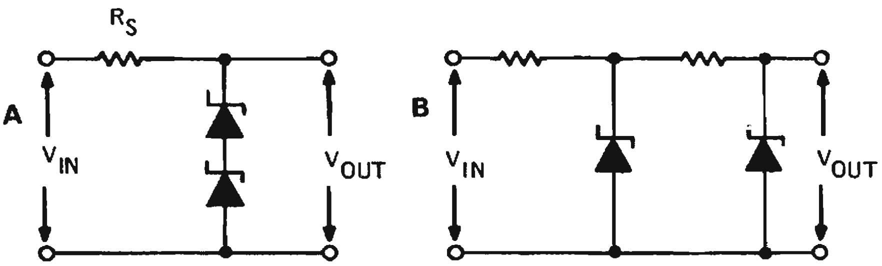


Fig. 3—Where voltage limits are beyond the range of a single zener diode, two or more can be combined, **A** in series and **B** in cascade.

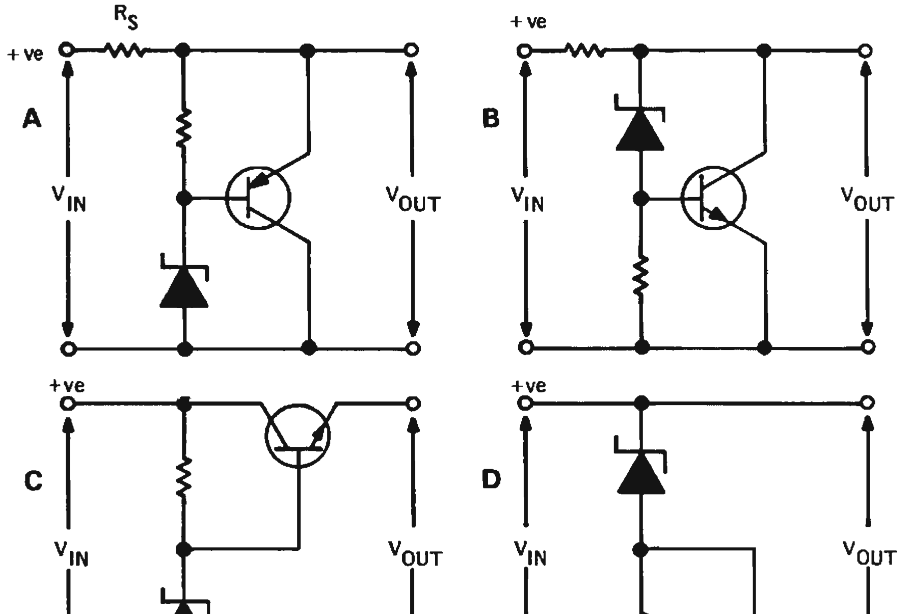


Fig. 4—Control via a bipolar transistor: **A** PNP in shunt; **B**, NPN in shunt; **C**, NPN in series, and **D**, PNP in series.

istic. This is the reason for that 0.01 μF or so capacitor shunting the diode in many circuits.

So much for what it is. How does it work? Well, a look at Fig. 2. shows the relationship of voltage and current for a particular zener diode, and here we see the characteristic hell-dive when the reverse voltage reaches the breakdown or zener point. In Fig. 1b there is a simple example of the circuit where these tendencies can be used. VIN is an unregulated input voltage. For the purpose of calculation we take the minimum VIN and say that Rs=(VIN-VOUT)+Imax. VOUT is the breakdown voltage of the zener and Imax the highest load current that will be demanded. So long as this load current is not exceeded (see Fig. 2.) VOUT will remain constant. The series resistor in a circuit like this will be chosen to give five to ten percent more than the specified load current, as a safety margin. As a further precaution, the relationship of VIN to VOUT will not be less than 1.5 to 1. By having a reasonable size of series resistor through which both load and diode currents will flow, the circuit is kept more stable. The equivalent circuit of this arrangement is shown in Fig. 1b. The dissipation rating of the diode is calculated from [VOUT (VINmax-VOUT)]+Rs.

Although the zener can be used primitively, like this, or stacked in series (Fig. 3a) or cascaded, as in Fig. 3b, a more realistic use is as a control device. Where the load current would be too great for a normal zener—or when the cost of a higher rated one forbids—the solution is to use a power transistor as the regulator and a zener diode to set its switching voltage. This can be connected as a shunt or a series regulator, and single-transistor circuits along these lines are shown in Fig. 4.

A shunt regulator (Fig. 4a) has a load current nearly as much as the product of the maximum rating of the zener and the gain of the transistor. If the load decreases and VOUT tends to rise, the base voltage of the transistor rises, relative to the emitter, and the zener in the circuit tends to keep the collector-to-base voltage constant. So the emitter current rises, the voltage across the series resistor goes up and VOUT remains reasonably constant. Only reasonably, for this circuit is still pretty Rube Goldberg, and its main virtue is that it is proof against short circuits. The output resistance is relatively high. But as with the previous circuit, full current is drawn from the power supply all the time, and a further drawback is that special precautions against temperature change have to be taken to prevent this affecting the output voltage. All the foregoing remarks apply as much to the PNP circuit as to the NPN circuit, Fig. 4b.

A more usual device is the series regulator, seen in its PNP and NPN versions in Figs. 4c and 4d. Here, the zener diode clamps the base relative to its appropriate line. Once more, a change in voltage because of demand is corrected by the transistor current varying, but now the current drain is proportional to the load current—or very nearly. So the efficiency of the circuit is greater. The disadvantage is that it is not proof against some Tom Fool short circuiting it, and even a fuse is not an adequate protection. The answer is a more sophisticated version of the same idea. Numerous circuits have been developed and we shall look at a few. These are mainly developments of common-collector high-gain Darlington pairs. The driver transistor provides most of the current gain and the main transistor takes care of the power demands.

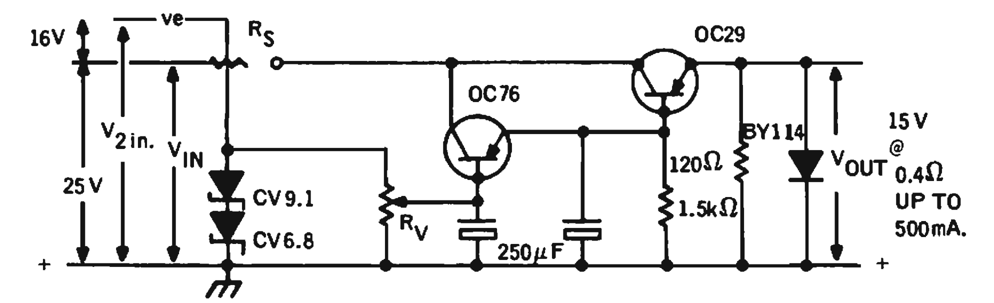


Fig. 5—A simple bench power supply circuit, with rudimentary control, effective over a limited range of values.

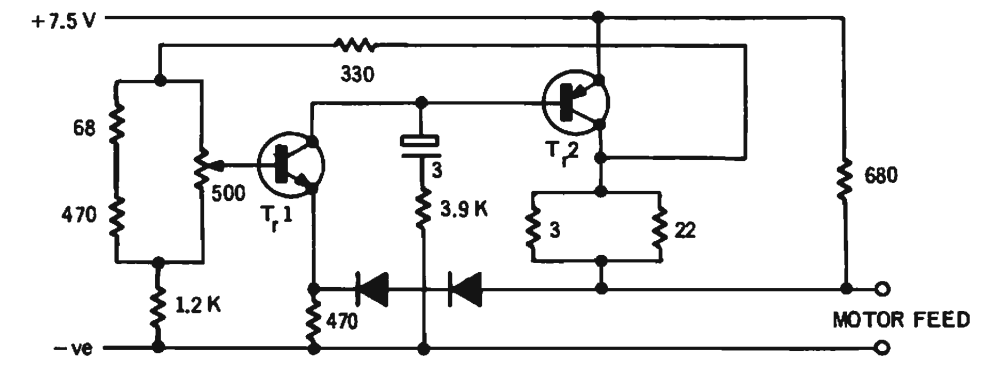


Fig. 6—A variant (by B. R. C. Ltd.) on the supply and motor regulation circuit of the Philips cassette portable shown in the Jan., 1971 issue of AUDIO, p. 28.

An introduction to practical tape recorder circuits is best made via the simple style of configuration used, not in a tape recorder, but in a bench supply unit. This little box of tricks was designed around a Mullard circuit and should give an output adjustable from zero to 15 volts at a load current of a half-amp. This is enough for some pre-amps, small radios and the like, and is chosen only to illustrate my points. The main supply is omitted to save space, and consists of a full-wave rectifier fed from a center-tapped transformer, with quite comprehensive filtering. The 16 volts to the zener diodes is thus negative to the main (zero) line. Part of the zener voltage is tapped off by the control potentiometer Rv and applied to the base of the driver transistor. The OC29 is in cascade so that the pair form a compound emitter follower. The output voltage at the emitter of the second transistor closely follows the base of the first, which is in turn controlled from the variable resistor, across which the zener diodes are connected to give a stable reference voltage.

Protection against short-circuits is limited, up to about 2 amps, by the value of Rs, shown dotted in Fig. 5. This has to be a high-wattage component (or combination of components), and an ohmic value of 7 ohms at 40 watts was used in the original design. As a protection against reverse voltages applied to the output terminals—easily done when testing sub-circuits in the electronic jungle—a reversed diode is fitted across the output. To keep the circuit stable, the series transistor has to be heat-sinked, allowing a thermal resistance of less than 2.5 degrees C/W up to 10 watts at low output voltages. The circuit is quite practical and has proved useful, but no claim for originality is made.

It is not always necessary to use zener diodes to set the base voltage of the

(Continued on page 53)

(Continued from page 24)

controlling transistor, as we saw from the circuit of the Philips design, at the end of the last article. By careful selection of the source voltage, i.e. its circuit derivation, the inherent flywheel action of regulation can be used to advantage. Figure 6 gives a British Radio Corp. modification to the Philips design shown in the Jan., 1971 issue of AUDIO. Note that here we have the control voltage for the base of the first transistor derived from a potentiometer network, with a preset variable, and the upper end of this chain is taken to the collector of Tr2. The circuit has two functions: voltage stabilization and motor regulation.

With everything normal, Tr2 base bias is determined by Tr1 collector current. This is set so that the correct voltage is fed to the motor while the effective forward resistance is lower than the 470 ohms in the emitter of Tr1. The two diodes, used here as temperature compensated devices and mounted near the heat-sink of Tr2, are forward biased to the constant voltage portion of their characteristics. A drop in supply voltage affects the emitter of Tr1 via the forward biased diodes. A smaller effect is felt at the base by the potentiometer chain so there is a net increase in forward bias. The collector current of Tr1 rises, driving Tr2 on harder and reducing its effective series resistance, which offsets the voltage drop in the supply.

As a motor governor, the circuit works the other way around, offsetting the change in potential across the motor which will be caused when a varying load affects the armature current. As the motor current rises, there will be an increased voltage drop across the paralleled pair of resistors (3 and 22 ohms). (Note that the 680 ohms resistor is here to give Tr1 its forward bias when the circuit is first switched on; without this, both transistors would remain cut-off.) Applied to the effective diode (base-emitter) of Tr1 via the two paths previously mentioned, it causes an increase again of forward bias, increases collector current, turns Tr2 on harder and feeds more voltage to the laboring motor. Variations of Tr2 base voltage around a mean value are ensured by the CR combination, the large value of the electrolytic helping to maintain the necessary average bias level.

Many more interesting circuits can be found, and in the next part we shall look at motor control and servo circuits in greater detail, using some of the information already gained and introducing one or two fresh concepts. Æ

SPLICING TAPES

ANDREW H. PERSOON*

THE ART OF SPLICING evolved very shortly after the first piece of recording tape broke. While seemingly unimportant, quite a bit goes into achieving the "ideal splice" —enough to assure cleaner recordings and better-functioning equipment when properly done.

A proper splice is one that will remain intact for an indefinate period of time. It must be mechanically strong, have a minimum of "adhesive escape" around the edges of the splice, and must in itself produce no audible disturbances.

While all these considerations are quite familiar to recordists, there are considerations from a manufacturing standpoint as well as some fundamental procedures which lead to correct splicing.

Tape Construction

Any pressure-sensitive tape takes two components into consideration in its design: Backing and adhesive coating. While there are many types of adhesive tape on the market, only one set of combinations is truly suitable for the splicing of magnetic recording tape.

The backing for a splicing tape has to be tough and durable while being as thin as possible. Paper is not suitable, so plastic is used instead. Both acetate and polyester backings are currently being produced for splicing tapes.

The adhesive presents other design problems. Three basic adhesive qualities must be considered: (1) Shear adhesion, (2) peel back or ASTM adhesion, and (3) wet grab.

Shear adhesion is the resistance of an adhesive to being parted from its adhered surface by a full force in what is called the shear direcion. (Fig. 1).

Peel-back (ASTM adhesion), is a measure of the adhesive's resistance to being peeled away from the surface to which it is adhered. (Fig. 2).

"Wet grab" is an elusive quality. It is more popularly referred to as "thumb appeal" or "quick stick." It is the quality of an adhesive to feel sticky. Oddly enough, it is not an important quality as far as the actual strength of the bond is concerned, but it is a quality readily noticeable by the user. Completely untrue is the statement that the stickier the tape feels, the better the splice.

A tape with high "wet grab" might improve peel back adhesion, but this apparent advantage may result in damage to the equipment and recording tape.

First of all, a sticky adhesive creates a bond on recording tape that produces excess "ooze" which may actually bond one layer of tape to another within the tape reel. The more pressure exerted on the splice, the more probable "ooze." The result might be the complete removal of oxide on the layer of tape above the splice on the reel, or worse, complete blocking of the reel.

Secondly, an increase in "wet grab" decreases shear strength. If this is reduced, the tightly butted ends of the recording tape—essential for an inaudible splice—could "creep," and may leave adhesive on the recorder's heads and tape guides—in addition to the problems already mentioned.

A properly designed splicing tape, then, does not feel "sticky" by intentional design—yet will produce a strong bond.

Splice Geometry

There are several variations in splice geometry, the right one depending on the conditions of use. Necessarily these considerations include the size of the spliced area and the angle at which tape ends meet each other.

The length of a splice depends on the amount of curvature it will have to sustain in its path from reel to reel, (Fig. 3). As a recording tape passes around the curved surface of a

Fig. 1—Shear adhesion (creep).

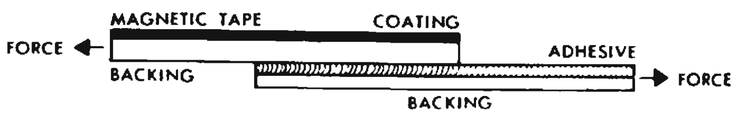
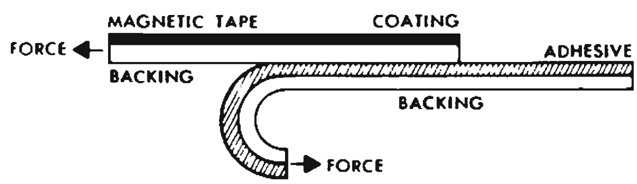


Fig. 2—ASTM adhesion (peel-back)



***Technical Director,**
Magnetic Products Div.,
3M Company.

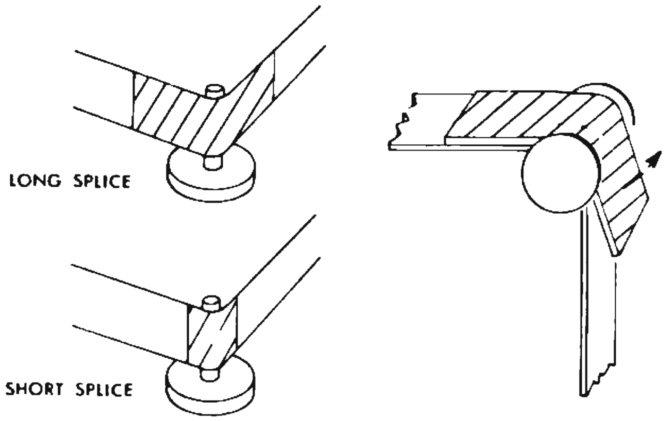


Fig. 3—Splice length and bend radius.

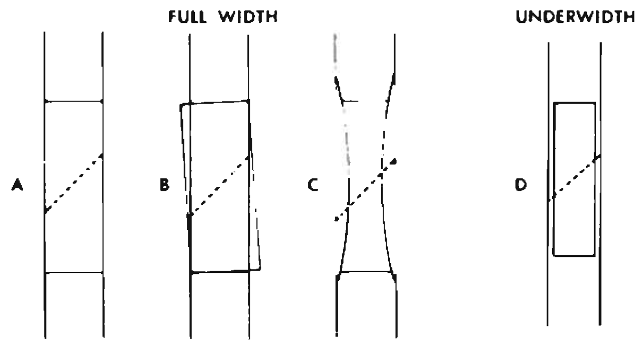


Fig. 4—Splicing widths: (A), full-width, properly done; (B), full-width, carelessly trimmed; (C), full-width, with an arc cut in the sides; and (D), under-width splice.

guide, as is shown at (A), there is a tendency for the leading edge of the splicing tape to continue in its original direction, as at (B). In effect, it is attempting to peel itself away from the recording tape, and is an example of one design parameter: Peelback (ASTM) adhesion. Here, the length of the splice has no effect on the tendency to peel, but is important for another reason.

A short splice, as at (C) Fig. 3 may tend to loosen if subjected to a tight bend because the area of peel may extend far enough into the tape's bond to free one end of the recording tape.

A longer splice solves this problem extending beyond the splice junction. Even if the splicing tape tends to peel at a guide, the junction remains undisturbed. Once the tape is wound again on a reel, pressure from succeeding wraps of recording tape secure the splice firmly by pressure.

Generally, this rule of thumb evolves: The smaller the radius of the bend expected, the longer the splice.

As we have mentioned, the tendency to "creep" is dependent on the shear strength of the splicing tape adhesive. The force that opposes this shear strength is, of course, the amount of tension the tape encounters on the transport and while wound on a reel during storage, (shear strength is constant for a given splicing tape). If subjected to a constant tension—for example a properly adjusted transport—the variable affecting "creep" is the area of the bond. The larger this area, the better the "creep" resistence.

Carrying this thought further, a splicing tape with a poor adhesive shear strength could be used if the area of the splice were increased. Limited by the width of recording tapes—¼-inch—the area could only be increased by additional length. A spliced bond of two or three feet would increase shear strength, but realistically would be impossible to execute mechanically. Since program material exhibits a drop in level of 2 to 4 dB in the area of the bond; the shorter the splice, the less disturbance during playback, a fact supporting the need for a splicing tape with high adhesive shear strength.

Splicing Tape Width

Should the splicing tape be the same or narrower in width than the recording tape? Several considerations affect this decision.

When using a full-width splicing tape, care must be taken not to overlap the edges of the recording tape being spliced. Adhesive may adhere to adjacent tape layers causing problems similar to those encountered with ooze. While splicing jigs are available which cut an arc into each side of the splice —to eliminate overhang—the possibility of adhesive "ooze" remains.

A narrow-width splicing tape offers a number of advantages with no apparent disadvantages: (1), Overlap is eliminated completely; (2), A simple splicing jig may be used, eliminating undercutting; and (3), Overall bond area is not materially affected.
(Figure 5 shows four examples of full- and under-width splices)

Splice Angles

Ideally, a recording tape to be spliced will be cleanly cut at an angle of 45 to 60°—measured with respect to the tape edge. As the angle increases above 60° toward the perpendicular, the amount of electrical disturbance is increased because the recorder head sees the discontinuity as an abrupt change—producing an annoying pop or click.

A shallower angle produces less disturbance, but as the angle is decreased below 45°, the pointed corners of the recording tape become extremely vulnerable to being peeled back or actually debonding.

Commercial jigs are available which produce the ideal 45° splice angle as well as affording a means for an even, clean tape cut.

Tape Handling

Cleanliness is probably the most important consideration in making a good splice. Hands especially should be free from oil or dirt as a single oily fingerprint can reduce output several dB. Also, contamination of the backing or adhesive can reduce the strength of the board—possibly inducing bond failure. The cut itself must be executed cleanly using a sharp *demagnetized* razor blade. When handling the pressure-sensitive splicing tape, care should be taken not to touch the adhesive more than necessary. After carefully aligning and splicing the tape, all air pockets should be removed with the fingernail to promote a secure bond. Æ

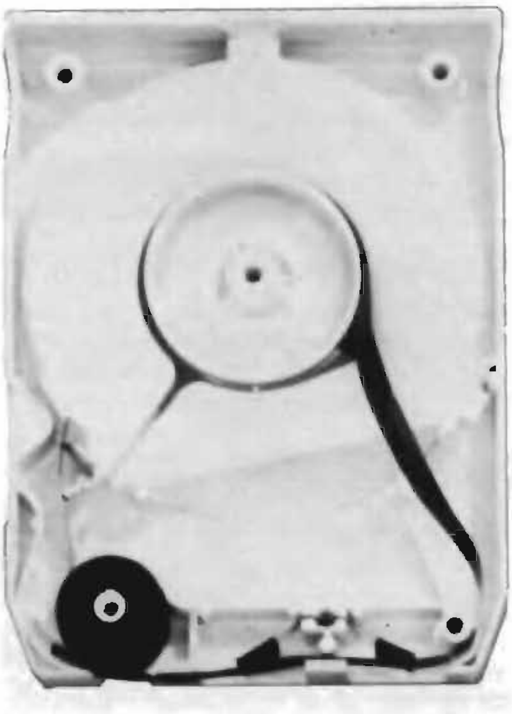
The 8-Track/Cassette Cold War Gets Warmer

Don Humphreys*

For several years now the eight-track-cartridge supporters and the cassette supporters have been playing down the existence of a conflict over which system will become the victor and which will be the vanquished in the consumer marketplace. Some observers feel that there will be no decisive winner, that each system will find its own niche, and that the two systems will co-exist as in the case of the 45-rpm disc versus the 33⅓-rpm disc. They further feel that this peaceful state of co-existence will also find room to tolerate the open reel-to-reel market, the four-track-cartridge system, and even the Playtape system. Other observers feel that one of the two (either eight-track or cassette) will bury all the other systems.

* Michigan Magnetics, Vermontville, Mich.

Fig. 1—Cutaway view of 8-track endless-loop cartridge.



As a neutral observer, we have noticed how each industry has been striving to add the features of the opponent's system to its own system. For instance: when the original battle lines were formed, eight-track featured: (1) Playback only; (2) Continuous play; (3) Rapid one-direction selection of four programs; (4) A formidable library of pre-recorded music from the top-of-the-line duplicator (record) giants. In contrast, cassettes featured: (1) Record/playback capabilities. (2) Fast forward and reverse search capabilities. (3) A more compact tape package.

The cassette people subsequently introduced: (1) Playback-only models to appease the recording/artists/copyright faction; (2) Continuous-loop winds in cassette cartridges; (3) Excellent libraries of pre-recorded tape releases to satisfy the dealers' requests for a continuing after market.

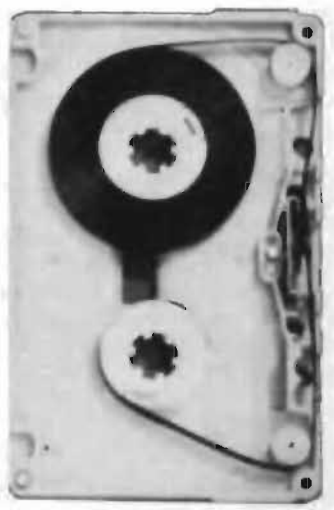
The eight-track people, not to be outdone, countered with: (1) Record/play models, (2) Smaller packages that matched the cassette in two dimensions (now they would only be thicker to accommodate the ¼" tape instead of the 0.150 tape).

The "Match Features" contest intensified as both systems displayed radios packaged into their cartridges that effectively converted the tape players into FM and AM receivers and the Audio Component people began showing the systems packaged with tuners, record changers, and portable transistor radios. One enterprising manufacturer even showed an eight-track cartridge that would accept and play a cassette.

Somewhere in the middle of all this, a "Starr-System" modification of the basic cassette system was introduced; and stack loaders and slot loaders were introduced by Phillips and several others. Shortly thereafter a company named Quatron entered the fray by introducing a carousel-type, eight-track unit that approximates the new stack-loader features of the opponent.

If you now compare the two systems, you can see that each industry has almost matched the other, feature for feature. Nevertheless, there still remained two salient differences: (1) Eight-track could not reverse or search (although several manufacturers are

Fig. 2—Cutaway view of enclosed reel-to-reel cassette.



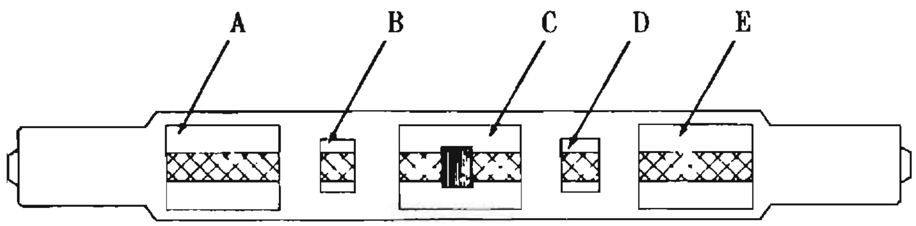


Fig. 3—Standard cassette openings are illustrated here. See text for references.

now promising this feature soon). (2) The continuous-loop cassette was not considered a bona fide solution to the continuous-play requirement.

For better than a year now, many cassette manufacturers have been promising automatic reversing machines to solve the continuous-play dilemma. In fact, at the last CES show in New York (June), Craig showed a playback-only version of an automatic-reverse unit; and Roberts showed a prototype of a unit that reached into the cassette, pulled out a loop of tape, and accomplished the bi-directional feat.

The record feature of the eight-track worked well, but potential users were frustrated in their recording attempts by the inability to reverse or pause conveniently during the recording process to eliminate commercials or to record over a selection no longer wanted on this program.

Eight-track remained the best continuous-playback medium, and cassette remained the best recording system.

A patent recently issued to Michigan Magnetics, and other patent applications in process, have now added the next chapter in the unfolding drama. The Michigan Magnetics "K-Set System" utilizes a shifting cassette head (similar to eight-track) that provides a solution to achieving a *bi-directional cassette* with full record and playback features.

The problem was a tricky one. As can be seen in Fig. 3, the standard cassette has only five openings. Openings A and E are taken up (in an automatic reversing deck) by pinch rollers, since one roller is needed for forward tape motion and another needed for reverse. For best time-base stability, the capstan and pinch roller should be downstream of the tape head to ensure adequate tape wrap and a pulling movement of tape instead of a pushing movement.

Openings B and D have no pressure pads behind the tape, so using a head in these holes was considered impractical.

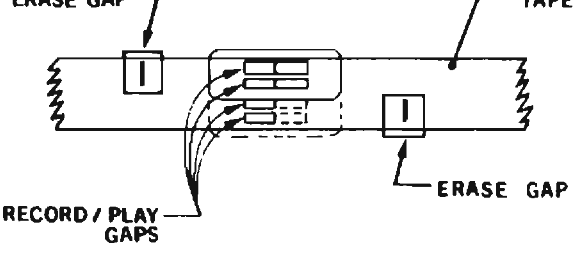
This leaves only opening C to accommodate three heads! Worse yet, the pressure pad in most cassettes is only 1/8" long. To use a four-channel combination head would require *six* gaps on a pad measuring 1/8" by 3/16". To date no one has been able to mass produce such a design. Note that a four-channel head would be needed to provide stereo pairs in both directions. This means up to six coils in a tiny head, with each coil acting on the others in a transformer fashion so that a signal in one would induce crosstalk in the others. A further complication arises when four channels replace two. In this instance, all coils must have less turns to make room for each other; and less turns mean less output. As the output of a head decreases, the signal-to-noise ratio of the system must suffer. If you are not yet impressed with the complicated nature of the beast, let me now add that a record/play head without an erase head upstream is almost useless. For effective recording technique, a system would have to have the six gaps positioned as shown in Fig. 4.

The Michigan Magnetics "K-Set System" proposes to solve the problem by using the proven eight-track shifting technique. Erase heads are inserted in openings B and D to erase the tape immediately before recording in either the forward- or reverse-drive direction. As indicated earlier, this would not be an ideal place to erase because there are no pressure pads behind the tape for good tape-to-head contact. Still true, but the difference is that this system uses a newly developed *ferrite* erase head that will take high-frequency (70 kHz) erase currents of sufficient amplitude to "bloom" well beyond the gap and erase previously saturated tape to a remarkably low level of −60 dB. The outboard erase heads have built-in precision tape guides that replace the fork type common on many heads.

Now we get into the heart of the system. Michigan Magnetics has announced that the shifting head (that will operate in opening C of Fig. 3 will have the same electrical specifications as its standard "K-Set" stereo head. This means no sacrifices in the 1-kHz sensitivity, or in the 1 to 10 kHz frequency-response ratio, or in the crosstalk specs. Perhaps even more important, it means a program-1-to-program-2 crosstalk level that is almost non-existent. One of the problems that confronted engineers considering the four-channel-head approach was the fact that every time the tape changed direction, the critical audiophile user would have to jump up and readjust his balance control since a new pair of head channels would be used. The Michigan Magnetics "K-Set System" obviously eliminates this problem, since only one pair of channels is used; and, once the balance is adjusted for one direction of tape travel, channel balance is the same in the opposite direction.

It will be interesting to see how the industry will accept the drop-in feature of the system. Æ

Fig. 4—For effective recording in a reversing cassette system, six gap positions are required.



Donald R. Hicke

BUILD YOUR OWN Tape eraser

Fig. 1

Fig. 2

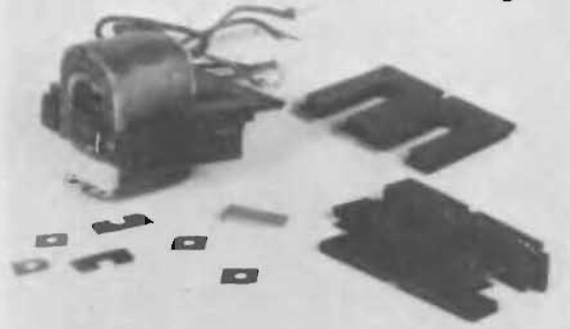


Fig. 3

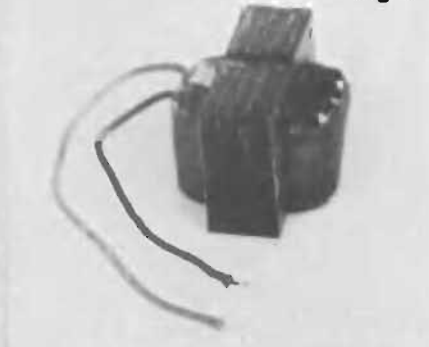
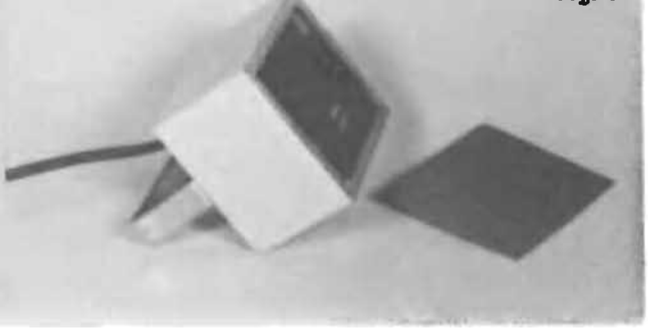


Fig. 4



THE NEXT best thing to using a fresh, new roll of tape for recording that important concert or other one-of-a-kind performances is to bulk erase an old reel of tape. This is especially important if the erase head on your recorder is not very efficient, or if you use high-output tape, which is difficult to erase. Otherwise, the previously-recorded material may be audible during the more quiet passages of the new recording.

A bulk tape eraser consists of a coil of wire wound around a laminated-iron frame open on one side. When the coil is connected to an a.c. source and held near a reel of tape, the magnetic circuit is completed through the iron oxide coating on the tape, effectively erasing any recorded material. Commercial tape erasers use a specially-designed coil, but you can make your own eraser using almost any old transformer. Good results have been obtained with filament transformers, audio output transformers, and power-supply chokes. All you have to do is take the transformer apart and rearrange the core pieces. The accompanying pictures will show you how.

To use the eraser, hold it directly against the reel of tape and turn on the power. I used a push-push switch in this version, but this is only a convenience item. You will hear and feel the tape vibrate at 60 Hz. Move the eraser all around the reel, then repeat on the other side. Slowly move the eraser several feet away from the tape before turning off the power. ●

Fig. 1—This transformer was salvaged from a vacuum-tube amplifier. Pry up and straighten the four tabs, and remove the frame. Pull out the wooden wedge(s), and remove the core pieces by prying them apart with a sturdy pocket knife. Do not cut into the core winding! **Fig. 2**—Reassemble the E-shaped pieces, facing them all the same way. Discard the I-shaped pieces. Clip off and discard any extra-length tabs on the two outside core pieces. **Fig. 3**—Reinsert the wedge(s), and clip off all wires except the primaries (plate-to-plate for an audio output transformer). **Fig. 4**—Pass the a.c. connection cord through a rubber strain-relief grommet and then through a hole in a suitable box. Connect the a.c. cord to the coil, making certain each connection is well taped and the whole is well insulated. Locate the grommet, and mount the coil. Cover the bottom of the box with cardboard. Fasten a handle with screws, and paint the box.

THE EFFECTIVENESS OF BULK ERASING

HOWARD A. ROBERSON

As many recordists know, we've had a long history of hearty recommendations to use bulk erasing, whenever possible, to ensure the lowest possible tape noise. Because it's been the thing to do for the past 30 years, it may be perfectly reasonable to believe that bulk erasing makes sense today. However, after seeing a number of very strong advertising claims and hearing several tales of woe from consumers, I thought the time was right to examine the performance of bulk erasers.

A bulk eraser, or degausser, is basically an accessory device that radiates a strong magnetic field onto a tape in order to erase it. How much erasure occurs depends on the strength of the magnetic field and on the tape's level of coercivity. The higher the coercivity, the stronger the demagnetizing field required for full erasure—whether that field comes from a recorder's erase head or from a separate bulk eraser.

Much of the higher performance of today's tapes comes from their higher coercivities. High-coercivity tapes require more bias but are also more resistant to self-erasure. Type I tapes have coercivities from around 300 Oe to over 500 Oe for some premium formulations. The Type II coercivities overlap those of the normal-bias tapes, ranging from less than 450 to at least 700 Oe for the recent Type II metal-particle tapes. Type IV formulations have coercivities of 1,000 Oe or more. This results in much greater resistance to erasure, accidental or intended.

I should point out that the stated coercivity for a given tape formulation is the average of all its particles. The tape actually includes particles with higher and lower coercivities, the occurrence of both types roughly following a normal distribution curve. This means that a tape with a 500-Oe rating would probably have some particles with 700-Oe coercivity, and *those* particles would establish the erasing-field strength needed for complete erasure. (I refer readers wanting further explanation to "The Mechanism of Magnetic Tape Erasure" by Peter Vogelgesang in the April 1981 issue of *Audio*.)

It is apparent that the erasing field required for metal-particle tapes is *much* stronger than the field required for many older tape formulations. In fact, with the continuing increase in the coercivity of tapes in general, I had to be skeptical, as I began these tests, at least of the older bulk erasers that had not been designed for such challenges. I noticed on one figure from Nortronics (now The Geneva Group) that the drop in field strength for several bulk erasers is 40% or more for *each*

When a tape is rotated through an eraser's field rather than run straight through, demagnetization is made easier.

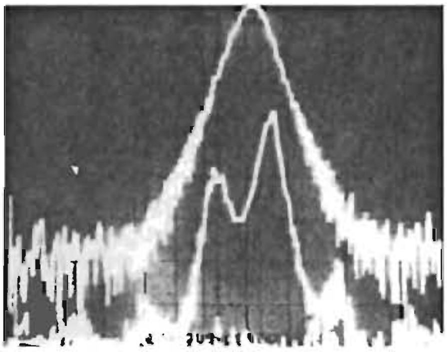


Fig. 1—Comparison of bulk and deck erasure on Maxell XLII-S (Type II) tape. Top: Spectrum analysis of 1-kHz tone recorded at 250 nWb/m. Middle: Same tone after degaussing by Tascam E-2A bulk eraser. Bottom: Same tone after erasure by Nakamichi 582 deck (see text). Scales: Vertical, 10 dB/div.; horizontal, 20 Hz/div.

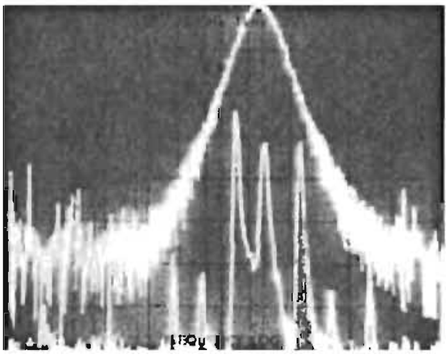


Fig. 2—Spectrum analysis of 1-kHz tone recorded at 250 nWb/m (top) and same tone after degaussing by Lafayette ML-120 bulk eraser (bottom). Scales: Same as Fig. 1.

doubling of distance. (The exact drop will depend on the shape of each eraser's field.) This fact of magnetic life requires that the eraser and the tape be in as intimate contact as possible, and perhaps that the tape be turned over to ensure magnetic saturation on each side.

Measurements

I have an old Lafayette ML-120 bench-type bulk eraser, and I really expected that it would suffer in comparison to newly designed units. I ran tests on it and on the following bulk erasers: The Radio Shack No. 44-233, a hand-held unit which the catalog says can be used for both audio and videotapes; the hand-held Geneva PF-211 video/audio eraser; the bench-type PF-250 Professional eraser from Geneva, and the bench-type Tascam E-2A. Both the Geneva PF-250 and Tascam E-2A have spindles for reel center-holes and adaptors for 10½-inch reels; the PF-250 also has a guide rail for audio and videocassettes.

I don't remember if I ever had any information on the magnetic field generated by my old Lafayette unit. There was no such information supplied with the Radio Shack or Tascam bulk erasers. On the other hand, the Geneva literature lists the surface flux intensity as 2,300 gauss for the PF-211 and 3,000 gauss for the PF-250. The manual with the PF-250 has figures of field versus distance and erasure depth versus signal wavelength. One table in this manual lists the flux intensities required to erase particular tape formulations. Another shows the maximum distances the field would need to reach, either in a single pass or in separate passes on each side, in order to erase common formats ranging from cassette up to 1-inch reels and half-inch videocassettes. The manual also points out that the erasing field in gauss should be 1.5 times the tape's coercivity, all the way out to the maximum distance required for erasure (at least halfway through the tape); this information is reflected in the field-distance table. Obviously, wide tapes may have to be turned over to achieve acceptable erasure. In general, the eraser must be brought as close to the tape as possible; other techniques will be discussed later.

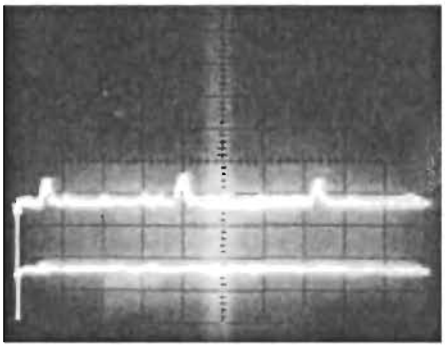


Fig. 3—Amplitude vs. time for playback of 400-Hz, 250-nWb/m tone on Nakamichi EXII (Type I) cassette after erasure by Lafayette ML-120 (top trace) and by Nakamichi CR-7A deck (bottom trace). Scales: Vertical, 10 dB/div.; horizontal, 1 S/div.

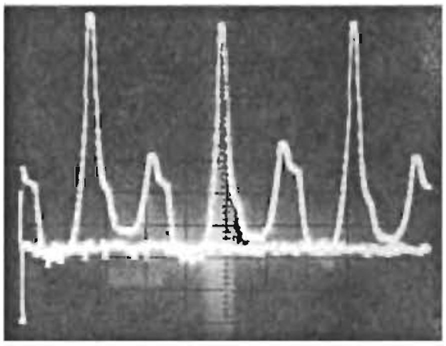


Fig. 4—Same as Fig. 3 but using Nakamichi SX (Type II) cassette.

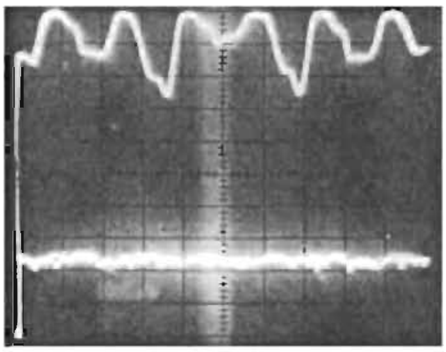


Fig. 5—Same as Fig. 3 but using Nakamichi ZX (Type IV) cassette.

For my first erasure test, I used Maxell XLII-S, a Type II cassette tape, on which I had recorded a 1-kHz tone at a level of 250 nWb/m using a Nakamichi 582 deck. A spectrum analyzer scan (the top trace in Fig. 1) was made of the tone in playback. Then the tape was erased on the Tascam E-2A by passing it through the field, flipping it over, and passing it through again. The middle trace of Fig. 1 shows the degree of erasure, which was rather poor—just 20 dB at one point. After erasure by the 582 deck (bottom trace), the analyzer showed no definite signal, just noise spikes 80 dB down. Figure 2 shows another scan of the 1-kHz tone as recorded (top) and after erasure by the Lafayette ML-120 unit (bottom). The Radio Shack 44-233 did not do as well as either the Lafayette or the Tascam bulk erasers, so it was dropped from the test program. Additional checks with cassette tapes showed that the much older Lafayette was at least as good as the new Tascam E-2A, which was then put aside for the time being.

I realized that the analyzer frequency-scan method was not the best for checking the level of erasure, so I switched to a zero-scan sweep, measuring amplitude over time at the test-signal frequency rather than measuring amplitude versus frequency. I also shifted to a 400-Hz test tone to make degaussing a little more difficult. Figure 3 shows the erasure achieved with a Type I tape (Nakamichi EXII) at 400 Hz for the Lafayette and for another Nakamichi deck, the CR-7A, which proved to be more convenient for these tests. The reference level is at the top of the 'scope's graticle, and the Lafayette trace was raised two divisions for clarity. (The vertical scale is 10 dB/division.) The actual erasure by the Lafayette, therefore, was about 82 dB (close to the noise limit of the analyzer), with the exception of the momentary peaks (more on these later). The erasure by the deck was at least 83 dB; that's about the noise limit of the analyzer.

Figure 4 shows what happened when I tried erasing another tape, the Type II Nakamichi SX, on the Lafayette unit and the CR-7A deck. In playback, there was a strong cyclical variation in the erasure by the Lafayette, with very brief periods of full erasure alternating with periods in which erasure fell to a mere 5 dB. Erasure by the deck, in contrast, was about 77 dB. Figure 5 shows that a metal-particle tape, Nakamichi ZX, almost completely resisted the erasing field of the Lafayette, while the deck managed an erasure of close to 77 dB.

Figure 6 shows the results obtained with the hand-held Geneva PF-211 bulk eraser on Nakamichi EXII, SX, and ZX tapes. A worthwhile improvement in erasure over the old Lafayette is immediately obvious. The residual signal on EXII (Type I) is buried in the noise floor, from which a few residual peaks emerge from the erased SX (Type II). The metal tape, ZX, remained untamed, with 50-dB erasure at best.

The basic test was repeated (Fig. 7) using the Geneva PF-250 bench-type eraser, this time comparing just the SX and ZX formulations. Surprisingly, the PF-250 erased the metal tape more completely than the hand-held PF-211 did, but its erasure of the Type II tape was poorer.

At this time, I considered the fact that bench-type bulk erasers come with spindles on which reels of tape can be easily rotated. When the tape is rotated through the eraser's field, particles from all points around the reel are oriented, at least part of the time, for easy demagnetization. By contrast, when a cassette tape pack is passed straight through a bulk eraser's field, there is a 90° variation in the orientation of the tape particles to the field. Some particles are oriented in the same direction as the field, and others are at angles to it. Figure 8 shows what happened when I rotated the SX and ZX tape packs in the center of the Geneva PF-250's top plate, rather than sliding them along the unit's cassette guide rail. The SX erasure was down into the

Among the bulk erasers examined were Geneva's PF-250, Teac's Tascam E-2A, and Geneva's PF-211.

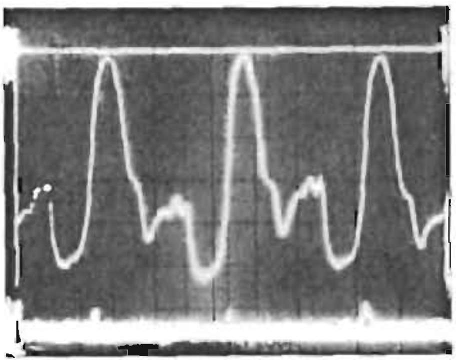


Fig. 6—Erasure of 400-Hz, 250-nWb/m tone on three tape types, rotating Geneva PF-211 hand-held bulk eraser. Top trace: Tone before erasure. Middle trace: Nakamichi ZX after erasure. Bottom: Nakamichi SX and EXII, overlapped (see text). Scales: Same as Fig. 3.

No bulk eraser I tried did a good job of erasing metal-particle tapes; cassette decks are much more effective at this.

noise, and the ZX erasure improved to 60 dB or better.

With the hand-held PF-211, it had been natural to rotate the cassette or to do a circular scan of the tape pack; the benefits of this can be seen by comparing the lower traces of Fig. 6 (the PF-211, rotated) with those of Fig. 7 (the PF-250, straight pass). Without some form of scanning, the degree of degaussing varies around the circumference of the tape pack, and the residual noise varies cyclically in playback.

Erasure varies with frequency too. Frequencies lower than 1 kHz or 400 Hz are harder to erase, and so I recorded pink noise on the ZX metal tape to permit seeing erasure across the band and its variation with time. The hand-held PF-211 was scanned around the cassette in a circle. Figure 9 shows that good erasure of this Type IV tape was not possible with the PF-211. The storage scope used here allowed me to record the entire range of erasure over time for all frequencies in the audio band. The best minimum erasure was 43 dB at 10 kHz, and the best maximum erasure was 54 dB at 5 kHz—really unacceptable performance at any frequency.

Figure 10A is a simplified representation of a bulk eraser with a spindle for use with open-reel tapes. The poles which supply the erasing flux are oriented so that the lines of flux lie in the plane of (are tangent to) the magnetic particles in the tape's layers. Because of the original alignment of the particles during manufacture, this relationship makes erasure easier. When a cassette is passed straight through these same pole pieces, however (Fig. 10B), most of the tape pack is *not* in good alignment for erasure. At two points, the orientation is very poor for effective degaussing. These poorly erased points are opposite each other in the tape pack, and in playback, the residual spikes might be heard every half-revolution of the pack. Many of the previous figures had the cyclic pattern discussed here, and Fig. 8 showed the great improvement in erasure that is possible with rotation of the cassette pack in the degaussing field. Experiments by R. E. Fayling, of 3M, have revealed that degaussing a tape the hard way, with the tape at right angles to the lines of flux, requires a degaussing field of 2.5 times the tape's coercivity, a field 67% stronger than that needed for degaussing it along the easy axis.

Figure 11 shows the erasure of a 1-kHz tone from a Maxell XL I open-reel tape by the Lafayette and Tascam units. Note that the erasure of this relatively high-frequency tone is a marginal 60 dB and that, with this fairly new tape, the Tascam unit shows no advantage over the old Lafayette. Figure 12 presents the different story resulting when the Geneva PF-211 and PF-250 units were used: Erasure of at least 77 dB was achieved. With the PF-211, I had made a careful, slow spiral scan from the outer edge in to the hub.

TDK SA open-reel tape, an EE type, was used for the final challenge. The Revox A77 recorder I used could erase the 1-kHz tone only 50 dB, although it had achieved at least 77 dB with Maxell XL I. I tried to do a speedy but careful scan with the hand-held PF-211. Figure 13 demonstrates that my spiral was not very smooth or all-covering. The bench-type PF-250 was superior in this test, with complete erasure of over 75 dB (noise limited).

Degaussing Videocassettes

Considering the growing use of Hi-Fi VCRs for audio and the claims of the eraser manufacturers for excellence in degaussing, I also tried the Lafayette and the two Geneva bulk erasers on videocassettes. I made a simple assessment of the erasure of picture and sound in normal (Beta III) mode, using high-grade formulations from BASF, PDMagnetics, Scotch, and TDK. Generally speaking, there was little differ-

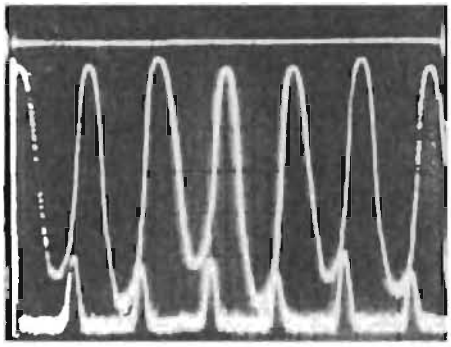


Fig. 7—Erasure of 400-Hz, 250-nWb/m tone on Nakamichi SX and ZX tapes, using Geneva PF-250 bench-type bulk eraser and with tapes moved along its guide rail. Top trace: Tone before erasure. Middle: ZX after erasure. Bottom: SX after erasure (see text). Scales: Same as Fig. 3.

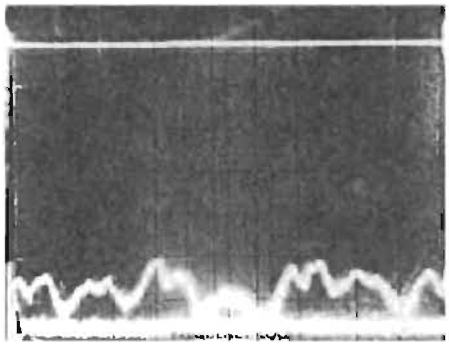


Fig. 8—Same as Fig. 7 but with tapes rotated in magnetic field (see text).

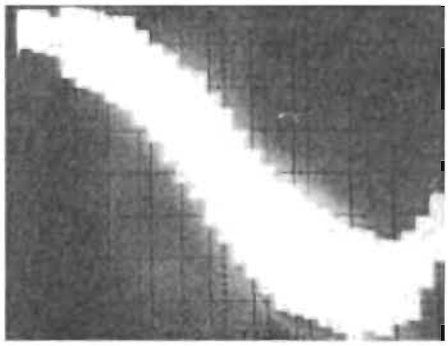


Fig. 9—Erasure of pink noise recorded on ZX tape with Dolby C NR, using Geneva PF-211 bulk eraser. Range of erasure over time in each third-octave band is shown by vertical spread of trace, with maximum erasure at bottom of trace. Vertical scale: 5 dB/div., with −15 dB at top of screen.

ence in erasure from one brand to another. In all cases, the degaussing field was applied to both sides of the videocassette.

The Lafayette was just fair at video erasure and was poor at erasing the sound. When the hand-held Geneva PF-211 was used for a spiral scan over both sides of each tape pack, the erasure was excellent for both video and audio on all formulations.

With the PF-250, I slid the tapes along the guide rail and got good to excellent results on the video and fair to excellent on the audio. When I slid the tapes more through the center of the degaussing area (slightly away from the rail), erasure of both picture and sound slightly improved. The poorly erased portions were cyclic and were time-related to position in the tape pack. I tried rotating the videocassettes on top of the PF-250 but had trouble doing it smoothly. The results were erratic; sometimes the erasure was poor for both video and audio.

The hand-held Geneva PF-211, with a good spiral scan, was able to get a higher effective degaussing field to all sections of the videocassette tape pack than the bench-type PF-250, whether I slid the tape straight through the PF-250's field or rotated the tape pack. In other words, it was less difficult to get easy-axis erasure with the PF-211.

Fig. 10A—Bulk erasure of open-reel tape. For best results, tape reel should be rotated on spindle as it is erased.

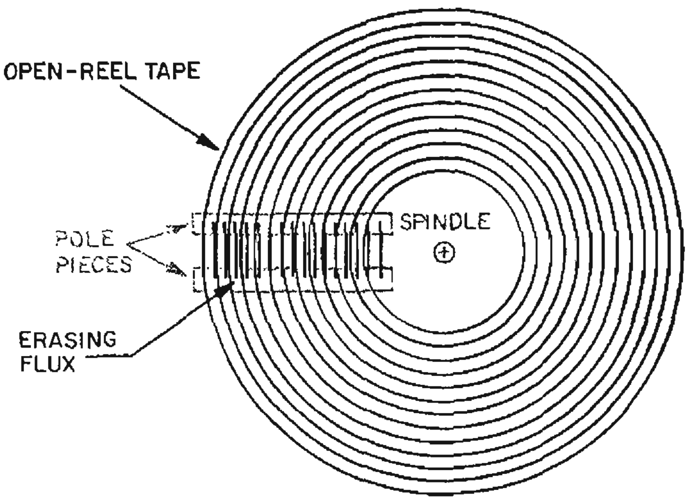
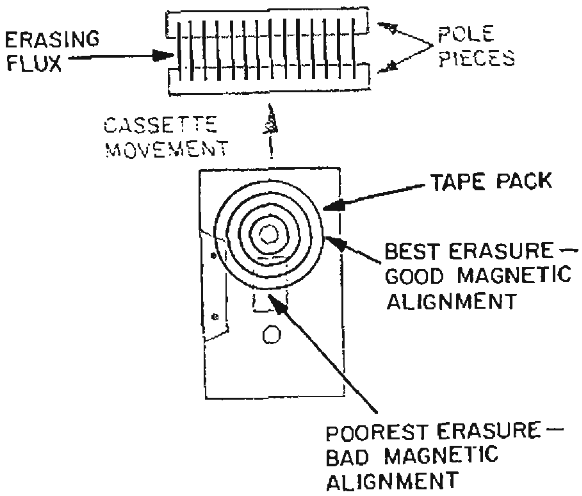


Fig. 10B—Cause of uneven erasure when cassettes are passed straight through degaussing field.



Conclusions and Recommendations

The conclusions I reached after all these experiments emphasize the limitations of bulk erasing (and the limitations of some older tape decks).

Most old and many new bulk erasers cannot adequately erase Type II cassettes, open-reel tapes of the EE type or in the same coercivity range as Maxell XL I, or high-grade videocassettes. No bulk eraser I tried did a good job in degaussing metal-particle cassette tape. Claims by *any* manufacturer that an inexpensive hand-held unit will erase "all audio tapes" should be rejected—especially as regards metal tapes.

Older open-reel decks, even if they can record on EE tape, might not be able to erase what they have recorded. Cassette decks are much more successful in degaussing Type IV tapes than even expensive bulk erasers. In general, but particularly for Type II and IV cassettes, erasure is best left to the recordist's deck.

Bulk erasing a tape before recording on it does not seem to improve the S/N of the recording. No matter whether the deck's erase head or the bulk eraser does a better job of degaussing the tape, the residual noise level on a rerecorded tape will stay about the same.

Of the erasers I tested, the hand-held Geneva PF-211 was the best overall, for both audio and videotape. Its price of $54.95 was judged very reasonable for what it can do. Geneva's bench-type PF-250 was the best of the group with EE-type open-reel tapes. Its $400+ price would be a very stiff one for most recordists, but it could be a good investment for someone who does a great deal of taping, particularly on open-reel tapes. Most of the other bulk erasers I tried would be completely acceptable for degaussing lower coercivity open-reel and cassette tapes, but none of the others earn a general recommendation for degaussing of magnetic media.

If you already have a bulk eraser, you should review the technique you use for erasing tapes. For open-reel

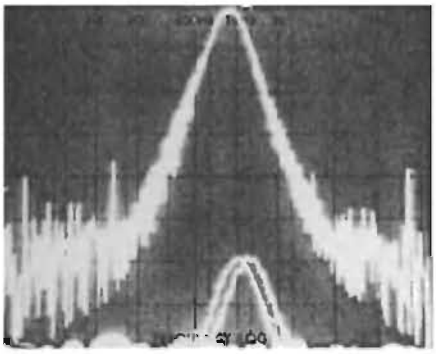


Fig. 11—Erasure of Maxell XL I open-reel tape. Top: Spectrum analysis of 1-kHz tone recorded at 200 nWb/m. Bottom: Same tone after degaussing by Tascam E-2A and Lafayette ML-120 erasers (traces partly overlap; see text). Scales: Vertical, 10 dB/div.; horizontal, 20 Hz/div.

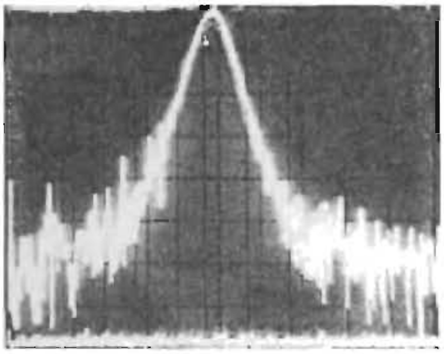


Fig. 12—Same as Fig. 11 but using Geneva PF-211 and PF-250 for bottom (overlapping) traces.

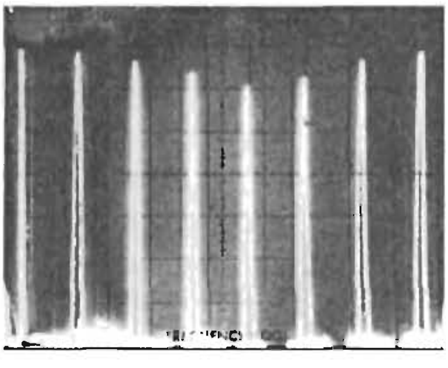


Fig. 13—Amplitude vs. time for erasure of 1-kHz, 200-nWb/m tone on EE open-reel tape (TDK SA), using Geneva PF-211. Note unerased sections, caused by failure to scan the tape properly (see text). Scales: Vertical, 10 dB/div.; horizontal, 2 S/div.

tapes, bench-type units with spindles will produce the most even demagnetizing fields, as long as rotation is smooth and slow.

It is also essential that the tape be 2 to 3 feet away from the eraser before you turn the eraser off! Otherwise, if the magnetic field collapse at switch-off occurs at the peak of the line-voltage waveform, you risk leaving a saturated signal on the tape.

When using a bulk eraser for cassette tapes, the best procedure is to wind all of the tape to one end, place the cassette so that its tape pack is in the center of the erasing area, rotate the tape pack at least one complete revolution, and remove the tape from the degausser before switching it off. If the tape is a Type II, repeat the process for the second side.

When operating a hand-held bulk eraser, scan the tape pack, following a slow spiral from one edge of the pack to the other, while maintaining intimate contact with the reel or the cassette shell. Repeat on the second side for higher coercivity tapes. If the hand-held unit is grasped with its erasing poles up, the cassette tape pack can be rested on the flat surface and the tape pack can be rotated. It might even be possible to make a simple fixture to help rotate reels in similar fashion.

With any procedure, of course, it is essential that the maximum field reach all parts of the tape. Keep the eraser as close to the tape as possible. (I've found people tremendously reducing the effectiveness of their erasers by trying to degauss cassettes which were still in their boxes!) To check your erasing technique, record a low-frequency tone (100 to 400 Hz) at a high level (+3 on your deck's meter). Listen to the playback after erasure. If the tone is low in level but steady, your technique is good, but the degausser does not have a strong enough field for that formulation. If the tone varies greatly with high and low levels during playback, the magnetic scanning was not smooth around the tape pack. It is also likely that the erasing field of a hand-held unit will have a particular orientation relative to the handle position: Erasure might be good when the handle is tangent to the tape pack and poor when the handle is perpendicular, or vice versa. To be on the safe side, scan the tape pack twice, holding the eraser the same way each time but rotating the cassette or reel 90° between scans.

If you get marginal degaussing with your present eraser, it is quite possible that the problems will disappear if you use one of the models recommended above.

If you don't have a bulk eraser, you may see little reason to get one now. A number of them, hand-held or bench-type, will not completely degauss many of the tape formulations in current use. It is definitely *not* true that by some magical process bulk erasing will "reduce wear on your recorder," as one exuberant vendor claimed, unless you regularly run tape through the recorder just to erase it.

While it might seem, from all of the above, that there are no reasons to bulk-erase, it offers advantages on some occasions. If you wish to record several individual selections on a used tape, bulk erasure will ensure that old bits of music will not pop through where you paused or stopped recording between the new selections. It is also possible, especially with 4-track/2-channel open-reel decks and also with the semi-pro 4-track/4-channel decks, that recordings made on one machine cannot be erased on another because of head-height discrepancies. Good bulk erasing will allow use of the tape without requiring that the erase head of the second deck match the recorded tracks of the first deck. And users of some professional tape-duplication decks which lack erase heads will find bulk erasing a necessity if tapes are to be recycled.

If you decide to get a bulk eraser, make certain that you can return it if it fails to meet your needs. Follow the recommended procedure given earlier for evaluating degausser performance, trying the higher performance tapes you expect to use. My own investigation showed that most advertising claims are not met and that few manufacturers provide technical data upon which to judge performance. Product information that gives field strength and lists specific media that can be erased, such as is available from Geneva, will help to provide a basis for judging all degaussers. A

110 dB Dynamic Range For Tape

Richard S. Burwen*

It is now widely recognized that the signal-to-noise ratio of even the best studio tape recorders is insufficient for much of the recording performed today. If a piano played fortissimo is recorded on four-track ½" tape with peaks reaching the 2% distortion level and then played back at the original acoustic level, the hiss is not only audible but objectionable. With other types of program material noise becomes objectionable after the several rerecordings common to the preparation of a prerecorded tape.

For this reason studios are turning to electronic methods for further processing the signal to reduce the noise. One system available is the Dynamic Noise Filter [1] manufactured by Burwen Laboratories. This system reduces noise by attenuating the high and low frequencies in a band-pass filter whose bandwidth is extended with each musical note and contracted between the notes. Due to the characteristics of the ear the noise that is present during each note is masked by the music.

Another signal processor available is the Dolby Laboratories Noise Reduction System [2] which is used in making new tape recordings but is not designed to improve existing noisy program material. The Dolby System compresses the signal in four separate frequency bands before recording in order to maintain a higher signal level on the tape in each band and thereby overcome the noise. A complementary four band expander used for playback restores the signal to its original level and flat response and reduces the noise between notes because lower level signals are played at reduced gain. A simplified high band version of the Dolby System is now coming into use in consumer equipment and prerecorded cassettes.

A new system to be described, the Burwen Laboratories Noise Eliminator, Fig. 1, is also based on the principle of signal compression before recording and expansion after recording. Whereas the Dolby Type "A" System reduces noise from 10 to 15 dB, the Burwen Laboratories Noise Eliminator reduces noise as much as 50 to 60 dB.

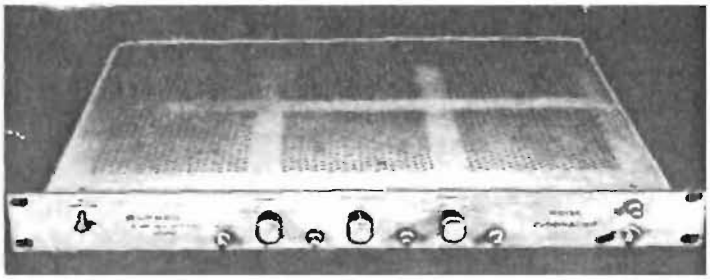


Fig. 1—Two-channel Noise Eliminator.

Effect of 50-60 dB Noise Reduction

With such a tremendous increase in the dynamic range the recording engineer gains a new freedom. No longer does the signal level have to be set so the VU meter reaches precisely into the red region and not beyond. An A-B comparison of the source and playback signals in headphones when recording a piano shows no audible difference. In fact the input level can be reduced 30 dB while the output is magnified 30 dB and the audible noise is less than that in a normal recording. When dubbing from a conventional tape a reduction in input level as much as 40 dB still produces a good dub. The greatest benefit, of course, is in making recordings at proper input levels in which case tape noise is audibly eliminated. Swishing noises are not apparent in spite of using a single wideband compressor because the instantaneous signal-to-noise ratio is very high for a wide range of input levels.

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All of this is predicated upon an input signal coming from noiseless studio equipment. In practice, of course, the extremely wide dynamic range of the tape recording system will simply uncover the noise in the studio console that was previously negligible. One possible solution when making multi-track recordings is to record directly from a high quality capacitor microphone preamplifier which can have a dynamic range in excess of 110 dB.

Compatibility

One facet of the Noise Eliminator system being investigated at the present time is the possibility of compressing the signal prior to broadcasting FM or AM or making reel to reel tapes, cassettes, and phonograph records. For highest quality reproduction of a compressed signal the audio hobbyist would have to own at least a simplified version of the playback expander. For those who do not own an expander listening to the compressed signal can still be quite pleasant.

Unlike the Dolby System which changes the frequency response of the compressed signal in accordance with its level, the frequency response of the Burwen Laboratories system is constant. Although the signal is greatly compressed and the VU meter hovers around zero, the psychological dynamic effect of the music is to a considerable extent restored by a moderate amount of low frequency and high frequency pre-emphasis included in the record signal processing. As background music the compressed signal sounds quite pleasant because even very low level passages can be heard.

The principal disadvantage in listening to the compressed signal is the substantial increase in the studio console amplifier noise that can be heard, particularly if the music stops altogether. This problem can be alleviated by occasionally turning down the gain on the compressed signal. Alternatively the maximum gain of the compressor can be limited so that the background noise in the absence of music will not be so high.

Provided the compressors in a stereo system are ganged to avoid changing the directional effect, compressed music, starting with a moderate dynamic range, can be described as fairly compatible. Further development work is under way with the aim of making wide dynamic range compressed music more compatible using automatic signal processing.

Basic System

As mentioned earlier the Burwen Laboratories system is based on a combination of pre-emphasis and compression in the record electronics and expansion and de-emphasis in the playback electronics. It takes advantage of advances in record-

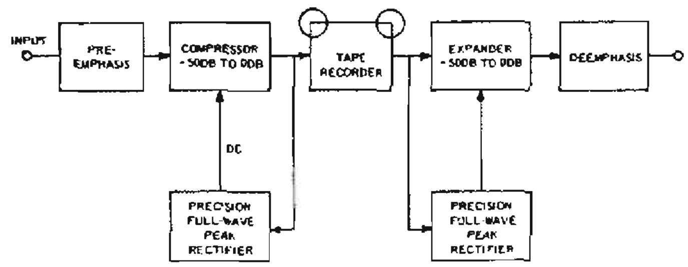

Fig. 2—Simplified system diagram.

ing tape technology made in recent years which result in increased flux density on the tape at high frequencies for a given record head current. Present day recorders having NAB playback response cannot properly take advantage of modern low noise tapes because of the standardized playback response curve.

When a high resolution, low noise tape is used the high frequency pre-emphasis in the record amplifier is reduced so that the flux recorded on the tape relative to middle frequencies is no greater than with ordinary tape. The net result is a very slight improvement due to the lower particle noise of the tape and possibly a 1 or 2 dB increase in overall output. At speeds of 7½ and 15 ips it is possible to improve the signal-to-noise ratio by from 6 to 12 dB simply by pre-emphasizing and de-emphasizing the high frequencies more than in a standard tape recorder.

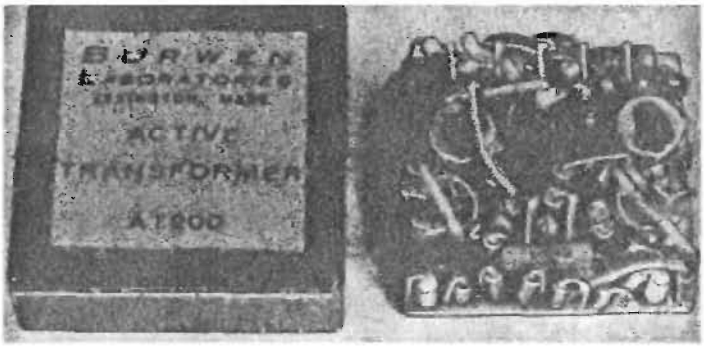

Fig. 3—The active transformer module.

In the basic system diagram, Fig. 2, the record compression system is shown on the left of the tape recorder and the playback expansion system on the right. The record signal first passes through a pre-emphasis network which increases the high and low frequency gain and then into a variable gain amplifier used to compress the signal by as much as 50 dB. The gain of the compressor is determined by its output signal which is fed to the record input of the tape recorder and simultaneously to a precision full wave peak rectifier.

The peak rectifier converts this signal to a d.c. gain control voltage which is smoothed by a multistage nonlinear filter. This circuit is designed for rapid compression when the input signal suddenly increases in amplitude but has a slow enough decay to prevent modulation of the signal at audio frequencies.

After the signal is recorded on the tape and played back at unity gain it is expanded in a complementary manner using a variable gain amplifier having a gain from -50 to 0 dB. The control voltage for this expander is derived from a precision full wave peak rectifier exactly the same as on the record side and which receives an input signal, the tape playback, similar to the record signal. Following the expander the high frequencies and low frequencies are de-emphasized to produce flat response for the entire system.

The combination of pre-emphasis and compressor gain of 50 dB or more increases the high frequency signal content on the tape by over 60 dB for low level signals. On the playback side the gain is reduced by the same amount and accordingly the unweighted noise level is reduced between 50 and 60 dB.

Because the compressor follows the pre-emphasis network its own noise is less significant and the tendency of the pre-emphasis network to overload the tape on high frequency input signals is greatly reduced because the compressor tends to hold the signal more nearly constant. Unlike the signal emerging from a conventional volume limiter the signal on the tape must have variations in peak amplitude. Otherwise the playback expansion side of the system could not determine how much to expand the signal. The action of the compressor and expander is smooth and precise and the dynamic error at any frequency and level is typically under 1 dB.

Of particular importance in this system is the fact that there is no alteration of the frequency response even if the gain of the tape recorder is not exactly 0 dB. The only effect is a change in output level and there is not even any appreciable expansion or compression of the signal. While it is necessary to maintain a reasonable signal level at the input of the expander if the system were used for FM reception for example, the signal level could easily be set by ear with sufficient accuracy without transmitting a calibration tone.

In the Noise Eliminator system, Fig. 1, two channels are contained on a single 19" x 1¾" rack panel. The system is built using encapsulated epoxy modules such as that in Fig. 3 which are also available for building into studio tape recorders. This is an automatically switched system in which the same components used for compression in the record electronics are used for expansion in the playback electronics. When the equipment is used for two-channel recording the compressed playback signal passes directly through the instrument so it can be used for monitoring from tape. Monitoring the compressed signal is quite useful in that it readily shows up defects in the program material. Switching from record to playback can be either via front panel switches or remotely from the tape recorder.

The input and output levels are +4 dBm at 0 VU but can be adjusted internally for other levels. A d.c. coupled differential input amplifier called the "Active Transformer" is used in place of the usual input transformer and the d.c. coupled system output will feed any load from 150 ohms to an open circuit with negligible change in frequency response.

Future Uses

The general usefulness of the Burwen Laboratories Noise Eliminator system has only begun to be explored. Besides reducing noise in studio masters and consumer recordings, processing the compressed signal through the studio console can produce some interesting effects. For example, boosting the bass and treble a few dB will result in expansion upon playback during orchestral crescendos which increase the dynamic effect of the music without causing the usual boominess due to excessive bass. Mixing compressed signals together can produce an effect whereby a predominant instrument can cause an increase in the level of all the others after playback expansion. Reverberation in a compressed signal sounds as though it has been increased because the number of dB the music level can decay in a given time has been reduced.

It is anticipated the availability of wide dynamic range recording will have a substantial impact upon the recording industry and it may even spark a new generation of low noise equipment. Æ

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Refacing Tape Recorder Heads

William B. Fraser

NOT LONG AGO, a customer brought me a tape recorder for repair. The complaint was poor and erratic sound.

The recorder was an expensive one and almost new. It didn't seem likely that the electronics were at fault. A check of the heads located the problem immediately. The playback head had been mounted incorrectly, so that the height was grossly out of adjustment. In the meantime, the machine had been operated enough so that an obvious tape groove was visible on the face of the head. Under these circumstances, it is usual to consider the head has been ruined. The conventional solution is to install a new head.

But a new head was expensive and not readily available. Further, the customer was anxious for speedy repairs. After some thought, I decided to attempt to salvage the head by refacing it. I have not heard of this being done in a service shop (or elsewhere for that matter) and approached the job with some apprehension. Fortunately, one thing was going for me. The head was ruined anyway, so what was there to lose?

The results of this first attempt were so satisfactory that subsequently I have refaced about a dozen other heads with 100% success. I have often wondered how long these reconditioned heads would last. So far it has been impossible to determine, simply because no customer has ever returned with a complaint. As a guess, perhaps the initial life expectancy of the head has been doubled.

Anything on the face of a tape recorder head which interferes with intimate contact between head and tape will cause unsatisfactory operation. A deep tape groove or a scratch or an erratic or incorrectly located wear pattern all cause the tape to lose contact with the head. It is these types of abnormalities which are considered in this article.

It takes me about one hour of grinding and polishing to reface even the worst head. Add to that the time to remove the head, then reinstall and align, and the entire job runs two to two and a half hours. This is a considerable saving to the customer over the cost of a new head. Occasionally, you will run into a head that is so severely worn that refacing should not be attempted. This is the time when you must depend upon your good judgment. In performance, a refaced head compares favorably with a new head in all important characteristics.

Here's how the job is done. First get the necessary materials. They are all inexpensive and readily available:

1. A wooden board on which to do the grinding and polishing. A small kitchen cutting board about 8 × 10 in. does nicely. It must have a smooth, flat surface.
2. One sheet wet sanding paper, 220 grit.
3. One sheet wet sanding paper, 400 grit.
4. One sheet wet sanding paper, 600 grit.
5. Jar of silver polishing cream, Wright's or equivalent.
6. Piece of soft flannel cloth about a foot square.
7. Magnifying glass, about 10 ×.

Having assembled the equipment, you are ready to start. Remove the head from the machine and disassemble any mounts or shields which interfere with access to the face. Examine the wear pattern to make sure the head can be salvaged and to determine what you have to do. Let us assume the wear consists of a grove about three or four times the thickness of a 1 mil tape. This amount of wear will require a good deal of grinding, and so we start with the coarse (220) paper. The grinding should be done wet, that is with a small flow of water on the grit paper. Wet grinding produces a smoother result than dry grinding. A pressure of no more than six to eight ounces should be used, as heavier pressure will make deep gouges in the face of the head. If you're not sure what a 6 ounce pressure is, get out a pressure gauge and find out. It's important. Hold the head by its sides with the face against the grit paper. As you look downward toward the work, use a rotary motion of the hand, making circles about three or four inches in diameter. Simultaneously with this rotary motion, rock the head back and forth so the entire face is exposed to the grinding action. At all times keep an even pressure on the head. Occasionally, turn the head end for end to insure even grinding. Examine the work frequently under the microscope. When the depth of the groove has been reduced to about half the thickness of a one mil tape, it is time to use the medium (400) paper. Keep up the grinding until the groove is almost eliminated, then finish grinding with the fine (600) paper. Avoid grinding more than necessary, but be certain all traces of the groove have been eliminated.

The next step is to polish the face. Put aside all abrasive papers and wash the cutting board. Fold the flannel cloth double, moisten it, apply a small amount of polishing cream, and polish, using the same techniques as employed during grinding. Fifteen minutes of polishing usually will produce an excellent surface finish—provided your grinding was correctly done.

Let us retrace our steps for a moment to the point when we made the initial examination of the head to determine the depth and extent of the wear pattern. If the wear is light, the grinding should start with the medium or fine grit rather than the coarse grit. In case the coarse grit is employed late in the grinding process, deep scratches will be left in the face of the head. These tend to accumulate debris and interfere with head-to-tape contact.

Well, that completes the job as far as refacing is concerned. Of course, you must reassemble the equipment, align the heads, and test the equipment. As far as the head is concerned, you should achieve specified performance of the recorder. Æ

A Dynamic Noise Filter For Mastering

Richard S. Burwen

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NOISE IS A PROBLEM! You can't escape it. It is in the environment, electronics used for recording, records, tapes, AM, FM, TV, microwave, and other audio communication media. Noise in studio tape recorders is too high for much of the musical program material being recorded today. Studios without noise reduction equipment usually overcome noise by such undesirable methods as (1) recording at such a high level that peaks far exceed the 1% or even the 10% distortion level of the tape, (2) limiting the signal before recording or (3) monitoring through a speaker system that attenuates the very high frequencies. The rather serious limitations of these methods have been overcome by electronic noise reduction.

Among the three principal methods of noise reduction available for studio mastering two of them, the Dolby Laboratories Noise Reduction System[3] and the Burwen Laboratories Noise Eliminator[2], involve compression of the signal before recording. As a result the signal recorded on the tape is nonstandard and a separate noise reduction channel must be used for each track in multitrack recording.

A third system, the Dynamic Noise Filter, Fig. 1, manufactured by Burwen Laboratories, is much simpler, requires far less equipment, eliminates the problems of nonstandard tapes, and is more versatile than either of the two systems mentioned above. For example, a single two-channel Dynamic Noise Filter operating on a two-track master tape output feeding a disc cutting system, as illustrated in Fig. 2, reduces the cumulative noise from *all* sources ahead of it. Thus two channels of the Dynamic Noise Filter do the job of 18 channels of the other types in the 16-track system of Fig. 2. One of the key advantages of the Dynamic Noise Filter is that it may be used to reduce the noise in any program material without the need for special processing and without audibly affecting the signals.

Principles of Operation

The operation of the Dynamic Noise Filter is based on two principles: (a) the noise output of an electronic system is dependent upon the system bandwidth and (b) the human auditory system "masks out" noise in the presence of the desired signal at frequencies in the vicinity of the signal frequency when the signal-to-noise ratio is sufficiently high. The Dynamic Noise Filter* can be described as an automatically variable bandpass filter whose bandwidth changes rapidly with each musical note and whose high and low frequency cutoffs are independently controlled by the spectral content of the input signal. Figure 3 illustrates the Filter's dynamic frequency response. Noise reduction is achieved by restricting the bandwidth at high and low frequencies when the signal level is very low (the minimum bandwidth is 800 Hz). At medium and high signal levels it passes the full 20 Hz to 20 kHz bandwidth. As in the other noise reduction systems, the Filter reduces noise only for low level signals; the noise is passed along with the high level signals. However, due to the characteristics of the ear, the noise that is present during each note is masked by the music.

The noise reduction, as shown by the minimum bandwidth curve in Fig. 3, is 25 dB @ 30 Hz and 22 dB @ 10 kHz. On 7½ or 15 ips unweighted tape noise the reduction measured typically 10 to 11 dB. Because noise reduction is attainable without any special preprocessing of the signal, the Dynamic Noise Filter is useful not only for a multitrack mix or a master tape playback but also for prerecorded tape, cartridges, cassettes, records, FM programs, or video tape sound.

This versatility comes, of course, at a sacrifice—the bandwidth is restricted for low level signals. However, as will be shown later full bandwidth is attained at such low signal levels and in such a short time that the effect of the Dynamic Noise Filter on most good quality program material is completely inaudible except for the reduction of noise. When musical instruments are played at such a low level that the bandwidth is restricted, they usually produce much less overtone output and little is lost by the attenuation of the high frequencies.

Design Simplification

The system described here is a simplification of an extremely flexible and highly complex Dynamic Noise Filter[1] system described in an earlier paper. The previous system, built in

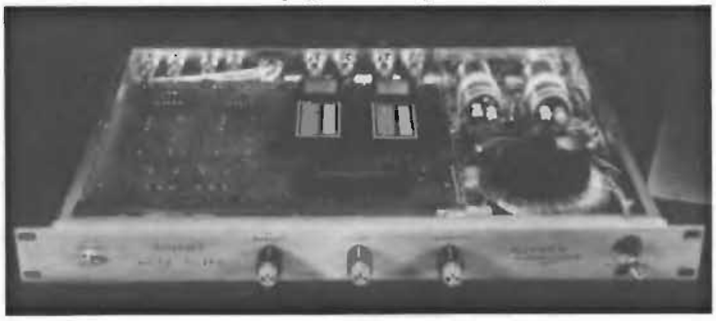

Fig. 1—The Dynamic Noise Filter chassis accommodates modules for 1, 2, 3, or 4 channels.

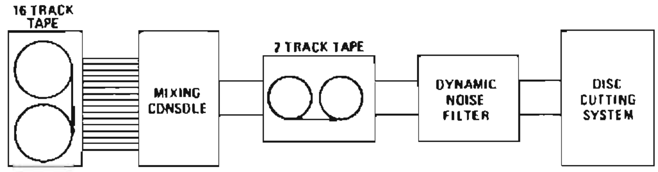

Fig. 2—Two-channel Dynamic Noise Filter reduces noise in a 16 track system.

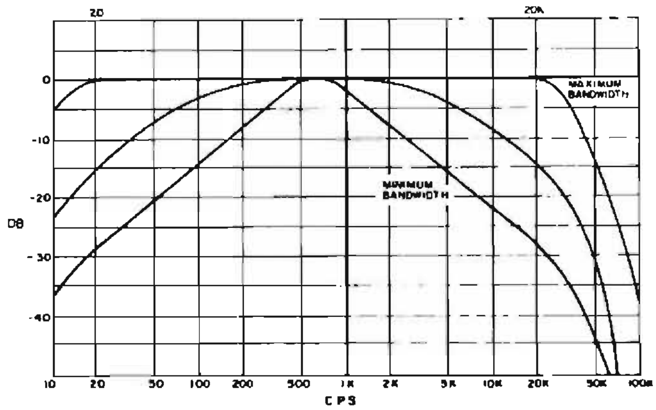

Fig. 3—The frequency response varies with the input signal level.

*Patent pending

Fig. 4—Rear view.

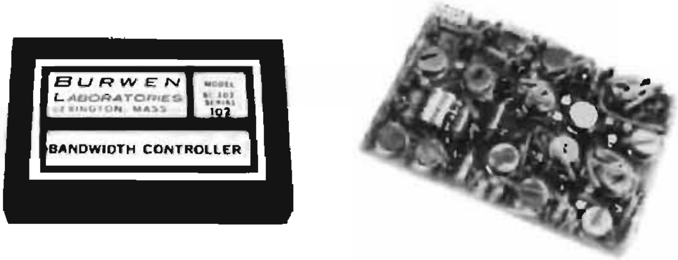

Fig. 5—A high performance module incorporating 15 operational amplifiers.

three channels, incorporated precision multipliers and integrators for bandwidth control and used a total of 320 integrated circuit operational amplifiers mounted on 57 plug-in circuit cards. It provided a choice of cutoff frequencies of 6, 12 or 18 dB/octave at high and low frequencies, a variety of bandwidth limiting functions, four-frequency notch filtering for hum and rumble, and a unique click limiter to eliminate noise impulses from phonograph records. Experiments made with this unit on a wide variety of program material indicated that the steeper slopes were preferred mainly for old material having high frequency distortion or completely lacking in the very high frequencies. For wide frequency range program material it was found that the same noise reduction could be achieved using a 6 dB/octave filter having a high cutoff frequency considerably below the minimum tolerable for 12 or 18 dB/octave filters. Furthermore, the 6 dB/octave filter seems to produce smoother operation on low level classical music.

In an instrument optimized for studio mastering, the notch filters for hum and rumble along with the click limiter and a number of the controls were deemed expendable luxuries. The use of common control circuits for two channels saved a considerable number of components while preserving the stereo balance. Thus a simplified Dynamic Noise Filter, Figs. 1 and 4, evolved having a minimum of controls and accommodating 1, 2, 3, or 4 channels on a single 1¾" high rack panel. Operational amplifiers are still used extensively to achieve a 100 dB dynamic range with 0.01% midband harmonic distortion. The system is built using high quality, reliable plug-in modules, such as is shown in Fig. 5, involving a total of 80 operational amplifiers for four channels with as many as 15 in a single module.

Block Diagram

The modules can be arranged into as many as four signal channels and the configuration for one channel is illustrated in Fig. 6. The input signal is first fed to the active transformer which is a unique differential input d.c. amplifier. It serves the same function as a conventional audio transformer and provides the same common mode rejection while overcoming the transformer's limitations in frequency response, distortion, and hum pickup. The signal from the active transformer is fed to two other modules, the voltage variable bandpass filter and the bandwidth controller, which form the heart of the noise reduction system. At the input to the bandwidth controller signals from two channels are added. Inside the module the high and low frequency components are selected out and individually rectified, filtered, and compressed to form two separate d.c. control voltages that adjust the cutoff frequencies of the voltage variable bandpass filter. The control voltages vary the bandwidth of the system in relationship to both the frequency content and level of the input program material.

The Voltage Variable Bandpass Filter

The key module in the noise reduction system, the voltage variable bandpass filter, is shown in block form in Fig. 7. The signal from the active transformer is brought to the input of the module where a feedback type pre-emphasis network increases the high frequency gain at 6 dB/octave above 3 kHz for the purpose of increasing the signal-to-noise ratio. At the output of the module the response is flattened by a complementary de-emphasis network and output buffer amplifier.

Following the pre-emphasis network the signal passes through a multiplier, X1, which attenuates the signal in accordance with the value of the high frequency d.c. control voltage produced by the bandwidth controller. The multiplier then feeds the main signal operational amplifier, A1, which in turn delivers its output to the de-emphasis network. Three separate feedback paths around A1 determine the high, middle, and low frequency gains of the module.

The first feedback path via capacitor C1 converts A1 into an integrator and reduces the high frequency gain. The gain from the input to the output at very high frequencies is the product of the gain of the multiplier X1 and the integrator A1. At middle and low frequencies, where the gain allowed by capacitor C1 is high, the closed loop gain is determined by the ratio of the values of resistors R6/R1. At low frequencies the feedback is further increased by the multiplier X2, the integrator A3, and the inverter A2. Combining the three feedback paths produces virtually infinite d.c. feedback and reduces the system gain to 0 at d.c.

Changing the high frequency cutoff point is achieved by varying the high frequency control voltage supplied by the bandwidth controller. Increasing the control voltage on multi-

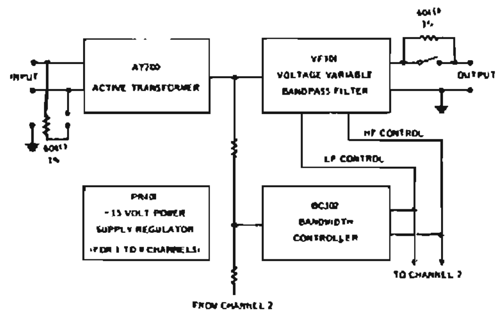

Fig. 6—Single channel system diagram.

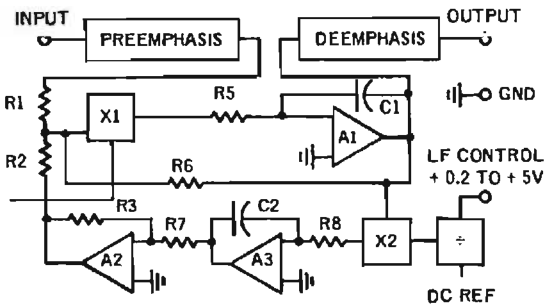

Fig. 7—Block diagram of Voltage Variable Bandpass Filter module.

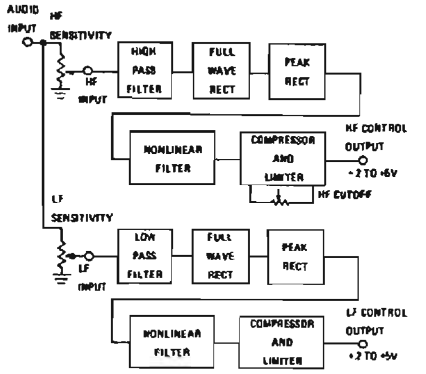

Fig. 8—Block diagram of the Bandwidth Controller.

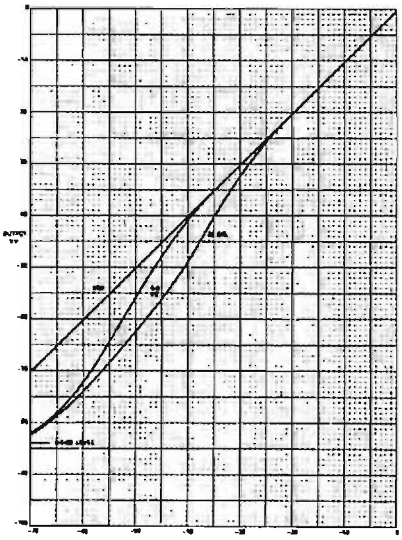

Fig. 9—Output vs input at 85 Hz, 650 Hz, and 6.6 kHz.

plier X1 effectively reduces the time constant of integrator A1 and raises the high frequency cutoff in direct proportion to the voltage.

The low frequency cutoff is changed by causing the feedback path made up of integrator A3 and multiplier X2 to predominate over R6 at low frequencies. Changes in the gain of the multiplier and the effective time constant of the integrator are caused by the low frequency control voltage generated by the bandwidth controller. Increasing the d.c. control voltage delivered to the multiplier X2 effectively reduces the time constant of the integrator A3 and raises the low cutoff frequency. What is desired is a reduction in low cutoff frequency for an increase in low frequency control voltage. To accomplish this, the value of the low frequency control voltage is inverted by dividing it into a d.c. reference in the divider block causing a value to +0.2 V to produce a 350 Hz frequency cutoff and +5 V to produce a 13 Hz low frequency cutoff.

The frequency response curves produced for d.c. control voltages of 0.2 V, 1 V, and 5 V at both control inputs are the same as in Fig. 3. Note that the midband gain is constant and determined by R6/R1 and the wideband response is flat within 0.2 dB from 20 Hz to 20 kHz. These curves are very similar to the curves that would be produced by feedback around A1 via R6 at middle frequencies, through a variable capacitor at high frequencies, and through a variable inductor at low frequencies if the remaining components were removed.

The Bandwidth Controller

The bandwidth controller which produces the high and low frequency d.c. control voltages is shown in more detail in Fig. 8. External to the module are high frequency sensitivity and low frequency sensitivity potentiometers which divide down the summed output of the active transformers in channels 1 and 2. The arm of the high frequency sensitivity potentiometer feeds a high pass filter which selects out frequencies primarily in the vicinity of 6.6 kHz and provides considerable amplification. The high frequencies are then full wave rectified and peak rectified using a precision feedback circuit that produces accuracy down to millivolt levels. A multiple section nonlinear filter is used to smooth the output of the peak rectifier sufficiently to eliminate modulation of the bandwidth at any audio frequency. The filter is a nonlinear feedback system which can charge rapidly and takes only 1 mS to reach full output. Its decay time is approximately 50 mS to within 10% of final value. The output of the nonlinear filter passes through a compressor and limiter circuit which increases the effect a small audio input has on the bandwidth. An external high frequency cutoff potentiometer places an upper limit on the +0.2 to +5 V control voltage anywhere within this range, and may be used to limit the bandwidth for poor quality or distorted program material.

The low frequency d.c. control voltage is generated in the same manner as the high frequency control voltage following the low frequency sensitivity potentiometer. In this part of the bandwidth controller the low frequencies primarily in the vicinity of 85 Hz are used to produce the d.c. control output and the time constants are longer. The low frequency attack time to produce +5 V output from an overdrive input is 10 mS and the decay time is approximately 500 mS to within 10% of the final value. The low frequency part of the controller is about 14 dB less sensitive than the high frequency section because music tends to have more low frequency energy than high frequency energy.

In choosing the attack and decay time constants for the high and low frequency sections of the bandwidth controller, an attack time of 1 mS to attain the full 32 kHz bandwidth was found to be short enough for no audible effect on musical transients. On the other hand, it was found undesirable to shorten the attack time further because this allowed the ticks on a record to actuate the filter. The decay time seemed to be optimum at about 50 mS to within 10% of final value. A shorter time reduced the effective reverberation in the program material and a longer time allowed more noise to be heard between the notes.

The low frequency attack and decay times appeared to be somewhat less critical but had to be longer to support the low frequency response and prevent more than 0.05% harmonic distortion on a low level 20 Hz signal.

Other Features

Each channel has a differential input with a 100k impedance which may be changed to 600 ohms by means of a rear panel switch. The overall voltage gain of each channel is 0 dB ±0.1 dB and the maximum output is ±11V instantaneous peak into an open circuit, +18 dBm into 600 ohms, or +16 dBm into 150 ohms. The output is single ended, grounded to the chassis and has an internal impedance of 0.4 ohms to which 600 ohms can be added by means of a rear panel switch.

A 0.01% regulated ±15 V power supply powers all four channels. The transformer is toroidal for low external magnetic field, and the positive and negative voltages are interlocked and overvoltage protected so that a supply failure will not damage the other modules.

Performance

A measurement of the output vs input, Fig. 9, shows that the Dynamic Noise Filter behaves as a linear amplifier at

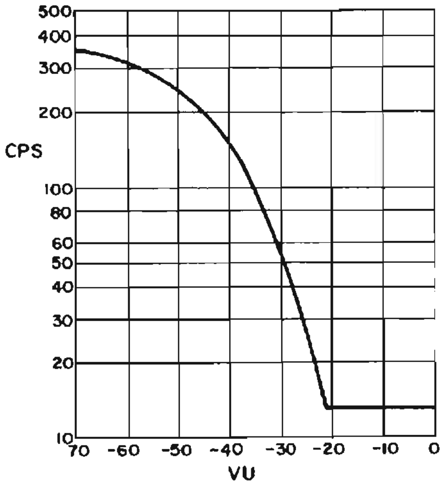

Fig. 10.—Low frequency cutoff vs input at 85 Hz.

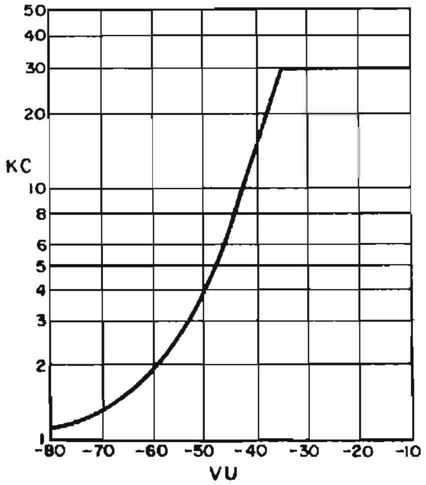

Fig. 11—High frequency cutoff vs input at 6.6 kHz.

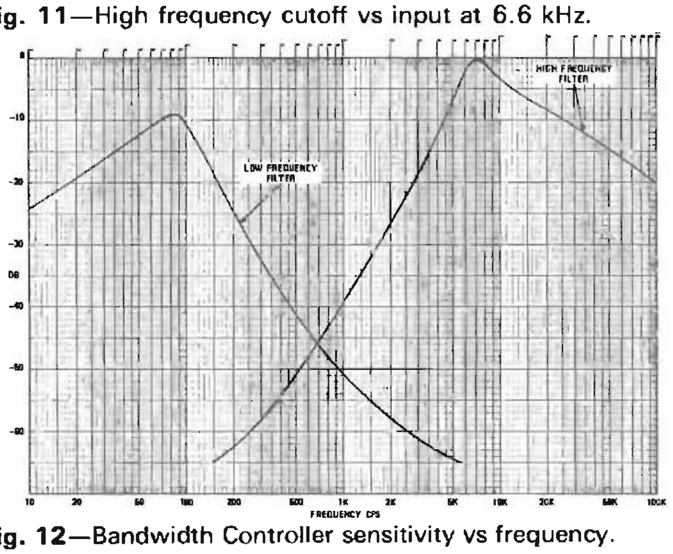

Fig. 12—Bandwidth Controller sensitivity vs frequency.

650 Hz but appears to be a low level expander at 85 Hz and 6.6 kHz. These measurements were made at the typical control setting used when playing a multitrack mix or a master tape. It can be seen that the expansion takes place at a very low level in the range of -75 to -35 VU at 6.6 kHz. What is really happening, of course, is that the bandwidth is being extended as the signal increases.

Another way of looking at this same effect is to plot the -3 dB bandwidth vs input as shown in Fig. 10 at 85 Hz and in Fig. 11 at 6.6 kHz. These frequencies are the points at which the bandwidth controller is most sensitive as shown by the response curves of the bandwidth controller filters in Fig. 12. At other test frequencies the bandwidth variation will occur at higher levels as determined by these curves.

The sensitivity can be varied over a wide range to shift the curves in Figs. 9, 10, and 11 up and down in level to suit the particular program material. Generally, the low frequency sensitivity is set so that rumble just begins to operate the low frequency filter and the high frequency sensitivity is set so that hiss just begins to operate the high frequency filter.

As in other types of noise reduction systems, the Dynamic Noise Filter discriminates best between the music and the noise when the signal-to-noise ratio is high initially. When used with extremely noisy program material such as a 78 rpm record, the controls have to be set for a compromise between noise reduction and degradation of the program material. Many 78 rpm records contain only noise and distortion in the region of 6.6 kHz and there is some advantage in cutting off the high frequencies at 12 dB/octave at a somewhat lower frequency ahead of the Dynamic Noise Filter. Since the filter attenuates the high frequencies at small signals as low as 1100 Hz, it produces worthwhile noise reduction in excess of that attainable with a fixed filter. The tone can be balanced for pleasing response following the Dynamic Noise Filter.

In listening tests an interesting psychological effect was observed. A reduction in tape hiss seems to be accompanied by an attenuation of the high frequency content in the program material even when it does not actually occur. This effect can be confirmed by adding hiss to a noise-free signal in which case the high frequency output seems to be somewhat increased.

Applications

The Dynamic Noise Filter achieves 10-11 dB of noise reduction on tape program material by performing as a constant gain filter whose instantaneous bandwidth is a function of the program content. It has the advantage over other noise reduction methods in its ability to reduce noise arising anywhere in a system ahead of the filter. In contrast, the Dolby Laboratories Noise Reduction System and the Burwen Laboratories Noise Eliminator are both designed to prevent the recording system from introducing noise into the program material but they are not designed to improve existing noisy program material.

For live recording on 16 tracks followed by a two-track tape master and then a two-channel disc, 18 channels of the Dolby Laboratories Noise Reduction System or the Burwen Laboratories Noise Eliminator are normally used. By recording nonstandard signals on the tape the former achieves 10 dB noise reduction and the latter as much as 50 dB. Alternatively, the signal can be recorded on the 16-track machine and then on the two-track machine in normal manner without noise reduction and played through a two-channel Dynamic Noise Filter inserted before the tape duplicating equipment or a disc cutter.

The range of applications for the Dynamic Noise Filter is just now being explored. Potential exists in the AM, FM, and television broadcast fields as well as in theatre sound systems, and rerecording of historic material. For the audio engineer the Dynamic Noise Filter is a new, versatile, and economic tool by which he can control the noise present in his environment, his equipment, his recordings, and his program material.

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A Peak-Reading VU Meter

With Compensation For Tape Saturation

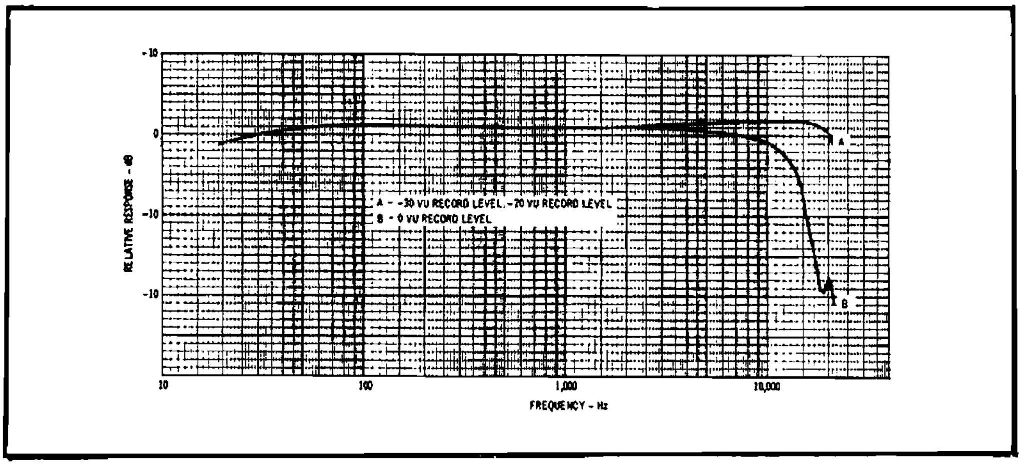

Fig. 1—Record-playback response at 7½ ips. 0 VU corresponds to a 400 Hz signal recorded 8 dB below the 3% THD level.

E.A. Ballik*

IT IS WIDELY RECOGNIZED that peak recording levels can be considerably greater than the average levels. The majority of tape recorders employ average-reading VU meters, primarily because of circuit economics. Futhermore, VU meters generally have a frequency response which is relatively uniform in the region of 20 Hz to 20 kHz. Unfortunately magnetic tape does not have a uniform frequency response, and therefore considerable recording experience is required with standard VU meters in order to achieve high signal-to-noise ratios together with low distortion.

This article describes a simple, and relatively inexpensive, peak-reading VU meter circuit which can be used either at the high level output of a tape recorder (PHONE or LINE) or which can be incorporated into the recorder inself. The circuit also provides compensation for the recording and saturation characteristics of the tape medium. This compensated peak-reading VU meter circuit makes it possible to achieve the maximum available signal-to-noise ratio from the tape and ensures that THD and intermodulation distortion are kept at acceptable levels over the whole of the audio spectrum.

A slightly modified two-track Revox A77 tape recorder and Scotch 206 tape were used for all the measurements reported below. The Revox meters were calibrated so that 0 VU corresponds to a 400 Hz signal recorded 8 dB below the 3% THD level. Playback response, as measured with an Ampex 21690010-01 two-track test tape, was well within ± 1 dB. The unweighted record-playback signal-to-noise ratio at 400 Hz was 60 dB for 1% THD.

Tape Record-Playback Characteristics

Figure 1 illustrates the relative record-playback response at 7½ ips, taken at constant input with frequency. Curve A corresponds to record levels of -30 VU and -20 VU. The only difference noticed during these two measurements was a decrease in the 20 kHz response of approximately ½ dB at the higher input level. Curve B corresponds to a record level of 0 VU. The decrease at higher frequencies results from tape saturation. Clearly curve A would be the response quoted in manufacturers' literature or in test reports.

Several parameters affect the high frequency response of the tape and consequently the saturation behaviour. Several of these are inherent in the tape. These include:

Oxide Layer: The oxide material, the particle size, and the coating thickness can all have a significant effect. These effects can only be controlled in the tape manufacturing processes. Oxide coatings have been slowly but constantly improved in recent years and are expected to continue to do so.

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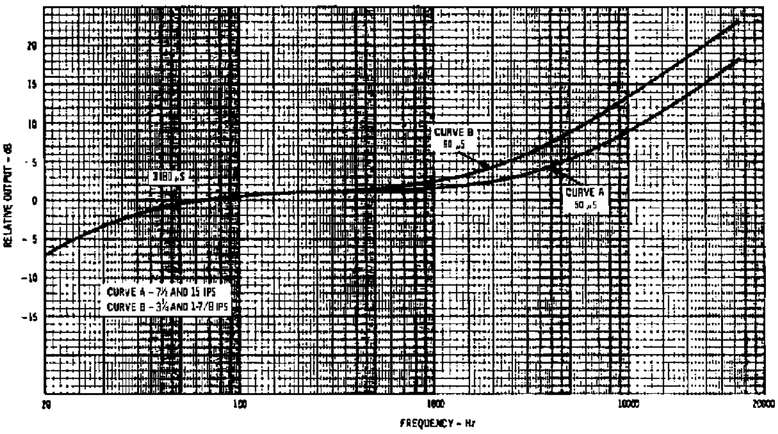


Fig. 2—NAB standard reproducing characteristic. Reproducing amplifier output for constant flux in the core of an ideal reproducing head.

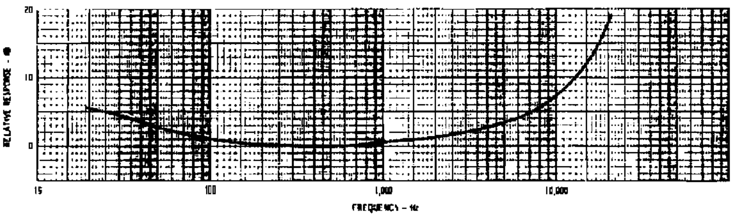


Fig. 3—Revox A77 record amplifier response. The curve is essentially the same for both 3¾ and 7½ ips.

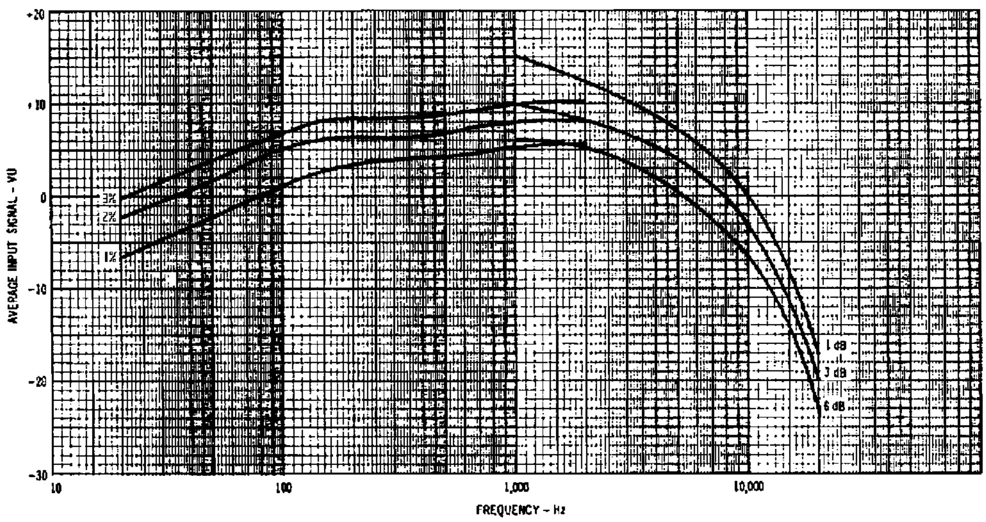


Fig. 4—Tape distortion-saturation data. Curves are given for 7½ ips. Those below 1 kHz correspond to THD; those above 1 kHz to the output in decibels below the peak saturation output.

Penetration Losses: At low frequencies the full depth of the tape is magnetized. With higher frequencies (i.e. short recorded wave-lengths) the depth of magnetization decreases, until only the surface of the oxide coating is magnetized. The high frequency recorded intensity is therefore decreased. Penetration losses for a given oxide can only be modified by varying the tape thickness. However this usually affects other parameters such as tape uniformity and signal-to-noise ratio, particularly at lower frequencies.

Self-Demagnetization: At high frequencies the recorded magnetic poles are closely spaced and thereby cause a decrease in the recorded signal. This effect is usually small. However it can increase with storage time of the recorded tape. Losses of a few decibels at the highest recorded frequencies are possible after storage of several years.

Other sources of high frequency losses are primarily caused by the tape recorder characteristics. These include:

Recording Demagnetization: Increasing the recording bias results in high frequency attenuation (i.e. self-erasure). A low bias cannot be used because it would result in high distortion levels. Generally the optimum bias is characteristic of the particular tape used.

Head Losses: The playback output drops to zero when the playback head gap is equal to a recorded wavelength. The effect can be made negligible by the use of an extremely narrow gap. This will result, however, in a very small output, with decreased signal-to-noise ratio. In practice a compromise is made in the gap size which results in significant head losses at the higher recorded frequencies.

Equalization to compensate for high frequency losses is carried out in *both* the record and the playback amplifiers. In order to allow interchangeability of recorded tapes a standard playback equalization is used in the better audiophile recorders and in professional equipment. The current standard reproducing characteristics used in North

America is the NAB standard, shown in Fig. 2. The curves are for a constant flux in the core of an ideal reproducing head (i.e. zero frequency-dependent head losses). Note that with zero playback equalization a constant flux would give a playback output which increases at the rate of 6 dB per octave.

The equalization illustrated in Fig. 2 provides a 6 dB per octave boost at high frequencies (3 dB up at 1.6 kHz for 3¾ and 1⅞ ips, and 3 dB up at 3.2 kHz for 7½ and 15 ips). In addition there is a 6 dB per octave cut at low frequencies (3 dB down at 50 Hz). Standard playback tapes are recorded in such a way that an ideal reproducing head will give constant output with frequency. Note that all recorders will have some additional playback equalization to correct for playback head losses.

It should be clear from Fig. 2 that the record amplifier requires a 6 dB per octave boost (3 dB up at 50 Hz) to compensate for the playback characteristics at low frequencies.

As mentioned previously the high frequency boost in Fig. 2 only partially compensates for the high frequency losses; additional boost is provided in the record amplifier. The record-amplifier response can be determined by measuring the record current in the record head. For this measurement the bias oscillator is disconnected. A low value of resistance is placed between the "ground" terminal of the record-head and "ground." The record-head current, in the record mode of the tape recorder, is proportional to the voltage developed across the resistance. A resistance value of 100 ohms or less can typically be used. It should, however, be made several times smaller than the d.c. resistance of the record-head, unless it is already an integral part of the circuit.

The measured record-amplifier response of the Revox recorder is shown in Fig. 3 and was essentially similar for both 3¾ and 7½ ips. The curve indicates that tape saturation will occur more easily at both the low and high frequencies in comparison with the mid-range. This is acceptable for many recording applications because most of the signal is *usually* in the mid-range of the audio spectrum.

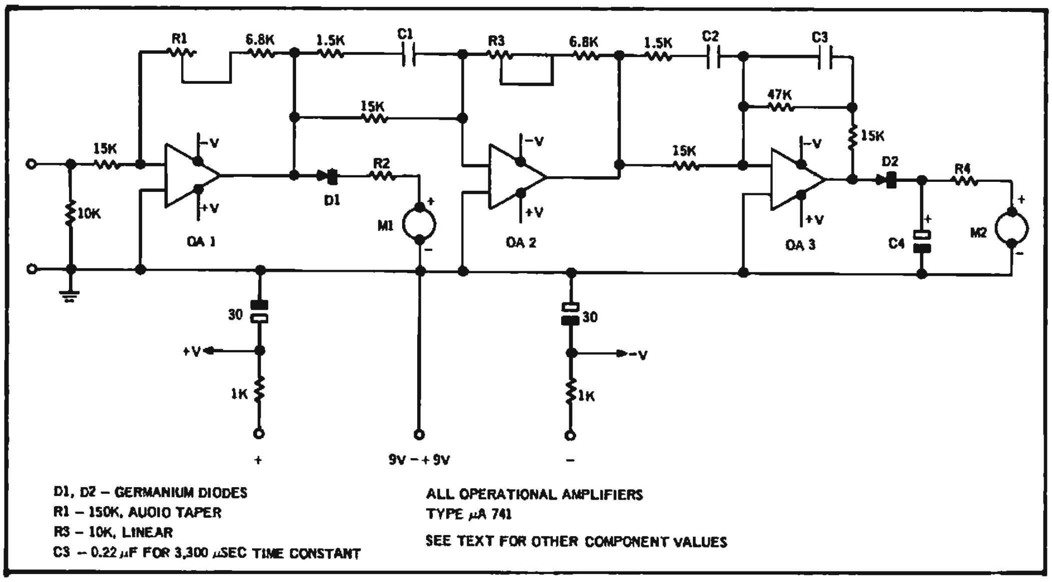


Fig. 5—Compensated peak-reading meter circuit. The circuit provides for both peak-reading and average reading capability.

Figure 4 illustrates record-playback distortion and saturation measurements at 7½ ips. These were made by monitoring during recording. The curves below 1 kHz are input levels in VU for 1, 2 and 3% THD. The curves above 1 kHz are for input levels 1, 3 and 6 dB below maximum output. These latter curves were obtained by increasing the signal input until the playback output started to decrease, and then *decreasing* the input until the output is decreased the appropriate decibels below the *peak* output level. In general the THD measurements have little value at the higher frequencies, and the saturation measurements have little value at the lower frequencies.

It can be seen that the 6 dB below maximum output curve makes a good match with the 2% THD curve. This combination was used for all subsequent work.

The advantages of peak-reading meters are obvious. It should also be obvious from the above that the meter amplifier response should compensate for the distortion-saturation curve of the tape. Such a meter circuit will allow the maximum possible recording levels while minimizing distortion caused by transients and tape saturation.

Comparison of Fig. 4 with Fig. 3 shows that the saturation response characteristics relate closely to the record-amplifier response characteristics. Thus either the record-amplifier characteristics or the distortion-saturation curves can be used to determine the compensation for the meter amplifier. The latter, however, is considerably more accurate because it includes the complete record-playback response.

reading capability is not desired. Alternatively a single meter can be switched to provide either average-reading or peak-reading.

The first stage (OA1) has an input impedance of 6 kilohms and provides a low output impedance drive for the average-reading meter and for the following stage. An input capacitor should be used if d.c. components are present in the input signal. Potentiometer RI provides a first stage gain which is variable from -7 dB to +20 dB. Potentiometer R3 gives a second stage gain in the range -7 dB to +½ dB. This is used to calibrate meter M2 relative to meter M1. Operational amplifier OA3 provides a low impedance drive for the peak-reading meter M2. Low frequency boost results from C3 in

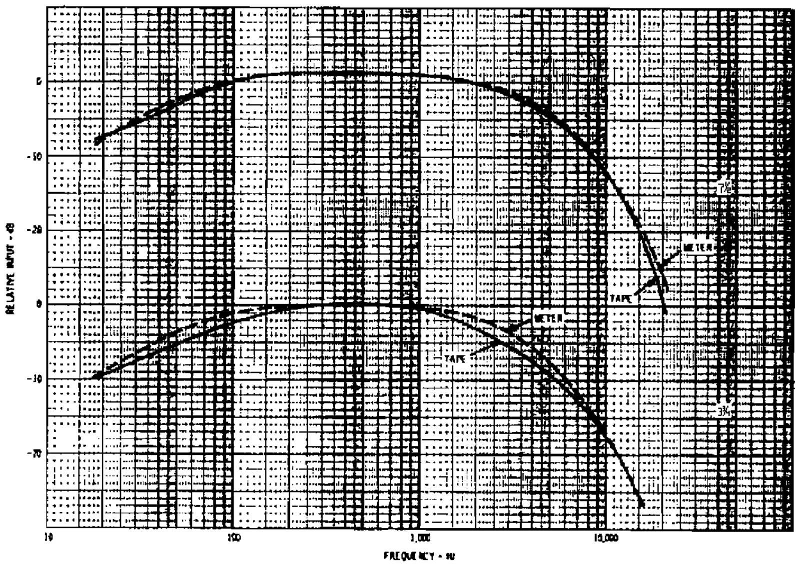


Fig. 6—Comparison of tape saturation characteristics with meter circuit compensation. The solid curves correspond to the distortion-saturation characteristics of the tape; the dashed lines correspond to the meter amplifier equalization.

Compensated Peak-reading Meter Circuit

The complete circuit is shown in Fig. 5 and includes both an average-reading and a compensated peak-reading meter. It is based on inexpensive operational amplifiers, namely the μA741, and simple RC networks. The μA741 is particularly useful because it is internally compensated and therefore requires a minimum of external components. No attempt has been made to correct for d.c. offset or bias current because of the relatively low d.c. gain. Meter Ml can be omitted if the average-

series with 15 kilohms. A 0.22 μF capacitor gives a time constant RC=3,300 μSec; which is close to the NAB standard of 3,180 μSec. The 47 kilohms resistance in parallel with C3 limits the extreme low frequency gain in order to prevent low frequency instability.

High frequency boost, at 6 dB per octave, results from C1 in parallel with 15 kilohms and by C2 in parallel with 15 kilohms. The ultimate boost is therefore at the rate of 12 dB per octave. The 1.5 kilohm resistors in series with C1 and C2 limit the maximum gain in order to prevent instability at the ex-

(Continued on page 81)

(Continued from page 54)

treme high frequencies. Capacitors C1 and C2 are chosen so as to give the best fit to the distortion saturation curve. This can be done either by calculation or by "trial and error."

Series meter resistors R2 and R4 should be chosen to allow full scale meter deflection (+3 VU) for an rms input signal of about 1 volt at the meter diode. The meters used in the design were inexpensive d.c. microammeters with VU scales. These had full scale sensitivities of 400 μA (for +3 VU) and d.c. resistances of 650 ohms. It was found that 470 ohms was a suitable value for R2 and 2.2 kilohms for R4. A value of C4=60 μF was found to be satisfactory. This provides a decay time-constant of RC=0.17 sec (since R=2.2 kilohms + 0.65 kilohms) for the peak-reading meter. A decay time-constant of about 0.2 sec appears to be the optimum value for most recording applications. The measured charge time-constant was about 15 mS and corresponds to an equivalent input resistance of about 250 ohms. The limitation here is the particular diode used for D2 (a small signal germanium) rather than the operational amplifier. The measured effective OA3 output impedance is considerably lower than 250 ohms. Although it doesn't appear to be necessary, the transient response can be improved by decreasing the charge time. This can be done either by use of low forward resistance germanium diodes or by the use of more sensitive meters. The latter will allow the use of a larger value of R4 (and also R2), and therefore a smaller value of C4.

A constant input with frequency gives a constant average reading. The relative input required to give a constant indication on the peak-reading meter is shown in Fig. 6 for both 7½ and 3¾ ips. The solid curves are distortion-saturation curves derived from Fig. 4 and similar data. The dashed curves represent measurements which give the relative input required for a constant peak-reading indication. For 7½ ips, C1 and C2 were each chosen to give time constants of RC=25 μSec. There is clearly an excellent match between the tape saturation curve and the peak-reading meter compensation. For 3¾ ips, C1 was chosen to give a time constand of RC=37 μSec. The match here is good but can obviously be improved by some adjustments of the time constants. Clearly the appropriate time constants are determined by the tape speed, the particular tape used, and by the tape-recorder characteristics.

It should be noted that the roll-off on the saturation curves occurs at lower frequencies with decreasing tape speeds. Thus frequency compensated meter circuits are particularly essential for quality recording at the lower tape speeds.

Performance

A two-channel system was built for stereo recording. Initial test recordings helped establish the optimum relative mid-range sensitivity between the average and peak-reading meters. As a result of these tests the peak-reading meter was calibrated to give a reading of -3 VU at 400 Hz for an average-reading meter indication of 0 VU. From inspection of Fig. 2 it can be seen that a peak reading of 0 VU corresponds to about 1% THD at all frequencies, and +3 VU corresponds to about 2% THD at all frequencies. Here the saturation is considered in terms of the equivalent THD because this will directly determine the magnitude of the intermodulation products.

Several live recordings have been made at 7½ ips and the results have been most gratifying. If the peak-reading value is *always* kept below about +2 to +3 VU, then there is negligible audible distortion. The distortion increases rapidly when the recording levels are increased above these levels.

Various types of live concerts were used. An organ recital provided an excellent check of extreme low frequency behavior. The highest average recording levels were reached with the pedal tones. The compensated peak recording levels were generally 4 to 5 VU above the average readings (actual difference is 7 to 8 dB!), even on sustained notes.

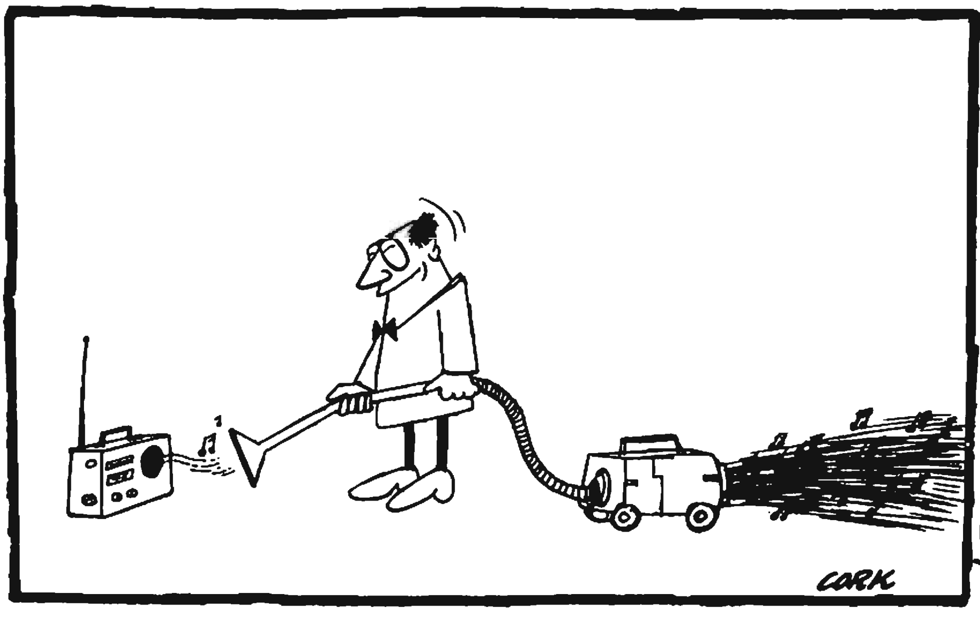
Several recordings of mixed choirs indicated that the average and compensated peak readings are generally comparable (actual difference 3 dB). Occasionally, however, the compensated peak readings were 3 to 4 VU above the average readings (actual difference 6 to 7 dB!).

With woodwind quintet it was found that both readings were also generally comparable (actual difference 3 dB). Only rarely did the peak indication exceed the average indication.

In summary it was found that average-reading meters on a professional quality tape recorder could achieve excellent recording quality at 7½ ips, provided that the 0 VU indication was set 8 dB below 3% THD at 400 Hz, and that the signal levels were *always* kept below 0 VU. With average program material, where the low and high frequency components were of relatively low intensity, this results in a sacrifice of 3 to 4 dB in signal-to-noise ratio. More dramatic differences will obviously occur at slower tape speeds because the high frequency roll-off occurs at lower frequencies.

A major advantage of the system is the complete confidence and ease of operation that it provides during recording because there exist well defined limits on the maximum recording levels. Thus the compensated peak-reading meters allow utilization of the full signal-to-noise capabilities of the tape under all recording conditions.

It should be noted that the cost per channel is quite modest even though the circuit contains more components than absolutely necessary. This was done in order to keep the design as simple as possible and also to achieve maximum flexibility. A more sophisticated design can obviously result in a considerable reduction in the number of components, and consequently in the cost. Æ



Noise Reduction Techniques

H. W. Hellyer*

LET'S TAKE A LOOK at one or two ventures into noise exclusion that have been at least a bit more ambitious than a mere clipping of playback peaks. One such system is Panasonic's NFD device. NFD, quite simply, mutes the line output unless the signals (on playback) are above a predetermined level and below a set frequency. This reduces hiss when the signal level is low. That is, you get what you want when you most want it.

In the RS 735US, there was a two-transistor, nine-diode circuit that gave very good results indeed. Figure 1 shows the basic configuration. Signal-to-noise ratio, when I tested it, with this noise filter employed, was as good as 66.5 dB. At 1 kHz, the improvement was a mere ¾ dB, but although at rated output level the NFD only made 1 dB difference to the S/N ratio, when the level of signal was down around the danger level, approaching what would have been obtrusive hiss, the circuit effectively blanked signal, and its action did not, as with so many compandor systems, provide an aural switchback.

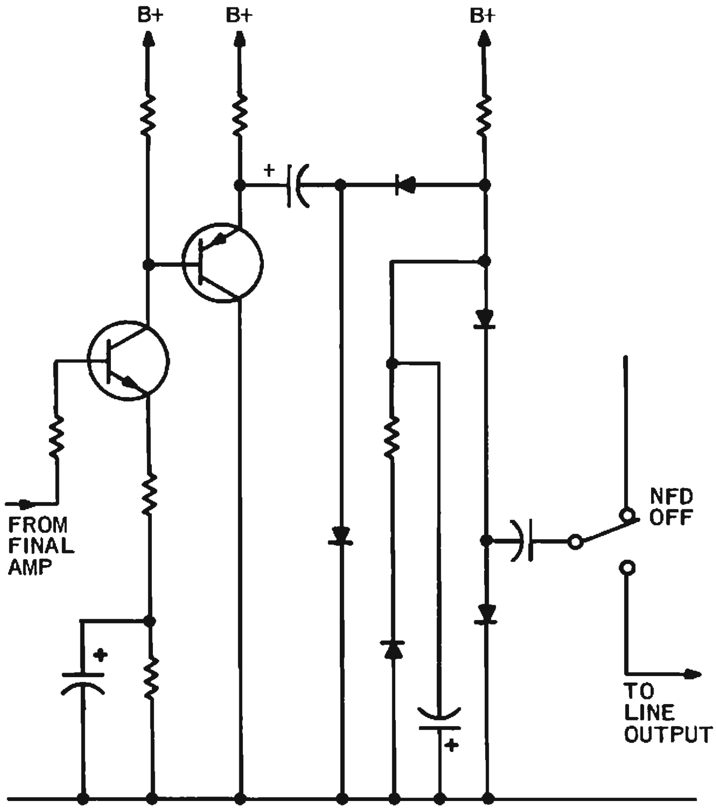


Fig. 1—A simple muting circuit used by Panasonic—simple, but effective, sensing the signal level and "killing" the line output when the signal drops dangerously near the noise level. The circuit shown is for one channel. The same two-transistor network is employed for the other channel, and this "commoning" can lead to problems.

*Bristol, England

Taking the replay system a step farther, Philips has the DNL innovation, which should make much cassette work with other folk's tapes a really feasible possibility.

DNL means Dynamic Noise Limiter, and Philips (Norelco to you) argues thus . . .

"When music is played softly, it is made up almost entirely of pure tones in the middle and low frequency ranges with hardly any harmonics. This is mainly because very few musical instruments produce tones whose fundamental frequencies are much higher than 4.5 kHz. Tape hiss, however, is made up of sounds in the higher frequencies so that it is during the soft passages and silent intervals that it becomes most noticeable.

"When music is played loudly, it not only contains the lower and middle frequency pure tones, but also a great deal of harmonics, which give character to the sound. It is in the loud passages that noise suppression is unnecessary as the high frequency harmonics hide the tape hiss. Any filter action would make the music sound dull and unnatural.

"Therefore, if tape noise or hiss is to be suppressed, it must be completely eliminated in periods of no music signal, reduced during the soft passages of music, and left unsuppressed during the loud passages."

Thus, the oracle—begging one or two questions, like: "Pure tones—all instruments played softly?" and "What happens to the soft tones of one instrument when another plays loudly?" and "How soon after the loud noise ends does the suppression take place?"

The Dynamic Noise Limiter acts on replay, the argument being that it therefore allows complete compatibility, giving the benefit of noise suppression even to those poor, deprived owners of untailored cassettes. It is, effectively, a steep, low-pass filter which acts when there are no high signal frequencies.

Philips has been rather clever about it, allowing high frequency signals that exceed a predetermined level to bypass the filter: so there are two signal chains. Fig. 2 shows the block diagram. From the splitter, the signal takes two paths, one path merely inverts the phase without affecting the linearity while the other passes it through the tailoring process.

This process chops off the lower and middle frequencies, leaving only those above 4 kHz (approximately—you can't do these chopping actions abruptly without introducing almost ineradicable distortions, whatever the advertising copywriters say). This remaining high frequency band is now monitored so that the quieter parts of higher frequency are boosted. Hence the variable attenuator—it is both level and frequency-conscious.

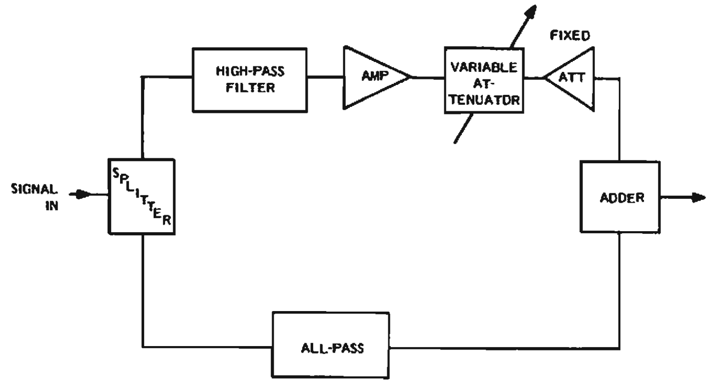


Fig. 2—Block diagram of the Philips (Norelco) Dynamic Noise Filter. The surprisingly effective though unsophisticated system acts on playback only and has the effect of an 18 dB/octave filter when the signal is low. A S/N ratio improvement of around 10 dB at 6 kHz and 20 dB at 10 kHz has been measured (unweighted). The high-pass filter takes effect above 4 kHz.

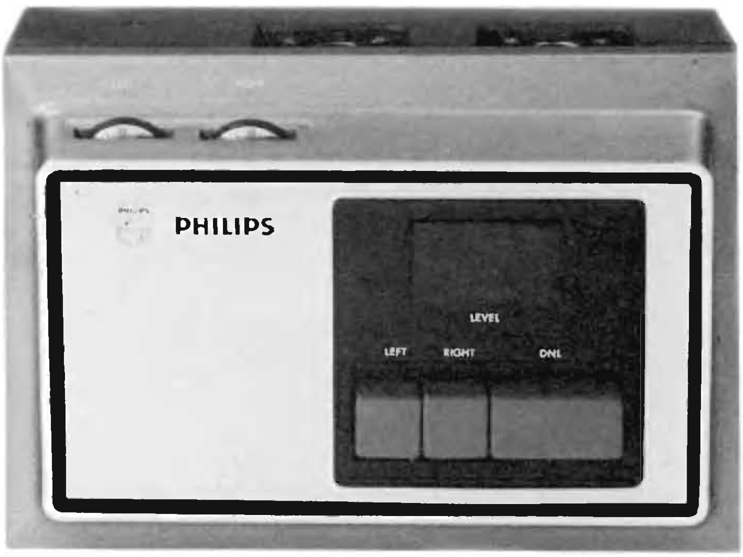


Fig. 3—The DNL circuit, four transistors, six diodes, and a handful of common components, can easily be made up into a neat set-side box—no bigger than a double pack of 20 cancer-sticks.

Adding together the processed and unprocessed chains should now, theoretically, give a signal whose low-level high frequencies have a quietened effect, while middle and low frequencies are unaltered and where the higher volume high frequencies are given their full, required weight. In theory, once again, the result should be a true replica of the original, but without the hiss.

And, I must admit, despite some initial misgivings because Philips demonstrated this device to us a year or so ago in an hotel room whose air-conditioning added some 30 dB to the ambient noise, the subjective effect is a cleaner sound, whatever the condition of the recording.

But I still feel that the answer is not to use a circuit that gives, as Philips claim, a 10 dB improvement of S/N ratio at 6 kHz and a 20 dB improvement at 10 kHz on replay, but to improve the overall record/replay process in such a way as to retain its original sound structure, not "tailor" it. Again, if you must have slow-speed, narrow-track recording, then you have to engineer out the hiss, not allow it to happen and *then* try to beat it.

So we come to Dolby and the now-famous stretching process that Dr. Ray Dolby pioneered. The original "A" process aimed at beating the "breathing" that compansion procedures forced disc users to suffer and cost more than some recording companies could afford. It begins its work during recording, splitting the audio path into a direct and a rectifier chain. But the expensive "A" system did this in four bites, carving up the frequency spectrum to give differential gain depending on signal level within the frequency bands. These are: below 80 Hz, from 80 to 3,000 Hz, above 3,000 and again above 9,000 Hz.

Both hiss and hum are present in the recording process, and while hum can be relegated to one low portion of the audio spectrum, hiss is a very different problem. It obtrudes into the very region where our ears happen to be most sensitive. It has measurable components that extend way upwards into what some engineer colleagues of mine call the "annoyance passband." Any crude way of militating against hiss will mutilate the upper frequencies which we need to preserve to get the clash and tingle of a full musical experience.

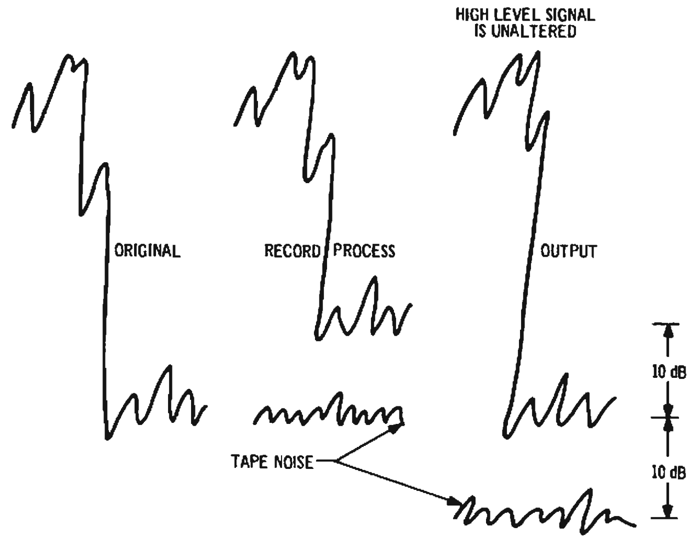


Fig. 4—One way of explaining the Dolby system: The original signal has its lower levels down around the system noise. Processing during record gains some 10 dB of S/N ratio. Replay retains this, raising the lower levels of signal that much above the noise.

Again, the procedure is to let the noise remain when the music is loud enough to mask it. Masking—as a technical term—is a peculiar business. It depends as much on relative frequencies as on loudness, and has some strange anomalies to do with time difference and phase factors. Subject for a later discourse, maybe. At present, please take my word for it that the phenomenon happens, and by letting the main, high level signals straight through the system, Dr. Dolby follows the method we have roughly outlined already.

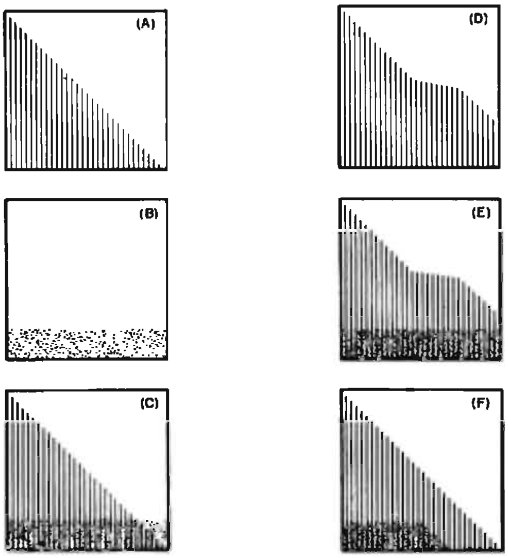


Fig. 5—An alternative explanation, as depicted by Dolby: A, music is made of sounds of different loudness with intervals of silence; B, noise of some kind is inescapable; C, when a tape recording is made and replayed the noise interferes with the low level signals, spoiling the program; D, the Dolby system boosts the lower signals during recording; E, those lower signals are still above the annoying noise during replay, as shown in F, the composite picture of the reconstituted sound with noise "reduced" by the carefully engineered boost and stretch system.

Fig. 6—Noise reduction units can be added quite easily to existing equipment. This Advent 100A has been enthusiastically received, despite the $250.00 price tag. My own special interest is harmonic distortion, and I was interested to note that the 100A was under 0.4% to 0 dB and less than 0.2% at lower levels. Output noise, -60 dB; noise reduction around 10 dB above 4 kHz, about 3 dB at 6 kHz. This is a stereo unit and well worth considering for slow-speed recording.

The subtlety lies in the treatment of the low-level signals, where noise is obtrusive. Dolby calls this the differential component, and this is, of course, relatively small—and hence more difficult to handle. It has to be remembered that the noise reduction system does not eradicate noise; it boosts weak signals to improve the signal-to-noise ratio, that's all.

That's all! Pause for hollow laughter! Arguable decisions are the threshold limit, below which noise-plus-signal will be processed, attack time, the response of filter circuitry to the information that a signal in need of treatment is coming along, the amount and nature of compression, and the way of ensuring a mirror image expansion and an avoidance of overshoot (which would process signals that did not need such treatment).

If the distortion has a duration of less than a millisecond, it will defeat the human ear. This is a smaller fraction than normal signal transients and our aural loudness-growth characteristic cannot distinguish the short-lived distortion.

The Dolby "B" system came into being when Ray Dolby was asked to dream up a modified noise reduction device for use with domestic equipment. The only feasible way to keep such a system within our budget was to forgo the technical requirement of four passbands and operate over the whole audio spectrum, this time making the sensor part of the apparatus listen for frequency as well as loudness, on a kind of sliding scale.

The system comes into action at about 600 Hz, with a maximum 3 dB effect. (O.K., so the ads say it extends above 2 kHz, but the sliding scale method means it really begins lower down). At 1.2 kHz it has a maximum 6 dB effect, has 9 dB at 2.4 kHz and reaches the advertised 10 dB above 4 kHz. The compression comes in about 45 dB below what has become known as the "Dolby level." This can be defined as a flux level on tape of 200 nanowebers per meter. Call this 0 VU.

In more technical terms, the differential chain splits into the rectifier path and into the linear path to the mixer for readdition to the main signal. The rectifier path contains boost circuits giving a 6 dB per octave flip to the higher frequencies. Then the output is rectified. This rectified signal effectively alters the dynamic resistance of an FET at the input end of the chain, and so gives a boost at low dynamic levels and practically no boost at high levels. By the simple device of driving the FET via a small coupling capacitor, Dr. Dolby achieved both a drop in gain with an increase in dynamic level and a change of the turnover frequency of the "threshold" as the level changes. The sliding scale, in fact.

At low levels the capacitor lets the FET see the full signal and gives a good 10 dB boost above 2 kHz. Increase the input level and the frequency above which this full boost is given begins to rise. Turn up the wick still more and the treble boost in the rectifier chain stops the over-saturation of the tape. To reinterpret, that means the tape is driven to its full limit when need be, at high dynamic levels (of original signal), but is allowed up to a 10 dB boost at lower signal levels. The replay mode is reciprocal.

The entire processed chain is inserted in a feedback loop around the main chain to subtract instead of add. The elegance of the system is that the same basic circuitry, and, indeed, a mirror-image printed circuit board makes production costs tumble and the add-on Dolby units now available should be within any enthusiast's purse. (Dolby IC chips are also coming soon.—Ed.)

Fig. 7—Slim, elegant, technically precise, one section of the Dolby A system as used by professional recording bodies throughout the world. Having had the chance to "rip one to bits," I can vouch for its engineering excellence.

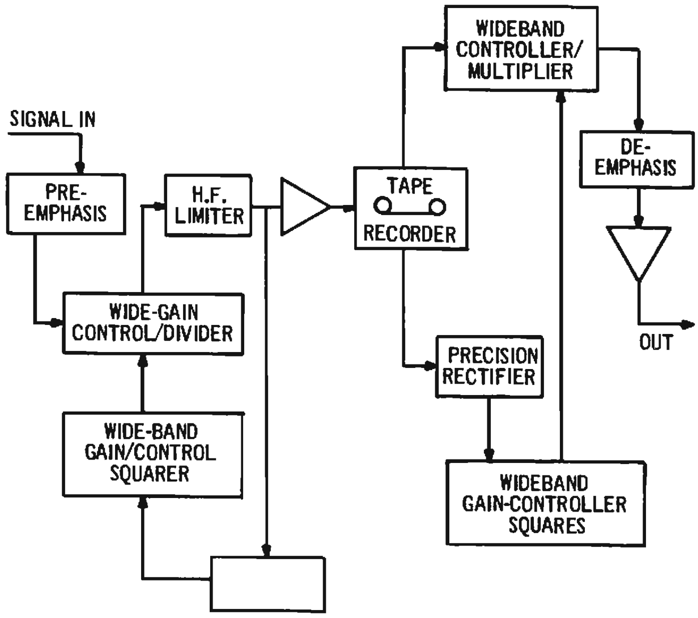


Fig. 8—Block diagram of the Burwen system, with refinements like active transformers and direct play equalizers omitted. The heart of the system is the rectifier module, monitoring the gain of two channels simultaneously in the "domestic" system. Operational amplifiers are used widely in this system with very high accuracy as a result.

My own tests with those available in the U.K. have confirmed that signal-processing of cassette-recorded music, speech, and sound effects have done wonders to guard against hiss and have not made detectable any audible worsening of the prime signal.

After Dolby, what? Well, according to Richard Burwen, quite a lot. In the December, 1971 issue of the Journal of the Audio Engineering Society, I came across the Design of a Noise Eliminator System which gave me much brain-searching and is at present exercising the pundits in those polite tomahawkeries of the erudite correspondence columns. (See also AUDIO, June, 1971.)

To begin with, the title of Richard S. Burwen's paper hits a sore point. The only way you eliminate noise, truly, is not to cause it. After the die is cast, all you can do is guard against it—which we have seen three different systems doing in the preceding notes.

Mr. Burwen took the critics by the ears at the 41st convention in New York on October 5, 1971. In February of that year, a paper of his entitled "A Dynamic Noise Filter" had aroused comment. He is more concerned with studio tape machines, just as Dr. Dolby was, and there seems little hope, at present, of such an elegant "domestic" solution to the noise reduction problem with a plain man's Burwen. But anyone who has been in the audio field as long as us (well, me) knows better than to say that something, anything, cannot be done.

So let's conclude with a brief look at Mr. Burwen's solution.

He set himself some pretty high parameters. His system was not, he told us, to exceed the present 1%, and preferably 0.5%, distortion level of good taping. He wants to record live music "with no audible noise whatsoever." So his first experiments were to determine peak recording levels.

Recording to +3 VU, a normal process, when 0 VU is the standard set limit and peaks above this as much as +6 VU are occasionally tolerated because of their short duration, meant that distortion on tape went over that critical 1%. He concluded—first point, and first stumbling block for his critics—that it is not always advisable to retain every peak.

Listening tests revealed that for noise to be negligible in the absence of program material, it had to be 90 dB or more below the 1% distortion level, i.e., better than -84 VU. Then he found that noise 65 dB down was audible with a 500 Hz sine wave but masked by frequencies above 3 kHz. You could reduce the bandwidth to about a half-octave centered on 500 Hz and get a pure tone—so the solution seemed to be split the waveband, per Dolby A.

But the multiband system, according to Richard Burwen, has the disadvantage of frequency response errors in the tape machine causing errors in the expansion process. The solution was to use the whole band but compress the 90 dB expected input to 30 dB at the tape. He then combined the principles of his dynamic noise filter (see June, 1972 issue of this magazine) with a single wideband compandor.

The dynamic noise filter acts as a low level expander at top and bottom of the frequency spectrum—again, something like we've seen before. Adding a high and low-frequency compression system seemed to be the answer, and high frequency pre-emphasis was intended to improve the S/N ratio. Some hellish problems raised themselves at this point, and Mr. Burwen went back to the drawing board. He finally produced three systems, A, B and C. Characteristic A is optimized for studio recording at 15 ips. It has a dynamic range of 110 dB and this is the one you'll see hailed in the ads! System B operates more modestly to give a 102 dB dynamic range at 7½ ips, and C is the one that may eventually interest us at 3¾ or 1⅞ ips for FM broadcasting, records or background music. If you want it in the words of the master: "The system . . . utilizes high and low frequency pre-emphasis and a single wideband cube root compressor to produce the recorded signal, and a complementary expandor and pre-emphasis for playback."

The important point slipped in later is that in the single-band system the frequency response is constant and is not affected by inaccuracy in the tape machine. Again, we shall leave the pundits to argue.

The high performance of the Burwen circuitry has been made possible by the low-noise two-quadrant multiplier/divider. Bettering Dolby by one magnitude in claim and applicable also to FM systems, it seems to offer possibilities, and we must wait and see what the outcome may be.

For my part, in this noise-polluted environment, I welcome any device that can help rid us of clamour. But noise is what you make it, and the tick of an obtrusive clock, as many an amateur recordist has found, can be as bothersome as a traction engine. The subjective results, applied to cassette, have been enormous—praise to the noise-breakers! Æ

The Tape Guide

All About Tape Recorder Equalization

Herman Burstein

FROM READERS' QUESTIONS and things that appear in the popular audio literature, it seems that tape recorder equalization is less well understood by audiophiles than its importance deserves. The mystery tends to be compounded by the variety of equalization characteristics necessitated by an assortment of tape formulations and tape speeds, as well as by the occasional promulgation or advocacy of new equalization standards.

Therefore the Tape Guide seeks to explain what tape recorder equalization is all about—why it is needed; how it is achieved; how it can be modified to optimize the interdependent requirements of extended treble response, low noise, and low distortion; the nature of the NAB equalization standards, and how equalization is affected by such things as tape speed, tape formulation, use of the Dolby system, etc.

For the most part the discussion assumes that tape speed is 7½ ips. Despite the greatly improved performance obtainable at lower speeds, 7½ ips is the NAB (National Association of Broadcasters) standard speed and is the one generally preferred for high quality home recording. (In fact, some home recordists prefer 15 ips.) In any event, what we have to say applies in principle to all tape speeds.

It is further assumed that the tape machine has: (1) clean and demagnetized heads so that no treble losses occur due to poor tape-to-head contact or magnetization; (2) heads in perfect azimuth alignment (gaps exactly at right angles to tape length) to eliminate treble losses owing to incorrect azimuth; (3) bias set for approximately minimum distortion at mid-frequencies, specifically at 400 Hz.

Why Equalization Is Necessary

Suppose that the tape machine we have just described has no equalization circuits to alter frequency response, and that it is employed to record and play a tape at 7½ ips. Further suppose that input signals of *constant level* are recorded throughout the 20-20,000 Hz range and then played back. A meter connected to the output would typically show the machine's record-playback response to be quite similar to Curve ABC in Fig. 1: Record-playback response climbs steadily at virtually 6 dB per octave (20 dB per decade, to be precise), reaches a peak around 3,500 Hz, and drops substantially thereafter.

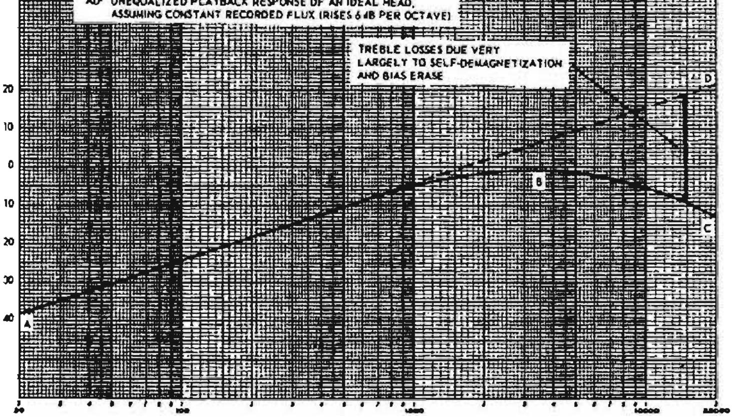


Fig. 1—Unequalized record-playback response of a tape recorder at 7½ ips.

Clearly, bass boost is needed to compensate for the drooping bass portion AB, and treble boost is needed to compensate for the drooping treble portion BC. That is the role of equalization—to provide bass and treble boost made necessary by the inherent nature of the tape recording process.

To help us see why Curve ABC is the way it is, Fig. 1 supplies line AD, which is the response of an "ideal" (perfect) playback head if a tape were recorded flat; that is, if the tape contained recorded flux of equal magnitude at all audio frequencies. At this point let us carefully note an important distinction between *applying* a flat signal to the tape and *recording* a flat signal on the tape. Losses, which we describe shortly, take place in the treble range of the recorded signal. However, line AD assumes there are no such losses so that a flat signal is recorded on the tape. In sum, AD is the playback response of an ideal head if the tape is *recorded* flat.

AD rises steadily at 6 dB per octave because the head is a velocity device. That is, the head is an electromagnetic generator with a voltage output proportional to the rate of change of the magnetic field of the tape. The field changes at a rate corresponding to the audio frequency. Hence the voltage output of the head is proportional to audio frequency. For example, at 10,000 Hz the playback head produces twice as much output as at 5,000 Hz, and 10 times as much as at 1,000 Hz. (That is why we say line AD rises 6 dB per octave or 20 dB per decade, since 6 dB represents (very nearly) a doubling of voltage, and 20 dB represents (exactly) a 10-fold increase).

Beyond approximately 800 Hz the record-playback curve fails to continue its 6 dB per octave climb due almost entirely to magnetic losses that occur *in recording* and become more severe as frequency increases. These losses are of two kinds, self-demagnetization and bias erase, and we shall return to them in a moment. There are also slight losses—especially slight at higher speeds—attributable to the playback head. Winding capacitance of the playback head may result in treble loss. And there may be some treble loss due to gap width (the wider the gap of the playback head and the slower the tape speed, the greater the loss.) But with a playback head that is well made and boasts a gap as narrow as 40 or 50 microinches, capacitance and gap losses at 7½ ips are usually quite negligible—on the order of 1 dB or less at 20,000 Hz. Thus we are principally concerned with recording losses described by the terms self-demagnetization and bias erase.

Self-demagnetization refers to the fact that the recorded signal on the tape in effect consists of a series of bar magnets end to end. The higher the frequency, the more bar magnets are recorded per inch of tape, so that each magnet is necessarily shorter. But the shorter the bar magnet, the closer together are its north and south poles, and the more their opposing magnetic fields tend to cancel; that is, the signal tends to self-demagnetize. In sum, with increasing frequency the strength of the recorded signal—the amount of magnetic flux on the tape—tends to weaken.

Bias erase is a side-effect of the high frequency signal, typically 75,000 Hz or higher, which is fed in moderate amount to the record head to minimize distortion and generally maximize the amplitude of the recorded signal; this is called bias current. In much greater quantity, about 10 times as much, the oscillator current powers the erase head. Unfortunately, bias current in the record head has the deleterious side effect of also accomplishing erasure—not nearly as effective as the erase head, but erasure nonetheless. Bias erase increases with frequency because the higher frequencies penetrate the tape less deeply and hence are more vulnerable to an erasing field. Altogether, bias current produces treble loss; the larger the bias current (for reduced distortion), the greater the treble loss.

The magnitude of the magnetic losses in recording is indicated by the interval between Line AD and Curve ABC. Recall that AD is the response of an ideal playback head in the absence of recording losses, and that an actual high quality playback head is close to ideal. Therefore the interval between AD and ABC represents recording losses. For example at 15,000 Hz the interval shows a loss of about 30 dB. Roughly 20 dB of this may be ascribed to self-demagnetization and the other 10 dB to bias erase.

Equalization for Flat Response

A major goal of high fidelity is of course flat frequency response, output signals having the same relative levels as

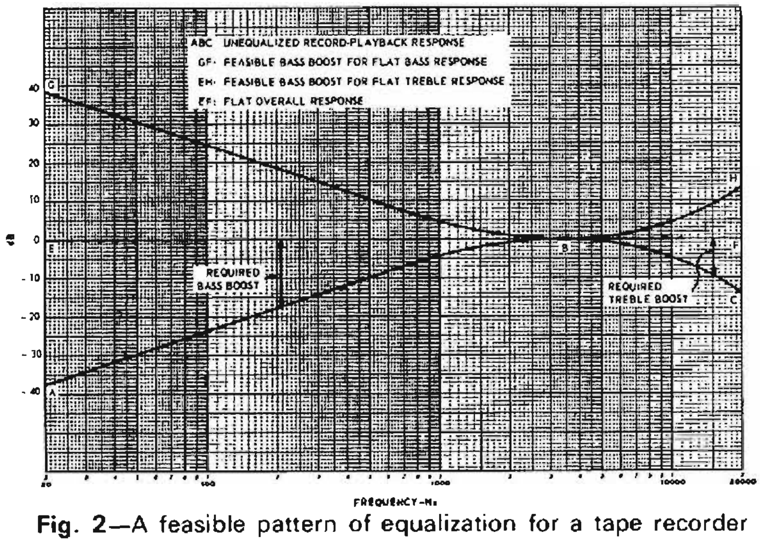


Fig. 2—A feasible pattern of equalization for a tape recorder at 7½ ips.

the input signals at all audio frequencies. In other words, for constant level input there should be constant level output in the range of approximately 20 to 20,000 Hz. Hence flat response is represented by a straight horizontal line, such as EF in Fig. 2.

Record-playback Curve ABC from Fig. 1 is repeated in Fig. 2, and a feasible scheme of equalization for flat response is straightaway evident. The interval between EB and AB may be interpreted as bass loss, and therefore represents the bass boost needed for flat bass response. Similarly the interval between BF and BC represents the needed treble boost. Accordingly, GF is a suitable bass equalization curve, rising in a fashion that mirrors the decline AB. And EH is a suitable treble equalization curve, rising in a fashion that mirrors the decline BC. Together, GF and EH complement record-playback Curve ABC to produce flat response.

Figure 2 is a workable scheme of equalization and something fairly like it is used. However, matters are not all that simple. In addition to flat response, high fidelity has low noise and low distortion as major goals. For reasons connected with improving the signal-to-noise ratio, actual equalization (the generally employed NAB standard) is a modified version of GF and EH in Fig. 2. But we must postpone, and pave the way for, discussion of NAB equalization in order to deal first with the question of where equalization takes place in a tape machine so as to best serve the triad of goals—flat response, low distortion, and low noise.

Where Equalization Takes Place

We begin with an important observation. Figure 1 shows great treble losses in recording, reaching about 30 dB at 15,000 Hz and 36 dB at 20,000 Hz. Yet Curve EH in Fig. 2 indicates that a treble equalization curve with only 10 dB of boost at 15,000 Hz is needed. The seeming paradox is explained by the fact that treble boost is not required to fully make up for treble losses. Only enough treble boost is needed to achieve flat response. Putting it differently, the rising response of the playback head (the portion of AD above 800 Hz in Fig. 1) compensates for a substantial part of the treble losses. Only the remainder of the treble losses must be made up by Curve EH in order to achieve flat response in the treble range. Thus we note that rising response of the playback head has a key role in treble compensation.

Where should equalization circuits be placed in the tape machine? One might expect that they could be placed in the record amplifier, or in the playback amplifier, or in a combination of the two. However, not just any combination will do, because some offer better results in terms of noise and distortion, while others offer worse.

Without yet explaining anything, one may offer a descriptive general rule: *Playback losses are equalized in playback and record losses in recording.* Thus Curve GF in Fig. 2 would be supplied by an equalization circuit in the playback amplifier. And curve EH would be supplied by an equalization circuit in the record amplifier.

Why this general rule? If the large amount of needed bass boost were supplied in recording, this would tend to apply excessive signal (magnetic field) to the tape and overload it, resulting in excessive distortion. Alternatively, one would have to greatly lower the recording level, resulting in a poor S/N ratio. Therefore bass boost is applied (largely or altogether) in playback. Moreover, Curve GF in Fig. 2 may be viewed in the guise of a treble cut characteristic. In this vein it serves to reduce noise of the entire tape recording system when used in playback.

Turning to treble boost, we must consider that noise, while prevalent at equal *amplitude* throughout the audio spectrum, is usually most evident from about 3,000 Hz upward. This is partly because of the human ear's sensitivity in the vicinity of 3,000 to 5,000 Hz. Mainly it is because of the increasing amount of random noise energy as one goes up each octave of the audio spectrum; the more frequencies per octave, the more must be the noise energy. (Clearly there are more frequencies between, say, 3,000 and 6,000 Hz than in the preceding octave of 1,500 to 3,000 Hz.) Thus we characterize tape and amplifier noise as high-pitched (hiss, spitting, frying, etc.) even though low-pitched noise is also present. If treble boost were applied in playback, it would magnify tape noise and noise of the record and playback amplifiers, resulting in a poorer S/N ratio. Therefore treble boost is applied instead (largely or altogether) in recording, where it only magnifies noise of the record amplifier.

A logical and important objection is in order at this point: Won't the treble boost in recording overload the tape (much as a large amount of bass boost might)? The answer would be yes if, for typical sounds, all frequencies had equal peak amplitudes. But for most recorded sounds desired by humans, particularly music, amplitude is usually a good deal less at high frequencies than at mid-range ones, as suggested in Fig. 3. This figure shows for a typical orchestral selection the relative peak levels of audio energy throughout the spectrum.

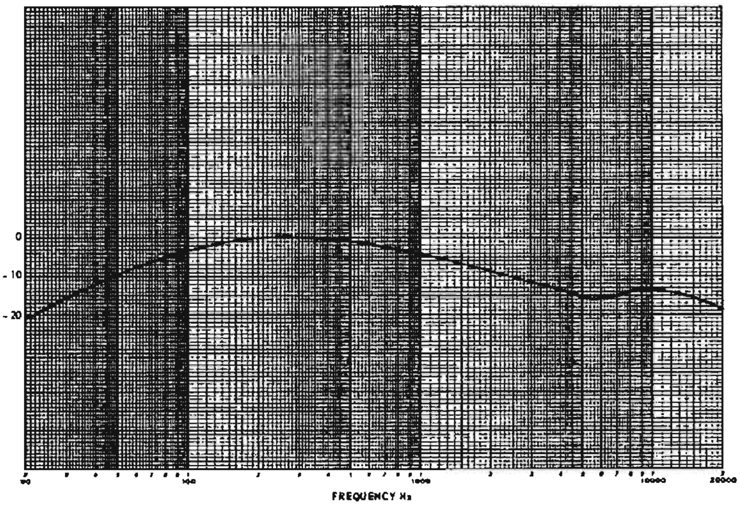


Fig. 3—Smoothed approximation of relative peak amplitudes for a typical orchestral selection.

Compared with peak amplitude at 400 Hz, there is a dropoff of 15 dB or more at higher frequencies.

Therefore, in dealing with the kind of sound generally recorded, a good deal of treble boost is feasible in recording. This boost is offset by the decline in amplitude of the higher frequencies, which helps prevent the tape from being overloaded. (Another preventative, when necessary, is reduction of recording level by the user.)

A Modified Pattern of Equalization

We have already noted that low noise is one of the three major goals of high fidelity. Put differently, we are interested in high S/N (signal-to-noise ratio). This can be achieved by recording more signal on the tape, especially at high frequencies, where the extra signal can mask the noise.

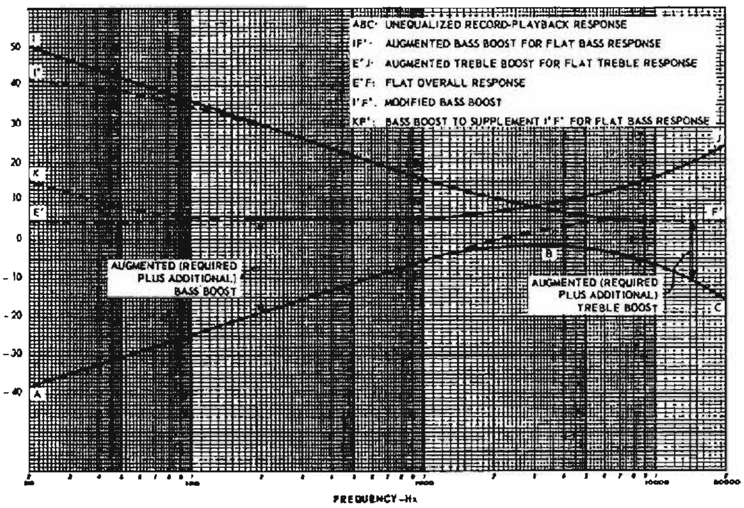


Fig. 4—A modified pattern of equalization to obtain improved signal-to-noise ratio (at 7½ ips).

Figure 4 shows a modified pattern of equalization that uses additional boost in recording yet results in flat response. Desired flat response is denoted by Line E'F'. It is 6 dB higher than the corresponding line EF in Fig. 2, reflecting an improvement of 6 dB in S/N ultimately achieved at higher frequencies.

Curve ABC, as before, is the unequalized record-playback response. The interval between AF' and ABC is the treble boost needed to approach flat response at the higher frequencies; that is, response 3 dB below flat at about 3,200 Hz, and increasingly flat as frequency rises. We may refer to this interval as "augmented" treble boost, consisting of the amount originally required in Fig. 2 plus an additional amount for higher S/N. (The interval between BC and the O dB line is the originally required treble boost, so that the remainder of the interval between AF' and ABC is the additional boost.) Thus Curve E'J is the augmented treble equalization needed for flat treble response. (E'J is the same distance from E'F' as ABC is from AF'.)

The required bass boost is the interval between flat response E'F' and drooping response AF' (keep in mind that AF' represents the response of an ideal head to a recording made with treble boost E'J). We may refer to this interval as "augmented" bass boost because it consists of the amount originally required in Fig. 2 plus an additional amount. (The interval between AB and the O dB line is the originally required bass boost, so that the remainder of the interval between AF' and E'F' is the additional bass boost. Thus

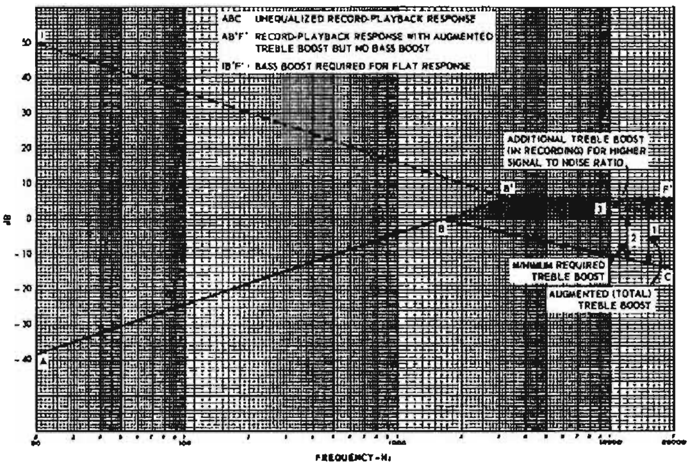


Fig. 4A—Basic scheme of the modified pattern of equalization.

Curve IF' is the augmented bass equalization needed for flat bass response. IF' is the same distance from E'F' as AF' is from E'F'.)

The story told by Fig. 4 is somewhat complex. It can be made more clear by presenting its essentials in simpler form in Fig. 4-A. (We haven't yet finished with Fig. 4 and shall return to it shortly.) ABC in Fig. 4-A represents in linear form the unequalized record-playback response. Line AB'F' shows the record-playback response that would result if there were only augmented treble boost and no bass boost. Augmented treble boost is depicted by Arrow 1, minimum required treble boost by Arrow 2, and additional treble boost by Arrow 3. Additional boost is further spelled out by the shaded area between BB'F' and BC. Given record-playback response AB'F', it remains to supply bass boost IB'F' in order to achieve overall flat response. AB'F' in Fig. 4-A corresponds to AF' in Fig. 4; and IB'F' in Fig. 4-A corresponds to IF' in Fig. 4.

Returning to Fig. 4, we may ask: Why stop at a 6 dB improvement in S/N at the higher frequencies? Why not add yet more treble boost in recording to achieve still higher S/N. The answer lies in Fig. 3, which shows that, relative to 400 Hz, the higher frequencies are down roughly 15 dB at 15,000 Hz and a bit more at 20,000 Hz. Correspondingly, this allows treble boost of about 15 dB at 15,000 Hz and a bit more at 20,000 Hz without excessive risk of serious tape distortion. Treble boost curve E'J does just about that, with no margin of safety to spare. In other words, in the present state of the art, treble boost in recording which approximates E'J is about as far as one dare go without risking excessive distortion in the upper end of the treble range. (Here lies the reason why some recordists still prefer 15 ips. Treble losses in recording are less than at 7½ ips, so that less treble boost is needed in recording and there is less risk of overloading the tape due to such boost. The recordist speaks of the greater "headroom"—margin between the amount of treble signal applied to the tape and the amount which causes tape saturation—available at 15 ips.)

It is difficult in practice for a tape amplifier to fully supply the amount of bass boost indicated by Curve IF' in Fig. 4. For one thing, an enormous amount of amplification is needed to achieve bass boost which at 20 Hz is up 44 dB from the reference line E'F' and still rising. High amplification is costly, may unduly magnify hum frequencies, and entails the risk of oscillation owing to phase shift or stray feedback. Therefore a preferred course is to allow bass boost to level off, as shown by I'F' in Fig. 4. Now bass boost is up about 35.5 dB at 20 Hz and soon reaches a maximum of 36 dB below 20 Hz.

To compensate for the levelling off of bass boost in playback, some bass boost may be introduced in recording, as shown be Curve KF' in Fig. 4. However, this isn't always necessary, because at low frequencies the playback head often tends to exhibit a slight rise in response owing to what is called the contour effect. Low frequencies correspond to long wavelengths (bar magnets) on the tape. In the presence of long wavelengths, the entire playback head, not only its gap, tends to react to the magnetic flux of the tape. The resultant rise in bass response may approximate KF' well enough to obviate the need for bass boost in the record amplifier.

If bass boost is supplied in recording because the contour effect is minimal, ordinarily this raises no problem of overloading the tape. Figure 3 shows that the typical decline of peak amplitudes at low frequencies would easily offset the bass boost of Curve KF'.

A final and key note on Fig. 4: Comparing this with Fig. 2, we observe that an equalization pattern which calls for increased treble boost in turn requires greater bass boost. Conversely, a decrease in treble boost is accompanied by a decrease in required bass boost. *In sum: A given recording characteristic implies a complementary playback characteristic; or a given playback characteristic implies a complementary recording characteristic.*

NAB Standard Equalization

The approach of Fig. 4 is followed by the NAB standards for tape recorder equalization. In fact, I'F' is the NAB playback equalization *characteristic* for 7½ ips (and for 15 ips as well). For greater clarity, I'F' is repeated in Fig. 5. We underscore the word "characteristic" because the NAB standard does not merely describe the frequency response of an equalization circuit in the playback amplifier (as is the case for RIAA phono playback equalization and for FM tuner equalization). True, measured frequency response of the tape

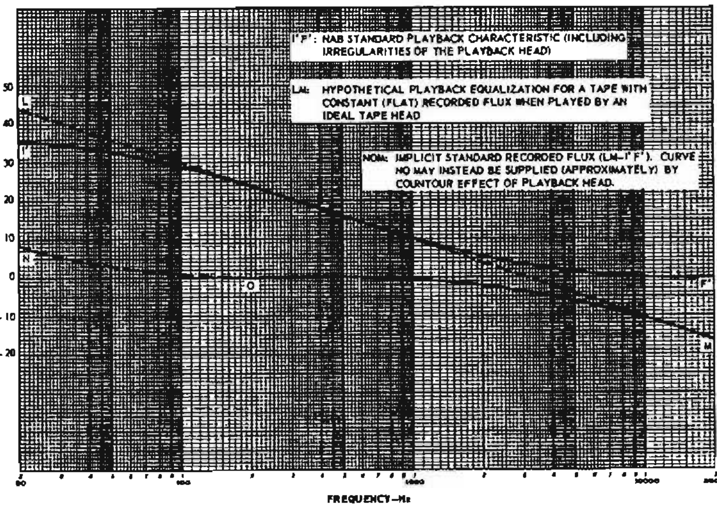


Fig. 5—NAB standard playback equalization and recorded flux.

playback amplifier is ordinarily quite close to Curve I'F', but it is not necessarily the same as I'F' in order to achieve flat response. The NAB playback characteristic in Fig. 5 is the *sum* of playback equalization provided by the tape amplifier plus irregularities in frequency response of the playback head. As already discussed, these irregularities tend to consist of some boost in the low bass region and a slight dropoff in the high treble region.

To illustrate, assume that a playback head is significantly deficient in high treble; that is, its output rises with frequency but at less than the theoretical 6 dB per octave rate of an ideal head throughout the audio spectrum. Then, if flat response is to be maintained, the playback amplifier must supply enough treble boost to compensate for the head's deficiency. *Together,* the amplifier and playback head supply the NAB playback characteristic, Curve I'F' in Fig. 5 (or something like I'F' if the contour effect is pronounced).

Without elaborate test equipment, how is one to ascertain whether a tape machine (playback amplifier plus playback head) provides the NAB playback characteristic? The answer lies in a standard test tape. (This is supposed to have been available from NAB by now, but hasn't yet been released. In its place, the Ampex test tape is customarily used.) The test tape contains a series of audio signals recorded at such relative levels that a tape machine with the NAB playback characteristic will provide flat response when playing this tape. That is, a meter connected to the machine's output will read equal output level for all the test frequencies.

Accordingly the manufacturer of a tape machine designs the playback equalization circuit to yield flat response when playing the standard test tape. The equalization circuit allows for bass and/or treble irregularities of the playback head he uses—that is, departures of the actual head from the response of an ideal head (AD in Fig. 1).

Some machines include adjustments which enable the technician or the user to touch up playback equalization on the basis of a test tape. It is then merely necessary to connect a meter to the machine's output, play the test tape (after making sure heads are cleaned, demagnetized, and aligned for azimuth), and touch up the playback equalization for flattest response as indicated by the meter. In some machines the VU meter can serve this purpose.

Now, what about treble boost in recording? Does NAB specify a treble recording characteristic in terms of what gets on the tape? Or does it specify a given amount of treble boost in the record amplifier, such as Curve E'J in Fig. 4? The answer is that, at least *directly,* NAB does neither. What NAB specifies is that, after playback equalization has been

(Continued on page 79)

(Continued from page 32)

adjusted for flat response when playing a test tape, the record equalization should be adjusted for flat record-playback response.

There are good reasons for NAB not specifying a specific treble boost curve in the record amplifier: (1) Treble losses in recording vary among types and brands of tape, so that the amount of required treble boost varies according to the tape which the manufacturer envisions will be used with his machine. (2) These losses vary according to the amount of bias the machine manufacturer elects to use. He tries to come fairly close to the bias which achieves minimum distortion. However, minimum-distortion bias may entail excessive treble loss, causing the manufacturer to reduce bias somewhat rather than increase treble boost in recording and thereby heighten the risk of overloading the tape. In other words, the manufacturer may find that the increased distortion resulting from a slight decrease in bias is not as bad as the increased distortion which would occur if more treble boost were used.

The NAB Recording Characteristic

Indirectly, NAB does stipulate a recording characteristic in terms of the relative levels of recorded flux on the tape at various frequencies in the audio range. To explain, let us repeat our dictum on recording/playback characteristics: "A given recording characteristic implies a complementary playback characteristic; or a given playback characteristic implies a complementary recording characteristic."

Hence the NAB recording characteristic is Curve NOM in Fig. 5. Given playback characteristic I'F' and flat response, then the flux recorded on the tape must vary with frequency in the manner portrayed by NOM.

NOM is derived as follows. We draw LM to show the hypothetical playback equalization required if the recording characteristic were flat and if an ideal playback head were used (or a playback head with its irregularities fully compensated in the playback amplifier). That is, LM declines 6 dB per octave with increasing frequency, thereby complementing the playback head's 6 dB rise per octave. However, any departure of the acutal playback characteristic (I'F') from LM implies a corresponding (complementary) departure of recorded flux from a flat characteristic. Thus NOM is the difference between hypothetical and actual equalization, namely between LM and I'F'.

At the low end, I'F' supplies less than the hypothetical bass boost. Therefore the deficiency must be made up in recording by bass boost NO (unless the deficiency is made up by the contour effect of the playback head). At the high end, I'F' does not drop as rapidly as LM; thus I'F' in effect is contributing treble boost. And a corresponding drop in treble must occur in recording, namely the treble decline of recording characteristic NOM.

The implied recording characteristic NOM—flux on the tape—is the sum of magnetic losses in recording, equalization in the record amplifier, and response irregularities of the record head. All told and together with playback characteristic I'F', they produce flat response.

If, after all, there is an NAB standard recording characteristic, why doesn't NAB specify this explicitly (in the way that the RIAA phono recording characteristic and the FM broadcast characteristic are specified)? The answer lies in the kind and quality of laboratory test equipment required in order to measure recorded flux. It is far easier for the manufacturer, technician, or user to check the playback characteristic with the aid of a test tape and meter than to measure recorded flux. Since a playback characteristic implies a matching recording characteristic when overall response is flat, then if the playback characteristic is known to meet the standard (on the basis of playing a test tape) the recording characteristic perforce also meets the standard.

This signifies that if my tape machine and yours both have flat record-playback response, and if both produce flat output when playing the standard test tape, mine will record tapes that play back flat on yours, and vice versa.

We have several times commented on NO in Fig. 5. To bring things together, at the cost of repetition, this slight bass boost may be achieved by an equalization circuit in the record amplifier, by the contour effect of the playback head, or by a combination of the two. Even if there is no appreciable contour effect, the manufacturer may choose not to supply bass boost in recording. This is consistent with the

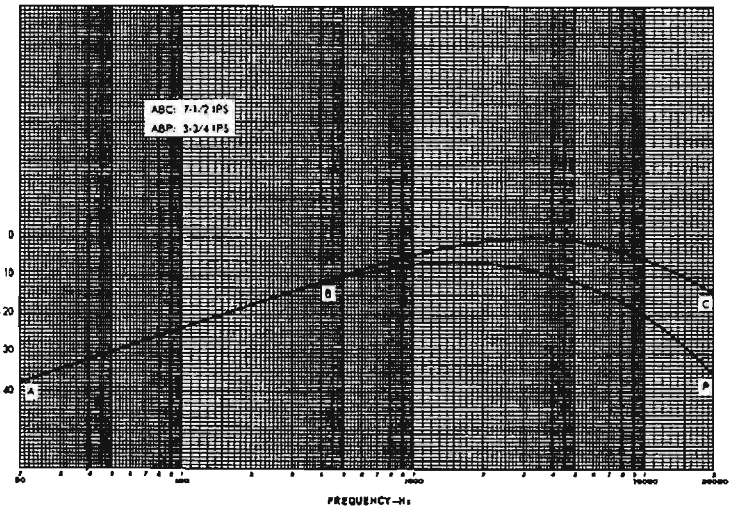


Fig. 6—Comparison of unequalized record-playback response at 7½ and 3¾ ips.

NAB standard for frequency response, which permits record-playback response to be down 3 dB at 50 Hz.

Other Speeds, Other Tapes, Other Matters

All that we have said about problems and techniques of tape equalization at 7½ ips also applies in essence to the lower (higher) speeds in home use. However, as speed is reduced, equalization must be changed, the principal reason being the greater magnetic losses that occur in recording at reduced tape speed. This is illustrated in Fig. 6, which compares unequalized record-playback response at two speeds, 7½ and 3¾ ips.

Why do recording losses increase with reduced speed? The answer lies in the fact that these losses actually depend on recorded wavelength rather than on frequency as such. We refer again to the length of the bar magnets that in effect are recorded on the tape. At a given frequency, a corresponding number of bar magnets are recorded per inch of tape. To illustrate, consider a 1,000 Hz tone, which is recorded as 2,000 bar magnets—one magnet for each positive portion of an audio cycle and one for each negative portion. At a tape speed of 7½ ips, the 2,000 magnets are recorded on 7½ inches of tape, so that the length of each magnet is 7.5/2,000 inches or .00375". But at 3¾ ips, the length is 3.75/2,000 inches or .001875"—half as long. Earlier we pointed out that the shorter the wavelength (bar magnet), the greater the loss due to self-demagnetization and bias erase. Thus at a given frequency the wavelength is reduced and the recording loss is increased as tape speed is reduced. In going, say, from 7½ to 3¾ ips, losses are of the same magnitude at 5,000 Hz as they were at 10,000 Hz at the higher speed; losses are of the same magnitude at 10,000 Hz as they were at 20,000 Hz; etc.

To offset the greater treble loss at low speed, more treble boost is needed in recording. But the requisite amount would

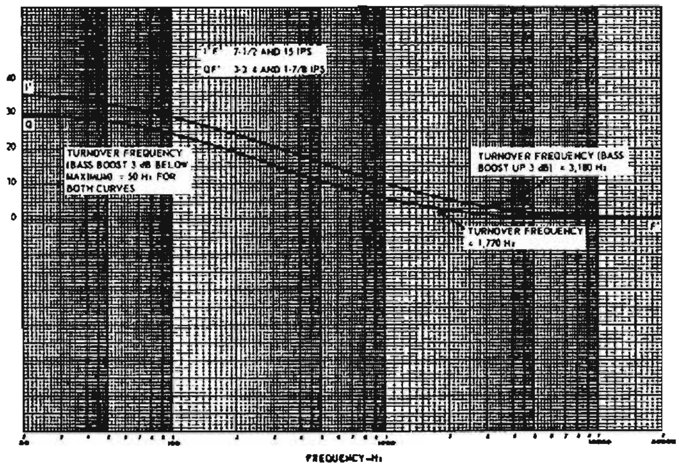


Fig. 7—NAB standard playback equalization curves (including playback head characteristics).

overload the tape and cause excessive distortion. A viable alternative is to reduce treble boost and match this with reduced bass boost in order to attain flat response. As pointed out in our dictum on record/playback characteristics (in conjunction with Figs. 2, 4, and 4-A), reduced treble boost correspondingly calls for reduced bass boost in order to achieve flat response.

In this light the NAB standard for 3¾ and 1⅞ ips calls for a playback characteristic with 5 dB less bass boost than the characteristic for 7½ ips (which is also the playback characteristic for 15 ips). Fig. 7 shows the two playback characteristics. Curve I'F' is the NAB standard characteristic for 7½ and 15 ips, and QF' is the standard for 3¾ and 1⅞ ips.

I'F' entails a total of 36 dB bass boost; QF', 31 dB. In technical terms, I'F' is described as having turnover frequencies of 3,180 and 50 Hz (or, respectively, time constants of 50 and 3,180 microseconds). This signifies that the curve has achieved 3 dB boost at 3,180 Hz, and at 50 Hz is 3 dB below maximum boost. QF' has turnover frequencies of 1,770 and 50 Hz (or, respectively, time constants of 90 and 3,180 microseconds), signifying 3 dB boost at 1,770 Hz, and boost 3 dB below maximum at 50 Hz. (The relationship between time constant t in microseconds and turnover frequency f in Hz is expressed by $t = 159{,}155/f$. The factor 159,155 derives from the technical relationship between the values of capacitance and resistance needed in an equalization circuit to achieve a 3 dB change in frequency response.)

According to Fig. 7, two playback equalization characteristics are used for four speeds, one characteristic for 7½ and 15 ips, and the other for 3-¾ and 1-⅞ ips. But the sense of this article is that each tape speed requires its own equalization in order to obtain optimum results with respect to frequency response, S/N, and distortion. In other words, why don't we have four standard equalization characteristics instead of two? The indicated answer is that four characteristics would complicate matters too much for those making tape machines, and perhaps for those using them. An equalization characteristic appropriate for a given speed can be used for the next higher speed without undue departure from optimum performance. So all in all, two playback characteristics for four speeds affords a practical compromise without unduly deleterious consequences.

What happens to equalization requirements as tape formulations change, resulting in higher output, lower noise, better treble, different bias requirements? So far as playback equalization is concerned, the answer in essence is *nothing*, or very little. Essentially it is only record equalization that changes (except in the unlikely event that changes in tape formulation would someday result in a new standard playback characteristic, which would be a very unhappy day for those with substantial collections of recorded tapes).

If a new tape has increased treble output, the tape machine manufacturer may choose not to extend treble response but to reduce the amount of treble boost supplied by the record amplifier, thereby lessening the chance of tape saturation. Alternatively, he can leave treble boost about the way it was, and increase bias to achieve less distortion, yet without reducing treble. Or, in increasing bias, his purpose may be not to reduce distortion but to permit recording at a higher level at the same distortion as before, thus improving S/N. Or the manufacturer may follow a compromise course which results in some combination of improvements in treble response, distortion, and S/N.

If a new tape has higher output, the machine manufacturer can in similar fashion take advantage of this to improve S/N, treble response, distortion, or a combination of them. Higher tape output enables the machine to apply somewhat less signal to the tape without sacrificing S/N; hence there is less danger of tape saturation, permitting more treble boost for better treble response; or permitting more treble boost together with more bias for lower distortion. Higher tape output enables the machine to reduce recording level for less tape distortion. Higher output enables the machine to keep bias and treble boost as before, with an increase in recording level to achieve higher S/N, yet without increase in distortion.

In similar ways, reduction in tape noise not only permits higher S/N but can also be translated into improvements in distortion and treble response, which might entail changes in treble boost, bias, or recording level. Changes in a tape's bias requirements do not in themselves entail changes in treble equalization. However, changes in bias requirements for a tape tend to accompany changes in noise, treble, and output characteristics of the tape, and it is then that the machine manufacturer may find it advisable to change record equalization.

Does use of the Dolby noise reduction system (or other noise reduction systems such as the Burwen) affect tape equalization? The answer, essentially, is only in recording if at all. Introduction of the Dolby system in a tape recorder installation does not necessarily call for a change in recording equalization. One records as before, except than the incoming signal first goes through a treble-boosting Dolby circuit; and one plays as before, except that the playback signal afterward goes through a matching treble-reducing Dolby circuit to achieve noise reduction and restoration of flat response. Overall record-playback response is not changed by Dolby, which therefore does not *compel* a change in tape equalization.

On the other hand, the Dolby (or a similar) system may *invite* changes in equalization for reasons already suggested here. To illustrate, assume that Dolby achieves a 10 dB reduction in noise. The tape machine manufacturer might decide to sacrifice part of this improvement, say 4 dB, in exchange for better treble response. He could adjust the tape machine (adjust the reading of the record-level indicator) to operate at 4 dB lower recording level, permitting 4 dB more treble boost without increase in danger of tape saturation. Or he could exchange part of the Dolby reduction in noise for improvement in distortion, by using more bias along with more treble boost, as well as a lower recording level. Or he could achieve some improvement in all three respects—noise, treble response, and distortion. Æ

LOGIC CONTROL OF TAPE RECORDERS

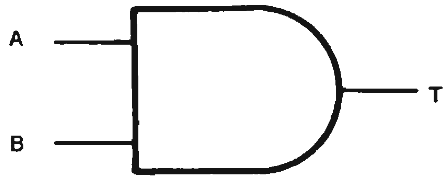


Fig. 1—Symbol for logic circuit controlled by DC voltage levels

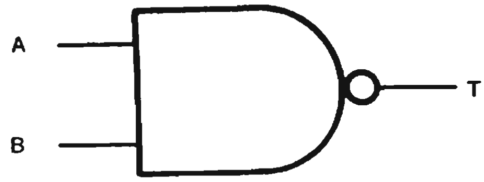


Fig. 2—Symbol for NOT-AND (NAND) gate

***Ole Melvold**

The seemingly simple process of transporting tape across the electronic magnetic heads in a tape recorder, so that signals can be recorded on and played back from the tape, relies upon a complex and often critical relationship between the electrical portion of the recorder and its mechanical parts.

The mechanical portion of this process is often complicated and is usually the source of the majority of malfunctions and failures in any recorder.

The problem faced by the designer is, then, to simplify this mechanical operation and, if possible, to replace as many purely mechanical functions by electrical ones to obtain greater reliability. The use of purely electronic controls will not only yield greater reliability but, of course, will provide faster, more convenient operation including all the tape handling processes, start, stop, wind, etc. Electronic, "finger tip" activated controls can, therefore, be used to go directly from any wind mode directly to a play mode without manually engaging the conventional "stop" function and with no additional stress imposed upon the tape, and with greatly reduced acoustical and electrical noise. The tape recorder lends itself to electronic pulse control which, in turn, facilitates new applications besides the preservation of audio signals. Recorded signals can be used

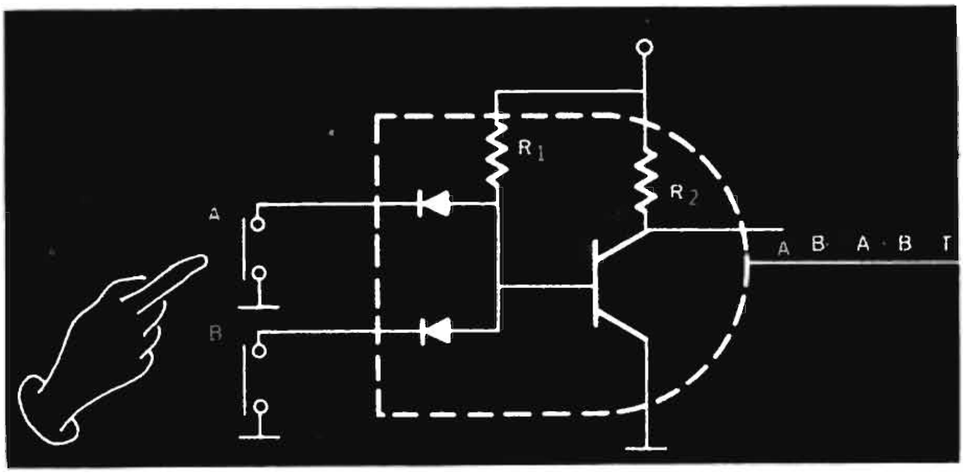


Fig. 3—Circuit using NAND gate

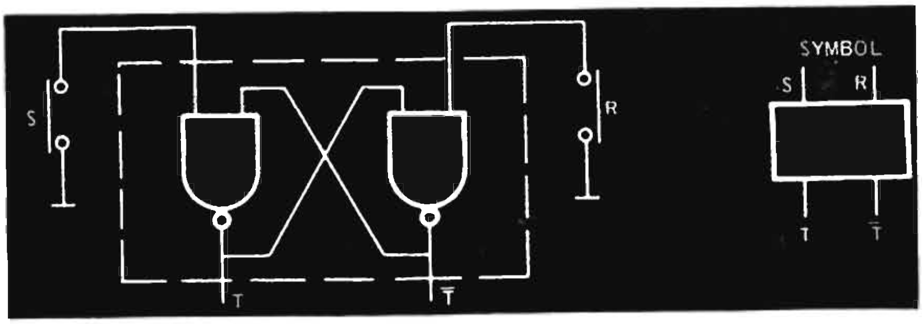


Fig. 4—Memory circuit

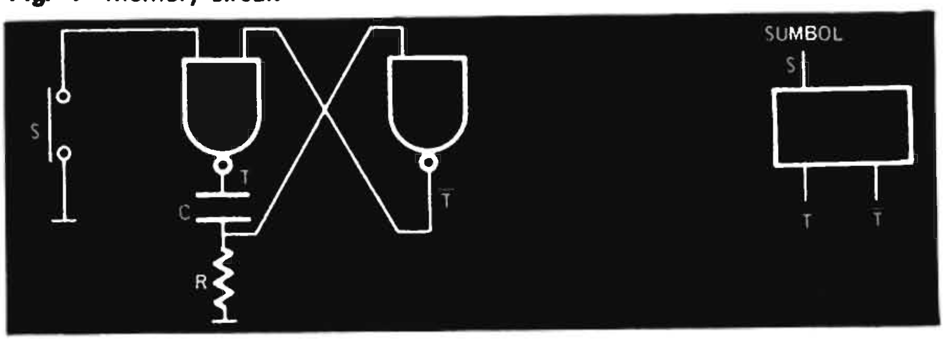


Fig. 5—Memory circuit with automatic re-set

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to control other electronic units and processes; remote control becomes available.

Electrical units can be used to replace many mechanical parts and eliminate purely manual operation of pinch roller, brakes, etc. Clutches can be replaced by motors whose speed in winding and braking can silently and easily be electronically controlled. Solenoids can be used for operating the pinch roller, brakes, etc.

All this assumes, however, some reliable form of control. The application of logic circuits in integrated form for control purposes can produce many advantages. The so-called High-level, specially integrated logic circuits lend themselves particularly well to tape recorder control, and this circuitry will be largely immune to possible electrical noise within or from a source close to the recorder. Control signals will originate essentially from the operating push buttons. By carefully analyzing the desired functions, it will be possible through couplings algebra to formulate these to a mathematical formula that can be realized by electronic circuits.

For newcomers to this technique, some explanation may be needed. A logic circuit is an electrical model of a logic function which tells us that certain conditions must be present for a change of condition (or for no change).

One symbol for a logic circuit controlled by DC voltage levels can be seen in Figure 1.

With high levels on both inputs A and B, output T will be high level, expressed $A \cdot B = T$.

Conversely, when either A or B is low, T will be low, expressed $\overline{A} \cdot \overline{B} = \overline{T}$. Additionally, $A \cdot \overline{B} = \overline{T}$ or $\overline{A} \cdot B = \overline{T}$.

The symbol $\cdot$ means in couplings algebra (Boolen Algebra) AND. The symbol + means OR. $\overline{A}$ may be called negation of A.

The connective AND indicates that T is high if, and only if, both A and B are high (conjunction).

The connective OR symbolizes that T is high if, and only if, either A is high or B is high or both are high (disjunction).

In electronic logic circuitry terminology, circuits as above may be described as AND-gates and OR-gates. As one in logic mathematics refers to a two-basic statement, true or not true, in electronics we refer to two levels, high or low (1 or 0). The conversion from one level to another is symbolized by a ring (NOT), as seen in Figure 2.

The practical logic circuit in Figure 2 is named a NAND-gate (NOT-AND-gate). When A and B represent high levels, T will represent a low level, $A \cdot B = \overline{T}$, $T = \overline{A \cdot B}$. To improve the understanding as to how a NAND-gate functions, we can look at Figure 3, where such a gate is visualized with discrete components.

The circuit is designed so that the transistor will be in saturation when neither A nor B are connected, which means that the level T is low. If the base current disappears, caused by connecting A or B to ground, the level T will become high. The circuit works then as an OR-gate with a converted output. One single NAND-gate cannot maintain any given information (when the finger in Figure 3 is removed, the high level will disappear). In order to obtain memory we have to use two circuits, as shown in Figure 4. A low level on the set-input will make T remain high. The circuit resets with a low level on reset-input. By adding a RC network, the reset can be automatic, as seen in Figure 5.

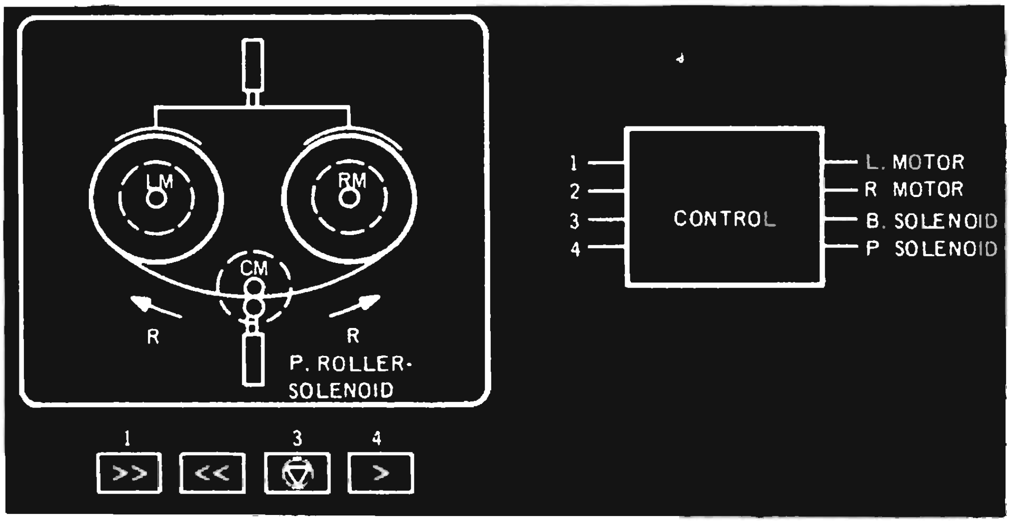


Fig. 6—9000X layout

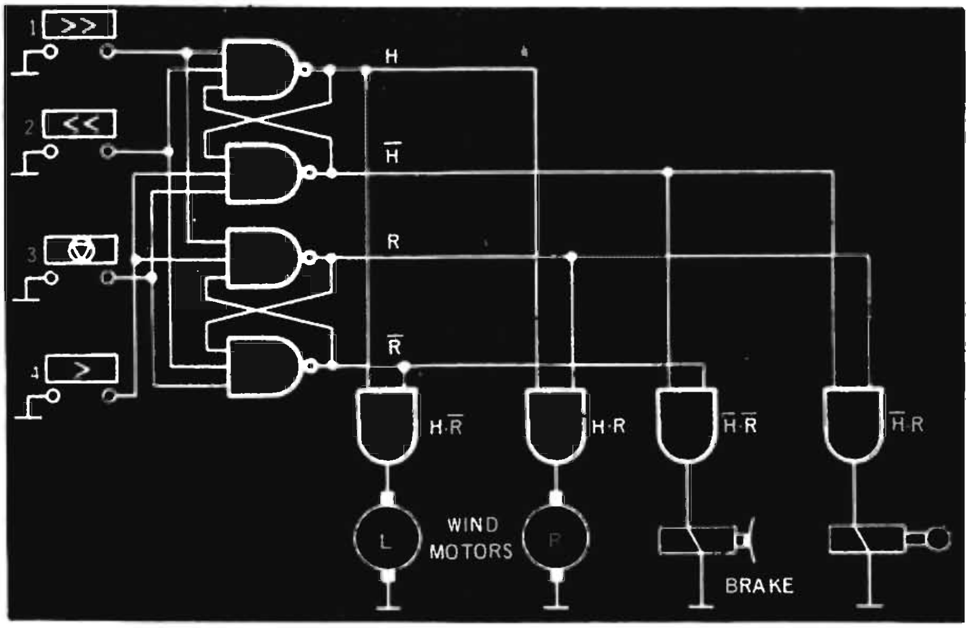


Fig. 7—Control circuit

A Description of the Logic Circuit System in the New Tandberg Series 9000X

In the design of Tandberg Series 9000X, efforts were made to take maximum practical advantage of this sophisticated type of control, through the use of integrated logic circuitry.

As in any tape deck, the rotation and radius of the capstan shaft will determine the tape speed. The tape is transported by the pinch roller (engaged by the pinch roller solenoid) moving toward the capstan. See Figure 6. Forward winding is caused by a strong torque on the right side motor (counter-clockwise), and a low torque for braking on the left side motor (clockwise). Rewinding will be established conversely.

In establishing the mathematical criteria for these functions, we will designate:

Tape movement to right as R
Tape movement to left as $\overline{R}$
Winding speed as H
Play speed as $\overline{H}$

The following conjunctions can then be formed:

Winding speed to right as H•R
Winding speed to left as H•$\overline{R}$
Stop as $\overline{H}$•$\overline{R}$
Play speed to right as $\overline{H}$•R

Two bistable circuits must accommodate four function criteria. The first circuit shall represent the speed H and $\overline{H}$, the second circuit shall represent the tape movement. The diagram relating directly to the operating buttons on the recorder can be seen in Figure 7.

This simplified control is, of course, not sufficient in a practical functional circuit design. Additional circuitry to operate the solenoid will be necessary, besides a memory to distinguish between playback and record. Further, there has to be a trigger network included to allow for braking time after completing winding before a play condition is set.

In addition to the mechanical braking, there was a requirement for electrical braking. Additionally, electronic end-stop pulses and stop-signal pulses, when turning on and off the AC switch (power-up reset-circuits), were also found necessary to eliminate the possibility of tape spillage. To prevent accidental erasing, the recording function was also given certain conditions that had to be met before becoming established.

As mentioned earlier, these functions can be formulated through couplings algebra. With a realization of this, one arrived at a solution as shown in Figure 8. This diagram fulfills the foregoing requirements.

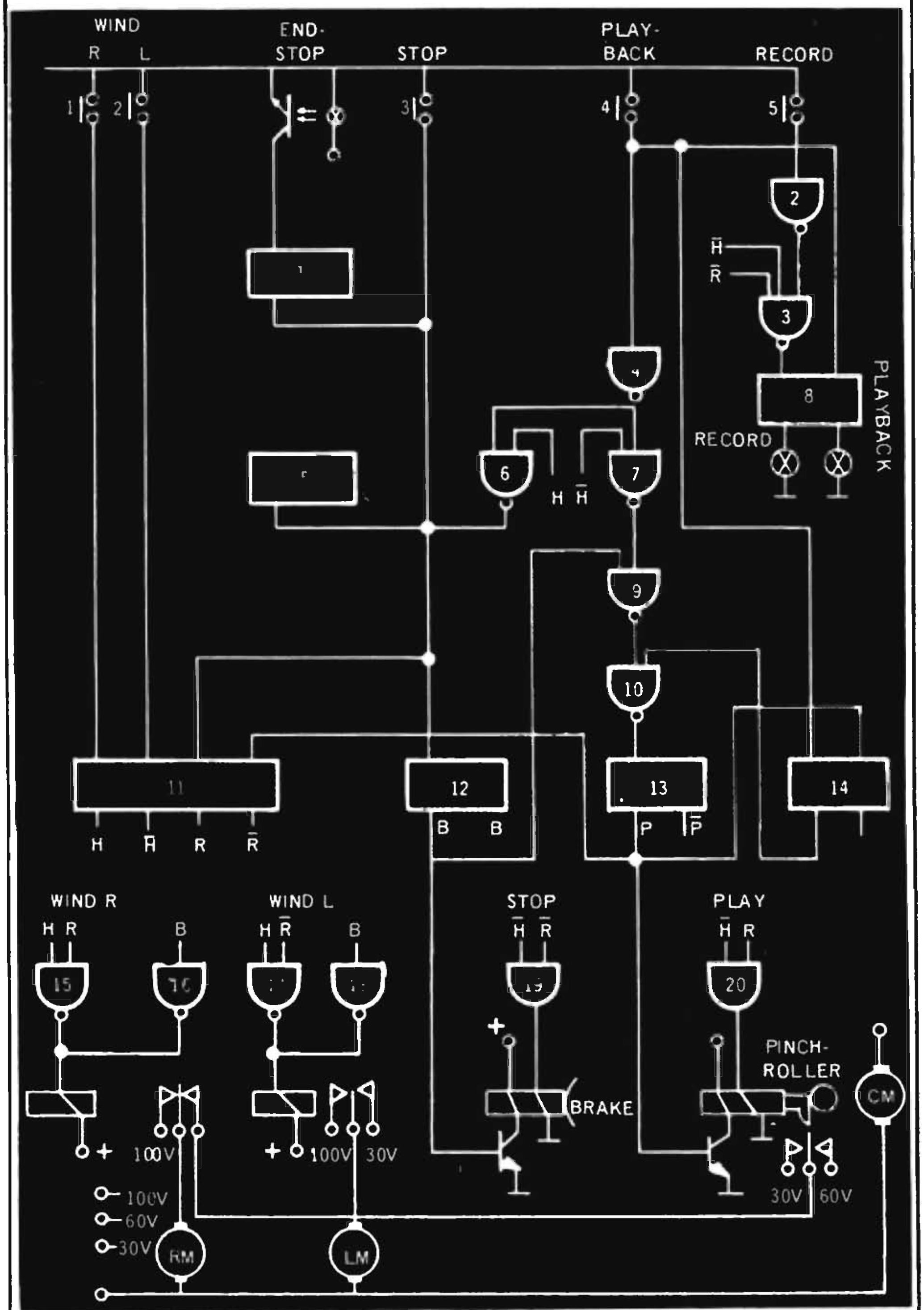


Fig. 8—Complete arrangement showing all functions

Circuit 11 in this diagram is basically the two bistable circuits providing the information that shall control the four tape conditions.

Circuits 12 and 13 are monostable and necessary for the control of, respectively, the brake and pinch roller solenoids.

Circuit 8 is bistable and has to distinguish between recording and playback modes.

The triggering Circuits 4, 6 and 7 control the braking phase when the tape mode changes from a wind position to a play position.

Circuits 9 and 10 shall make triggering of Circuit 13 feasible by Circuit 12.

Circuit 14 shall eliminate this possibility when stop condition is set.

Circuits 2 and 3 prevent the tape deck from being set in record mode prior to the presence of the stop condition.

Circuit 1 shall be a Schmitt-trigger that gives end-stop pulse.

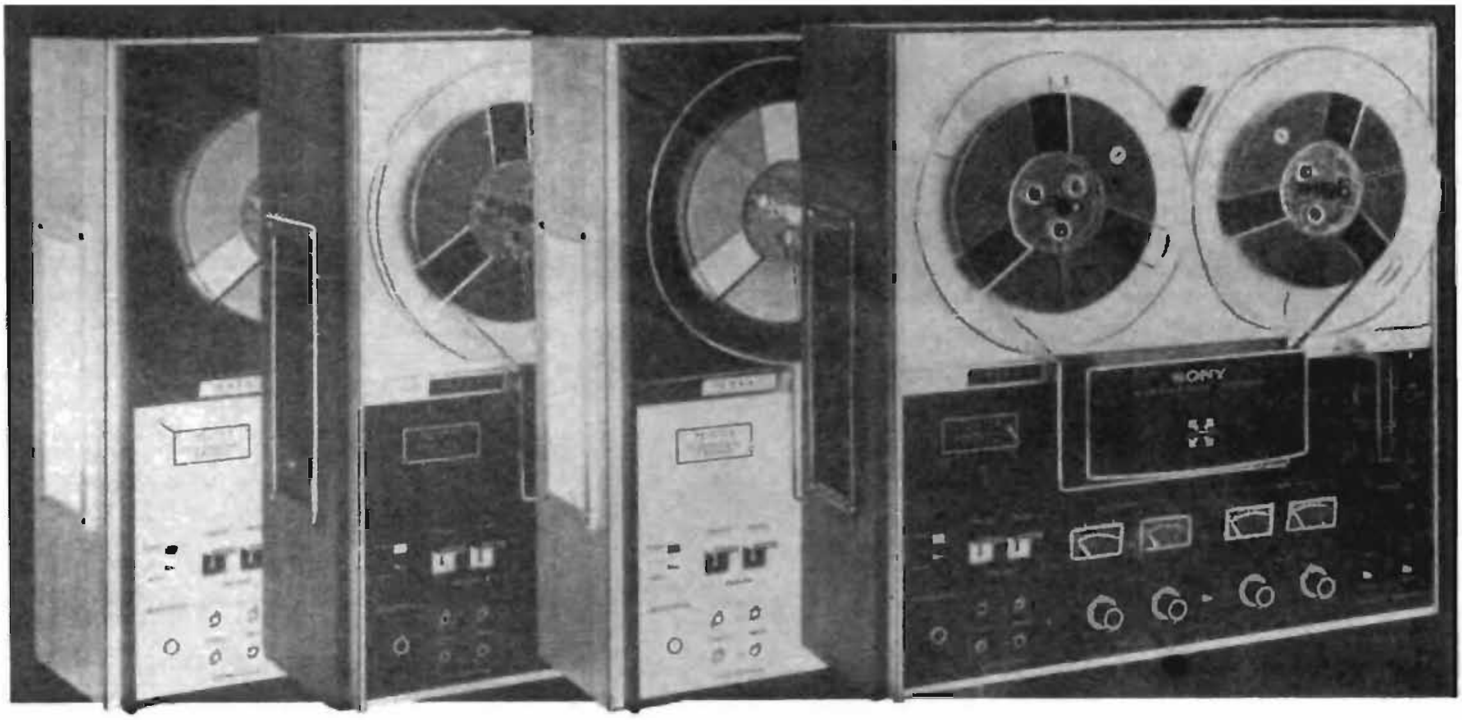
Circuit 5 is a "power-up" reset-circuit.

Circuits 15 and 17 transfer the information to the winding motors of their respective relays.

Circuits 16 and 18 give electrical braking during the braking phase.

Conclusion

The use of logical analysis to determine a logically correct engineering solution for a consumer tape deck opens many possibilities. Besides excellent electronic and mechanical specifications, safe, fast and easy operation are just some of the advantages. This type of logic control system can be applied toward timed and sequential automatic operations, remote controls, etc., even in environments with extensive electrical noise.



THE CHANGING FACE OF OPEN-REEL

market and product trends

*by G. L. Taylor

Once it was thought that the magnetic tape had a strong chance of replacing the vinyl record in the hearts and collections of music lovers. Today some probably still hold that hope—or fear—but their numbers are few. A glance into almost any current publication devoted to audio arts will reveal the continued prevalence of Record Power.

Yet, magnetic tape does hold an important place in the audio industry, due to its special capabilities which no record on the market today can equal.

The electronics industry is a fast-changing one. Innovation follows innovation: new products appear, old ones fade away. As products change, they attract new and different markets. Conversely, as the markets change, the products must change to accommodate them. This importance of market, then, is a primary concern in the study of trends in open reel tape recorders.

A strong demand for the versatility and high performance standards of open reel tape will always exist in the professional audio industry: broadcasting studios, recording studios, production companies, tape duplicators, wherever a high-quality master is necessary. These same characteristics continue to make open reel an invaluable part of computer operation and the sciences.

As an example of the latter, in a major motion picture, a bank of open reel tape decks is used in a program to translate the speech of dolphins. As with all good fiction, this movie has its feet planted firmly in fact: the ability of dolphins to communicate and their high level of intelligence long ago intrigued real-life scientists sufficiently enough for them to initiate similar programs at marine laboratories and universities around the world.

More importantly, the same qualities that keep open reel an entity in the professional market will serve it well in the consumer market, for a variety of reasons.

1. Increasing sophistication of music lovers. There are those who demand the best simply because it is the best. Others have weighted cassettes and cartridges in the balance and found them wanting. Many have started out with cassettes or cartridge units and are ready to step up. All of these people recognize and salute the well-deserved superior status of open reel.

2. Versatility of open reel. Cassettes and cartridges have as their principal advocates those whose primary interest is listening to music. For them, versatility is likely to be the ability to cut out commercials when recording off the air—if indeed they're interested in recording at all.

However, those whose interests extend to creating their own source material often demand more: three heads for editing and monitoring; special effects capabilities, such as echo, sound-on-sound, sound-with-sound, mic-line mixing. These are functions of open reel units.

3. Transition of music from vicarious listening experience to gut-level participatory experience. It used to be that composing and performing were arts that look incredible talent and many years to develop. Consequently, direct participation was limited to a relative few. No longer. Today music has become a more direct, personal expression of raw emotion open to nearly everyone with the urge to express. It is a trend characterized not so much by talent as by sincerity. This is evidenced by soaring sales of musical instruments in the past decade and by the tremendous outgrowth of new musical groups, each performing its own music. The fact that some have hit fantastic success almost overnight has encouraged countless others to leap on the music bandwagon.

An open reel tape recorder is vital for any group of burgeoning composer-musicians hoping to succeed. For creative

*Sony-Superscope Inc.

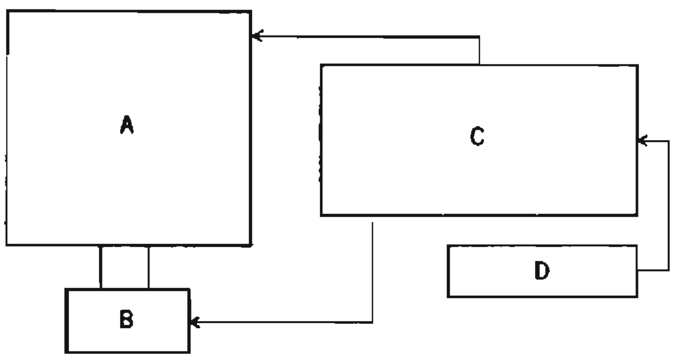


Fig. 1—Block diagram of a servo-control motor.

editing and special effects are as much a part of this music as electric guitars, earnest vocalists and arcane lyrics.

4. Money to spend. All of the foregoing market trends would be worthless in an indigent society. People can pay more for open reel equipment simply because they have more to spend. Despite unemployment and inflation, disposable income seems to be on the rise.

These changing market trends have naturally had their effect upon products. But the situation is not without a certain irony, for open reel, in its increasing sophistication, seems to run counter to the prevailing trend toward simplicity in tape recorders today.

For dozens of years, a prime objective of the tape recorder industry was miniaturization, heralded in by the transistor and the integrated circuit. This trend was reflected in May 1967, in a series of prognostications published by AUDIO MAGAZINE on the occasion of its 20th anniversary. Tape recorders, said industry leaders, would get smaller and start using slower speeds, with better results than the higher speeds of older models.

Coincidentally (?) the same issue of AUDIO carried an article on the Phillips cassette and two rival forms of the tape cartridge: Fidelipac, a standard of the broadcast industry, and Lear-Jet, created by that manufacturer of personal jet aircraft in conjunction with Ford Motor Company. In 1967, none of these configurations were viable competitors for either the record or open reel tape. Their FI was admittedly far from HI: the article considered them adequate for automotive use, but not "sonically attractive enough to warrant the attention of the serious audio buff."

Today, of course, that's all changed. Serious audio buffs by the hundreds are buying cassette and cartridge units—and not just for their cars. Technological advances have apparently fulfilled the prophesies of the industry's augurs. Innovations such as Dolby and chromium dioxide tape coatings have put cassette performance, even at 1⅞ ips, on a par with records. Four-channel sound and a wide selection of software have made cartridge units a force to contend with in the home entertainment field. Ultimate miniaturization is nigh. Already palm-sized cassette units exist that give better performance than behemoth open reel models of yesteryear. At least one of these, SONY's TC-55, features a select switch that enables the unit to capture music with adequate, though not high, fidelity. Further refinements will undoubtedly follow.

So the cassette and cartridge are now respectable. Anyone with doubts can check prices, for one thing in this ever-changing industry remains constant: good performance still costs more than mediocre performance. The result is that cartridge and cassette tape units have taken over the market previously held by low- and medium-priced open reel units

Consequently, as cassette and cartridge units become more compact and convenient, open reel units have been forced to become more complex and versatile. Features designed to provide professional-quality performance are finding their way more and more into consumer models.

1. Higher speeds. Paradoxically, as one faction of the industry moves toward the slower speeds predicted in 1967, another faction seems intent on defying destiny. Units with 15 ips speed settings are on the increase—corresponding to demand for the wide frequency response that's a function of these higher speeds.

2. Large reel capacity. To circumnavigate the reduced playing time that results from using higher speeds, an increasing number of recorders are being designed to accommodate 10½ inch reels. In addition, large reel capacity at the lower speeds can be useful to provide greatly extended continuous play.

3. More sophisticated drive systems. Uneven tape motion has long been a bugbear of the tape recorder. Now the sophisticated audiophile can reduce it and its attendant ills —wow-and-flutter and poor tape-to-head contact—with any of a growing number of 3-motor tape decks on the consumer market.

Innovative motor design is also helping to solve the problem of uneven tape motion. Typical is the servo-control motor, which prevents capstan speed variations due to normal voltage or load changes. (Figure 1.)

Connected directly to the capstan motor (A) is a frequency generator (B), whose frequency is dependent upon the motor RPMs. The frequency generated is relayed to a servo control board (C) which compensates for variations in the speed of the capstan motor by either increasing or decreasing voltage (D). The result is highly accurate motor speed and consequently, consistent tape motion.

Further sophistication can be found in yet another feature previously restricted by cost to the professional, but now making its way to the consumer: closed loop dual capstan tape drive.

In this system, two capstans isolate the tape from external vibration and abnormal reel movement by forming a "closed loop" of tape around the head assembly. Two current 3-motor tape decks featuring the system, SONY models TC-854-4S and 850, can attest to its efficacy with extremely low wow-and-flutter specifications of 0.03%. (Figure 2.)

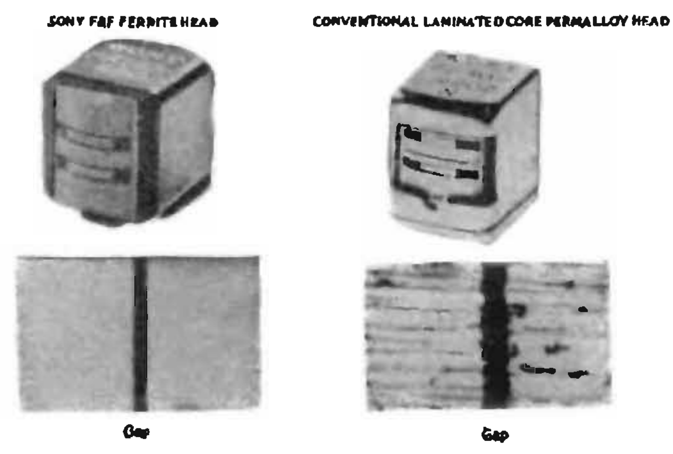


Fig. 2—Ferrite head vs Permalloy head, demonstrating the effect of laminations upon gap width and core smoothness.

5. Ferrite heads. Much has been written concerning the hardness of the ferrite head and its resultant ability to resist abrasion. There are other advantages. For example, the higher internal resistance of the ferrite material used in their manufacture allows ferrite heads to be molded of one solid piece of material. By contrast, the permalloy head must be built up of laminations in order to cut down eddy current losses.

Fig. 3—Sony Model 850, a professional quality 3-motor deck.

The single surface of the ferrite head material can be lapped to a sharper, more precise edge than can the laminations of the permalloy, permitting more even pole pieces, and ultimately a narrower head gap. (Figure 3.) The effect on frequency response is obvious.

Natural stereo and 4-channel separation is virtually dependent upon even pole pieces and straight headgaps to prevent sound drop-outs from phase shifting. It's not surprising, then, to see the ferrite head making an appearance on the more sophisticated open reel units.

What else does the future hold? Knowing that the tendency is toward professionalism, we can make educated guesses—or think wishful thoughts. Chromium dioxide tape for reel to reel? Why not, at least at the slower speeds. Refinements in ferrite heads will certainly aid the cause. So will special lubricants to reduce friction, and its resultant abrasion.

Abrasion is one of tape's most insistant drawbacks. Hopefully, the time will come when it's eliminated altogether, or at least reduced to a nominal level. A more efficient method of reading the magnetic pattern on tape would help—especially one that does not depend on tape-to-head contact. As incentive, consider the recently developed phono cartridge that uses a photo cell instead of a needle.

This is a transitory period for open reel. It has lost sales to cassette and cartridge simply because its new identity and market are not yet firmly established. That time is coming.

To be sure, the market for the sophistication provided by open reel is smaller than that for the convenience of its plastic-enclosed cousins. But price differential minimizes that problem—and furthermore, each purchaser of a cassette or cartridge unit is a potential step-up to open reel.

One thing is certain: no matter how far cassettes and cartridges progress, open reel is still the ideal they're measured against. As long as that's true, open reel will remain an important factor in the consumer audio industry.

Dolby B-Type Noise Reduction System

Robert Berkovitz and Kenneth Gundry*

An unsatisfactory signal-to-noise ratio has remained the major obstacle to attaining an adequate level of performance from consumer media for music reproduction. This is especially true of the music-cassette, because of its slow tape speed and narrow track width, but it is also true of stereo FM broadcasting and the phonograph record. Although hopes were raised in recent years that further development of magnetic tape would eliminate its inherent noise as a problem, these hopes have been frustrated by the relatively modest gains achieved, and by studies which indicate that the available signal-to-noise ratio of present-day tapes is very near the maximum value imposed by theory.

It is therefore not surprising that numerous attempts have been made to devise methods of noise reduction satisfactory for professional and consumer use. However, almost all of the methods proposed have had unacceptable drawbacks.

The effectiveness of single-ended (non-complementary) systems, for example, which are designed to be used only during playback, extends only as far as the listener's willingness to sacrifice musical information. In principle, all playback-only systems depend upon the idea that the signal and the objectionable noise occupy separate domains; if this is correct, then the problem of noise reduction is one of defining the boundary between the domains, in terms of frequency and/or level, and designing a circuit to suppress everything on the "noise" side of the boundary. However, if the noise spectrum of ferric oxide cassette tape is taken as an example (see Fig. 1), it is seen that the noise, when passed through a standard DIN weighting network simulating the ear's sensitivity, remains considerable in the 1-4 kHz range. Since this range includes many of the lower harmonics and upper fundamental tones in music, it is not possible to suppress it, even at low listening levels, without serious loss of information. On the other hand, the noise within this range is so disturbing that if it is not reduced by such a circuit, the amount of subjective improvement obtained is minimal.

Complementary methods, i.e., those which require some signal processing or encoding during both recording and playback, offer greater promise, but can also present difficulties when put into practice. Pre- and de-emphasis schemes, for example, in which high frequencies are increased during recording and decreased by the same amount during playback, are only of limited value. Even in FM broadcasting, where such standardized pre-emphasis has been employed for many years, the usefulness of its continued application is in doubt. The primary problem is that modern microphones and recording equipment now routinely reproduce high frequencies at amplitudes so high that they were considered unlikely when current FM standards were set. Broadcasters are now forced to use limiters to prevent overmodulation, if they also wish to maintain reasonable levels at middle and low frequencies. In magnetic tape recording, pre-emphasis is difficult to use because tape saturation occurs at lower levels at high frequencies. Since high-frequency signals already present problems in cassette recording because of their short wavelength, added pre-emphasis would complicate a task which is already difficult.

The compandor type of noise reduction system, in which the dynamic range of the signal is compressed during recording and expanded again during playback, offers more promise. However, a simple compandor, even if precise in its action, also presents problems. In recordings and broadcasting, one of the most serious drawbacks of the compandor is the danger of signal overshoot, which can result in distortion or overmodulation of a transmitter.

An even more serious problem of compandors, from the listener's point of view, is noise modulation. When a conventional full-band compandor is used, low-level passages are recorded at a level higher than normal. They are then played back at reduced level, restoring correct signal dynamics and reducing noise at the same time (see Fig. 2). There can be no noise reduction effect during high-level passages, because this would require increasing the level of such passages during recording, resulting in overload. The simple compandor therefore requires that one assume that noise is not objectionable when the signal level is high. However, this is not always the case. A high-level bass drum beat, for example, does not mask high-frequency tape hiss; as a result the drum and other instruments introduce noise modulation during playback—each note is accompanied by a "swish" as the noise level rises for the duration of the note. While it is not audible with all types of program material, noise modulation limits the usefulness of the compandor considerably.

The extreme diversity of available source material and the high quality of present-day master recordings are the factors which really determine the conditions to be met by a satisfactory noise reduction system for home use. It must be remembered that many home listeners own playback equipment with very low distortion and wide frequency range, disclosing audible effects which might have passed unnoticed

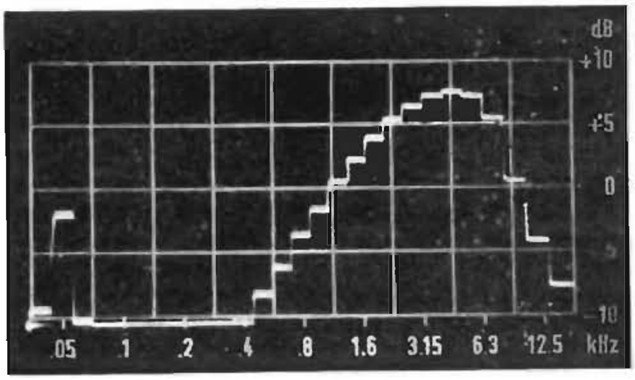


Fig. 1—DIN-weighted noise spectrum of low-noise ferric oxide cassette tape displayed on screen of real-time analyzer. Vertical scale is relative only; DIN noise measurement of sample shown was —47 dB with respect to 200 nWb/m. A substantial amount of the noise present falls well within the range of musical fundamental tones.

*Dolby Laboratories, Inc.

in earlier times. Therefore, it is especially important that the program be recovered accurately after noise reduction, without addition of any audible sound. For the listener's sake accuracy of recovery and effectiveness of the system should not require adjustment of system parameters to match various kinds of program material. At the same time, the size and cost of the system should introduce no obstacle to its use. Furthermore, as a practical matter for the industry, it is clear that the system should require no modification of present professional practice in master recording, duplicating, or broadcasting.

(To Be Continued)

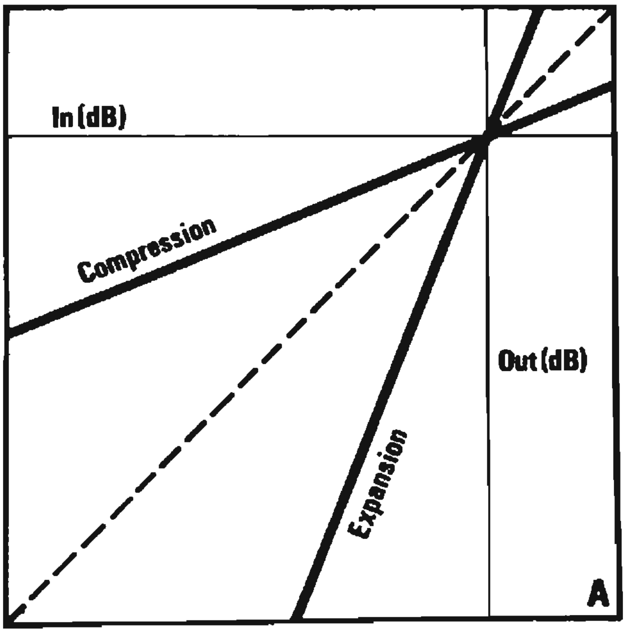


Fig. 2—Transfer characteristics of two conventional compandors (solid lines). A, constant-slope type; B, high-level type. Since compression and expansion are functions of signal amplitude only, in a single frequency band, such compandors fail to suppress noise whenever natural masking fails (see text).

Dolby B-Type Noise Reduction System - Part 2

Robert Berkovitz and Kenneth Gundry*

The Dolby B-Type Noise Reduction System

The Dolby B-Type circuit is a specialized form of compandor which avoids the usual deficiencies of compandors. The operational principle of the B-Type system is complementary low-level compression and expansion in a frequency range which varies in bandwidth as the signal changes.

Most objectionable noise encountered in home listening is at middle and high frequencies, from about 500 Hz to the upper limit of audibility. In the interest of circuit economy, the action of the B-Type circuit has therefore been limited to this range. A feedback control circuit adjusts system parameters automatically as a function of signal level and spectrum, so that the system's action complements the psychoacoustic masking of noise which occurs naturally in the course of the program. A block diagram of a Dolby type of noise reduction system is shown in Fig. 3. The circuits used for encoding (during recording or transmission) and decoding (during playback or reception) are quite similar and can be considered as the same circuit, switched to operate in either mode.

The compression and expansion characteristics of the Dolby B-System are fixed and are referred to Dolby Level, a specific internationally standardized reference level. In the case of cassette tape. Dolby Level is a flux of 200 nWb/m; in FM broadcasting, Dolby Level is ± 37.5 kHz deviation.

Figure 4 is a block diagram of a switchable (encode-decode) B-Type circuit. There are two paths which the input signal follows: a *main path* (at the lower part of the figure) in which no change other than linear amplification occurs, and a *secondary path*, a variable filter through which only low-level, high frequency components of the input signal are allowed to pass. To encode the signal, the output of the secondary path is combined with signal in the main path *additively*; this boosts low-level, high frequency portions of the signal. Decoding is accomplished by feeding the secondary path from the circuit output, which is opposite in phase to the input (note phase inverter in Fig. 4); the secondary path is then part of an a.c. negative feedback loop which reduces output, i.e., the output of the secondary path is combined with the main path *subtractively*. In the decode mode, therefore, the circuit reduces the level of precisely the same information which was increased in level during encoding.

*Dolby Laboratories, Inc.

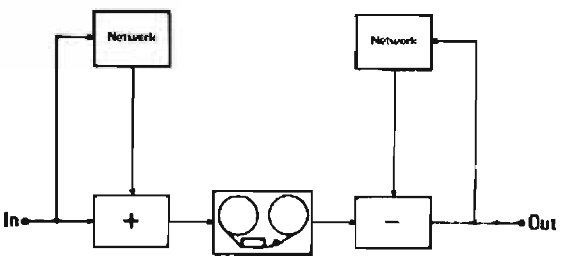


Fig. 3—Block diagram of Dolby type noise reduction circuit as used in typical record-reproduce chain.

As Fig. 4 indicates, the action of the B-Type circuit is controlled by the output of the filter in the secondary path. Above a fixed threshold level, the bandpass of the filter, in turn, is modified by the d.c. feedback loop.

At very low levels, i.e., below the threshold, which at high frequencies is about 40 dB below Dolby Level, the output of the filter is not sufficient to generate d.c. feedback; consequently, the output of the secondary path is simply proportional to signal level within the filter pass band. The output of the circuit is then essentially as shown in Fig. 5.

As signal level rises above the threshold level, the rectified filter output is returned to the FET gate where it is applied as negative feedback, raising the filter cutoff frequency so that the output of the secondary path, while still increasing, no longer does so in proportion to the change in signal level. As signal level becomes even larger, the increasing d.c.

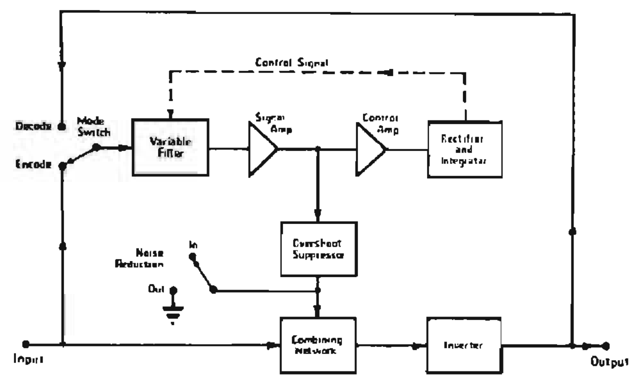


Fig. 4—Block diagram of Dolby B-Type noise reduction circuit. The configuration shown can be switched to encode or decode the signal.

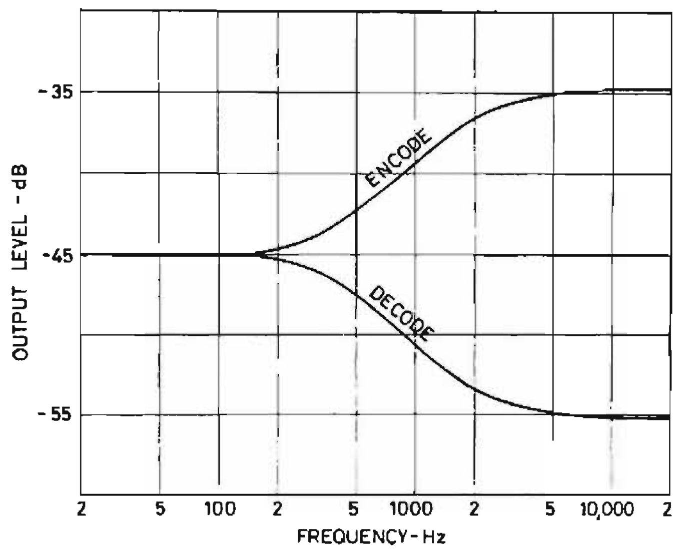


Fig. 5—Output of B-Type encoder and decoder circuits under low-level input signal conditions. The two operations are symmetrical and the result is an overall frequency response which is level.

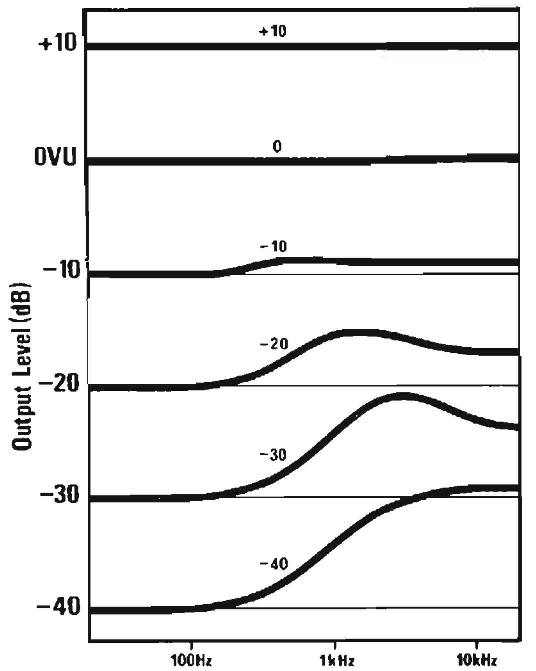


Fig. 6—Characteristics of encoding processor at several levels. The gradual reduction in boost with increasing level avoids possible tape overload.

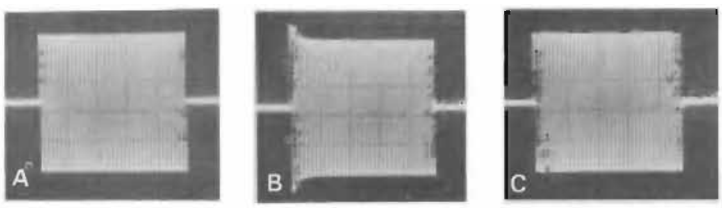


Fig. 7—Effect of the B-Type circuit on a tone burst; frequency = 3 kHz; burst duration, 12 milliseconds; low level = 40 dB; high level = +6 dB; (A) Input to system; (B) Encoded, (C) Encoded and Decoded.

feedback generated restricts the filter bandwidth further, and near Dolby Level the output of the secondary path remains relatively constant. The net effect is that the secondary path has no audible effect on output at low frequencies, and increasing effect with increasing frequency and decreasing level to about 40 dB below Dolby Level. At high levels, the effect of the extra signal is so small as to have no significance; at low levels, in the spectral region in which noise reduction is required, the increase during encoding is as much as 10 dB, and is of considerable importance.

The manner in which the secondary path changes from constant-gain to constant-output is determined by the adjustment of gain within the feedback loop. In addition, the exact variation in filter bandpass with changing level is set optimally by making the control amplifier frequency-dependent. The overall frequency response of a B-Type encoder circuit for different input levels is shown in Fig. 6.

A compandor operating over a wide frequency range must be designed to take into account the problem of noise modulation discussed above. If some high-level passages in the program differ sufficiently in frequency content from the noise components present, the latter will remain audible during the program in many cases. However, these passages cannot be increased in level when encoded, because of the danger of overmodulation. Under these conditions, compression may be applied intermittently, and high-frequency noise modulated audibly by mid-frequency components of the signal. The B-Type circuit overcomes this problem because it continues to function when a high-level signal occurs within its operating range; instead the feedback control shifts the range upward in frequency. This avoids the danger of overmodulation, but retains full noise reduction at frequencies higher than those masked by the signal.

The attack time of the B-Type circuit is dependent on the amount and rapidity of the signal change, due to the nonlinear design of the integrator, varying from about 100 milliseconds to as little as 1 millisecond. The recovery time of the rectifier-integrator is shorter than that of the human hearing system, about 100 milliseconds.

All compressors exhibit overshoot, including the B-Type

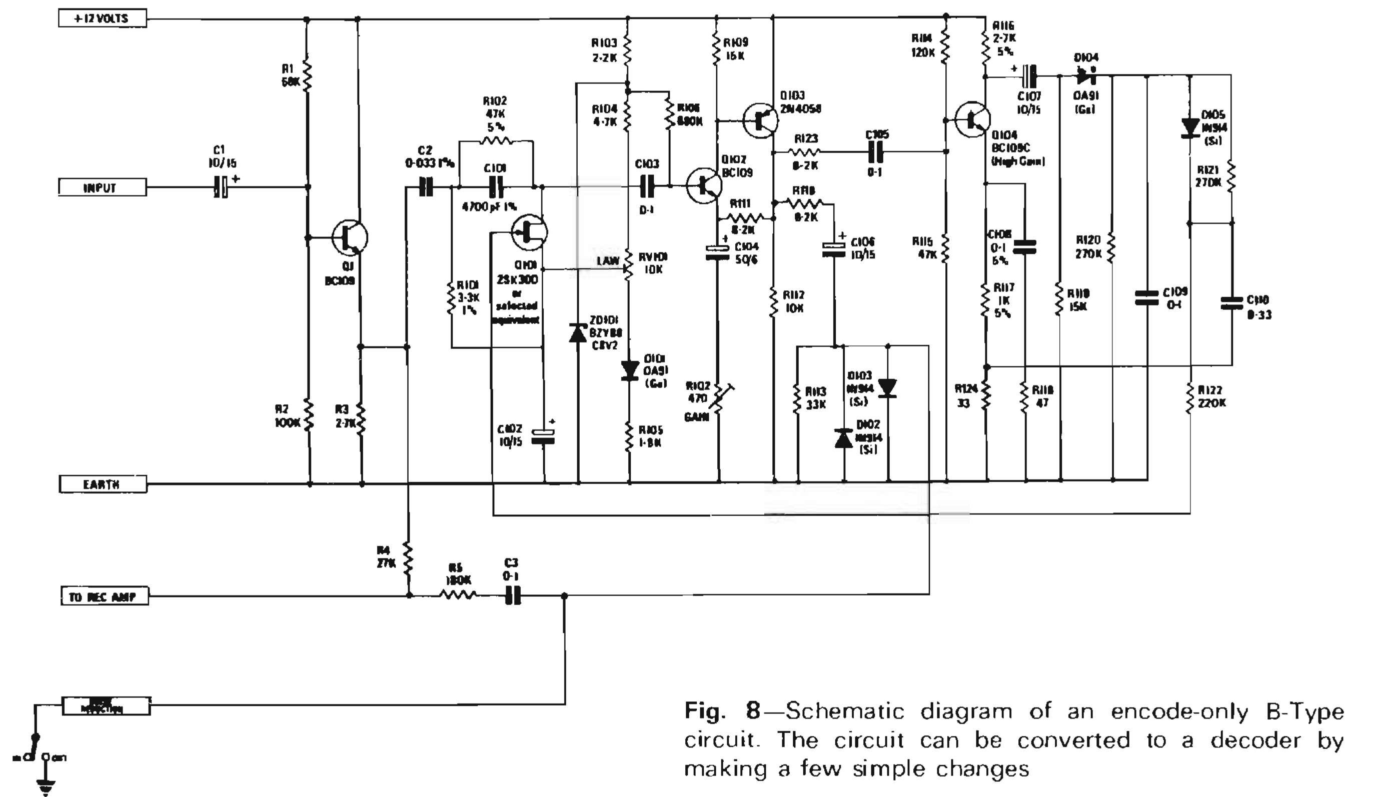


Fig. 8—Schematic diagram of an encode-only B-Type circuit. The circuit can be converted to a decoder by making a few simple changes

circuit. However, the dual-path approach used makes it possible to reduce the amplitude of overshoots significantly. Overshoot, which can occur only in the secondary path (where it can be suppressed without affecting the main signal) is comparatively small, and essentially disappears when the signal is decoded again. When signal levels are low, or when changes in signal level take place slowly, there is no overshoot problem; when signal changes are large and rapid, diodes in the overshoot suppressor stage limit the peaks of the overshoot. Since this takes place in the secondary path, the result of the suppressor action is to limit overshoot to a relatively small fraction of the full-level main path signal. Further, by restricting overshoot suppression to the secondary path, it is possible to avoid introducing audible distortion to the encoded signal. Because a complementary action takes place during decoding, the small remaining overshoot in the encoded signal is eliminated, and as with other effects produced during encoding, the original signal is restored. Figure 7 shows the result of encoding and decoding a short burst of 3 kHz, which changes in level from −40 dB to +6 dB.

Figure 8 is a typical schematic diagram of an encode-only B-Type circuit; the circuit for decoding-only is similar. As can be seen, only five transistors plus an FET are required; the parts cost of the circuit is approximately $2.40.

Figure 9 is the schematic diagram of a B-Type processor which has been designed to integrate noise reduction with other tape recorder electronics requirements as much as possible. The resulting circuit provides 26 dB of gain, whether or not noise reduction is in use, bias and multiplex filtering, and meter and monitor amplifiers. In fact, the only additional electronics needed to complete the recorder are a bias oscillator, recording amplifier (one transistor) and a microphone and head amplifier (two transistors). With the active elements used in the record/play switchable processor shown (eight transistors and one FET), the total used in the recorder, for two channels, is 22 transistors and two FET's. The cost to a manufacturer of the components shown in Fig. 9 is about $3.20, excluding the bias and multiplex filter components, which are, of course, necessary in the circuits of any properly designed tuner and recorder.

Dolby Laboratories and Signetics have collaborated in the development of an integrated-circuit version of the B-Type circuit. The IC is expected to offer manufacturers economy of assembly, elimination of adjustments, and somewhat smaller space requirement than the discrete-component version.

The characteristics of Dolby B-Type noise reduction can be summarized as follows:

1. Program recovery characteristics, with regard to frequency response, phase response, transients, and signal dynamics, are theoretically perfect; in practice, this ideal is attainable to any desired accuracy. Distortion in practical B-Type circuitry is considerably lower than that of the tape recorders or tuners with which it is used. Any type of program material can be encoded and decoded without audible loss.

2. The circuit is simple, inexpensive, and small in size, either in discrete-component or IC form.

3. The circuit is easy to manufacture and use because of the absence of critical components or adjustments. The circuit can be quickly and easily calibrated during manufacture, after which further calibration is not required. In use, only a simple level adjustment is necessary if tape of significantly different sensitivity is substituted for that formerly used.

4. No modification of broadcasting or duplicating practice is required to incorporate B-Type encoding. The use of the noise reduction system often makes worthwhile other im-

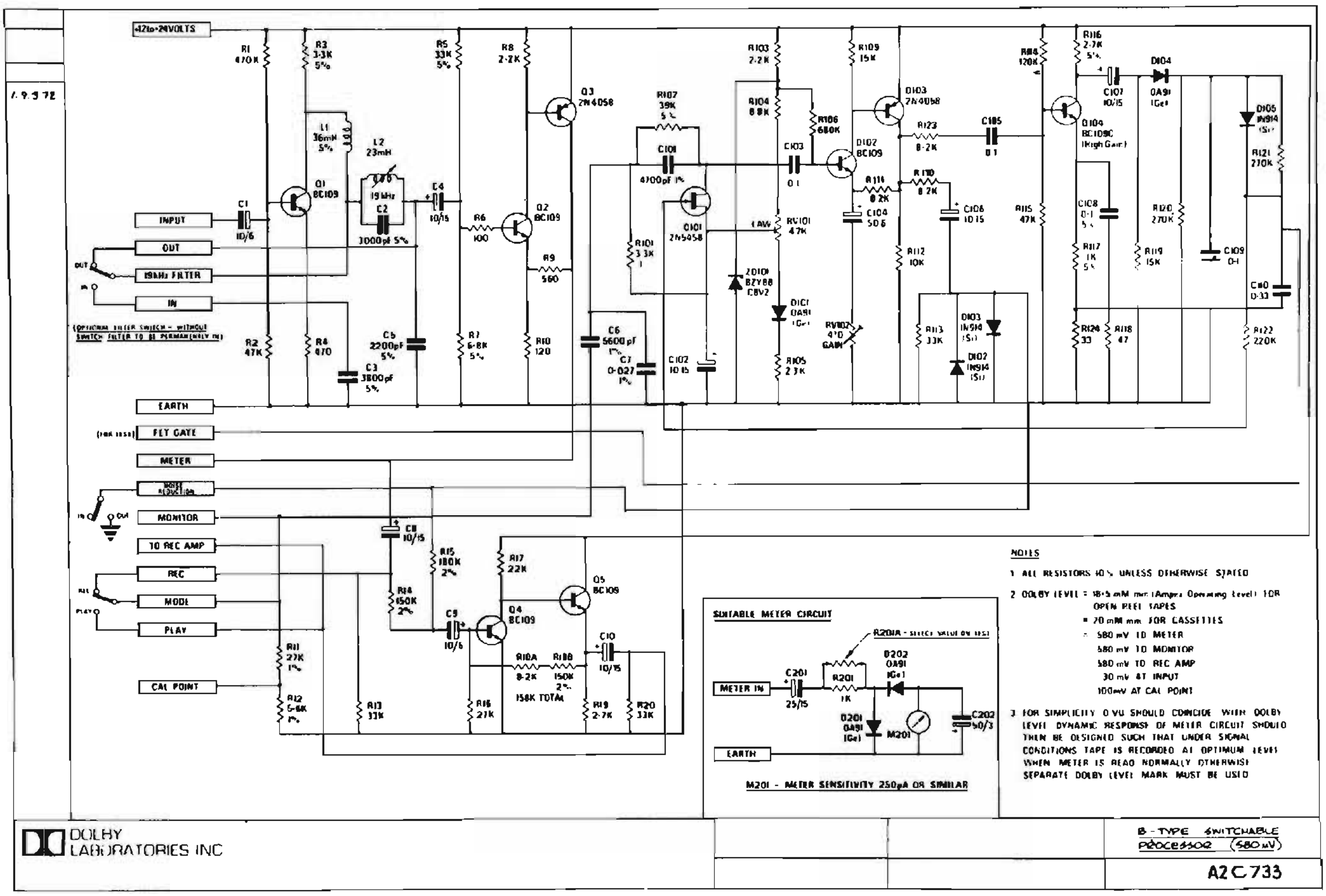


Fig. 9—Schematic diagram of complete switchable encode/decode circuit for use in tape recorder, including HF filtering, meter drive, monitor output and 26 dB of gain. Only a few more parts need be added to make complete record/play electronics for one channel of the recorder.

provements, however, such as extension of frequency response and dynamic range, or reduction of distortion by use of lower modulation levels, or some combination of these.

Effects Upon Noise Spectra

Figure 10 is a multiple exposure of the screen of a 1/3-octave real-time analyzer, allowing a direct comparison of the noise spectra at the output of a high-quality cassette recorder when different kinds of tape were used with and without the Dolby B-Type noise reduction circuit. Curve 1 is that produced by C90 ferric oxide tape; curve 2 is that of C90 chromium dioxide tape; curve 3 is produced by the same tape used for curve 1, but the B-Type circuit is switched "in," and curve 4 represents the noise spectrum of the chromium dioxide tape with the circut in. The tapes shown were biased before the measurements were made; no changes in gain or other control settings were made during the tests, other than to set equalization differently for the chromium dioxide tape from (70 microsecond). In fact, most of the improvement in noise level obtained when chromium dioxide tape is used appears to be due to the change in equalization; if this change is not made, there is little advantage in chromium dioxide tape from a noise point of view. On the other hand, the combination of chromium dioxide tape, 70 microsecond equalization, and B-Type noise reduction results in an excellent noise figure, 57 dB below Dolby Level in the example in the photograph (DIN 45405).

The advantages of B-Type noise reduction are also obtained when the system is used for FM broadcast transmission and reception, i.e., the improvement in signal-to-noise ratio obtained by use of the B-Type circuit is approximately the same as that produced by a 10 dB increase in field strength. The significance of this improvement can be appreciated when it is realized that such an increase would usually require an increase in transmitter power by a factor of ten. Considerable experimentation and broadcast experience in the USA have demonstrated, as one would expect, that the area in which listening is satisfactory is greatly extended by use of the B-Type noise reduction system. Several American classical music FM stations are already broadcasting full-time using Dolby B-encoding.

Compatibility

When any improvement is made in a system as widely used as the compact cassette system, it is highly important that the new development should be fully compatible with existing equipment. Improved cassettes must be playable on any machine which can play old-type cassettes, and fortunately this is true of Dolby B-Type cassettes. Such cassettes are subjectively compatible (i.e., generally pleasing to the listener) when played without decoding circuitry, to a great extent because of the unique approach taken in the B-Type circuit. Because most low-cost cassette machines are deficient in high-frequency response, the increase in low-level high frequency content in a B-Type cassette is usually welcomed by listeners with such equipment. Cassette recorders of higher quality, or the associated equipment with which they are used, contain tone controls which permit the balance to be adjusted to suit the taste of the listener. It is quite likely that many of the millions of B-Type encoded cassettes which have been made commercially are owned and played by listeners who are unaware of the special nature of the program material they hear. In any case, the subjective difference between encoded and other cassettes is sufficiently unobtrusive that none of the recording companies offering "Dolbyized" cassettes have found it necessary to offer old-type cassettes as an alternative.

It is worth noting that almost all pre-recorded cassettes are already compressed, for only in this way can the audibility of low-level passages be preserved in programs of wide dynamic range. B-Type cassettes differ mainly in that the listener now is able to remove the compression by pushing a button on his cassette machine restoring program dynamics and reducing noise. This is only possible because B-Type compression is standardized, while other types of compression vary considerably.

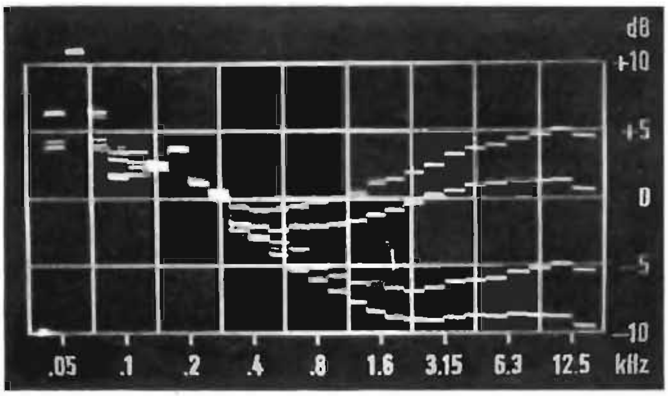


Fig. 10—Noise spectra at output of cassette recorder during playback of four kinds of tape. All tape was biased and control settings were the same for the four tests. (1) Ferric oxide C90 low-noise, −45.7 dB DIN 45405; (2) Chromium dioxide C90, −48 dB; (3) Same as (1) with B-Type noise reduction, −55 dB; (4) Same as (2) with B-Type noise reduction, −57.1 dB. All noise measurements referred to Dolby Level = 200 nWb/m.

Commercial Use

Within a few years of its introduction, the Dolby B-Type noise reduction system has been licensed to most major manufacturers of consumer tape recorders. At the present time there are more than 40 licensees manufacturing over 100 different B-Type products. Licensee payments for use of the circuit are on a sliding scale, based on quantity, from a maximum of 50¢ (U.S.) to 10¢ per circuit. Royalty charges are typically 60¢ per stereo unit for a major manufacturer.

In addition, most of the pre-recorded cassettes now made in the United States, the United Kingdom and Japan are "Dolbyized," and many of the largest recording companies issue their cassette output in this form, among them Ampex and CBS in the United States, Decca and RCA in England, and CBS-Sony, Nippon Columbia, King, and Apollon in Japan. Pre-recorded open-reel tapes and 8-track cartridges are also becoming available. In the United States, a number of FM stations have already started to broadcast regularly in B-Type encoded form, and this procedure is under study in other countries as well. There is no royalty payable for encoding cassettes or other tape recordings, or broadcasts.

Conclusions

The reduction of background noise by the Dolby B-Type noise reduction system has contributed importantly to the improvement in quality of home tape recording and playback. It has helped to make the extension of frequency response, the reduction of wow and flutter, and other improvements worthwhile, particularly in cassettes. The unique characteristics of the B-Type system permit excellent noise reduction without program losses, noise modulation and other drawbacks which have afflicted earlier attempts to solve the noise problem. The simplicity and economy of the B-Type circuit facilitate its use in consumer products at all price levels. Æ

HIGH FIDELITY FROM CASSETTE SYSTEMS

M. B. Martin*

THE PURPOSE of this article is to discuss a relatively small number of factors which affect the quality of recording and reproduction from cassette tape. The discussion is confined to considerations which came to light as part of the work of developing a new cassette tape and no attempt is made to completely analyze the cassette recording system.

The modern cassette tape system has reached a point where high fidelity sound recording and reproduction is a proven fact and effective competition with the phonograph disc is technically feasible. The development from the high noise, low quality system to the present state has been unusually rapid; one of the reasons being that the standards of tape speed and recorded track width have been adhered to, thus permitting technical development to be applied to improving quality and not to achieving greater economy of tape or providing a larger number of tracks per unit width. In the past, technical improvements in magnetic recording have, to a large extent, been applied to the economics of the system; whereas, with phonograph records, the standards have been fixed over long periods of time permitting developments to be applied to the improvement of quality thus, at the consumer level, the phonograph record has always been able to compete with tape from a quality of sound viewpoint, as well as being a more easily handled medium at a lower price per playing minute.

As part of the design project for a new cassette tape, the cassette recording system was analyzed to better understand demands made upon the recording media by the hardware and current recording standards. The work included a study into the effects of noise reduction systems, the relationships between recording head gap length and coating thickness, and some brief investigation into the energy spectra of music. The latter investigation confirmed the belief that, in many ways, the best method of testing to provide the most meaningful results, in relation to music recording, is to use as the signal source pink noise, the energy of which reduces at the rate of three dB per octave as frequency increases .

The Test System

Much of the data presented was generated by the use of white or pink noise as the signal source and a General Radio Real Time Analyzer, Type 1921, as the detection system. Frequency response curves, music spectra, and spectrum analysis of noise are printed out by the analyzer on an X-Y Plotter. When white noise is used as the signal source, the analyzer is adjusted to have a sensitivity which reduces by 1 dB for every third octave with increasing frequency; under these conditions a system with a flat frequency response will produce a horizontal line printout. When pink noise is used, the analyzer is set to a flat response so that a system with a flat response will also produce a horizontal line printout.

The frequency response data presented here was analyzed using an integration time of eight seconds, system and tape noise spectra were taken with an integration time of eight seconds, and a variety of integration times up to 32 seconds were used for the analysis of music spectra. This method of taking data has a number of significant advantages; two worthy of mentioning are:

1. Families of related curves can be plotted in a period of time short enough to permit the exclusion of system drift effects from consideration as affecting measurement accuracies.

2. The use of pink noise tests the tape and system under conditions which are a good approximation to those generated by modern music incorporating electronic synthesizers, heavy percussion, and electronically assisted string instruments.

All data presented in this article was taken on recorders which have very low electronics noise and, therefore, the signal-to-noise performance is dependent on the tape characteristics alone. Unfortunately, in real life, this is not always the case; the author has seen more than a few so-called high fidelity cassette recorders where the electronics noise predominates. With modern solid state circuitry, this is unforgivable particularly when, as so often happens, the recording amplifier is so noisy that the recorded noise completely obscures the bias noise of the tape. Obviously, with such a machine, there is no way that a better tape can improve the situation.

Tapes

Reasonable high fidelity recording and reproduction can be achieved with four classes of tape. Listed in order of appearance on the market as cassette tape, they are:

1. Low noise, high output tapes;
2. Chromium dioxide tapes;
3. Chemically modified gamma ferric oxide tapes—cobalt or Fe_3O_4 (magnetite) doped particles, and
4. Highly developed gamma ferric oxide tape such as MRX_2.

The characteristics of the tapes are determined by the magnetic particle used. Within each of the four categories, there will be differences in performance from manufacturer to manufacturer determined by the differences in the processing and formulations of the binder system used by each company.

Cateogry 1: Low noise, high output tapes, use a magnetic particle which is unmodified gamma ferric oxide (γF_2O_3). The improvement in performance over the earlier ferric oxide tapes is achieved by a reduction of particle size and some improvement in shape. The particle still is troubled by the

*Memorex Corporation

presence of protuberances known as dendrites and holes and the important length/width ratio varies from 4:1 to 6:1.

Category 2: Chromium dioxide[2], is a synthetic compound with magnetic properties that are, in some ways, superior to those of ferric oxide. The fundamental particle size is approximately the same as the irom oxide particles in category 1, but its shape is almost perfect, being a single crystal with a length to width ratio of 8:1 with no dendrites or holes. In addition, the coercivity is higher, 500 oersted as compared with 300 approximately. As a result of the better shape, the particles can be more accurately aligned in the direction of tape travel which, with the high coercivity, results in a much improved magnetic performance at the short wavelengths: i.e., high frequencies.

Category 3: The chemically modified gamma ferric oxide particles, are, in size and form, the same as the pure ferric oxide used in category 1. The improvement in performance is obtained by the addition of carefully controlled small amounts of impurities; either metallic cobalt or magnetite (Fe_3O_4).

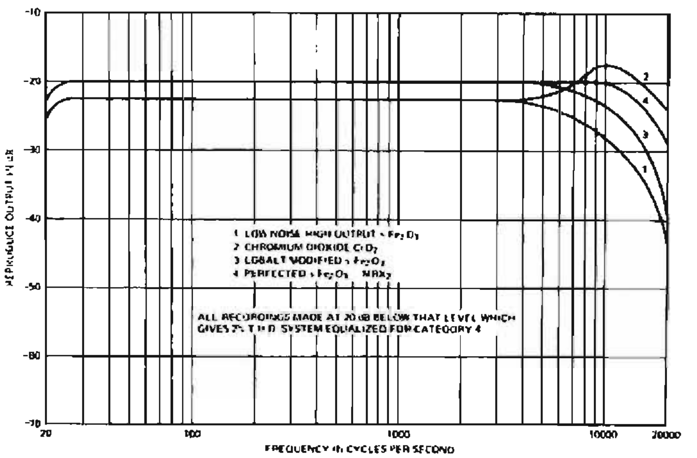


Fig. 1—Comparison of frequency response.

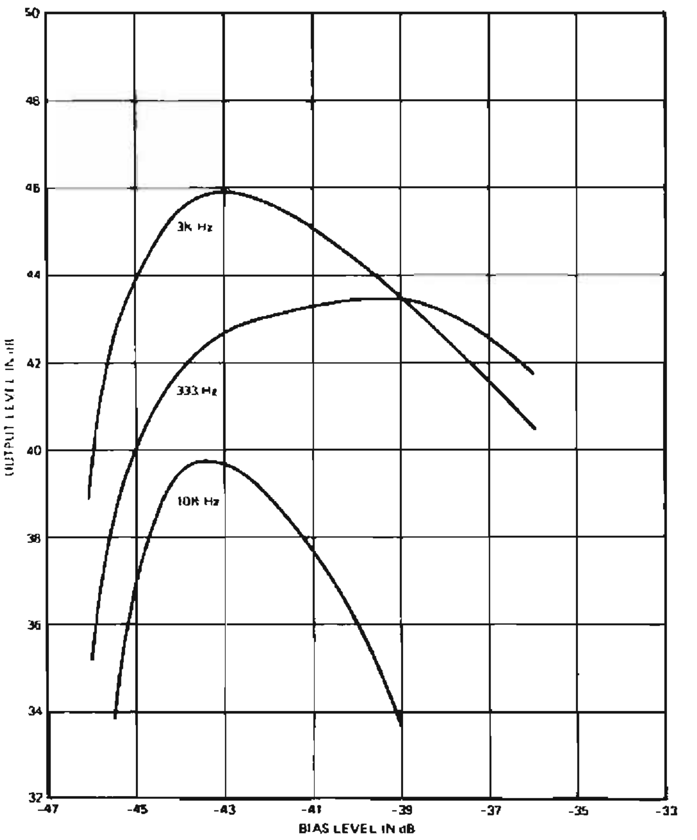


Fig. 2.1—Output/bias level for low noise, high output tape.

another oxide of iron. The effect of these impurities is to raise the coercivity of the particle and increase its magnetic efficiency. The formation of the crystal is not improved. Additionally, there is a tendency for chemical instability which results in some magnetic instabilities under certain conditions which can easily occur in practical use. There have been several short-lived attempts to make tapes from these types of particles over the past 38 years. Time alone will tell whether today's particle chemists have solved the problems.

Category 4 uses a pure ferric oxide particle chemically identical with that used in category 1 which means that it has all the inherent stability and other properties which have made γFe_2O_3 the only wholly successful magnetic compound of iron for tape since its introduction in 1936. The improved performance is obtained entirely because of a perfected crystal shape with a length/width ratio of approximately 10:1. The absence of dendrites and holes gives the tape designer the capability of increasing the magnetic density and, hence, the magnetic efficiency of the coating. Much better orientation is also achieved and the resulting tape is considerably more efficient at all frequencies. Because of the better particle packing, dispersion, and orientation, the undesirable modulation noise effects caused by magnetic discontinuity are significantly reduced. At the time of writing, we believe MRX_2 is the only cassette tape containing this magnetic oxide.

Figure 1 shows the differences in the frequency response of these four categories of tape at 1⅞ ips when recorded with the bias carefully optimized for each and the signal recorded at a level 20 dB below that level which gives 2% total harmonic distortion at low frequencies. For the purpose of showing the differences in response between these tapes, the recording pre-emphasis was maintained at the optimum for the perfected gamma ferric oxide. The chromium dioxide essentially has the same output: i.e., the same sensitivity at low frequencies as the low noise, high output tape, whereas the cobalt modified and the perfected particle have a higher output at the long wavelengths resulting from approximately 2 dB greater sensitivity and the ability to accept a higher recording signal. The perfected particle also has a greater efficiency at the high frequency or short wavelengths which result in up to 8 dB more sensitivity at 10 kHz at 1⅞ ips when compared with high output, low noise ferric oxide tapes and about 2.5 dB less sensitivity than chromium dioxide tape.

Figure 2 gives typical bias output curves for each of the four types of tape at three signal frequencies, 333 Hz, 3kHz and 10 kHz. The optimization points for the three ferric oxide tapes are very similar provided the criteria of optimization is that over-bias which reduces 10 kHz signal by 3½ dB. As is well known, chromium dioxide requires approximately 40% more bias current to provide adequate biasing field. Decreasing the bias slightly would obviously improve the high frequency performance; however, this is undesirable from the point of view of long wavelength distortion and it also increases the susceptibility to drop-outs caused by surface asperities.

As with any other magnetic recording system, the highest biasing frequency possible should be used to minimize modulation noise and beat effects. The data given later in this article was taken with a bias frequency of 102 kHz and the even harmonic distortion present in the bias waveform was 0.05% second harmonic. This low even-order harmonic distortion is essential to minimize the effects of d.c. noise and second harmonic distortion of the signal due to unbalanced bias waveform.

Equalization

The standard replay equalization for cassettes operating at 1⅞ ips has a bass roll-off created by a circuit with a time constant of 1590 microseconds and a high frequency boost with a 120 microsecond time constant. Recently, a second equalization standard has been proposed to permit fuller use of the characteristics of modern tapes, specifically chromium dioxide. The new proposed standard has a low frequency roll-off of 3180 microseconds with a 70 microsecond equalization curve at the high frequency and recorders are now on the market which use this proposed standard.

The two replay characteristic curves are shown in Fig. 3. The old standard has the advantage that with improved high frequency performance of tapes, the high frequency compression generated by tape overload is significantly reduced because of the reduced recording pre-emphasis required to produce a flat frequency response. However, under these conditions, the use of chromium dioxide would not significantly improve signal-to-noise ratio of the system when compared with the same system using low noise tape; it would only result in an extended frequency response and reduced modulation noise. The proposed new standard improves the signal-to-noise ratio at the expense of the greater risk of high frequency compression; however, with chromium dioxide, this compression is no worse than with low noise, high output ferric tapes using the 120 micro-second equalization curve. Excellent results can be achieved by using the same recording pre-emphasis for both chromium dioxide and low noise tape

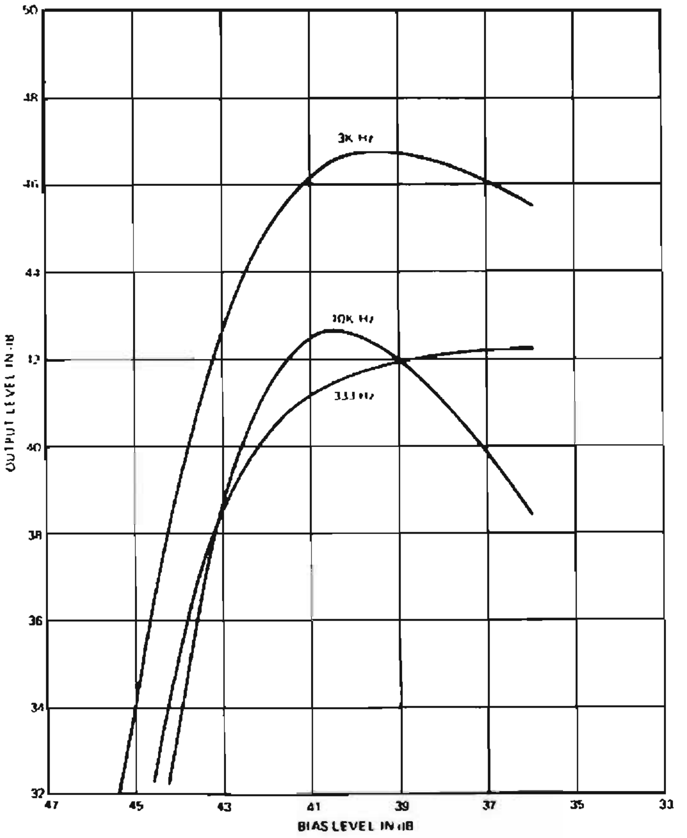


Fig. 2.2—Output/bias level for chromium dioxide tape.

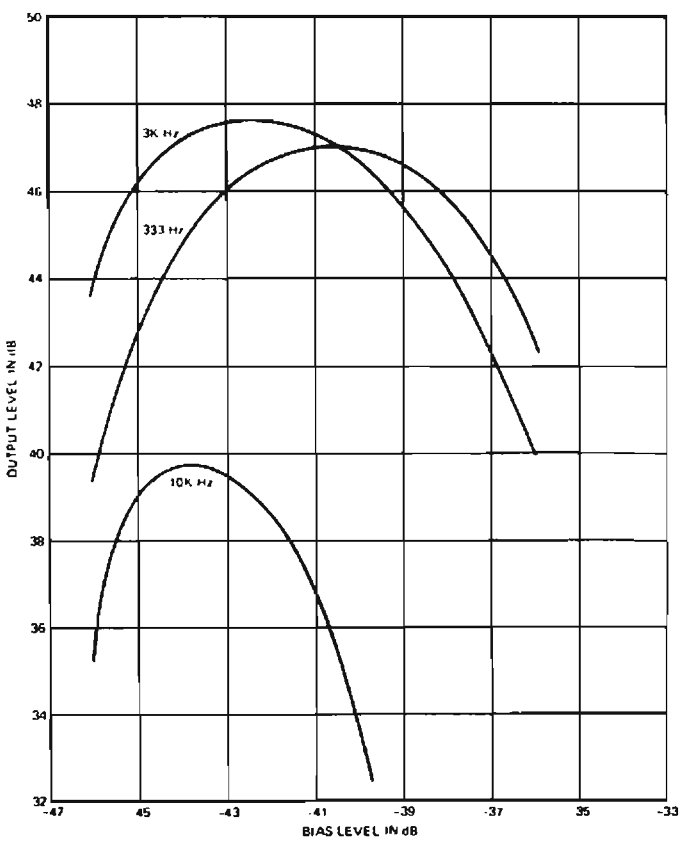


Fig. 2.3—Output/bias level for cobalt modified tape.

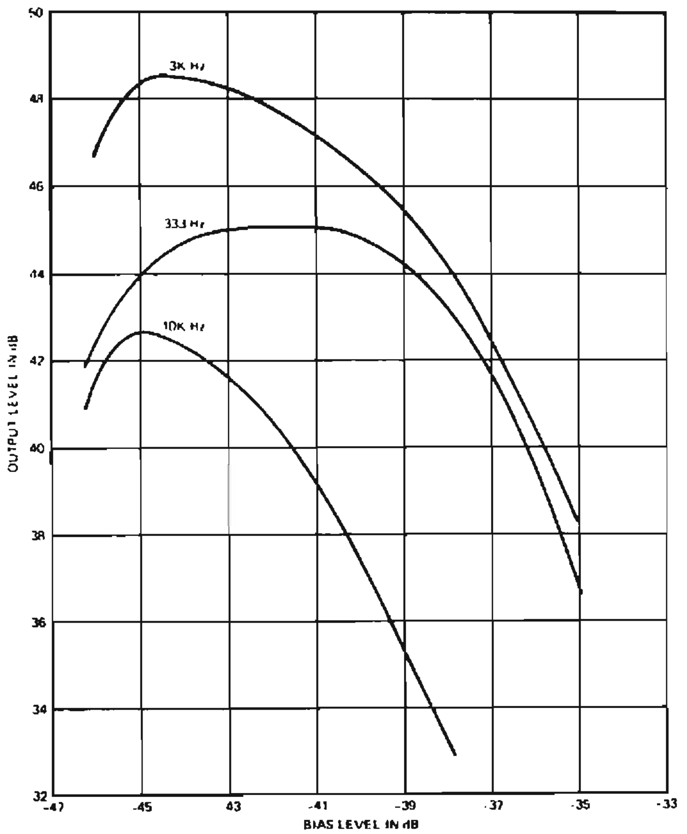


Fig. 2.4—Output/bias level for improved gamma ferric oxide tape.

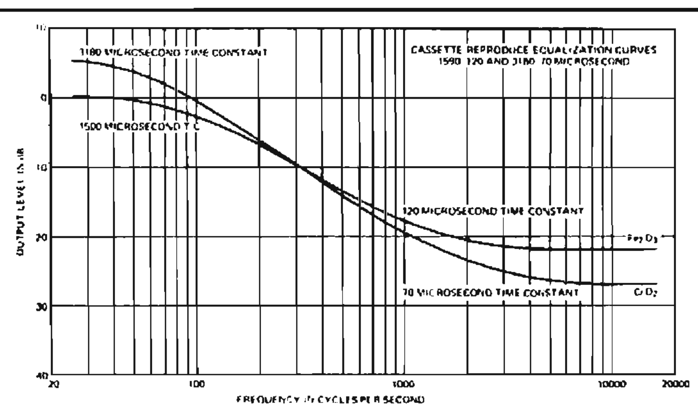


Fig. 3—Equalization energy standards.

by switching the bias and replay equalization leaving the recording pre-emphasis the same for both tape types.

The change at the low frequency end reduces the risk of low frequency distortion. A good case can be made for eliminating all low frequency pre-emphasis in the recording process and, thus, removing the need for the roll-off at low frequencies on replay. With modern solid state circuitry, the elimination of power line generated noise is relatively simple and inexpensive. The reason for the low frequency de-emphasis on replay was to simplify the electronics designers' problems with hum. The reason no longer exists with cassette tape and the heavy bass which is characteristic of much modern music makes it painfully difficult for a duplicator to record a satisfactory cassette without low frequency overload. The elimination of this bass equalization would significantly assist in this problem.

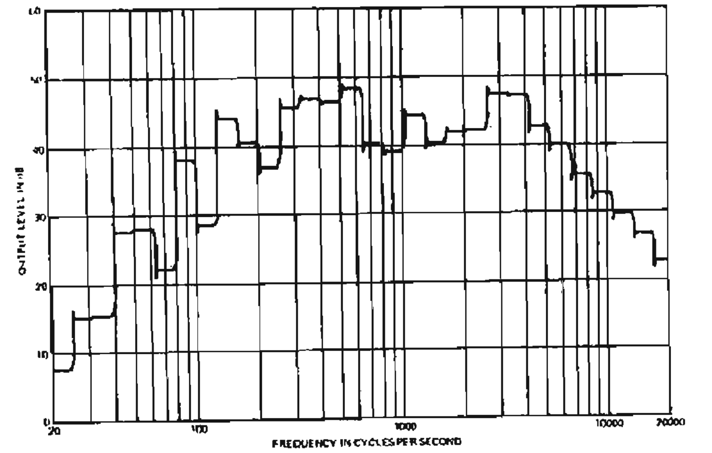


Fig. 4—Recorded frequency spectrum for cymbals in Holst's *Planets*, DG 2530-102.

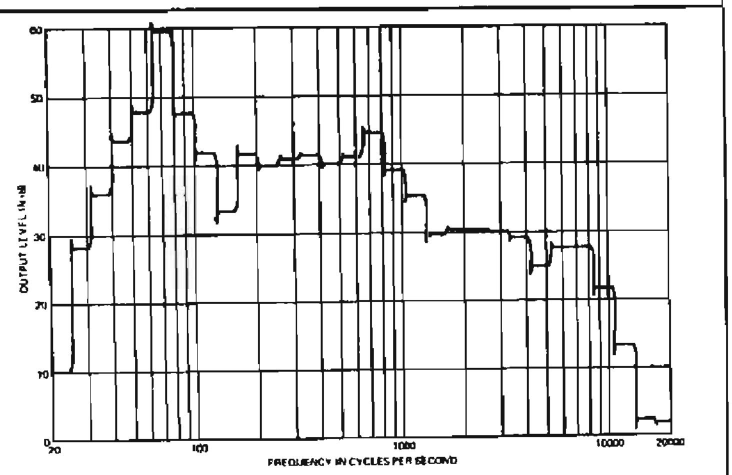


Fig. 5—Recorded energy spectrum for *Mendocino*, Polydor 24-4508.

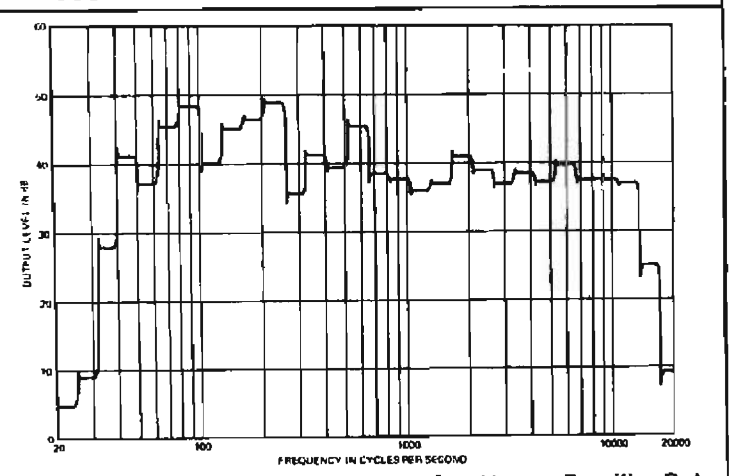


Fig. 6—Recorded energy spectrum for *Happy Brasilia*, Polydor 24-4509

Music

Before considering further the demands placed upon cassette tape by equalization and signal-to-noise ratio improvement systems, it is appropriate to examine the energy spectra of the musical sources available. The most likely source of high quality signal within the scope of the home user is the phonograph record; few users have a better source of quality music, such as high speed master tapes, and with the average standard or quality exhibited by today's FM broadcasters, even when they are transmitting from tape, the transmission quality is such that it rarely reaches the fidelity available from even moderate quality discs.

Analysis of the spectra of two or three disc selections by means of the GR Real Time Analyzer gave the spectra shown in Figs. 4 through 6. Figure 4 is the spectrum of a cymbal crash from Deutche Grammophon's recording of the Boston Symphony/Steinberg performance of the Holst *Planets* Suite. As can be seen, there is considerable high frequency energy to the limits of the analyzer at 20 kHz and the energy from 125 Hz through 5 kHz approaches a horizontal line which, with the setting of the analyzer used, means that the energy was reducing at the rate of 3 dB per octave with rising frequency. Figure 5 is from a record made by a combination using a wide variety of percussion instruments with very strong electronically generated bass. In this record, energy is concentrated around the bass tones at 80 Hz and falls off fairly rapidly up to the limits of 10 kHz where apparently the record cuts off. Figure 6 is of some Latin American music, using heavy orchestration with percussion, electronic instruments and brass; this disc has an energy spectrum approaching that of pink noise. These examples by no means represent an exhaustive study; however, they do point to the fact that discs can easily be found with a very wide recorded bandwidth and high energy levels at the extremes of the band. The duplicator of music cassettes obviously has to cope with tape masters having energy at high levels over the whole of audible band which present a formidable problem to him.

It would appear from these analyses that the use of pink noise to study the behavior of a recording system is a test technique with greater validity than the use of pure sine waves at discrete frequencies.

The use of recording pre-emphasis which rises at high frequencies at a rate greater than 3 dB per octave will eventually result in tape overload when trying to record, from records such as those analyzed, if the record level indicators do not take account of the modified frequency characteristic created by the pre-emphasis. "Flat" level indication presumably is used by equipment designers on the assumption that musical spectra still conform to the classical spectra published in most of the literature which show considerably reduced energies at the very low frequencies and at frequencies above 5 kHz. Modern orchestration involving the use of synthesizers and electronically reinforced instruments has changed the picture.

The Compromise

The problem of establishing good high fidelity performance and the choice of equalization resolves itself into a compromise between tape overload or compression at the short wavelengths and a good signal-to-noise ratio. Pre-emphasis in the recording mode reduces the replay equalization necessary at the price of the reduction in high frequency performance at high signal levels with, consequently, high intermodulation distortion; the benefit of this choice is that the reproduced tape noise is lower than with a system where most of the equalization for high frequency losses is done on replay.

Within the limits of the existing standards, the biggest contribution the tape designer can make is to increase the sensitivity of the tape and/or maximum usable output from the tape at all frequencies, without deteriorating the fundamental bias noise of the tape or the frequency response and, thus, provide greater output on playback. The tapes developed with this aim include categories with chemically modified particles and the improved gamma ferric oxide particle. As has been stated, chromium dioxide does not increase the sensitivity of the tape over the whole band but does provide much improved performance at the very short wavelengths (i.e., the high frequencies); therefore, it does not meet the goal. The cobalt and magnetite doped gamma ferric oxide particles provide a much increased sensitivity at all frequencies and the improved gamma ferric oxide, of the MRX_2 type, gives a greater improvement in the performance at the short wavelengths. All three types will give an improved signal-to-noise ratio by virtue of replay output which is increased by as much as 4 dB.

The improved gamma ferric oxide tape of category four has the added advantage of significantly improved short wavelength performance which enables the recording pre-emphasis to be reduced by up to 8 dB at 10 kHz at 1⅞ ips. Thus, with this type of tape, not only is there an improvement in signal-to-noise ratio, there is an improvement in high frequency overload or compression. As will be seen in the following discussion, this reduction in compression improves the situation when signal-to-noise reduction systems such as the B Dolby are used; it results in improvements in system tracking when compared with the response errors which can occur with tapes which have significant compression problems.

Noise Reduction Systems

For practical purposes, this discussion is limited to the B Dolby signal-to-noise improvement system, since other systems are either similar in behavior or are not seriously affected by the behavior of the recording system. Also, the majority of the machines equipped with a noise reduction system use Dr. Dolby's circuitry and the only "stretched" pre-recorded cassettes in production by duplicators use the B Dolby mode.

During the recording process, the Dolby circuit detects the high frequency levels of the incoming signal. When these signals are below a pre-determined level, the gain of the amplifier is increased to boost the high frequencies before they are recorded; in addition, the frequency at which the boost starts is varied in relation to the HF signal level. The maximum boost at the lowest HF signal level is in the order of 10 dB. No account is taken of the low frequency signal level; low frequencies are recorded unmodified. On replay of the recording, the process is reversed.

From the viewpoint of the tape, the Dolby provides a variable high frequency pre-emphasis, the degree of which is dependent on the high frequency signal level; the lower the level the greater the pre-emphasis. A difficulty with this system is that the degree of tape magnetization does not take account of frequency, but responds to the sum of the energy at all frequencies at any given instant. Therefore, if one has the situation where the low frequency signal level is very high, approaching the usable recording limits, and riding on this high level of bass there is a high frequency signal at lower level, such as sibilance on a voice or a quietly brushed cymbal, the Dolby circuit will boost the level of these high frequencies and can drive the tape further into high frequency compression. On replay, because of the recording errors, the high frequency signal level is lower than it would have been if there had been no compression; therefore, the Dolby will react to this low level and reduce the gain at high frequencies by an amount which is greater than the boost which was applied during the recording process. The result of this tracking error is a loss of brilliance and an increase in distortion which is not a fundamental fault of the recording system, neither is it a malfunction of the signal-to-noise improvement device.

Figure 7, Curve A, shows the transfer characteristics of a cassette system at 10 kHz using low noise, high output tape. Curve B is the transfer characteristic of 10 kHz recorded and played on the same system in the presence of an 80 Hz tone recorded at "0" level, that is, at the same level as the pronounced energy peak shown in Fig. 5. Curve C is the transfer characteristic of the same 10 kHz signal in the presence of 80 Hz at "0" level but with the use of B Dolby. The increase in compression at the "0" level at 10 kHz caused by the presence of the 80 Hz signal is 1.0 dB and the use of Dolby gives a further response error of 2.0 dB. A more significant problem is probably the increase in distortion; the lower frequencies will produce audible harmonic distortion and the high frequencies whose harmonics are outside the system pass band produce intermodulation products within the replayed bandwidth.

If compression effects described above are to be avoided using conventional tapes and a Dolby stretcher, the recording level must be reduced. This, in turn, reduces the replay level and decreases the basic signal-to-noise ratio which, of course, reduces the effective improvement achieved by the use of the Dolby.

Another effect, which can easily be avoided with the self-contained recorder, but is a little more difficult to establish control over with pre-recorded cassettes using the B Dolby characteristic, is the effect generated when the recorded bandwidth is greater than that which can be reproduced. Most recording systems are capable of recording to shorter wave-

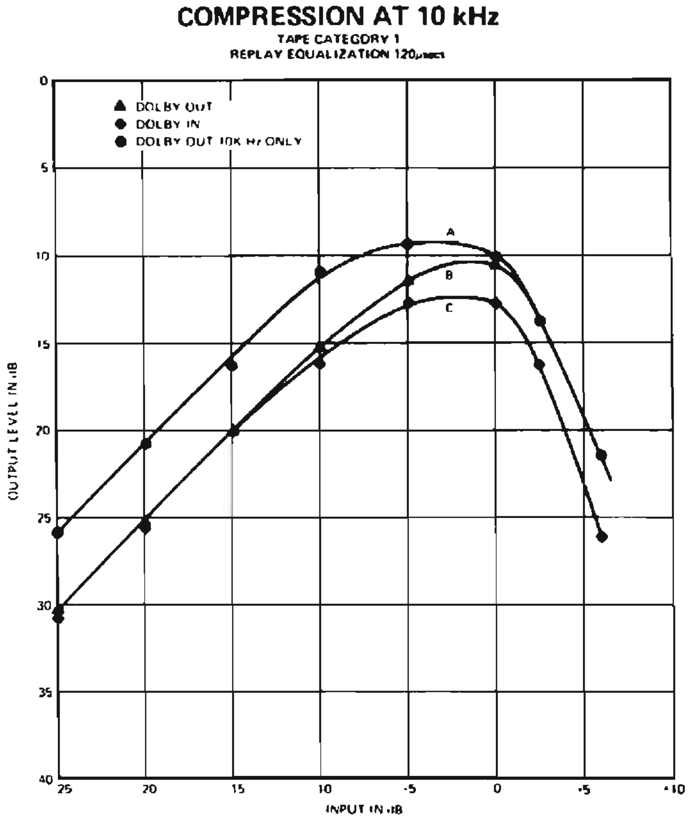


Fig. 7—Dolby B System transfer characteristics.

lengths than the replay channel of the recorder can satisfactorily reproduce; the limitation being the replay head gap length. If a wide band signal is received by the recorder such as that shown in Fig. 4, the lower high frequency levels; i.e., from 12 to 20 kHz, will be sensed by the Dolby and be pre-emphasized before recording. If now the recorder only reproduces up to 12 kHz, these signals will not be received by the Dolby circuit on replay. Therefore, the Dolby loop is not correctly closed and there is no corresponding reaction from the replay circuit to correct the level change generated in the recording mode. In a severe case, this tracking error results in a frequency response with a significant dip at low levels in frequencies around 5 kHz as shown in Fig. 8. The frequency responses shown in Fig. 8 are taken at 10 dB intervals with the top one at a level equivalent to maximum recording level. Responses B and D are taken with the recording bandwidth wide open and A and C were the controlled recording bandwidth. The mid-frequencies would not be significantly boosted during the recording process, but on replay the B Dolby HF gain reduces to its minimum because of the much reduced high frequency energy in the replay signal. A possible solution to this problem for the duplicator is to limit the bandwidth which activates the Dolby, during the recording of music cassettes, to about 10 kHz. With a cassette recorder, another solution is to design the recording amplifier to have the same bandwidth as the replay system.

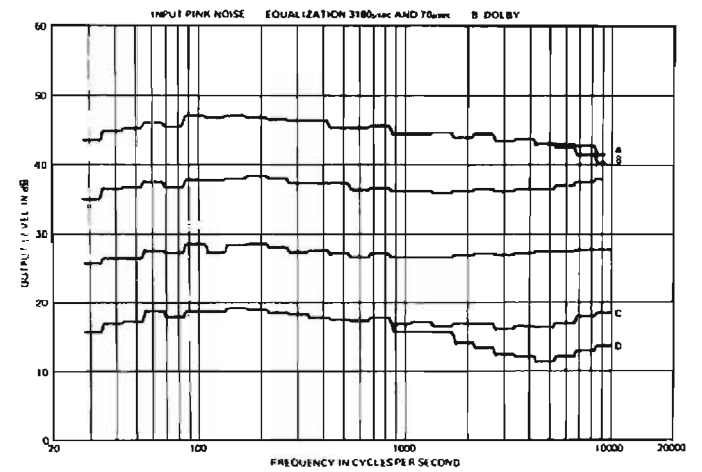


Fig. 8—Bandwidth mistracking.

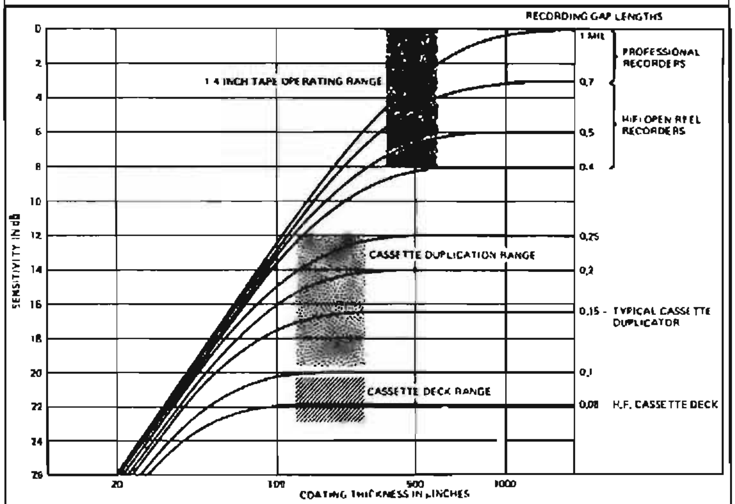


Fig. 9—Recording gap length effects.

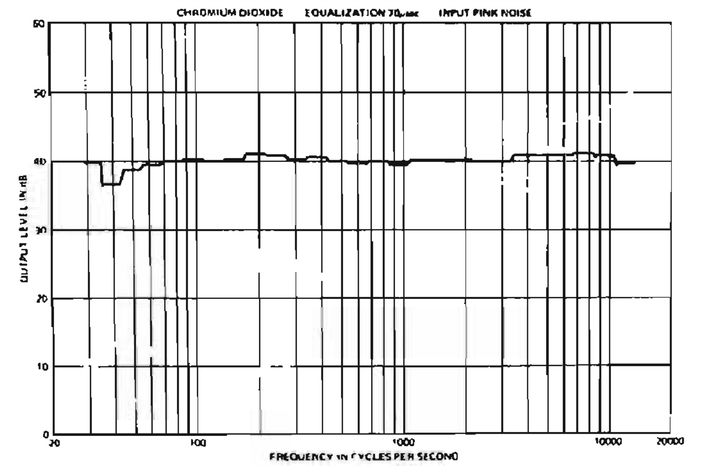


Fig. 10.1—Frequency response for chromium dioxide with pink noise input.

Recording Gap Length

The full presentation of the study into the gap-length/coating thickness relationship will be published as a separate article. However, it is appropriate for the purposes of this article to publish the theoretical relationships shown in Fig. 9. These have been verified experimentally. As can be seen with recording gaps which are shorter than the coating thickness, the performance of the tape/record head combination is not dependent on the coating thickness because they do not utilize the whole coating. Gaps which are significantly longer than the coating thickness record through the whole of the magnetic layer and, therefore, the sensitivity of the system at long wavelengths is coating thickness dependent.

Apart from the fact that a duplicator operates at speeds which are much higher than 1⅞ ips, the principal difference between recording on a duplicator and a consumer cassette recorder lies in the dimension of the recording head gap. Most duplicators use special record heads whose gaps are in the region of 150 microinches to 400 microinches long, whereas the consumer machines use dual purpose heads which have a gap whose dimensions are determined by the desired replay performance. On high fidelity machines intended to record and play frequencies up to 15 kHz, an 80 microinch or smaller gap is essential. Typical cassette tape coating thicknesses lie in the range from 120 microinches for C-120 product to 250 microinches for some C-60's.

The development of higher efficiency ferric oxide particles of the type used in MRX_2 tape gives the tape designer the freedom to optimize the coating thickness for overall performance on a duplicator at coating thicknesses considerably thinner than has been the former practice. This has several advantages:

1. The coating thickness can optimize to the biasing requirements at the short wavelengths without sacrificing distortion and output at the long wavelengths and this bias can be adjusted to be approximately the same as with conventional gamma ferric oxide tapes when using a typical duplicator record head. Because of the improvements in the oxide, the output available from the thinner coating is 4 dB greater than with high output low noise tapes at low frequencies and 8 dB at high frequencies.
2. The same coating thickness can be used for all configurations.
3. The thin coating of approximately 130 microinches does not sacrifice any performance when used in a blank cassette on a consumer machine.

Practical Systems

Two separate high fidelity systems have been used for tape evaluation and the parameters chosen for both systems are based on the study described and utilized consumer type cassette decks carefully adjusted to meet our requirements. Most of the listening tests and demonstrations of recorded quality have been performed without the use of any noise reduction system; although some testing has been carried out to determine whether the data presented earlier is, in fact, important in relation to what is heard. The recordings used were made from very high quality 15 ips stereo masters and which have recorded signals at significant levels to 20 kHz. There is little doubt that where the high frequency energy is present in the input signal, the variable frequency response generated by the B Dolby System is audibly worse than with the same tape and recorder used without the Dolby in circuit. Apart from this reason, the noise reduction system was not used because the objective of our study was to evaluate tapes under development; for this purpose it is better to compare tape performance with as little intrusion from electronics as possible.

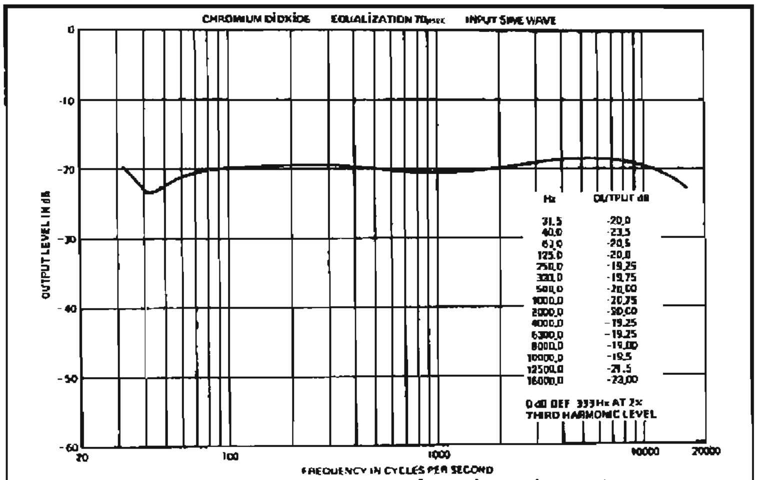


Fig. 10.2—Frequency response for chromium dioxide with sine wave input.

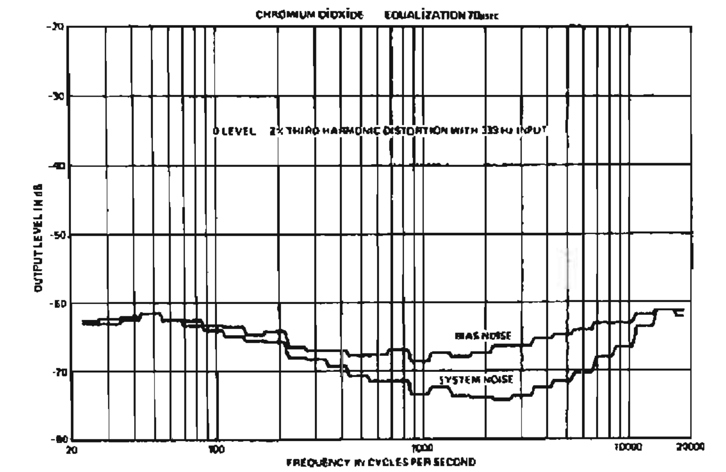


Fig. 11—System and bias noise for chromium dioxide.

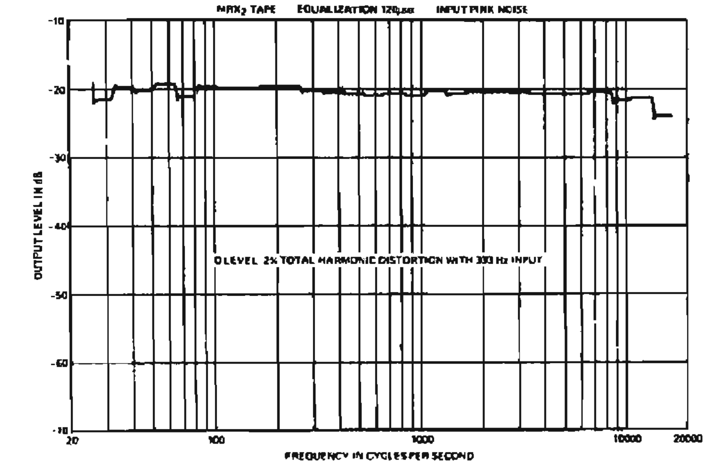


Fig. 12—Frequency response for improved gamma ferric oxide tape

The first system was designed for chromium dioxide and incorporated the new proposed replay equalization at 3180 microsecond bass curve and 70 microsecond treble curve. Figures 10 and 11 respectively give the frequency response and the noise spectra of this system using Memorex chromium dioxide tape. As a matter of interest, the response is presented in the form generated by the noise/analyzer system and the more conventional presentation taken manually with sine wave signals. The dip in the response at 40 Hz is caused by the contour of the record/play head. The unweighted signal-to-noise ratio is 53 dB referred to 333 Hz at the level which gives 2% total harmonic distortion. At mid-frequencies, the slot noise is –65 dB.

The second system used for the improved gamma ferric oxide MRX_2 used the standard replay system of 1590 microseconds bass curve and 120 microsecond HF curve. The response of this system is shown in Fig. 12; Fig. 13 shows the spectra of the bias and the system noise. The unweighted signal-to-noise ratio is 52.5 dB referred to 333 Hz and 2% total harmonic distortion and the slot noise is 71 dB at mid-frequencies. The excellent signal-to-noise ratio of the MRX_2 ferric oxide system is due to the 2 dB extra sensitivity of this tape at long wavelengths plus the capability of accepting 2 dB more recording drive without the bias noise having been deteriorated in comparison to low noise high output tapes. Thus, the unweighted signal-to-noise ratio is 4 dB better than one would obtain from standard ferric oxide particles. The slot noise at mid-frequency is 6 dB better than with chromium dioxide but because the 120 microsecond replay equalization was used for MRX_2 and the 70 microsecond for chromium dioxide, the final signal/noise ratios are approximately the same. However, MRX_2 exhibits less high frequency compression than chromium dioxide when the two tapes are equalized in these differing manners.

Acknowledgements

The author wishes to acknowledge the excellent experimental work carried out by Mr. Roy F. Nelson of Memorex Corporation, Audio/Video Group, without which this article would not have been possible. The curves for the recording head gap/coating thickness relationships were calculated by Mr. E. D. Daniel.

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[1] *Acoustical Tests and Measurements*, Davis; Published by Howard W. Sams and Company, Inc.; Page 60.

[2] *Chromium Dioxide Audio Cassette Tape*, Jordan, Kerr and Dickens; *Journal of the Audio Engineering Society*, January/February 1972.

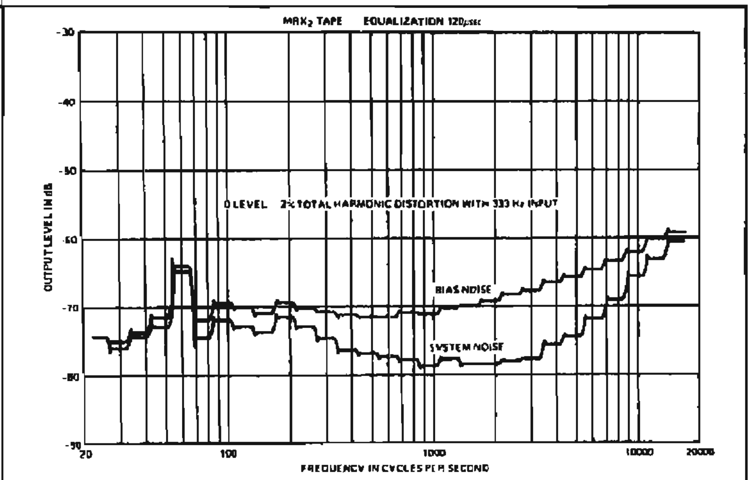

Fig. 13—System and bias noise for improved gamma ferric oxide tape.

CASSETTE TAPE RECORDING BIAS

Martin Clifford

IN TUBE and transistor circuits, the word bias means a voltage, generally d.c., applied to some element of an active component to produce a linear output. There are exceptions, of course, most notably in the case of class-C amplifiers functioning as frequency multipliers. However, for audio applications, the role of bias is to help ensure undistorted output from tubes or transistors.

For a component such as a tube, for example, bias in the form of a negative voltage determines the quiescent or operating point Q, as shown in Fig. 1. The graph or transfer characteristic is a plot of grid voltage vs. plate current. This first drawing shows that a bias of -2.5 volts results in a plate current of slightly more than 4 milliamperes. When a signal voltage (Fig. 2) is applied, its effect is to increase and decrease the amount of bias, producing an equivalent variation of plate current. Because the swing is between points A and B, the linear portion of the tube's plate current—grid voltage characteristic, the variation in plate current (the output) is also linear. However, if the bias is incorrect, either too large or too small, the output waveform is distorted. Fig. 3a shows the effect of insufficient negative bias; Fig. 3b excessive bias.

This isn't as far removed from bias for magnetic tape as you might think, for one of the functions of tape recorder bias is to help produce linear output. But there the similarity ends, for tubes and transistors are amplifying devices; magnetic tape is not. Magnetic tape, though, possesses an ability that tubes and transistors do not have—the ability to retain the signal, pending amplification. However, whether this retention is linear or nonlinear depends on the way the tape is biased during recording. It may seem strange that magnetic tape can be biased in a manner reminiscent of tubes and transistors, but not when you consider the way in which tapes are magnetized and demagnetized.

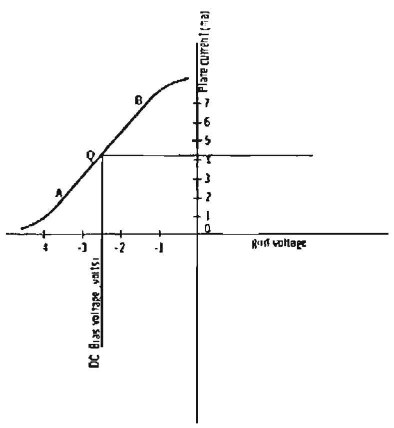

Fig. 1—Grid voltage vs. plate current curve. The d.c. bias voltage determines the operating point, Q.

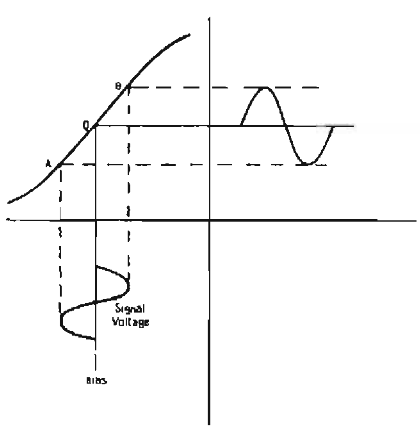

Fig. 2—Because of the presence of d.c. bias, operation is along the linear portion of the characteristic curve.

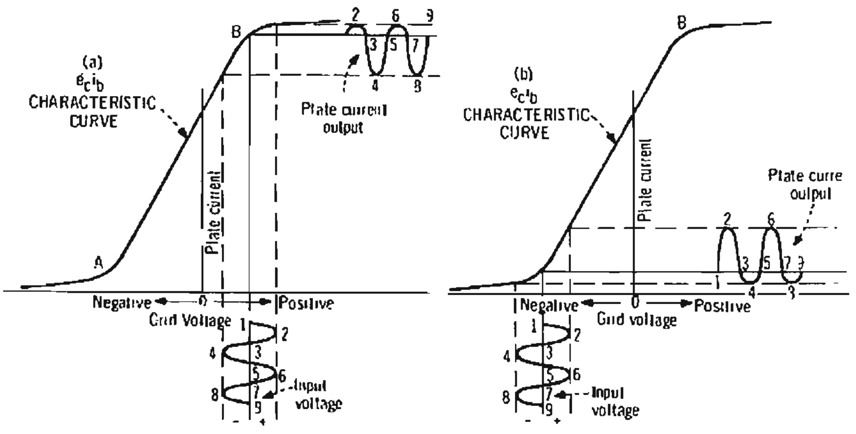

Fig. 3—Distortion occurs when bias is moved in an excessively positive direction (a) and also when it is made excessively negative.

Hysteresis Loops

The magnetic behavior of substances can be graphed, just as it is possible to plot the characteristics of tubes and transistors. The number of available magnetic flux lines or flux density per unit area (represented by the letter B) depends on the permeability of the material. Various substances have different amounts of reluctance to the presence of magnetic lines, much as they also have differing amounts of resistance to the passage of an electric current. A simple example would be a horseshoe magnet with a given magnetic strength, H. The number of magnetic lines of force existing between the adjacent north and south poles of this magnet would depend on the material placed between the poles. For iron there would be more lines of flux; for air, fewer.

A permanent magnet represents a condition in which the magnetizing force, H, is relatively constant. The magnetizing force, however, could be variable, as in the case of an electromagnet, produced by a varying alternating current flowing through a coil. A ferrous substance surrounded by the magnetic field around the coil would become magnetized, first in one direction, and then in the other. The poles of the ferrous substance, possibly a small iron bar, would keep reversing, in step with the frequency of the magnetizing current flowing through the coil. Further, the resulting magnet would also vary in strength, possibly ranging from weak to strong.

Figure 4 is a graph of the behavior of the ferrous material. At first nothing happens when the magnetizing force, H, is increased from 0 to 1, moving to the right along the horizontal axis representing the magnetizing force. This "non-action" can be considered in the same way as applying a force to a stalled heavy object, such as an automobile, in an effort to get it moving. There will be no action until the applied force can overcome the inertia of the car. In the same way, the magnetizing force H must overcome the inertia of groups of iron atoms to magnetization. Once the magnetization process starts, though, the flux density, B, of the substance rises rapidly. Note the distance along the H axis from 1 to 2 is about the same as from 0 to 1—that is, each of these distances represents an equivalent amount of magnetizing force. From 0 to 1 nothing happens, yet from 1 to 2 the value of B rises rapidly.

As H is increased, B increases, but not indefinitely. At point C on the graph, any further increases in H will produce only a small increment in B, and so we call this the saturation point. If the graph were to continue beyond point C, it would start to assume a slope parallel to the H axis.

If the magnetizing force is now decreased, the level of the flux density of the material that was magnetized will also decrease but some magnetic flux will remain, even if the magnetizing force H is removed. At point D on the graph, for example, the value of H is zero, but the substance is still partially magnetized. We can, of course, return the material to its original, unmagnetized condition, but only by applying a magnetizing force in the opposite direction. The point at which the graph crosses the -H axis (-1) indicates complete demagnetization, but note that we are now on the -H part of the horizontal axis. If the magnetization is continued, the substance will become more and more magnetized, but the limit will be reached at point A. Here the application of the magnetizing force will not result in much of an increase in the flux lines around the object being magnetized. Again, this is a saturation point.

At point A we can gradually reduce the amount of magnetizing force until it reaches zero. At this juncture, the graph crosses the vertical axis at point E. If we were to stop here, the magnetizing force would once again be zero, but the substance being worked on would still be a magnet—that is, it would be surrounded by its own magnetic lines of flux. This is comparable to the situation that prevailed at point D on the graph, but with one difference. The poles of the substance being magnetized have been transposed.

If we now apply a magnetizing force of +H, similar to that used originally, we will reach point 1 on the graph. The resulting graph looks somewhat like a loop and is called a hysteresis loop. The shape of the loop depends on the kind of material being magnetized.

Shape of the Hysteresis Loop

The hysteresis loop of Fig. 4 is a basic diagram used by engineers to indicate magnetic properties. In general, the closer the loop approximates a square—that is, the greater the area enclosed by the curve—the better this characteristic will be for recording purposes (Fig. 5). The vertical axis of the graph represents retentivity, while the horizontal axis is coercivity. Retentivity accounts for higher output and better low-frequency response; coercivity is responsible for extended high frequencies. Coercive force is the force required to reduce magnetism to zero; it can be regarded as a magnetizing force applied in a negative direction. Retentivity is the magnetic flux that remains in tape after saturation with the magnetizing force returned to zero.

Saturation Curve

When a substance is subjected to a magnetizing force, there is at first a slow increase in the amount of magnetization, followed by a linear rise in which the amount of magnetization is proportional to the magnetizing force. The remainder of the curve becomes nonlinear as magnetic saturation is approached. Known as a normal saturation or transconduction curve (as indicated in Fig. 6), it is derived from the graph of the hysteresis loop.

The normal saturation curve can be drawn without its accompanying hysteresis loop, as in Fig. 7. The lines between

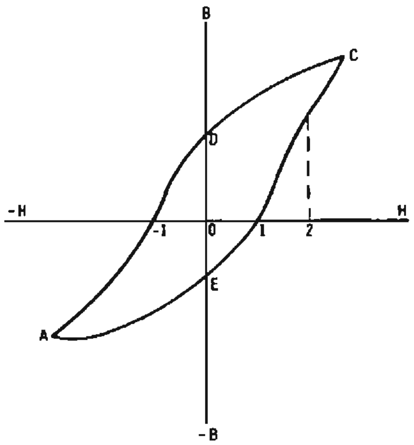

Fig. 4—Graph of the magnetizing and demagnetizing behavior of a ferrous material.

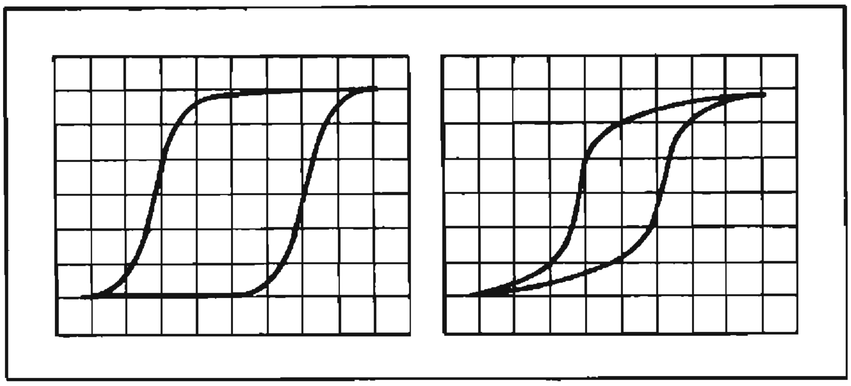
Fig. 5—Conventional hysteresis loop (right); more desirable curve (left)

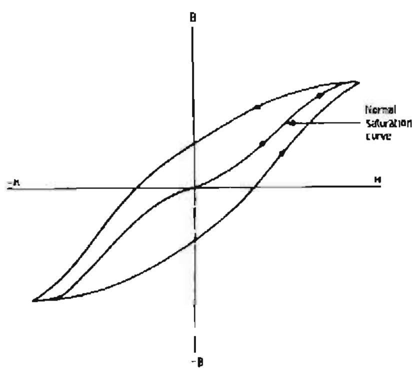

Fig. 6—Saturation or transconduction curve.

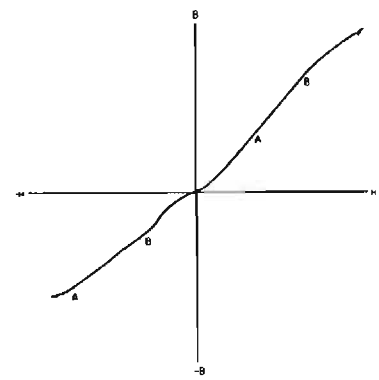

Fig. 7—Saturation curve without the hysteresis loop from which it was derived. A-B and A'-B' are linear portions.

points A and B and A' and B' are the linear portions of the curve. Note the similarity between this curve and the transfer characteristic shown earlier in Fig. 1. We can now use this curve to show the effect of recording a signal on tape.

In Fig. 8 the normal saturation curve is shown above and below the H axis. If magnetization in the reverse direction has the same force as forward magnetization, the lower half of the curve will be a mirror image of the upper half. Note, in Fig. 8, there is no bias and the only input is that of the signal itself. While the input is a sine wave, the output is a distorted waveform since just the nonlinear portion of the graph is being used. To overcome this condition, bias can be applied to put the operating point on the linear portion of the curve. The curve, however, has two linear sections, one above the H axis and the other below it. The bias could be d.c. and with one polarity would utilize the lower portion of the curve or with the opposite polarity, the upper linear portion. In early tape recorders that is what was done. This kind of biasing technique, however, takes advantage of only one small section of the saturation curve, and as a result, d.c. biased tape recorders had a restricted dynamic range. The modern technique is to use sinusoidal a.c. for bias.

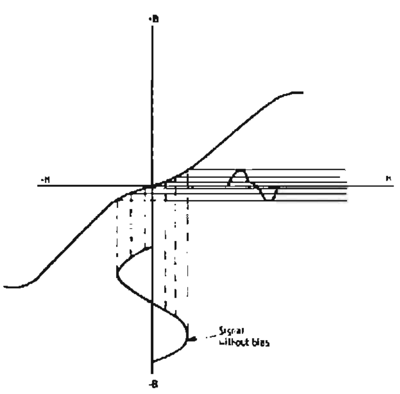

Fig. 8—Saturation curve and applied signal without bias.

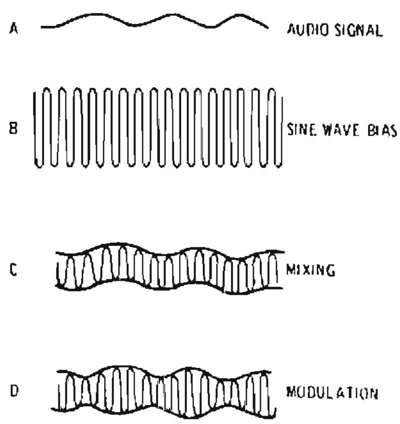

Fig. 9—A.c. bias plus signal is a mixing process, not modulation.

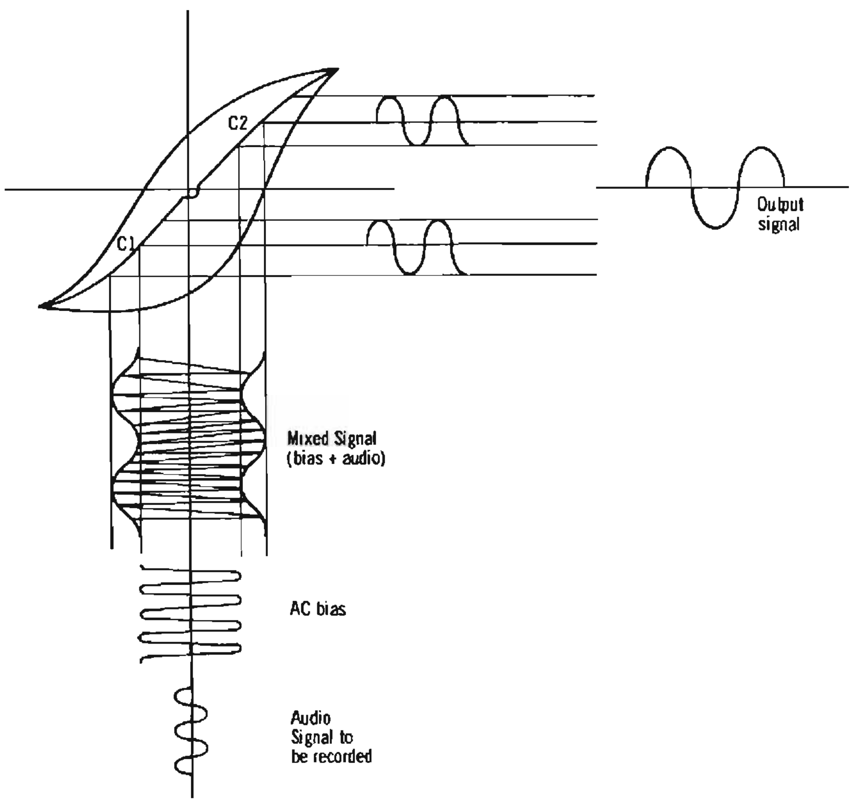

Fig. 10—Effect of a.c. bias is to put signal on linear portions of saturation curve.

Mixing vs Modulation

In broadcasting, the process of loading an audio signal on a sine wave carrier of much higher frequency is called modulation. However, in tape recorders, the audio signal does *not* modulate the bias but mixes with it. Figure 9A shows an audio signal while 9B in the same drawing represents sine wave bias. With mixing the amplitude of the bias remains constant, while in modulation the instantaneous values of the carrier keep changing. However, with either mixing or modulation, if we join the peaks by an imaginary line we will have a graph of the audio signal. Since there are two peaks—a positive and a negative peak for each cycle of bias—the result produces the effect of a duplication of the audio signal.

Biasing Tape

Using sine wave bias instead of d.c. now presents us with a technique for working with both linear portions of the transconduction curve. Figure 10 shows the complete action. Here we have the hysteresis loop and its resultant magnetization curve. The audio signal is mixed with an a.c. bias current and it is this mixture that is applied to the tape during recording. Note that the mixed signal has an upper and lower audio component and that each of these components is a replica of the original audio signal. Our magnetization curve, because of the presence of the a.c. bias, now has two operating points, both of which are centered on the two linear portions of the characteristic. This depends on the amount of bias current which must be such that the audio signal portion of the mixed signal is applied to *both* linear portions of the magnetization curve. Essentially, there are two outputs—one for the lower part of the curve and one for the upper portion. Both combine to supply a single output signal.

Now what about the nonlinear section of the magnetization curve? The bias is being applied to both portions—that above the H axis and that below it. This means there will also be nonlinear outputs. However, these cancel since they are out of phase. The technique is the same as that used to get harmonic cancellation in the output of a pushpull amplifier.

The a.c. bias, for linear output, must not only be a pure sine wave, but must be evenly distributed around its X axis. This means there must be no d.c. component present in the bias since this would have the effect of pushing the operating point up or down on the magnetization curve, depending on the d.c. polarity.

Linearity of output is also dependent on the lengths of the straight line portions of the magnetization curve, and these, in turn, depend on the way the cassette tape is manufactured. If the straight line portions are small relative to the amplitude of the recording signal, the result will be use of the nonlinear sections of the curve, producing distorted output. If, however, the input audio signal is deliberately restricted to avoid this possibility, then the linear portion may be underutilized, resulting in limited dynamic range.

More About Bias

Bias, then, is a constant high-frequency signal that is mixed with the signal to be recorded. Its prime function is to permit recording a signal on magnetic tape in such a way that the output is distortionless, while at the same time supplying a good dynamic range. But this is by no means the whole story. The bias frequency is in the supersonic range and is usually somewhere between 30 kHz and 100 kHz, or possibly a bit higher. As a general rule of thumb, cassette recorder manufacturers establish the bias frequency at about five times (or more) than the highest recorded audio frequency to avoid beats between harmonics of the audio signal and the bias.

Frequency response is affected by bias. This means that on fixed bias cassette decks, control of this important factor is out of the hands of the user. Bias is set at the factory by the recorder manufacturers. An examination of existing cassette machines shows there is roughly a 30% plus and minus variation among all cassette recorders in bias level settings for what can be considered "normal" or "zero" bias. These variations in bias, as mentioned earlier, affect a tape's frequency response characteristics.

The ability of a tape to perform properly over a wide range of bias settings is called bias tolerance or bias range. This is one of the factors to look for when buying cassette tapes. Bias range should be as broad as possible. The wider the bias tolerance, the more likely the cassette will perform well in all cassette players, with or without a bias selector switch.

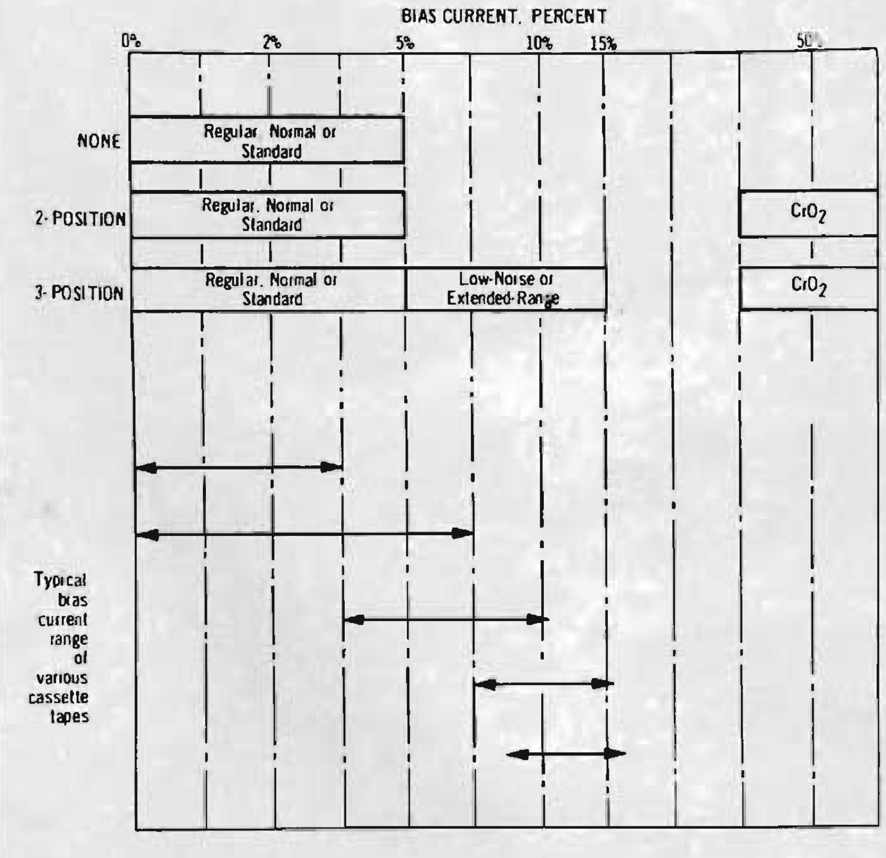

Fig. 11—Bias current ranges of various cassettes

Bias noise is the major contributor to overall tape noise and hiss. It is present on all tapes, even when no signal is present. Obviously, it should be as low as possible; the ideal is zero.

Some cassette equipment manufacturers calibrate their bias oscillator output with a specific tape in mind. In all tape systems, the noise level (hiss) is also a function of the recorder itself. If low-level signals are recorded at higher levels and then played back at much lower levels, there will be a considerable reduction in hiss. If the equipment has Dolby noise reduction circuitry, the noise level can be diminished even more.

Tapes and Bias Current

The amount of bias current (Fig. 11) required by a cassette tape depends on the manufacturing processes used in making the tape. Inexpensive recorders do not have a bias switch and so the user has no way of varying the bias current. Bias in such recorders is sometimes referred to as "normal" or "standard" or "regular." On a scale of 0 to 100, fixed bias recorders operate with a bias current of 5 percent or less. However, the fact that a recorder uses fixed bias does not mean that all "regular" tapes made by all cassette manufacturers will produce the same results. Correct bias will produce linear output, but only if the properly formulated cassette tape is used with it. As an example, TDK's Super Dynamic and Extra Dynamic tapes can be put in fixed bias recorders, but they are also designed to work well with bias currents as high as 10-15 per cent. Recorders with no bias switch can also use TDK's LN or F-series cassettes.

A more flexible type of cassette recorder is one that has a two-position bias switch. One position is for ferric oxide (FeO) tapes and is generally marked standard, normal, or regular. With the switch in this position, the recorder works in the same way as recorders that do not have a bias switch.

With cassette decks that have a two-position bias switch, the second position is generally marked CrO_2, the chemical symbol for chromium dioxide. These tapes, as the name indicates, use particles of chromium dioxide instead of ferric oxide, as the magnetic particles on tape. While chromium dioxide tapes do represent a forward step to high fidelity since they give a better high-frequency response (or as good as ED), the tape as originally manufactured was excessively abrasive and

(Continued on page 105)

(Continued from page 46)
had a pronounced wearing effect on cassette heads. Chromium dioxide tapes also require about 40%–50% additional bias current.

The most sophisticated cassette recorder-players have a three-position switch. In addition to "regular" and "CrO_2" positions, there is a third position for "low noise" or "extended range" cassettes, such as TDK's ED or SD cassettes. While these tapes, as mentioned earlier, can be used with the bias switch set to the "regular" position, on cassette recorder players having a 3-position switch, it is preferable to set the bias switch to its "low noise" position to take advantage of the higher bias current provided.

Equalization

The bias selector switch doesn't just change the bias; it also changes equalization, and this is a whole 'nother story. Equalization is electronic compensation made for the tape's frequency response curve. If the curve droops sharply at the high end, equalization circuitry will boost this drooping section electronically. If it rises or drops too sharply at some other point, equalization will flatten the variation.

Ordinarily, equalization is standard on cassette tape decks. For regular and premium-grade tapes, it's there during both recording and playback. The playback conforms closely to the NAB standard for 7½ ips open-reel tapes and because of this, cassettes recorded on one machine can usually be played back on any other cassette recorder with no significant differences in equalization characteristics.

With chromium dioxide tapes you may have to be a bit more careful. Each manufacturer has his own standards for equalization and bias changeover. While the bias selector switch also changes the equalization, there is no machine-to-machine standard and so the safest procedure is to record and playback on the same machine.

Which Tape To Use

With the proliferation of cassette tapes and adjustable bias controls on cassette recorder players, there is an excellent opportunity for confusion. The choice, however, is quite simple. If the recorder has a CrO_2 bias position, use this only when playing CrO_2 tapes. If the recorder has either fixed bias or a three-position switch, you can use cassette tapes such as TDK's ED or SD with the bias switch positioned to low noise or extended range. It is important to remember to record and playback at the same bias level. Æ

CASSETTE TRANSPORT PROBLEMS

Neal Rayborn*

THERE HAS been significant technological advancement and improvement in cassettes and cassette machines over the past several years. The interface between the cassette and machine, however, is an area that has received very little attention or discussion—yet this interface is critical to the successful functioning of the system.

The secret to successful interface is mechanical alignment, in addition to the usual azimuth-zenith head alignment. The area of mechanical alignment most critical to proper tape handling and guidance, which is essential in reliable usage of longer-length quality cassettes, includes the following components: (1) record/play head guide; (2) erase head guide; (3) capstan, and (4) pinch roller. If one or more of these components are not correctly aligned, they can render a premium quality cassette useless after one pass.

The observations made in this article are based on several years of experience gained in our product test lab. For the tests during this period we purchased, through retail outlets, numerous cassette machines produced by a wide range of manufacturers.

Overall, we found 50 percent of the machines we purchased were improperly aligned. As might be suspected, the percentage varies from the most expensive decks to the more inexpensive portables, but even among the most expensive decks, 25 percent were in need of realignment.

This article is intended to provide cassette users with: (1) factual information on proper alignment of cassette machines, (2) the means to detect and diagnose the various types of alignment problems, and (3) guidelines for what to do once the problem is discovered.

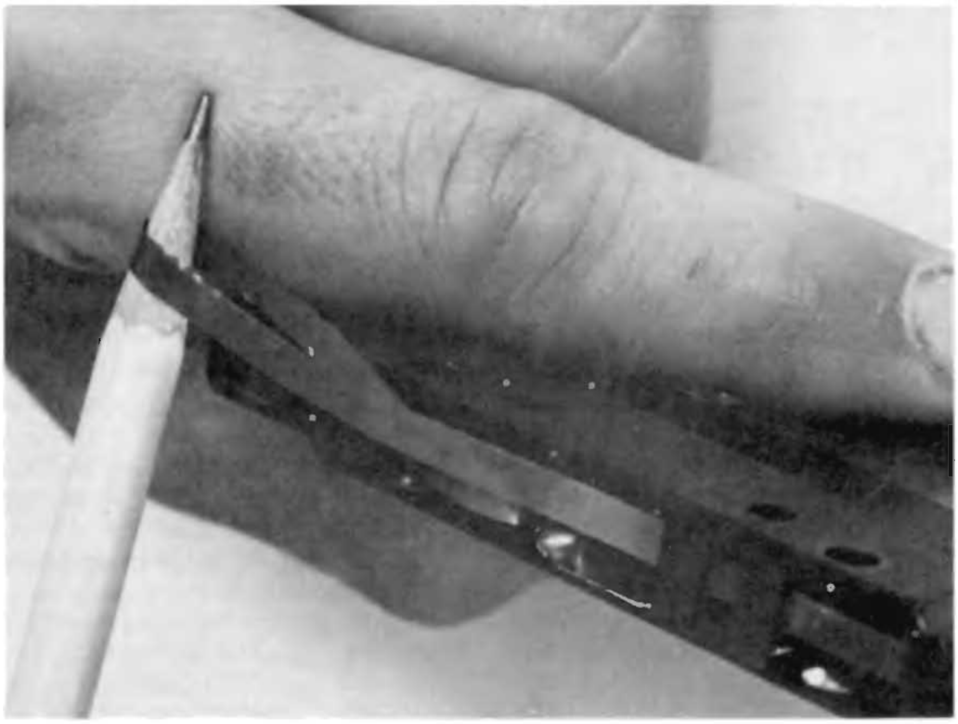

Fig. 1—Tape with a ragged edge is a common indicator of a machine alignment problem. Severe edge damage can result in a seized or jammed cassette.

*Consumer Plant Manager,
Memorex Corporation

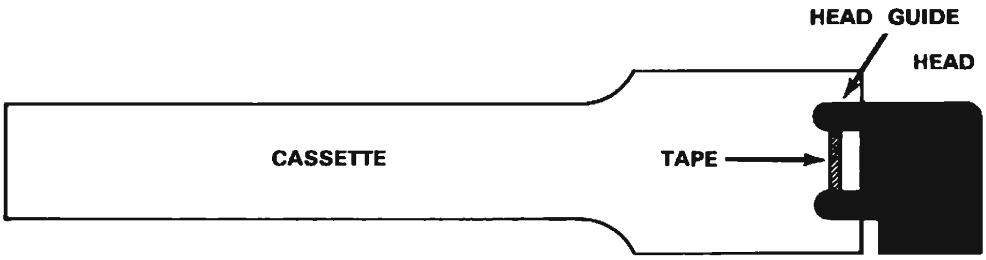

Fig. 2—An accurately aligned head guide will position the tape exactly in the center of the cassette, as shown here.

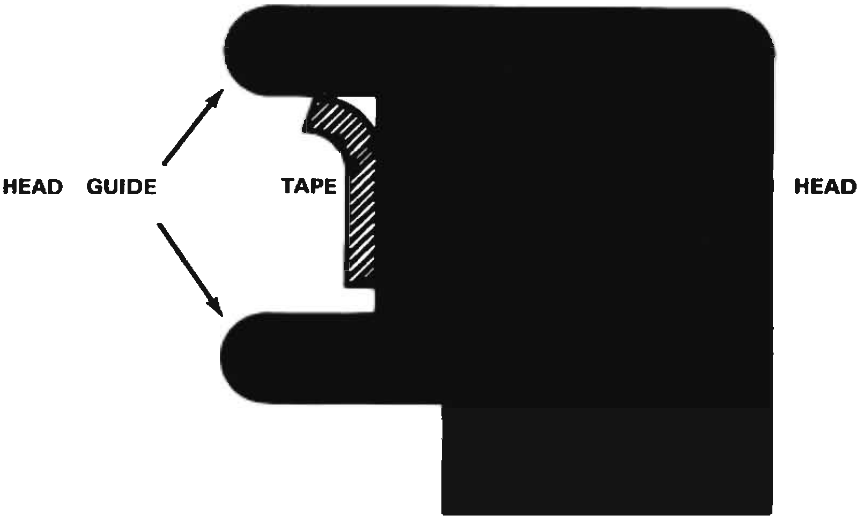

Fig. 3—If the head guide is either too low or too high, it will stretch or bend the tape edge.

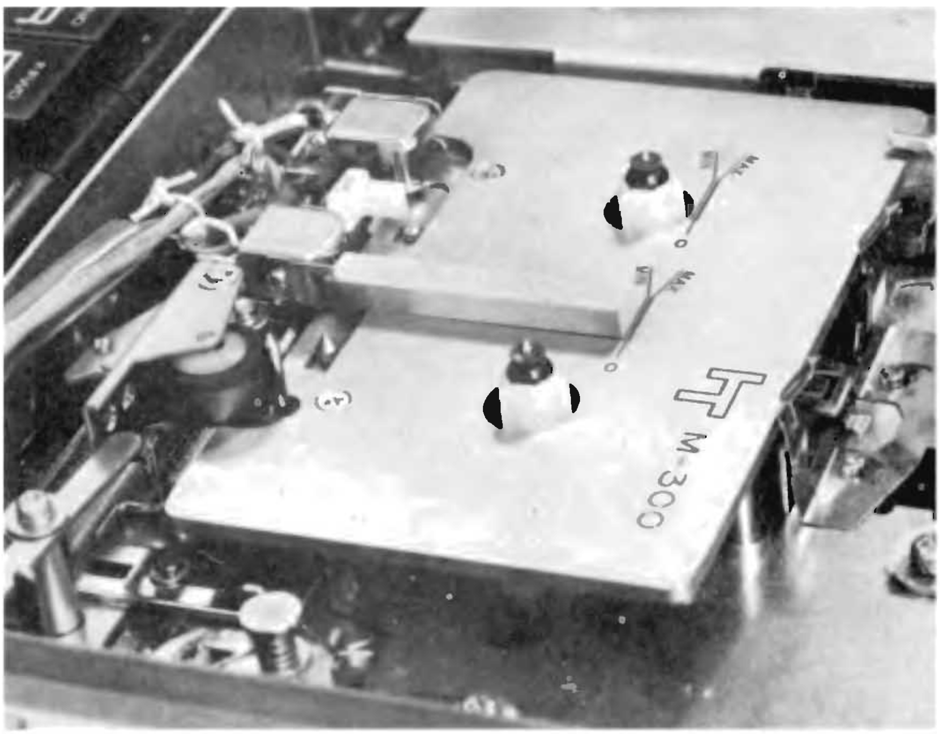

Fig. 4—The metal gauge finger on the mechanical alignment fixture serves two purposes. It checks head guide alignment by sliding between the head guides when correctly aligned. Secondly, it checks head penetration, indicating proper penetration on the lines scribed on the surface of the plate.

Detecting and Diagnosing Alignment Problems

The first and most important step in the diagnosis procedure is to test the machine with a premium quality cassette. A poorly-constructed, bargain-basement variety cassette can produce symptoms similar to those of a misaligned cassette machine, making accurate diagnosis difficult, if not impossible. Just as machine alignment and tape path are critical, so is the tape path and guidance system within the cassette. It is the thinness of cassette tape that makes it very susceptible to edge damage, and therefore necessitates a precisely-aligned guidance system.

The major indicator of a machine alignment problem is tape edge damage. This may initially appear as a signal variation in one channel or can be visually observed as an uneven or ruffled edge on the tape. (See Fig. 1.) Edge damage, if severe enough, can result in a seized or jammed cassette, often characterized by tape spilling out of the cassette and being wound around the capstan or pinch roller. This type of total failure is most common with the 90- and 120-minute cassettes, since they have thinner tape than the 30-, 45-, and 60-minute cassettes.

It seems popular among some cassette machine manufacturers to condemn the longer lengths (C-90, C-120) as unreliable and to discourage their use. This is true if 3-for-$1 variety cassettes are used and/or machine alignment problems exist. There is, however, no reason to avoid using the longer lengths if they are of high quality construction, and if the machine is well aligned.

Another type of misalignment, which may or may not produce the usual edge damage, (i.e., stretch edge), is capstan-pinch roller misalignment. A severe misalignment of these two vital components will usually drive the tape off one side of the pinch roller, often creasing or folding the tape in the process. This usually interrupts the recording to a very noticeable degree.

Head Guide Alignment

An accurately aligned head guide will position the tape exactly in the center of the cassette, as illustrated in Fig. 2. If the guide is too low or too high, it will stretch or bend the tape edge (See Fig. 3). The easiest way to check the alignment of the head guides is with a mechanical alignment fixture. Information Terminal Corp., 323 The Soqual Way, Sunnyvale, Calif. 94080, sells such a fixture (M-300) for approximately $85.00. The fixture is simply a metal plate that is positioned in the machine where a cassette is normally placed. The top of the plate should line up with the lower guide. There is a metal gauge finger which serves two purposes. First, it checks head guide alignment. If the head guide is not correctly aligned, the gauge will not slide between the head guides (See Fig. 4). Second, it is a check on head penetration. There are lines scribed on the surface of the reference plate which, when used in conjunction with the gauge finger, will detect excessive or insufficient head penetration.

If the head guides are incorrectly positioned, they must be repositioned by adjusting the head either up or down. However, the azimuth of the head must be maintained; azimuth being the perpendicularity of the head gap to the tape edge. An alignment tape must be used to check the azimuth. This is a reiterative process, requiring that one go back and forth between the mechanical alignment gauge and the alignment tape.

The process of aligning the head should only be attempted by those with proper equipment and experience since it is a delicate process.

Capstan—Pinch Roller Alignment

The ideal machine has the axis of the capstan and pinch roller perpendicular to the plane of the cassette. Deviations from true perpendicularity will result in the tape being driven either up or down by the action of the capstan pinch roller (See Fig. 5). Typical tape damage resulting from such misalignment is illustrated in Fig. 6.

This type of misalignment is very difficult to diagnose, since it is often erratic and inconsistent. It depends upon many variables, such as tape thickness, pack size, and take-up tension of the machine. The best way to diagnose this problem is to watch the tape pass between the capstan and pinch roller. The tape should track in an even line and should not oscillate up and down or track with part of the tape above or below the pinch roller. Improper tracking is often most pronounced just after starting to play a cassette in the middle. With some machine designs it will be difficult or impossible to observe the tape passing between the capstan and pinch roller. The only alternative is to repeatedly start and stop a cassette in the center of the pack, then, using a pencil placed through the hub, slowly rewind the tape while watching for damaged sections, such as that shown in Fig. 6.

If your machine has evidence of capstan/pinch roller misalignment, correcting the problem can be a major job. The easiest problem to correct is a worn pinch roller or bent pinch roller assembly. The roller and supporting bracket are often serviced as an assembly, and replacement of this assembly may correct the alignment problem. If it does not, the misalignment may be caused by a worn capstan bearing, a bent capstan shaft, or misalignment of the assembly as it came from the manufacturer. Correction of this more serious problem may require shimming the support plates for the capstan or pinch roller, bearing replacement, etc.—all of which should be left to those with experience and skill in this area.

Selecting a Competent Service Technician

Most tape machine repair facilities do not possess an alignment gauge nor the knowledge to use one. Certainly before submitting your machine to a repair service for realignment, you should determine the competency of the person who will be doing the work. If the service man cannot satisfactorily answer questions on head guide and capstan/pinch roller alignment and the resulting effects on tape damage, don't hand over your machine. You may have it returned in worse condition than before "servicing."

It is possible to check the alignment by purchasing an alignment gauge, as previously noted. Although somewhat expensive, the gauge may pay for itself by reducing tape damage or by saving an irreplaceable recording.

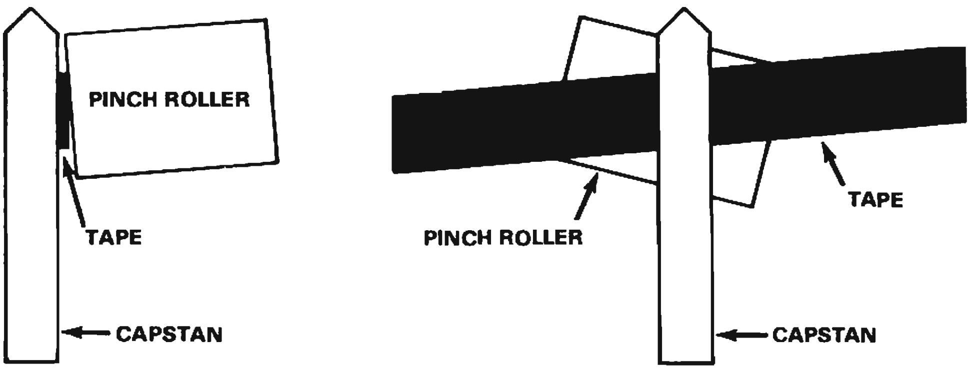

Fig. 5—Both the capstan and pinch roller should be perfectly perpendicular to the plane of the machine. Deviations will result in the tape being driven up or down, as shown here.

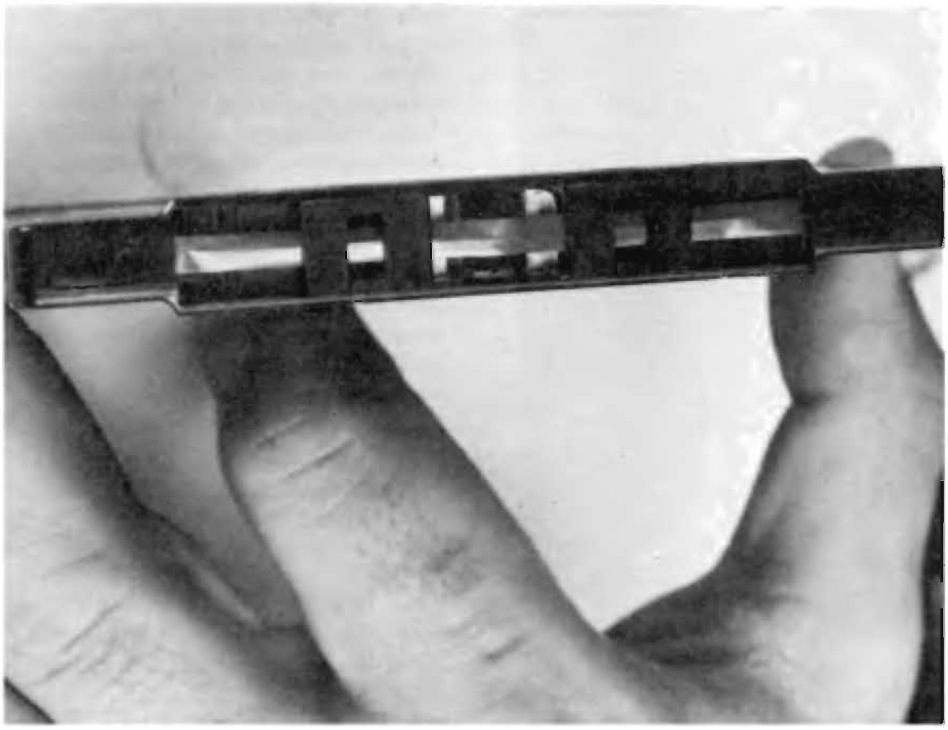

Fig. 6—A crease along the center of the tape is an indication of tape damage resulting from misalignment of capstan and pinch roller.

A Primer On Choosing Tape

William A. Manly*

WHICH TAPE DO I BUY? is an oft-heard question which has never been fully answered. This is attested to by the number of pamphlets, booklets, and articles aimed at an answer. The reef which most of the unwary ground upon is that the selection of a tape rightly starts with the machine! Many people asking this question have made the wrong first choice, and so they will never be completely satisfied with the second. An important point to remember is that no one else can make your choice for you if you are a discriminating user.

Before we get to the choice-making process, a little painless (we hope!) education is needed to define some terms, and to describe what we will call the "Audio Box" approach to understanding tape performance specifications.

Audio from Tape—A Noisy Signal

The first term is "noise." The dictionary defines this as "any disagreeable sound." This is true, but we want to get a little more specific than that, since noise is the floor of our box. We will separate noise into two components, called "frequency" and "amplitude." The noise in any system can be represented by a single line on a graph where vertical directions (in mathematical terms the "ordinate") represent the amplitude, and horizontal directions (the "abscissa") the frequency. This type of graph is called a "Bode Plot" for the man who invented it. Units on the abscissa are in Hertz, or cycles per second, and on the ordinate, usually in decibels (abbreviated "dB"). The Hertz (abbreviated Hz) is easy to understand as this represents the pitch of any audible note. The dB is a rather slippery character, as it represents the logarithm of a ratio, and is, therefore, a relative quantity—but relative to what? There's the kicker. The answer is that it is relative (or "referenced") to anything you want it to be relative to, which is what makes it such a stinker. We'll make it a bit easier by generally referring other levels to the "Standard Output Level" defined by several organizations, which defines this level as "Zero dB." Level set and frequency response standard tapes are available with this level recorded upon them.

Noise is generated by the tape, by the electronics of the tape machine, and in the heads. If the machine is well designed, tape noise will always have a greater amplitude than head and electronics noise. For all but first quality machines the reverse is true, making low-noise tapes a waste of money, so here is the first maxim: Be certain that your machine *deserves* low noise tape before you buy it. Look at Fig. 1. This is a picture of noise, plotted amplitude vs. time rather than frequency. Figure 2 shows two sine (single frequency) waves of different frequencies. Now look back at Fig. 1. The noise is seen to be the sum of a large number of sine waves of differing frequency. These can be separated from the noise one at a time and their amplitudes plotted against their frequency. For a

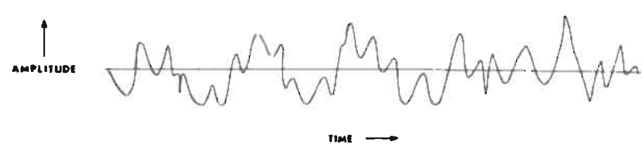

Fig. 1—A noise waveform, frequency range about an octave.

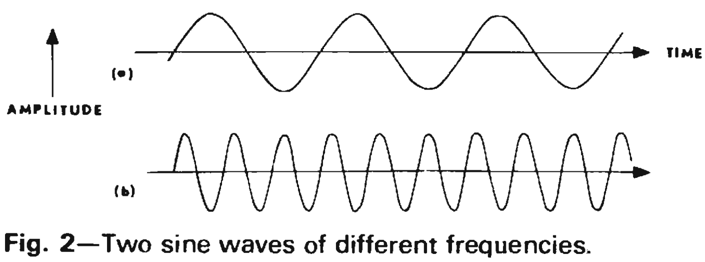

Fig. 2—Two sine waves of different frequencies.

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typical tape-limited audio system, the result is like Fig. 3. The frequencies are plotted logarithmically, meaning that the distance from 20 Hz to 200 Hz is the same as the distance between 2000 Hz and 20,000 Hz. This line is the lower limit to sounds we can hear, since once the signal amplitude drops to that of the noise, the signal tends to get masked by the noise. Thus we have established the floor of our "Audio Box."

Particles and Paint

Let us digress a bit to find out how the tape noise comes about, so that you can understand just what a "Low Noise" tape is. Almost all tapes are composed of a very high quality paint (similar to a lacquer), pigmented (colored) with tiny acicular (needlelike) particles of magnetic material, and coated on a base material of plastic film. The most common types of plastic for the base are cellulose acetate (or just acetate) and polyester terephthalate (or polyester for short). At

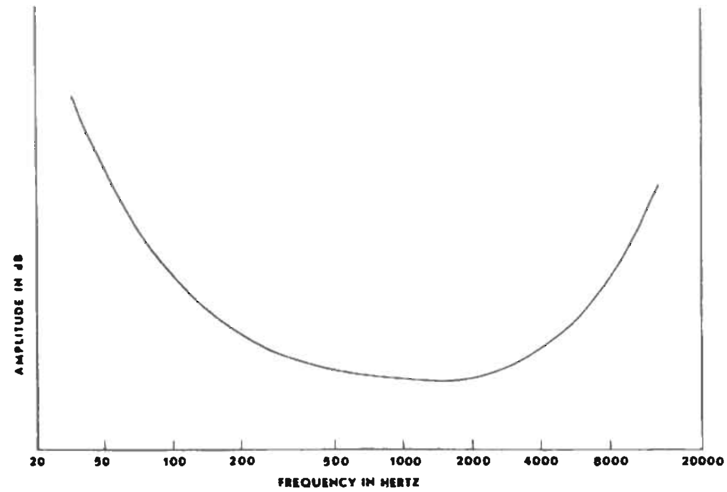

Fig. 3—Bode plot of noise amplitudes vs. frequency for a typical "tape limited" audio tape system.

present, most tapes are made with polyester base. The needlelike particles have a length ranging from about 8 microinches (200 nanometers) to about 50 microinches (1¼ micrometers—pronounced micro-meters). These particles are so small that they are single-domain, i.e., always permanently magnetized with the north-seeking pole at one end and the south-seeking pole at the other. Application of an external magnetic field can cause the poles to swap ends, but can never demagnetize them. These particles are *oriented* in the tape, meaning that they are arranged so that they point in the direction of tape movement across the head. When the *tape* is demagnetized, half of the particles have their N-pole in one direction, and half in the other. When all of the particles have their N-poles pointing in the same direction, this is called *saturation*, and is the strongest signal that is possible on this tape. Other signals, of varying strength, are somewhere in between these two extremes.

The Picture-Sound Analog

At any one time, the magnetic field going through the playback head comes only from those particles near the head gap. In a standard ¼-inch tape, using a full-track head, the number of particles involved is about 10 million. In a 4-track (stereo) cassette, the number is about ½ million This change by a factor of 20 is the primary reason that a good signal-to-noise ratio is more difficult to get with cassettes. In the same way that a newspaper picture is built up with individual dots, or a television picture with individual lines, the audio signal from a tape is built up with the individual fields from each of these particles. Even with the tape completely demagnetized, there is some field fluctuation which goes into the playback head, and we call this tape noise. If the particles are smaller, we can increase these numbers given above, and we call this a "Low Noise" tape. If we cut the track width, as in going from full-track ¼ in. to quarter-track ¼ in., we cut the number of particles and thus decrease the signal-to-noise ratio. If the speed of the tape decreases, the number of particles "seen" per second decreases, and this decreases the signal-to-noise

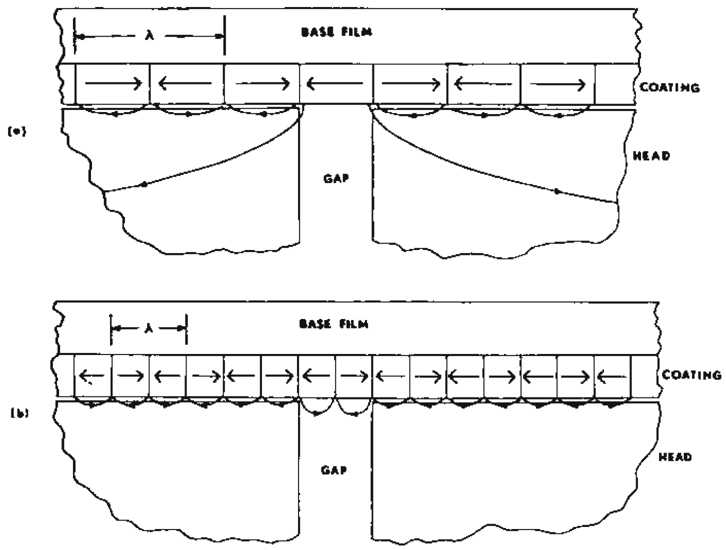

Fig. 4—(A) Simplified picture of a recorded signal passing a playback head. Wavelength is twice the playback gap length. (B), Same, but with wavelength equal to the playback gap length.

ratio. So—low speeds, narrow tracks, and large particles give a poorer signal-to-noise situation.

The sides of the "Audio Box" are shown in Fig. 3. They are simply the lower and upper frequency limits to the sounds we can hear. Their exact location depends on the age and sex of the listener and a number of other individual characteristics, but the outside limits are generally given as 20 Hz and 20,000 Hz.

The Top of the Box

The top of our "Audio Box" is defined by wavelength-sensitive magnetic effects in both the recording and playback processes. A simplified picture of the recorded signal can be imagined as a series of bar magnets passing by the playback head gap (see Fig. 4-a). The wavelength of the recorded signal, denoted by the Greek letter Lambda (λ), is comprised of two of the bar magnets end-to-end. The North-seeking poles are at the arrow heads. The magnetic flux coming out of each of the magnets is shorted through the head surface and returned to the magnets, except for one magnet over the gap, which is sending its flux through the head, and is thus being played back. Now note Fig. 4-b. If the wavelength is no longer than the playback head gap length, no signal is sent through the head, and the signal, though recorded, is not played back. This sets a short wavelength limit on the recorded signal, which amounts to an upper frequency at a given tape speed. Playback head gap lengths are from 100 millionths of an inch (2.5 micrometers) long on higher speed machines down to about 40 millionths of an inch (1 micrometer) long on cassette machines.

Now look at the recording process, where we create those little bar magnets in the tape coating. The first thing to remember is that except for machines which use a combination record/playback head, the record gap lengths are much longer—500 millionths of an inch (12.5 micrometers) to 1 thousandth of an inch (25 micrometers).

The audio signal is recorded by adding it to a large supersonic signal called bias and sending both to the record head. The bias acts in an analogous way to the power steering on your automobile. Bias does all the heavy work in magnetizing the coating, while the audio signal does the steering to control the amount and direction of magnetization. Like power steering, bias must be in proper adjustment, and we will see why it must be adjusted to match the tape in use.

In order to magnetize a piece of ferromagnetic material (like a tape coating), one must expose it to a field greater than a threshold field called the coercivity. Coercivity is sometimes sloppily called coercive force. The bias, which is about 10 times larger than the maximum signal, is used to raise the peak field produced by the head to the point where the smaller signal will record on the tape. The bias is necessary, since in its absence only the very largest signals would be recorded.

Returning to Fig. 4, the bar magnets are pictured as being rather elongated, which is the best shape for bar magnets. All bar magnets try to demagnetize themselves, and this effect gets worse the shorter and fatter the magnet. Here is where we lose a lot of short wavelength (high frequency) signal. As we shall see, making elongated, slim bar magnets is more difficult the shorter the wavelength, and it finally becomes impossible.

Now look at Fig. 5. All four parts show a record head in contact with a tape moving in the direction of the large arrow. The cucumber shape at the trailing edge of the gap is the zone where the bias field peaks go from values higher than the coercivity to values lower than the coercivity. This shape is called the "recording zone" and it changes drastically with the bias amplitude, as Figs. 5-a through 5-d show. The amount of bias is nearly always (on audio machines) adjusted for either maximum sensitivity (sensitivity is playback level divided by record level) or lowest distortion, or some combination of the two. When the coating thickness is about half the record gap length, both these conditions occur nearly simultaneously, as in Fig. 5-c. This is sometimes called "optimum bias," but this is a poor term. The best short wavelength bar magnets are formed under conditions as in Fig. 5-a, where the bias is small or non-existent. The maximum output

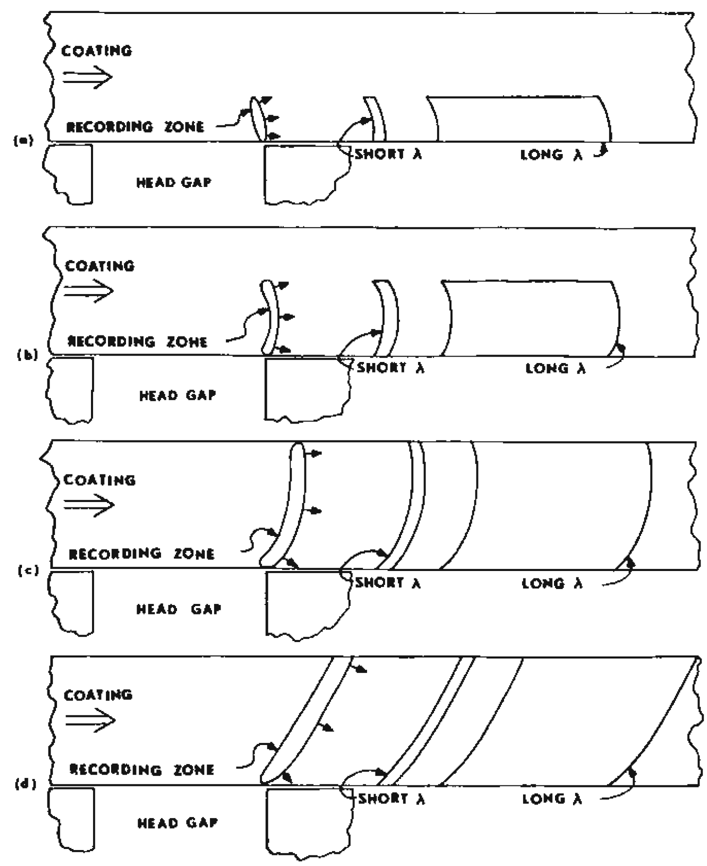

Fig. 5—(A) Tape passing the record head, with recording in progress. Bias very small. Note the shape of the recorded signals as a function of wavelength. (B) Same, but with a moderate underbias condition. (C) Same, but with bias at maximum sensitivity or lower distortion setting (there is very little difference between these two with this ratio of coating thickness-to-record gap length). (D) Same, but with tape in overbias condition.

at medium wavelengths comes in condition Fig. 5-c when all the particles are being magnetized in the same direction.

Now, this part is a little involved, so pay close attention. Since the bias is used to get over the coercivity threshold, the shape of the recording zone depends not only on the amount of bias, but upon the coercivity of the tape. High coercivity tapes are normally called "High Energy," but this term is not necessarily accurate when used this way. If the bias is adjusted per Fig. 5-c with a normal tape, and a high coercivity tape is placed on the machine without re-adjustment, the recording zone will look like Fig. 5-b or 5-a. The result is higher distortion, lower sensitivity, and an excessive output at short wavelengths (high frequencies). Thus, high energy tapes can improve the frequency response, but a price is paid. By re-adjusting the bias (sometimes done with a front-panel switch), the recording zone goes back to Fig. 5-c, giving a smaller high-frequency gain (or sometimes none at all), but reducing the distortion and increasing the sensitivity. Some high-energy tapes have the best of both situations by using a high-coercivity layer next to the head and a standard coating next to the base film. Such a tape is compatible with standard tapes, but has an increased short wavelength response because the external layer next to the head is biased similarly to Fig. 5-a, while the thicker internal layer is biased similarly to Fig. 5-c. This is at a single bias setting, and is a consequence only of the difference in coercivity between the two layers.

A few more items about the playback process, and we are ready to put all this together to form the top of our box. Figure 6 shows a similar situation to Fig. 4, but the tape and the head are not in intimate contact. The loss of signal is very pronounced in this case. A separation of 1 wavelength causes a loss of over 50 dB. This is about equal to the total dynamic range of your recording system! In the case of cassettes, where 15 kHz is equal to a wavelength of 50 millionths of an inch, a tiny dust particle on the tape can momentarily lose the whole high end. 50 millionths of an inch is about equal to two wavelengths of visible light. It is a distance so tiny as to be

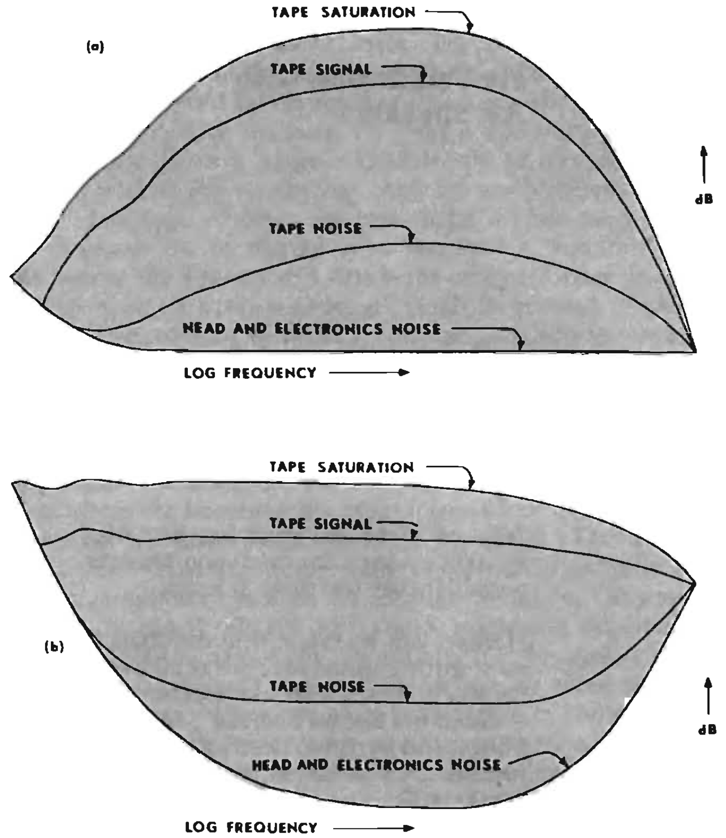

Fig. 7—(A) The complete "Audio Box" in an unequalized condition. **(B)** Same, but equalized.

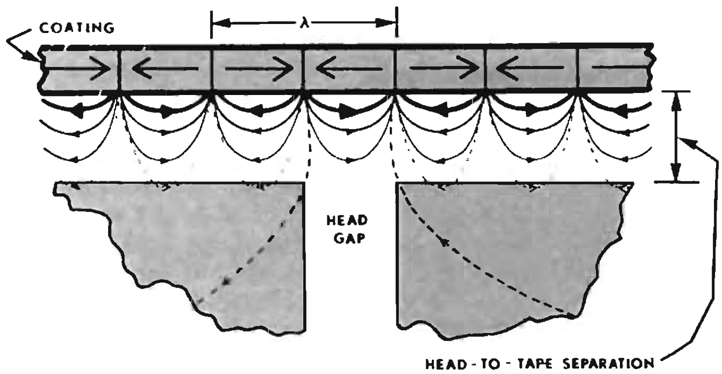

Fig. 6—Tape passing the playback head, illustrating extreme losses due to head-to-tape separation.

visible in the best of optical microscopes only with great difficulty. What this means as far as the tape is concerned, is that the surface of a slow-speed tape should be very smooth, even shiny. If surface roughness is visible, it will not have a good "high end." Also, a tape which sheds oxide that collects on the head will have large intermittent losses of the high frequency signals.

Playback heads have an inherent loss of signal at the low-frequency end because they are sensitive to the rate-of-change of the signal (which is less the lower the frequency). At extremely long wavelengths, there is an additional loss due to the fact that only a small fraction of the wavelength is in contact with the head at any one time, and much of the flux escapes to the air. There are also some interference effects at long wavelengths which cause a wavy frequency response (these waves are called "head bumps").

You can see by the last two paragraphs that the output is eventually going to fall off to zero at both the long wavelength (low frequency) end and the short wavelength (high frequency) end. The noise output falls off as well, but all outputs eventually get so low that the electronics and head noise predominate. The frequency response can be corrected by proper electronic equalization, but the things we have talked about control the dB range between noise and maximum output at any frequency.

At midrange, the maximum output is controlled by the saturation of the material, explained previously. The long and short wavelength phenomena discussed above are further limitations, causing severe roll-off in the frequency response at both ends of the spectrum. These roll-offs are then corrected by electronic equalization, but such equalization brings up the noise as well, and this finally causes the signal-to-noise ratio to go to zero dB at some high and low frequencies. Figure 7 shows the completed audio box; 7-a is the situation before equalization and 7-b after equalization. Note that the signal-to-noise ratios are unchanged by equalization. The equalization only serves to flatten the frequency response. Note that the dynamic range (distance from noise to saturation) is severely limited at the high frequency end and less severely at the low frequency end. This high frequency limitation gets worse with slower speeds. It deteriorates over the whole spectrum when the track is narrower. A thinner tape coating will sacrifice some of the low and middle frequency dynamic range to get an increase at the high end. Note that in all these, the tape is not generally used all the way to saturation in the low and middle frequency range because of a large increase in distortion. There is usually little distortion

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for a saturation level high frequency signal, but its presence may cause distortion in any lower frequency signals present.

Happiness Is a Box That Fits!

Now that you have your box with its four adjustable sides, note there is only one principle for its use: The box should be large enough to contain all the audio that you want to record. Except for the more restricted uses, this is an ideal that is never quite reached, but the closer you get, the better the sound will be. A corollary to this principle is that buying more box than you need is not only a gross waste of money, but can actually give poorer results.

Some of the uses and box sizes are listed in Table I. Note that the low and high ranges, when multiplied, give a number close to 400,000. This is required for a balanced sound. The dynamic range required can be achieved either by adjusting the bottom or the top of the Audio Box, or both. Now note that the sizes of the Audio Boxes in Table I fall neatly into three categories: Restricted (first two), Medium (next three), and High Fidelity (last two). Now these are all rectangular boxes, so that odd-shaped thing in Fig. 7-b is going to have to be larger than these in order not to lop the corners off when the boxes in Table I are fitted in. This is where the advertisers can fool you, if they rate their systems so highly that all the corners are gone from the boxes.

TABLE I - Audio Uses and Box Sizes

Use	*Low Freq. Range, Hz*	*High Freq. Range, Hz*	*Dynamic Range, dB*
Voice: Tape letters, other controlled uses	200	2,000	20
Voice: Speeches & plays	200	2,000	30
Voice: Speech analysis & language study	80	5,000	30
Music: Background	70	6,000	25
Music: Popular & most Rock	60	7,000	30
Music: Classical & Specialty	25	15,000	50
Sound effects	25	15,000	50

TABLE II - Systems rated for Audio Box Size

Rating Scheme		
	HF	High Fidelity
	M	Medium
	R	Restricted
	+	Somewhat better
	-	Somewhat worse

Note: R+ and M- are the same; M+ and HF- are the same.

System	Rating
Battery-powered cassette (lowest cost)	R- to R
Battery-powered cassette (medium cost)	R to R+
Cassette stereo (depends on cost)	M- to HF
8-track cartridge	M- to M+
Open reel, 1⅞ - 3¾ ips (low cost)	R
Open reel @ 1⅞ ips	R
Open reel @ 3¾ ips	M
Open reel @ 7½ ips	HF- to HF
Open reel @ 15 ips (mastering)	HF+

Table II shows the approximate rating of most of the system types in use. Price in each category controls the exact placement. If the machine is a portable, subtract half a rating unit to a whole rating unit (this does not apply to the first two which are all portable).

Tape Selection

It is possible, with proper tape selection, to raise and lower the ratings in Table II by about half a unit. Some selections may only adjust the frequency range, while others may adjust mainly the dynamic range, but some will do both.

There are only so many things which can be done to adjust a tape for different characteristics. We will start off with a "standard" ¼-in. tape that has these characteristics:

Base film thickness—1 mil (25 micrometers)
Magnetic Coating thickness—0.4 mil (10 micrometers)
Magnetic material—gamma ferric oxide
Particle length—32 microinches (0.8 micrometers)
Saturation remanence—1000 gauss
Coercive Force—280 oersteds

Of these chracteristics, we have not mentioned the saturation remanence. This is a measure of the magnetic strength of the particles, their concentration, and how closely they approach being perfectly oriented. This measurement is proportional to the medium wavelength saturation output divided by the coating thickness. The coating and base thicknesses for a standard cassette tape are about half those given above.

There is one last tape problem we have not yet discussed. When the tape containing a signal is wound up in a reel or cassette, some of the signal will gradually record onto the adjacent layers. This is called "layer-to-layer signal transfer," or "print-through." It is made worse by time, high temperatures, high winding tension, thinner base films, and high head-to-tape speeds. Only the smallest of the particles are involved, so the "low-noise" (small particle) materials tend to have poorer print-through characteristics, but not necessarily. Obtaining a "low-noise, low-print" material is one of the triumphs of the magnetic material maker's art.

Base Film Material The advantates of acetate are low cost and the fact that it breaks cleanly and does not stretch. 1 mil thickness is the thinnest acetate available. It is getting difficult to buy tapes on acetate as most tapes are now coated on polyester, either tensilized or non-tensilized. The tensilized polyester stretches less than the non-tensilized. Other plastic films such as PVC are intermediate between acetate and polyester. The very thinnest bases (1/2 mil and less) must be on tensilized polyester.

Base Film Thickness The thicker bases are stronger and easier to handle and have less print-through, but less can be wound on a reel or cassette. As a rough rule of thumb, dividing the tape speed by two will allow a base film of half the thickness to be used, while maintaining about the same *audible* print-through. Remember that the longer lengths of tape are normally achieved with a thinner base material.

Magnetic Coating Thickness This has a complex effect on all the tape properties, and there is an interaction with the magnetic properties of the coating and the gap length of the record head. Generally speaking though, with everything else the same, a thinner coating buys an extended short wavelength (high frequency) response and dynamic range at the expense of higher distortion and reduced medium and long wavelength dynamic range. Cassette tapes have thinner coatings, and so may "long-play" types of tapes.

Magnetic Material As opposed to plain gamma ferric oxide, from which most tapes are made, other materials may give higher coercive force, or high specific output (a stronger magnet from a given size particle), or a smaller particle, or some combination of the three. Higher coercivity will give

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more short wavelength dynamic range, at the expense of higher bias requirements. Higher specific output gives an increased output of both noise and signal, independent of wavelength, which results in an increase in the dynamic range at both extremes of wavelength. A smaller particle gives a lower noise characteristic, which increases the midband dynamic range. Of the three main alternatives to plain gamma ferric oxide, cobalt doping (generally used in "high energy" tapes) gives a higher coercivity only; chromium dioxide gives a controllably higher coercivity and a higher specific output; and metal particles give all three in a controllable fashion. There are also small particle gamma ferric oxides with a higher coercivity. Lastly, an effect similar to a higher specific output can be obtained by putting "more pigment in the paint," called a "higher loading" by the paint chemists. This has the disadvantage of lowering the coercivity of the coating. The "high output" tapes normally use this method, plus a thicker coating.

Sorting Out the Sales Blurbs

Trying to decide which of these approaches (or combination of approaches) has been used in a particular case may be confusing even to the experts. Many times an analysis requires the availability of a complete test laboratory. Nevertheless, we will try to list some commonly used marketing names with the various improvements used.

Base Film material Usually named, but may use trade names.

Base film thickness The thinner tapes use such names as "long play" or "LP." The more tape on a reel or cassette, the thinner the base (usually).

Magnetic coating thickness A "long play" tape may have a thinner coating, as do "slow speed" and cassette tapes.

Magnetic material "Chromium dioxide" is self-explanatory. DuPont's trade name is "Crolyn," and several other names are used. "High energy" or "HE" usually means cobalt-doped iron oxide. It's a bit too early to know all the names which will be used for metal particle tapes—if they become available.

Other terms "Super Dynamic," "Extra Dynamic," "Ultra Dynamic," "Extended Range," "Ultra-High Fidelity (UHF)" and the like, usually signify a compromise of a smaller, high coercivity particle; a somewhat thinner coating; and a very smooth surface. The noise is usually lower and the short wavelength response extended. With proper choice of material, higher distortion can be avoided. "Professional" usually has about the same properties as "General Purpose," but the former are closer to being alike from reel to reel. These are standard tapes with no enhanced qualities. "Low Noise" is self-descriptive, but such tapes sometimes have extended frequency response and higher distortion. "High Output" is also self-descriptive, but usually has degraded frequency response and low distortion. There are a few "Low Noise, High Output" tapes, which are nearly standard but with increased dynamic range (higher top and lower bottom on the box). "Low Print" tapes have a special magnetic material with very few extra-fine particles. It is standard except for the low print-through characteristics.

Machine Format Selection

Playback Only For this application, you have little or no choice of the tape. Generally speaking, you should check to see what is available in each of the formats: open reel, cartridge, or cassette. Open reel still offers the best fidelity, then cassette, then cartridge (cartridge has more capability, but it is usually not exploited). Cassette units are more portable, and cartridge units are best for one-hand operation, needed in automobiles.

Recording Use If you will be doing any editing, then open reel is for you. The higher the speed, the easier the editing. Cassettes are very difficult to edit, and cartridges all but impossible. High fidelity also indicates open reels and high speeds. It takes a very good recordist to get anything but an amateurish sounding recording on a cassette, and very few cartridge machines record at all. Sound-on-sound and other musical effects also usually call for open-reel.

For recording in the field, the portability needs should be carefully balanced against the need for fidelity. Cassette units are very portable, and nothing else should be considered for voice recording. Some cassette units have automatic volume control, and are especially useful for recording conferences. Musical events are best recorded on reel-to-reel units, unless the portability of a cassette unit is an overwhelming consideration.

Fitting the Tape to the Machine

Working basically from Table II, we can now use our knowledge of tapes to enhance or degrade the quality ratings given. Why should we want to degrade? One word—cost. If you are merely sending taped letters, it's ridiculous to use a high quality tape. An inexpensive one will do as well. There is one trouble with "white box" or off-brand tapes though—you never know what you're getting. If you find one that is shedding badly or otherwise coming apart, hanging up in the machine, squealing, etc., it's best to discard it.

To employ the machine at its rated use according to Table II, utilize one of the "General Purpose" or "Super Dynamic" tapes. More consistent results will be had from a "Professional" or "Mastering" type tape. "Low Noise," "Low Noise, High Output," "Extended Range," "Extra Dynamic" or "Slow Speed" tapes can be used on top line machines with an "M" or "HF" rating to extend their rating about half a step. It's a waste of money to use these tapes on the lower cost machines.

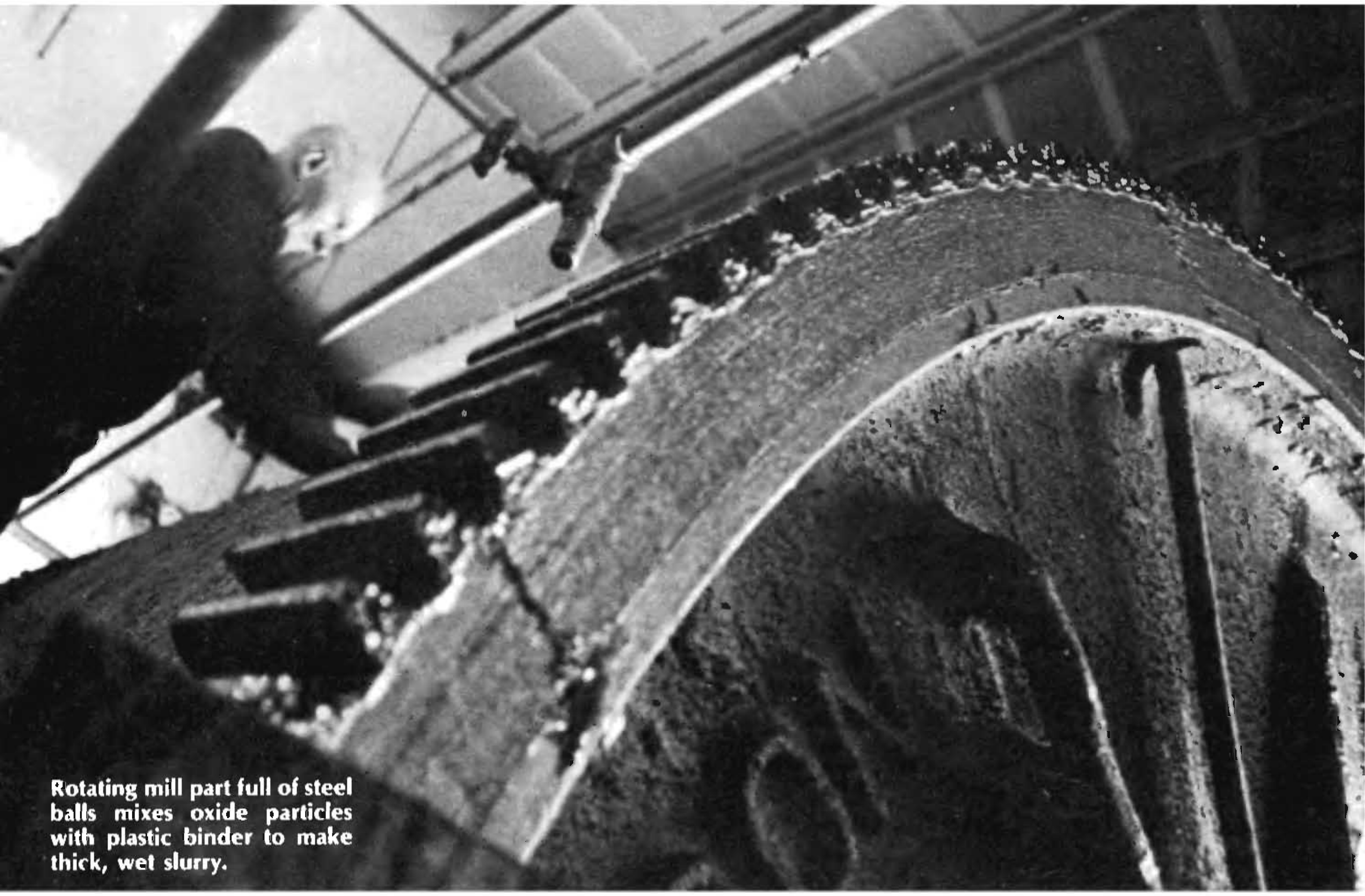
If you need maximum playing time, go to a thinner base or "Long Play" tape. If the recording must be stored a length of times, a "Low Print" tape should be used. If you cut the speed to get longer playing time, use a "High Energy," "Ultra Dynamic," "Ultra High Fidelity," "Slow Speed," "Chromium Dioxide," or other "Extended Range" tape to try to maintain fidelity to some extent. Make sure that the tape is compatible with the machine. Some can't handle chromium dioxide. "High Output" tapes are useful on recorders which have run out of gain control or are marginal for recorded volume. "Low Noise, High Output" tapes should be used only on the better machines in this category.

Experimentation Is Necessary

Nothing beats making a trial run of a different type of tape. If you can't hear any audible difference, use the lower cost tape. Always make sure that the tape you use does not squeal or deposit its oxide in globs on the heads and guides. A very small amount of dry shed in powder form is all right, but very much of this is aggravating. Use a brand of cassettes or cartridges which run smoothly and do not hang up. Good luck on your selections and recordings!

Acknowledgement: To William H. Orr, without whose encouragement this article would not have been written.

Credits: This was written when William A. Manly was Senior Physicist at Orrox Corporation. As of April 1, 1974 Mr. Manly is the Director of Product Development for The Cobaloy Company (A Division of Graham Magnetics, Inc.). It was written primarily as a booklet to be distributed by the International Tape Association, Inc., World Tape Center, Tucson International Airport, Tucson, Arizona 85734 (Executive Director, Larry Finley).



Rotating mill part full of steel balls mixes oxide particles with plastic binder to make thick, wet slurry.

Making Tape

Joseph Kempler*

CREDIT FOR the current high standards of tape recording should be shared by the tape recording equipment manufacturers and by the blank tape manufacturers, for while the equipment maker has contributed major improvements in his machines' heads, electronics, and transports, the tape manufacturer has produced higher energy tapes with precision construction through improved manufacturing technique. To help the reader understand the tape makers' rôle, this article offers relatively simple descriptions of most of the processes we use to make premium quality tapes designed to meet the widest variety of consumer and equipment requirements. While today's magnetic tape is produced for a great number of specific needs, this discussion will be confined to the manufacture of audio tapes for the consumer.

Magnetic tape is produced by a continuous manufacturing process in which rather few basic ingredients are transformed into a recording tape with precisely defined performance properties. Each step in the chain-like process is critical, however, and is dependent on the previous step. Nothing can be permitted to go wrong at any point in the process, or the entire batch of tape may be out of specification with little, if any, opportunity for correction. This means that strict controls must be maintained during the entire process, for any testing which is done on the finished product is at best a confirmation that everything went well.

Raw Materials

Four basic raw materials are used in the production of tape: magnetic oxides, binder components, solvents, and the base film.

Magnetic Oxide. The most common and still the most popular magnetic material for tape is iron oxide, used exclusively for all open-reel tapes, 8-track cartridges, and a vast majority of cassettes. Although chromium dioxide enjoyed an upsurge for a while in cassettes, its popularity appears to be on a plateau. One of the reasons is that new iron oxides and novel methods of processing have made possible new iron oxide cassettes which are in many respects indistinguishable from chromium dioxide and yet are less expensive and have other desirable features. Chromium dioxide, of course, really has been instrumental in bringing the art of cassette recording to present day standards.

This article will concentrate then, on the manufacture of single-coating, iron oxide tape.

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When the magnetic oxides first arrive at our manufacturing facility, they are subjected to thorough testing. Using electron microscopes, quality control inspectors check particle size and distribution. Particles vary between five and 25 millionths of an inch in length and have a needle-like shape because the length of each particle is about five to 20 times its diameter. Particle size must be checked because different sizes are used for different products; the smaller the particle, the lower the bias, a situation which contributes to audible hiss.

The inspectors also look for "cleanliness" of particles. "Unclean" particles have arm-like branches and tiny cavities which result in lower tape quality. All material not passing inspection standards is returned to the manufacturer.

The iron oxide is also examined for its magnetic properties. Probably the most important parameter subject to intentionally large variation is the coercive force, which is the magnetizing force necessary to reduce the residual magnetism to zero. Oxide which has a high coercive force will require higher bias and record levels than a lower coercive force material. It will also require a higher erase current for the same amount of erasure. The lower recording efficiency of such a material is compensated for by its greater resistance to demagnetization, which helps strengthen the demagnetization-prone high frequencies. As a result, high coercivity tapes usually have better high-frequency sensitivity and overload properties than low coercivity materials. Coercive force and other magnetic properties, such as remanent and saturation magnetization which determine the material's potential storage capabilities, are measured with various instruments such as vibrating sample magnetometers and hysteresis loop tracers.

While still in their powder form, the magnetic materials are also subjected to a variety of chemical and physical tests to determine ease of dispersion, acidity, impurity level, and similar variables which affect the subsequent manufacturing processes and the tape quality.

Binder. The binder consists of one or more plastic resins which determine the physical properties of the tape coating and, to some extent, recording properties. Its purpose is to bind all the oxide particles together into a srrong but flexible coating and to provide a permanent bond between the coating and the film base.

Depending on the application, a number of additives may be used in the binder to modify certain properties: plasticizers can be used to make the coating more flexible, lubricants to reduce friction and wear, conductive agents to reduce static charges. Wetting agents, stabilizers, and fungicides may also be added.

The binder resins and the additives are quality inspected in the raw material form with infra-red spectrometers which accurately analyze each ingredient for its composition and freedom from impurities.

Solvents. The purpose of the solvents is to completely dissolve the binder resins, producing a liquid dispersion suitable for coating on the polyester film base. The solvent serves as a temporary vehicle to make the dispersion and coating possible. After this is done, the solvent is evaporated entirely out of the coating.

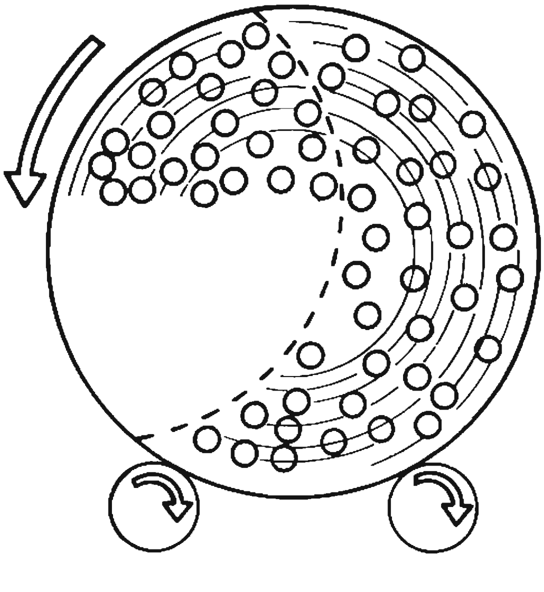
Incoming inspection for solvents includes checks for contaminants, boiling points, and solubility; gas chromatographs and similar devices are used.

Base. Polyester film is the most widely used base for magnetic tapes today because of its superior strength, stable dimensions (even when exposed to extreme climatic conditions), and its resistance to attack by chemicals which destroy other plastics.

Polyester film is available in a number of thicknesses, each designed for a specific type of tape product. For instance, 1 mil and 1.5 mil thick polyester is used exclusively for open-reel tapes. The new standard thickness for 8-track cartridges is 0.75 mil. Cassette tapes are 0.5, 0.3, and 0.25 mil thickness for 60-, 90-, and 120-minute lengths, respectively.

Because of the extreme thinness of film used in cassettes, the polyester for this application is tensilized. This is a special pre-stretching process performed by the film manufacturer which increases the tensile strength nearly two times.

Base film is also subjected to a large assortment of quality control measures before manufacture ever starts. It is tested for strength, thickness, smoothness, cleanliness, freedom from wrinkles and physical stresses.



Milling Media and Oxide-Slurry Preparation

MILLING ACTION DURING MILLING CYCLE

Manufacturing Process

Once all the inspection and quality control tests are completed, the raw materials are cleared for production.

Milling. The first step in the manufacturing process for recording tape is the blending and mixing of all the formulation ingredients. We do this in a ball mill, a large rotating drum partially filled with steel balls. When the drum is set into motion, the balls cascade through the mixture, called slurry, and create the type of agitation and blending action necessary to produce the desired dispersion.

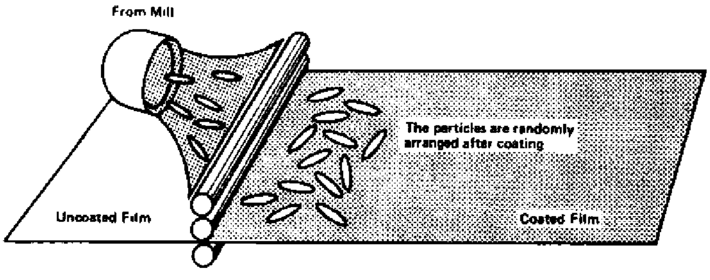
The ultimate purpose of milling is to produce a dispersion where each oxide particle is coated with a thin layer of binder and separated from all other oxide particles. In spite of the massive nature of this operation, it is indeed a delicate and precise process. Insufficient milling, for instance, results in an incomplete dispersion where large groups of iron oxide particles are clustered together, causing undesirable magnetic interactions. These interactions manifest themselves as increased hiss level, lower output, variations in uniformity, dropouts, and even local weak spots in the coating, which can eventually cause shedding, wear, and additional dropouts. Excessive milling can be just as harmful and may show up as a loss of high-frequency response, increase in layer-to-layer signal transfer (print through), and, depending on the formulation, may also cause a weakening of the entire coating.

Tapes have been greatly improved in recent years by progress in milling technology. Current methods of dispersion have resulted in reduced interaction losses and permitted a substantial increase in coating density without weakening either the physical or the electrical properties.

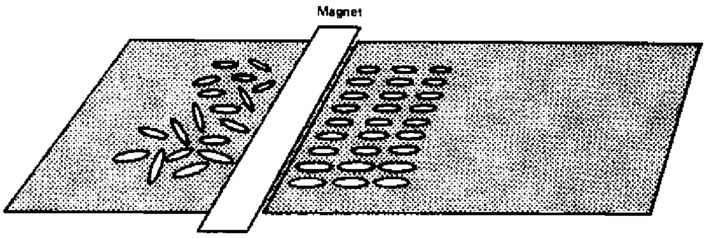
Coating. Once the slurry has reached its optimum dispersion level, it is brought to the coating machine to be joined with base film.

THE COATING OPERATION

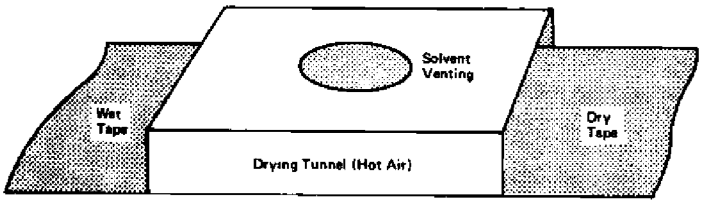
Film Is Coated



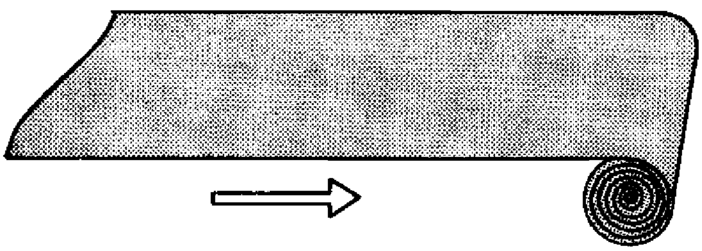
Orientation



Drying Zone



Take Up Zone



The coater is an imposing piece of equipment closely resembling a printing press. At the front end of the machine, large rolls of polyester—10,000 or more feet long and as wide as 60 inches in width—are loaded into position. The film is then threaded through a multiplicity of rollers, guides, and elevators which transport the film through the machine at high speed and low tension.

In the coater, many different processes are performed at the same time. The film will be coated on one side with one or more layers of magnetic coating, and for certain types of tape, a layer of special coating may also be added on the reverse side of the film. The film will also be destaticized and cleaned, have its wrinkles pressed out, and be stabilized. Particles will be oriented, the solvent will be removed from the coating, and the tape will be dried. The tape will then be polished, densified, and wound up evenly and smoothly on the other end of the coater. This amazing series of operations is done on a continuous basis without even slowing down to change the rolls of polyester film.

As in all other parts of tape manufacture, the precision with which the operation is accomplished affects final quality. The thickness of the magnetic coating determines the low- and mid-frequency sensitivity and compression, harmonic distortion, and the optimum operating bias and record level. Thus, the thickness has to be layed down with great accuracy and uniformity and within close tolerances. A typical coating thickness for open-reel tapes is about 400 microinches (millionths of an inch). Cassette tapes range from a maximum of about 250 microinches for 60-minute tapes to less than 140 microinches for 120-minute cassettes.

Coating thickness must be constant and uniform from one end of the roll to the other, across the full width. For proper control of uniformity, it is necessary to monitor the thickness continuously while coating is in progress. This must be done without physical contact with the wet coating or damage will be done to the tape surface. This is accomplished by measuring the absorption of radioactive energy or x-rays in the coating and converting the absorption data into thickness or mass.

In the case of a cassette tape, close tolerances must also be maintained. With a nominal thickness of 200 microinches, a realistic thickness tolerance of ±5 percent requires a thickness control of ±10 millionths of an inch. To visualize how thin this is, look at the typical cellophane shrink-wrap used to package tape products. It is literally one hundred times thicker than the coating tolerance on cassette tape.

As mentioned above, some tapes also have a special nonmagnetic coating on the back of the base film, which must be added during the coating process. Lubricated tapes used in 8-track cartridges, for instance, have a layer of a very slippery, dry lubricant on the back, approximately 50 microinches thick, which allows the use of this tape in continuous loops.

Another special coating applied to the back of the tape is a textured, carbon layer like Capitol's "cushion-aire" backcoating used for The Music Tape cassettes and open-reel tapes. Backcoating provides many valuable functions. In cassettes, it materially reduces the incidence of jamming and other failures. It also helps eliminate wow, frequently caused by slippage of non-backcoated tapes between the capstan and pinch roller in some cassette machines. In open-reel tapes a backcoating provides very smooth winds even at high speeds and eliminates damage which may occur as a result of tight winds, scatter winds, or storage under improper conditions.

Another operation which takes place during coating is orientation, a process designed to orient or arrange all oxide particles parallel to the length of tape. This is the desired direction because it is the same as the longitudinal component of the recording field around the recording head, and thus enhances all the recording properties in the direction of normal tape motion.

In a slurry, the needle-shaped particles are randomly distributed and point in all directions. Many undesirable performance conditions would result if the particles were allowed to stay in this disarranged state: output would drop; noise, particularly d.c. noise, would rise sharply; bias requirements would shift, and high frequency demagnetization would also be adversely affected. Owing to the shape of the particle, clearly defined north and south poles are produced, making it relatively easy to rotate them in one direction during the coating operation by passing the slurry through an external magnetic field.

Immediately after orientation, the coated tape enters drying tunnels where hot air evaporates the solvent from the coating. The drying tunnels are divided into a number of zones with individual means of adjusting the drying state by controlling air temperature, velocity, and volume. A drying rate which is too fast or too hot can cause some of the solvents to evaporate or boil off too rapidly, producing tiny pinholes and craters in the coating, which in turn will cause dropouts or noise pops. On the other hand, if the drying cycle is too slow or incomplete, the tape may be tacky and stick to itself, or even worse, may cause build-up on the recording heads and produce severe losses of output.

Surface Polishing. Before the tape leaves the coater, it is squeezed between two or more very highly polished rollers under great pressure and high temperature. This process uses the super-smooth finish of the calendering rolls to create a nearly glasslike surface on the tape which has an average finish depth of only a few microinches.

Not all audio tapes require the same degree of polish. The finest polish is usually reserved for premium cassette tapes in order to improve output at high frequencies, since for best high-frequency response, the tape must maintain intimate contact with the record and play heads. Even the slightest spacing between the head and the tape will introduce severe losses. For instance, a 10-kHz signal recorded on a casette at 1⅞ ips will suffer a combined record and playback loss of about 6 dB if there is only a ten microinch spacing between the tape and the heads. This type of separation can easily result from an unpolished surface since any roughness, even sub-microscopic in size, causes only the high points on a tape surface to come into contact with the heads. This also illustrates the need for cleaning and maintaining the heads of the recorder.

Slitting. When all operations performed on the coating machines are completed, the tape is wound onto large cores. These enormous rolls are then transported to another section of the plant where they are slit into hundreds of reels of tape in a single operation utilizing rotary blades. Slitting is done at very high speeds but under stringent control so that the film is cut in a perfectly straight line.

Consumer sound recording tape is slit to either of two widths. All open reels and lubricated tapes for 8-track cartridges are slit to a width of 0.246 inches ±2 mils. Cassette tape is slit to 0.149 inches ±1 mil.

Good guiding and trouble-free operation on the recorder is to a large extent dependent on the accuracy of slitting. For instance, width tolerances must be accurately maintained. Clearly, a tape which is too wide will stick in the guides or become damaged at the edges if forced to run under these conditions. Loss of output and dropouts on the edge track will very likely result as well. A tape which is too narrow may mistrack and cause output variations in edge channels and, in some cases, cause crosstalk.

Even if the tape is slit to the right width, but not guided through the slitting knives in a straight line, the tape will be skewed or "snaky." Snakiness of this type will produce large variations in the high frequency signal because the tape is moving past the head with a constantly varying angle with respect to the head gap, changing the azimuth correspondingly.

After the tape is slit, the individual pieces are usually wiped on both sides to clean off any loose debris generated during slitting. It should be pointed out that a slight amount of oxide deposit on the heads or on the pinch roller is not abnormal however. Some dirt accumulation must be expected because no matter how cleanly the edges are slit, they can never be as smooth and as free of debris as the surface. Also, as the tape edges rub against the guides, heads, and reel flanges, a slight edge polishing takes place, causing new debris accumulation, which is then deposited on the heads or pinch rollers. This is one of the many reasons why regular cleaning and maintenance of the recorder is necessary.

On the take-up side of the slitting machine, the slit and cleaned tape is wound onto the various reels. Most 7½-in. open-reel tape for home recording is wound directly onto plastic reels. Tape for 10½-in. reels, however, is normally wound on plastic or aluminum NAB hubs, with the metal flanges attached later.

Lubricated tape for 8-track cartridges is wound on large, 14-in. diameter pancakes, which hold 8,400 or more feet. These pancakes, without flanges, are then transferred to the cartridge assembly department where they are loaded into 8-track plastic housing. The same pancakes, incidentally, are also shipped to music duplicators throughout the world who record music on them at 120 inches per second, then load them into cartridges, and make them available as prerecorded 8-tracks.

Cassette tape used for loading into blank cassettes must be specially prepared before shipping to arrange for the provision of leaders. To accomplish leadering in an efficient way, the jumbo rolls of cassette tape are rewound, and at specific intervals of length, a section of full-width leader material is spliced in. The intervals depend on the desired length of tape in the cassette to be manufactured. For instance, if 60-minute cassettes are to be manufactured, the leader is spliced in every 282 ft.; for 90-minute cassettes, the length becomes 422 ft., and so on.

Final Assembly

Once slitting is completed, final assembly can begin. For each format the process is different.

Open Reel. Since reel tape is already on hubs, it is nearly ready for shipping. First it is visually inspected, and then bulk-erased to bring the noise level down to the virgin state.

8-Track Cartridge Assembly. Since the 8-track cartridge is designed to operate continuously in one direction, without rewinding, the cartridge has no reels and the tape is wound on a platform in a closed loop, with no ends. In operation, the tape feeds from the inside of the pack, driven in clockwise direction by the built-in pinch roller and the recorder's capstan, and travels past the various guides, the heads, the pinch roller, and back onto the outside of the tape pack.

Because pack diameters differ between take-off and take-up points, all tape layers within the pack of a cartridge must continuously slide on each other to adjust the running tension and also to provide wow-free, flutter-free motion. A good cartridge should run for hundreds of hours with no increase in tension or motion disturbance. To accomplish this, we employ a special lubricated coating on the back of the tape which substantially reduces friction and extends cartridge life.

This lubricant must also run cleanly. Any dirt deposited on the heads causes severe high frequency losses. It must not cake up on the pinch roller either, since this produces slippage, wow, and speed changes. Finally, the lubricant must not transfer to the oxide side of the tape, or it would cause dropouts.

The assembly of 8-track cartridges begins by unwinding a precise length of tape from the 14-in. pancake directly onto the cartridge platform. The platform and various components of the cartridge including the pressure pad and pinch rollers are then assembled into the base and the cover is snapped into place. The two ends of the tape hanging out from the cartridge opening are cut to a specific length to form a "drop loop." These ends are then spliced together on the oxide side using a metallic splicing tape, the track switching foil, which is designed to automatically switch between the four pairs of stereo programs recorded in the 8-track cartridge. All recorders and players for 8-track are equipped with a head that moves up and down in response to the switching action. Many users record right over the splice since the interruption in sound lasts less than half a second.

Cassette Assembly. In the assembly of cassettes, the pre-leadered tape on the large reels is transferred to the cassette hub. The leader is securely fastened to one hub and the tape is wound on the hub until the following leader appears. This leader is cut and the end fastened securely to the other hub. The two hubs are then inserted into the cassette half, along with other components, such as anti-friction liners, rollers, shield, and pressure pad. The cover is put on next and is screwed or welded together to form the finished cassette. The labels are attached last and the cassettes are packaged into the various type of boxes.

Final Testing

In addition to important process control tests performed during manufacturing, quality-assurance tests are performed on the finished product to check the product for conformance to specifications. All tapes undergo this testing, including tape which will be loaded into cartridges and cassettes.

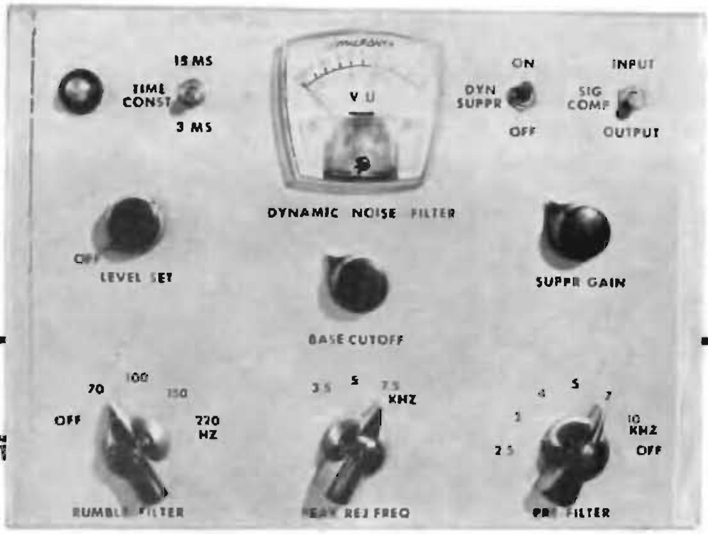
The three kinds of tests conducted are *physical* tests, to measure such parameters as width, thickness, density, tensile strength, friction, scratch resistance, surface smoothness, head abrasion, and temperature and humidity stability; *magnetic* tests, to check coercive force, residual flux (remanence), residual flux density (retentivity), and the squareness of the hysteresis loop; and *recording performance* tests, to check frequency response at various speeds, distortion, uniformity, noise, dropouts, print-through, and even the compatability of the tape with various recorders and their specific bias settings.

While physical and magnetic tests are standardized and do not vary with the application, recording performance tests are always geared to the particular use for which the tape is intended. Recording equipment used for performance tests consists of professional recorders equipped with heads and adjustments typical of home use conditions. In addition, some tests are performed on a large variety of different home recorders to see how the tapes will behave under somewhat less than ideal laboratory conditions.

Besides tests which are done on the tapes themselves prior to assembly, finished blank cartridges and cassettes are also subjected to complete quality assurance testing. Eight-track cartridges, for example, are tested for wow, flutter, and tension using new blank cartridges as well as ones hundreds of hours old. The tests are done under ambient conditions as well as the extreme temperature-humidity conditions which will be encountered in cars, boats, and other situations where cartridges are commonly used. Since 8-track players are often given little preventive maintenance by their owners, primarily because cartridge players are used principally away from home, particular attention is paid to accurate and trouble-free operation under unfavorable conditions of use and maintenance.

Cassettes, too, are checked for wow and flutter, torque requirements, smoothness of motion, and life. They are also checked for recording performance in the cassette since this cannot be accurately determined by pre-testing because the cassette plastic housing serves as the alignment surface in the machine and thus, at least partly, determines the azimuth adjustment between the head and the tape. A good cassette housing is made with precision molds from stable, warp-resistant plastics, and is assembled with great care to maintain the alignment of which the plastic parts are capable. Response tests measuring the effects of plastics are performed on the number of cassette recorders ranging from the simplest decks to the most elaborate three-head machines.

By now, most of you have probably concluded that a lot of effort goes into production of quality magnetic tape. Indeed, it takes a keen attention to manufacturing controls, a great deal of experience, and extensive quality control testing to produce this miraculous recording medium.



Construct

Maxwell G. Strange

THE SERIOUS collector of older recordings faces the challenge of getting the best possible sound from imperfect originals. Most of these records are quite noisy by today's standards. For example, 78 rpm commercial discs, even though in mint condition, will have a typical signal-to-noise ratio of only 30 to 35 dB due to the somewhat abrasive nature of the record material.

Most collectors dub their best records onto tape. This way they may be played as often as desired—and conveniently shared with other collectors—while the often irreplaceable originals are safely preserved. Also, the sound can often be improved considerably during the copying process through equalization and filtering. I'm going to describe a flexible, low-cost noise filter designed for taping records with a maximum "fidelity-to-noise" ratio. It can be duplicated by the serious electronics hobbyist for about $60, or slightly less if certain features or ranges won't be needed. Although not recommended as a beginner's project, the experimenter with some circuit experience should have no difficulty. Minimum equipment requirements are an oscilloscope, sine wave generator, and multimeter.

The heart of this circuit is a dynamic noise suppressor with frequency characteristics and convenience features which are optimized for its intended use. The concept of dynamic noise suppression has existed for many years. Workable circuits were designed by H.H. Scott in 1946, and their performance was improved by Scott and others in 1947 and 1948. Then, with the advent of the vinyl microgroove record and the rapidly increasing use of tape, both of which offered a considerable noise improvement over the 78 rpm system, the dynamic noise suppressor was almost forgotten. Recently, R. Burwen has revived this principle and applied it primarily to tape playback. Taking full advantage of modern integrated circuits, Burwen has designed highly sophisticated and flexible systems with impressive specifications. These, however, are too expensive for many hobbyists and do not have frequency characteristics optimized specifically for old, intrinsically band-limited material.

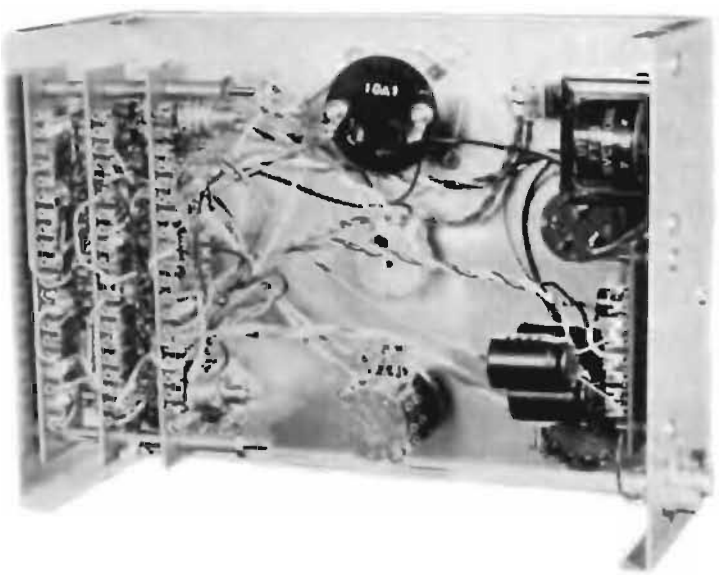


Fig. 1—Interior of the dynamic filter.

Theory

Dynamic noise suppression is simple in concept. Record surface noise varies in spectral content, but the higher frequencies (above 1 or 2 kHz) predominate. Low-pass filtering is commonly used to limit noise. But unless used sparingly, this type of filtering band-limits the program material, making it sound muffled and lifeless. The dynamic filter, however, provides a method by which a signal can be effectively extracted from the noise (at least subjectively) when signal and noise occupy overlapping frequency ranges.

Operation of the dynamic noise suppressor depends upon a characteristic of the human auditory apparatus. If two signals occupying well-separated frequency ranges are present simultaneously, they are clearly perceived as individual entities. (This effect is often used to advantage in public address systems for noisy environments. If considerable high-frequency boost is used, voice announcements will seem to cut through ambient noise of predominately lower frequency without having to be excessively loud.) This is the case, at least for a large portion of the time, for a typical recorded signal with attendant surface noise; hence, the annoyance of the noise. However, if two simultaneous signals occupy substantially the *same* frequency ranges, the ear will tend to hear only the louder signal and ignore the weaker one. A level difference of only a few dB is sufficient for one signal to effectively override, or mask, the other. Operation of the dynamic noise suppressor depends upon this masking effect.

The dynamic filter has a fairly steep low-pass character-

A Dynamic Noise Filter

istic which, in the absence of signal, starts cutting off at about 1 kHz. This very effectively rejects the noise spectrum. When a signal having high-frequency components at sufficient amplitude comes along, the filter is made to "open up"; that is, its cutoff frequency is quickly raised. As the high-frequency program content drops in frequency and/or amplitude, bandwidth contracts. The idea is that when high-frequency signal components are present, they will tend to mask the accompanying noise. When highs are not present, the wide bandwidth is not needed. Admittedly, the recovered signal is not as faithful as a noise-free original would be. For example, high-frequency content in low-level passages may be lost. Of some help here is the fact that many musical instruments tend to have less harmonic content at low acoustic levels. In spite of this compromise, the processed signal is usually far more pleasing to the ear than the noisy input signal.

The bandwidth control signal is derived by separating the high-frequency program components from the signal-plus-noise. Unless the signal level is consistently higher than the noise to begin with, this becomes impossible. Thus, there is a minimum signal-to-noise requirement below which no improvement is possible. As the original S/N improves, the dynamic suppressor's performance improves also.

Ideally, the signal frequency range to which bandwidth is most sensitive should correspond to the frequency range of maximum noise. The optimum filter characteristic for separating the bandwidth-control signal from the noisy input thus varies widely with the characteristics of the noise with which we are dealing. Bandwidth control sensitivity (or

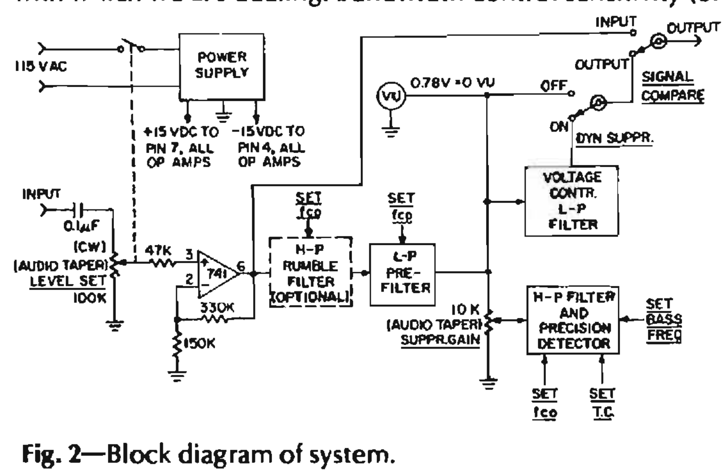


Fig. 2—Block diagram of system.

gain) must be set properly for the incoming signal level and noise properties. Bandwidth should respond rapidly to signal changes to avoid loss of transients and to prevent audible "swishing" sounds which can be produced by delayed bandwidth contraction.

Design Approach

I have tried to implement the basic requirements outlined above as completely as possible in an easy-to-use, low-cost unit. A dynamic high-pass filter stage was considered but later dropped, as high-frequency noise predominates on most older records. It is my experience that low-frequency noise can usually be handled adequately with a simple manually-set rumble filter.

Figure 2 shows an overall block diagram of the noise filter. Operational amplifier A1 is connected as a non-inverting amplifier with a voltage gain of 3.2 (10 dB), enabling the system to be driven to 0 VU with an input level of 0.25 volt. This amplifier also serves as a buffer, providing an input impedance of 100 kilohms for compatibility with virtually any signal source.

Amplifier A1 drives the *rumble filter,* which could be

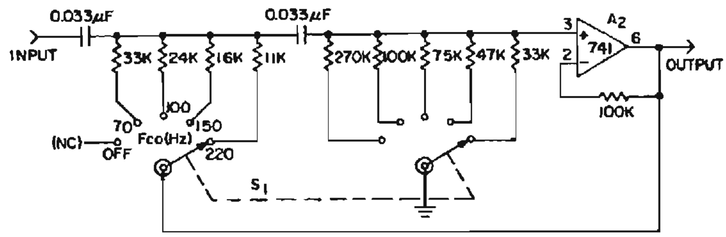


Fig. 3—Optional high-pass rumble filter schematic.

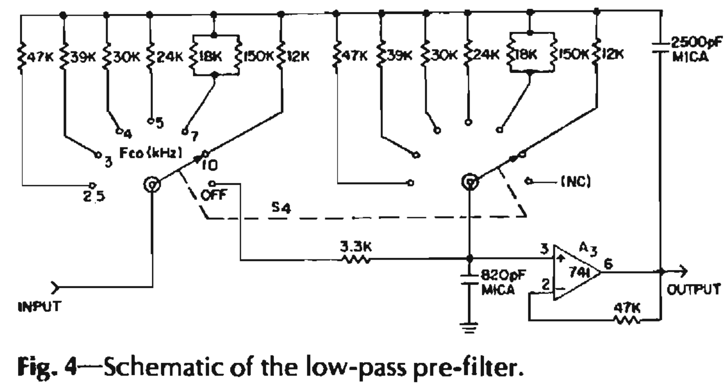


Fig. 4—Schematic of the low-pass pre-filter.

omitted if one is available in the associated external equipment. Following this is the *pre-filter,* which is simply a low-pass filter with a manually-set cutoff. This filter is important for several reasons. First, it removes noise which is above the frequency range of the recorded signal. Many recordings have no signal content above 4 or 5 kHz (even lower for acoustic records), and no program content is lost by cutting off the upper range. Thus, the total noise voltage is lowered, often appreciably, permitting the use of higher suppression gain settings as will be seen later. Another reason for this filter is that the dynamic filter can do nothing to reduce the annoyance of high-frequency distortion. Furthermore, since a limited-bandwidth signal cannot effectively mask higher-frequency noise, removal of the latter helps to eliminate audible evidence of the continually changing bandwidth.

From the pre-filter output the signal passes to the *voltage-controlled l-p filter* and, via the *suppression gain* control, to the *h-p filter/precision detector* whose function is to derive the bandwidth control signal. This point additionally goes to a switch which permits the dynamic filter to be bypassed at will so that its effect with various control settings may be easily judged. Another switch permits the output to be compared with the "raw" input signal.

All of the filters used in this system, including the voltage-controlled filter, are of the 2-pole active type, giving a 12 dB/octave rolloff slope. The damping factor is chosen (with one exception) for a Butterworth response, which produces the steepest possible slope beyond cutoff with no peaking in the passband. (High-pass filters with 3-dB peaking were tried, but these produced a slightly rough, "grainy" sound compared to the flat-passband version.) The design approaches are widely published and need no further discussion here. The rumble filter (Fig. 3) and the pre-filter (Fig. 4) are of this type; their response curves are shown in Fig. 5. The rumble filter is not essential to proper suppressor operation, but is convenient in case an effective low-cut filter is not included with the associated preamplifier in the copying setup. The design shown here has rather high settings intended primarily for acoustic records.

The bandwidth control signal is derived with the circuit of Fig. 6, which consists of a high-pass filter followed by a precision detector. The filter damping factor is made low in order to produce a pronounced peak and more rapid low-frequency rolloff (Fig. 7). Three selectable cutoffs produce peaks at 3.5, 5, and 7.5 kHz; these were empirically determined to best accommodate a wide range of noise characteristics and recorded bandwidths. The filter output is coupled to the detector via a small capacitor to make the low-frequency rolloff even steeper below 1.6 kHz. The precision full-wave detector uses diodes in the feedback circuit of an op-amp to effectively produce ideal rectification characteristics down to the millivolt region. The output amplifier doubles as a post-detection filter. Resistor R determines the gain, and capacitor C makes this stage behave as an operational integrator with time constant RC. A switch is provided for increasing the time constant by paralleling capacitor C1; this is helpful with sources having sharp impulse noise. The output of the detector/filter circuit controls the bandwidth of the dynamic suppression filter according to the curve of Fig. 9.

Early experiments showed that it is undesirable to make the no-signal cutoff lower than absolutely necessary to substantially reduce noise with a particular signal source. When the cutoff is made lower than actually needed, weak signals are unnecessarily band-limited and the dynamic filter produces such a level-dependent bandwidth contrast that its action is much more likely to be audible. Hence a BASE CUTOFF (not "BASS CUTOFF") control was found to be desirable. This control is simply a pot which offsets the detector output at zero signal level by applying a variable reference voltage to the op amp non-inverting inputs. This voltage, variable from about −1 volt to −6 volts, establishes a "starting point" or base cutoff frequency which can be set just low enough to virtually eliminate no-signal noise.

The variable-cutoff filter, Fig. 8, is the very heart of the system. Since there is some part selection and adjustment necessary, it must be checked out separately. The basic configuration is similar to that of the pre-filter, except the latter's switch-selected resistors have been replaced by field-effect transistors (FETs). FET channel resistance R_{DS} changes as a function of gate voltage V_{GS} as shown in Fig. 11, thus varying cutoff frequency. A resistor across each FET establishes a solid lower cutoff limit and smooths the control characteristic as the FETs approach their "off" state. The gate circuit network, consisting of diode D1 and resistors R1 through R5, is used to empirically shape the control curve (Fig. 9) for best audible results. Diode D1 prevents excessive positive gate drive, maintaining isolation between the gate and signal circuits.

An input attenuator (R10 and R11) limits the signal amplitude presented to the FETs to about 0.1 volt p-p at 0 VU to ensure low distortion. Output amplifier A7 makes up exactly for this loss. An op amp having external frequency compensation was used here so that this relatively high-gain stage could be tailored for flat response to 15 kHz (a µa741 could be used, but would roll off slightly above 10 kHz). Resistors R16 and R17 attenuate the output signal by an amount equal to the gain, so that this amplifier doubles as the unity-gain buffer required for filter operation. The highest cutoff frequency is dictated by minimum FET resistance and capacitors C1 and C2. The latter should have values in a ratio of about 3:1 to produce the desired Butterworth response. Figure 10 shows the measured response of the complete filter for four values of control voltage.

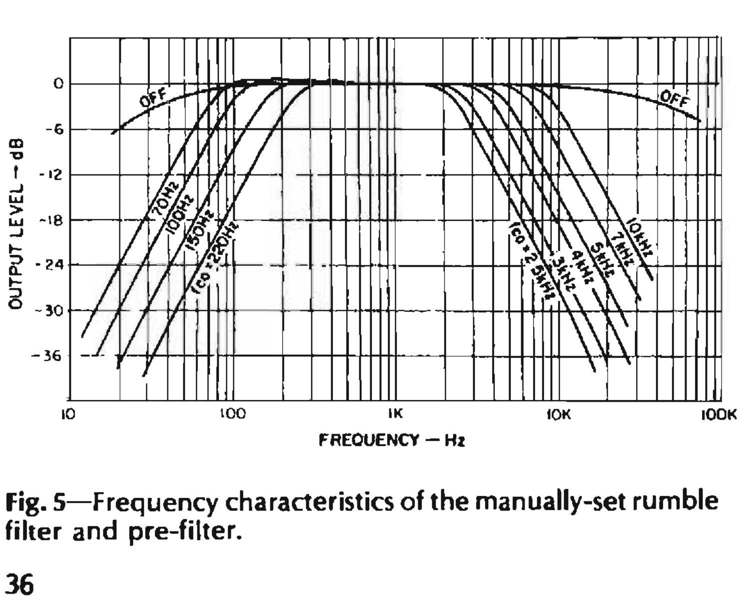


Fig. 5—Frequency characteristics of the manually-set rumble filter and pre-filter.

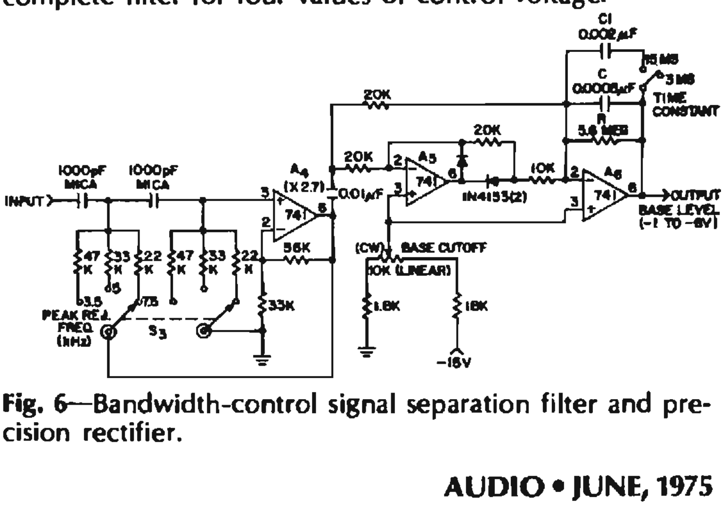


Fig. 6—Bandwidth-control signal separation filter and precision rectifier.

Unfortunately, FETs vary widely in characteristics, even between units of the same type, so these devices must be selected. The two FETs must be reasonably well matched over a 15:1 R_{DS} range for a 15:1 range in cutoff frequency (15 kHz to 1 kHz). (Dual matched FETs are available, but are more expensive and not necessarily matched for the parameter of interest here.) A transistor curve tracer is most convenient for this purpose and permits selection for best linearity as well as matching. I used N-channel 2N4220s on hand ($1.50 each) and selected the best matched pair out of a group of 6 units. Figure 11 shows the V-I characteristics of one of these. There are many other inexpensive FETs which should work as well, such as the 2N5484, 2N5716, and 2N5717 at under $1 each. In fact, any general-purpose, depletion-type FET with fairly low zero-bias current (I_{DSS}) and pinch-off voltage (V_p) should be usable. P-channel units would require reversing diodes D1 and D2 and the polarity of the control voltage.

If a curve tracer is not available, the setup of Fig. 12 can be used. A transistor socket will facilitate changing FETs. A good procedure is to first measure R_{DS} at $V_{GS} = 0$. Then increase V_{GS} (negatively for N-channel FETs) until R_{DS} is about three times the zero-bias value; this corresponds to a mid-range cutoff frequency where matching is the most critical. With this V_{GS} setting try different FETs until a 10 percent or better match is found. If R_{DS} values seem to cluster higher or lower, try another unit as a reference and try matching to it. When matched units are found, check the match at minimum R_{DS} ($V_{GS} = +0.5$ V) and at 10 times this value of R_{DS}. A 20 percent mismatch can be tolerated at these extremes. My 2N4220s measure 610 ohms at zero bias, 360 ohms at $V_{GS} = +0.5$ V., and about 8 kilohms at $V_{GS} = -0.7$ V. R11 and R12 are chosen for a cutoff of between 800 Hz and 1 kHz with the control voltage at its maximum negative value of about −6 volts. Circuit cutoff at zero FET bias should be roughly 12 kHz (see Fig. 9). A slight forward bias, limited to about +0.5 volt at the FET gates by diode D2, then boosts the cutoff to at least 15 kHz with maximum positive output from the precision detector.

Resistors R6, R7, R18, R19, and R20 reduce harmonic distortion significantly. R6 and R7 feed some signal to the FET gate circuit so that signal voltage does not appear between source and gate, which would make R_{DS} vary slightly with instantaneous low-frequency signal amplitude and polarity. R18, R19, and R20 feed back some output signal to the gates to further reduce distortion (this is a cancellation effect, not true negative feedback).

Distortion settings are best made in the vicinity of cutoff, where FET linearity is the most critical. Connect a variable-voltage d.c. source (the slider of a 5K pot temporarily connected between −15 V and ground will suffice) to the bandwidth control input and set it for a cutoff frequency of 2 kHz. Then, with a 2 kHz sinusoidal input at about 0 VU (2.2 V p-p), set trimpots R7 and R18 for lowest harmonic distortion at the output. It should be possible to sharply null the total harmonic content, which consists primarily of the 2nd and 3rd harmonics, to at least 60 dB below 0 VU. Then vary the cutoff frequency and make sure distortion is low for all settings. Of course, the filter itself will reduce harmonic distortion appreciably at its lower cutoff values. Lacking a distortion meter or wave analyzer, these adjustments can be made quite well by driving the input at 7 volts p-p (10 dB above 0 VU) to accentuate the distortion and setting very carefully for a symmetrical output waveform as monitored by a 'scope. Fixed resistors, determined by two decade boxes (the settings interact somewhat), could replace the pots. These adjustments, once made, are permanent unless the FETs are changed.

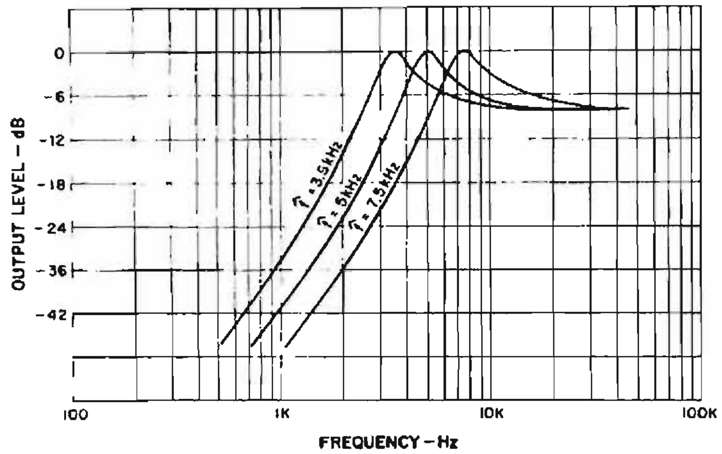


Fig. 7—Frequency characteristics of the filter used to derive the bandwidth control signal.

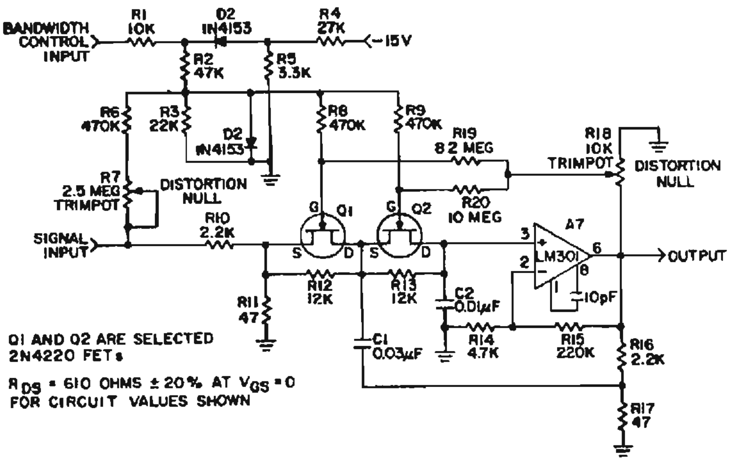


Fig. 8—Voltage-controlled filter schematic. FETs Q1 and Q2 are critical and must be selected (see text).

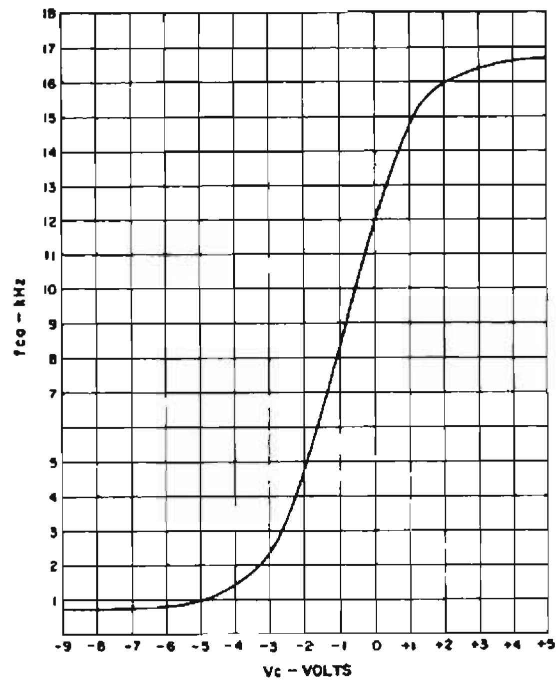


Fig. 9—Variable-bandwidth filter cutoff frequency vs. control voltage.

Fig. 13 shows the distortion of the complete noise filter measured at two fixed values of bandwidth control voltage. At normal levels, distortion is so low that it is largely a measurement of the harmonic distortion of the test oscillator. The large margin above 0 VU passes the highest program peaks ever likely to be encountered without clipping.

The simple power supply of Fig. 14 easily supplies the power requirement of ±15 volts at about 10 mA.

Construction

The entire filter can be duplicated for about $60 with new parts. Sources of the major components are shown in the Parts List; substitutes can be used in most cases. Quarter-watt, 5 percent composition resistors are suitable. Layout is not critical, since signal levels are high and impedances are relatively low. I strongly recommend that each of the functional blocks of Fig. 2 be built and checked for reasonable conformance with the curves before integration into the system. This makes troubleshooting for errors and occasional bad components much easier, practically ensuring success. My unit (Fig. 1 and lead photo) is a "breadboard in a box." The circuit is still undergoing occasional changes, even though it is a third-generation model. Parts are mounted on terminal boards which were on hand. A neater approach would be to use the commercially-available perfboard with snap-in terminals.

Operation

After checking the wiring, apply power to the unit and check for proper power supply voltages. Positive and negative supplies should both be between 14 and 16 volts with respect to ground. Much lower values would indicate a short circuit or bad op amp. Current drain should be on the order of 10 mA.

The noise filter can be conveniently connected to your audio system by means of the *Tape In* and *Tape Out* jacks included on most preamplifiers. An advantage to this connection is that the processed signal passes through the preamp tone controls, which can be set for the most pleasing final balance. For taping, the recorder input is paralleled with the output which drives the power amplifier.

For initial set-up experience, a record having a good frequency range and moderate, steady surface hiss is desirable. (A slightly noisy FM station can also be used, but results will not be quite as good because of the latter's flatter noise spectrum.) Initial control settings should be:

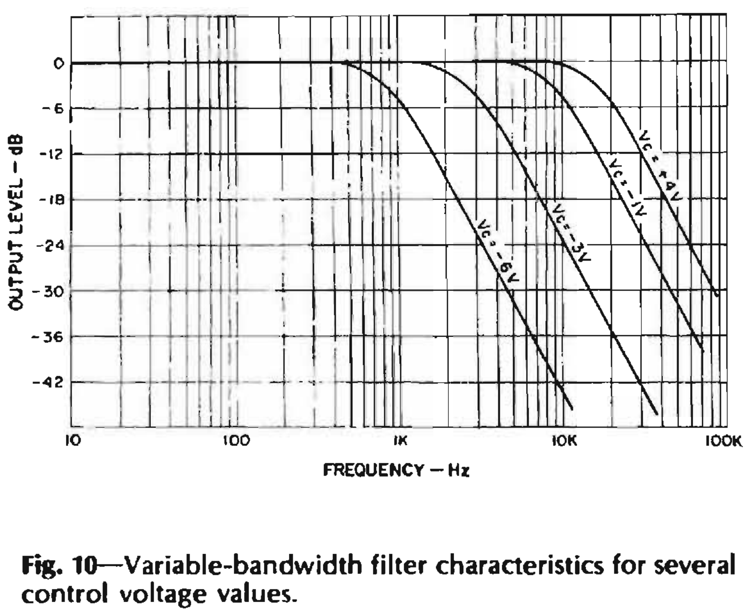


Fig. 10—Variable-bandwidth filter characteristics for several control voltage values.

Pre-Filter: Off
Rumble Filter: Off
Time Const: Off
Peak Rej. Freq.: 5 kHz
Base Cutoff: CCW
Suppr. Gain: CCW
Dyn. Suppr.: Off
Sig. Compare: Input

The signal should now pass through the unit unaffected, except the *Level Set* control will vary the gain from zero to 3.2 (10 dB). Set the level for 0 VU on signal peaks as you would set a recording level. Whenever the source is changed, the signal level should be reset as necessary.

Now switch the *Sig. Compare* switch to "output." The signal is now passing through the rumble filter (if used) and pre-filter, but bypassing the dynamic filter. Lowering the *Pre-Filter* cutoff setting should progressively cut off the highs. At the lower settings, which are primarily for acoustic records, the signal will sound severely band-limited. The best setting is the lowest cutoff which does not significantly affect the recorded bandwidth. I have found that with vocal music, the unfiltered sibilant sounds provide a means of judging bandwidth. If sibilants are quite strong and natural, a 7 kHz or higher cutoff is indicated. If they are weak or have a slight "whistling" sound, the upper limit is about 5 kHz. If sibilants are lacking, a 4 kHz or lower setting is best. Of course, the presence of high-frequency distortion may dictate a compromise setting a notch or two lower than indicated above. The filtered and unfiltered sounds may be compared at any time by means of the *Sig. Compare* switch. The optional rumble filter is used for the occasional records which have warpage or bumps or low-frequency noise in the recording. For acoustic records it can be routinely left at 150 Hz, as nothing is recorded below about 200 Hz.

Next flip the *Dyn Suppr* switch to "on," putting the dynamic suppressor in the circuit. The sound should become very dull and lifeless, as the high-frequency cutoff is now 1 kHz or less. Increase the *Base Cutoff* setting until record noise just begins to be audible. The signal will probably still be quite lacking in high-frequency content (if it is not, only the pre-filter may be needed for this particular source). Now turn up the *Suppr Gain* slowly. This should "magically" restore the highs without increasing the noise level. The highest possible setting which does not noticeably increase the noise is normally best.

At this point it is edifying to monitor the bandwidth control input signal to the variable-bandwidth filter with a d.c.-coupled oscilloscope. The instantaneous voltage here is a measure of high-frequency program amplitude and dynamic filter bandwidth (see Fig. 9). It should follow transients rapidly and may reach saturation (about +14 volts) on musical passages having high harmonic content and on strong voice sibilants.

The *Peak Rej. Freq* switch selects the frequency of peak rejection by choosing the appropriate filter curve (Fig. 7) for separating the bandwidth-control voltage from the input signal. The *5 kHz* position is used for most electrical 78 rpm records. For acoustic records or very noisy electrical 78s where the pre-filter is set for 4 kHz or less, the *3.5 kHz* position gives better results. Here the *Time Const.* switch can be set for *15 mS.* The longer time constant also helps to attenuate sharp clicks and pops occuring in quiet passages, as it prevents the bandwidth from increasing rapidly enough to follow their steep wavefronts. The *7.5 kHz* position is used for wideband recordings and tape.

With a little practice, you will be able to set the controls quickly for optimum performance. It is often best to set the

Base Cutoff for a significant improvement, rather than to try to eliminate the noise completely. This will minimize low-level band limiting, and the suppressor will be less likely to betray its presence with obvious bandwidth changes.

Performance

Figures 5 and 10 indicate the bandwidth ranges available. The pre-filter and dynamic filter (slope is 24 dB/octave above both cutoffs) can together provide well over 60 dB of noise attenuation at 10 kHz and over 40 dB at 5 kHz. The overall improvement in signal-to-noise ratio is strongly determined by the character and spectrum of the noise, which varies greatly with records. With the steady hiss typical of new electrical recordings on shellac, an average improvement of 8 dB (unweighted) is realized from the dynamic filter alone. Including the effects of the rumble filter and pre-filter on band-limited material, S/N improvement can be more than 12 dB. The apparent improvement is even greater, since the ear heavily weighs the higher frequencies where record noise is concentrated. The effect of the noise filter is surprisingly great on records which were originally thought to be quiet without filtering. It is a little weird at first to hear a familiar old record with realistic strings and brass and clear voice sibilants, but with the background suddenly rendered deadly quiet. I have spent many hours listening to the records and tapes in my collection and enjoying them anew.

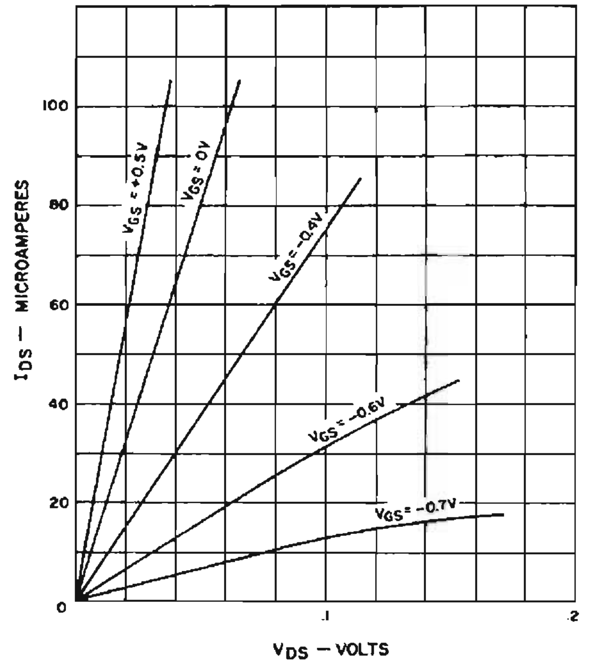


Fig. 11—Variable-resistance characteristics of a junction field-effect transistor with low values of drain-to-source voltage.

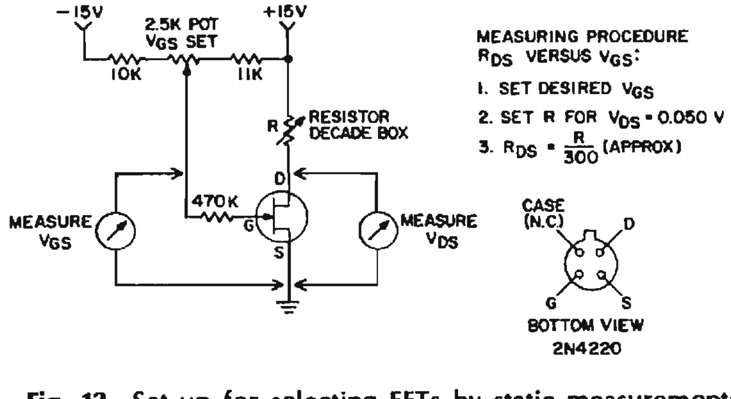


Fig. 12—Set-up for selecting FETs by static measurements (see text). Small 15 V batteries or the power supply of Fig. 13 may be used.

The noise filter works very well on tape noise, providing at least 8 dB total S/N improvement. A stereo version built for tape only could be simplified considerably, as only the *Level Set, Base Cutoff, Suppr. Gain,* and *Sig. Compare* controls would be needed. The power supply as shown can easily handle two channels.

The noise level of the filter itself depends mostly on output amplifier A7. Of several units I tried, the noise level ranged from 62 to 68 dB below 0 VU.

A few tips on the mechanical aspects of copying records are in order here. The importance of good tracking cannot be overemphasized. More can be gained here than with any amount of electronic processing. Groove radius, depth, and angle were not standardized on early discs, and experimentation with tracking force and stylus size, if possible, may yield a considerable improvement in both noise and distortion. The playback stylus should, of course, ride on the sides of the groove. If it is too small it may ride the bottom of the groove and skate from side to side in a partially uncontrolled manner, creating severe distortion. If too large, it will ride high in the groove where it is more sensitive to surface blemishes. Also, larger styli cannot follow high-frequency modulation as well, especially on the inner record grooves. Elliptical styli are helpful on relatively wide-range 78s if the latter have not been damaged by previous playings.

Acoustic records (1925 and earlier) tend to have a larger groove, since with acoustic playback the mechanically-imparted stylus motion had to supply all the sound power. For these, a stylus of 4-mil (.004") radius may produce better results than the standard 3-mil size. Custom-made styli with a

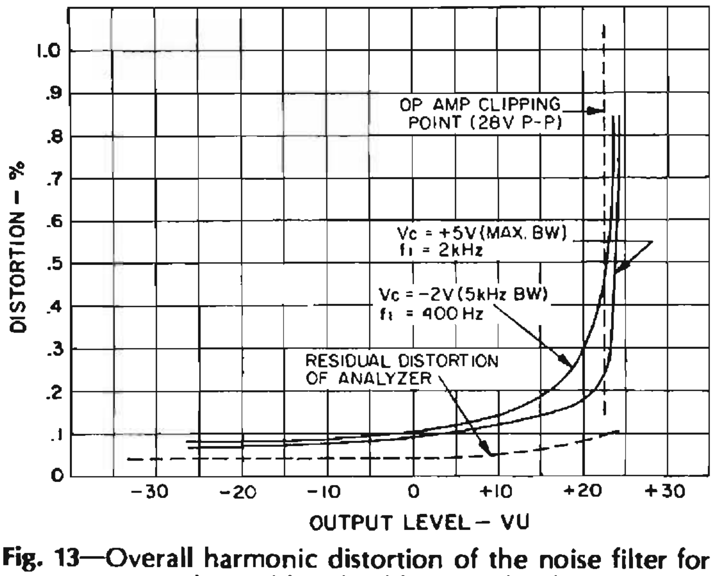


Fig. 13—Overall harmonic distortion of the noise filter for two constant values of bandwidth-control voltage.

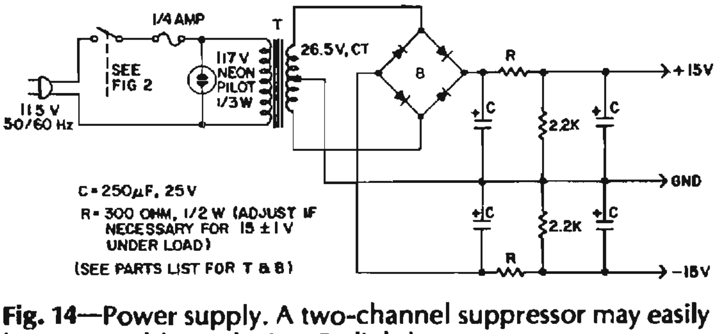


Fig. 14—Power supply. A two-channel suppressor may easily be powered by reducing *R* slightly.

"truncated" tip (really a smooth transition from a 2- or 3-mil radius to about a 4-mil radius at the very tip) have been used to track the groove sides of 78s properly while avoiding contact with the bottom. (Truncated and other special styli are available from International Observatory Instruments, 5401 Wakefield Drive, Nashville, Tenn. 37220.) Although not a cure-all, these can give dramatic results on selected discs. A 2.5-mil stylus is best for most post-1946 transcriptions. Obviously, the pickup should have adequate lateral compliance and should produce no output for vertical motion. Incidentally, electrical recordings made before the mid-1940s are mostly recorded flat, that is, they have no high-frequency pre-emphasis, while later records have pre-emphasis of as much as 16 dB at 10 kHz.

Edison cylinders (160 rpm) and discs (80 rpm), some Pathé discs, and some early wax transcriptions are vertically modulated. Here the stylus does ride on the groove bottom, and the pickup should have only vertical response. This can be obtained (as can lateral-only response) from a suitably-phased stereo cartridge. Stylus radii of 4 to 10 mils are typical here; as always, experimentation is in order.

Future Development

The experimenter may want to try to improve the performance of the circuit described. Of course, additional types of processing can be added, such as more effective click suppression at the filter input or multi-channel equalization at the output. These would be electrically independent of the noise filter, and beyond the scope of this paper. However, there are some possibilities for improving the noise filter itself. Many of these, unfortunately, would require an incongruous increase in complexity and cost.

Sharper filter cutoffs give a marginal improvement on very noisy material, but setup adjustments become more critical. Dynamic high-pass (low-cut) filtering using a simple 6 dB/octave slope might be a reasonable addition. Since the noise-rejection frequency band of the low-pass dynamic filter should complement the noise spectrum of the signal, a statistical study of record and tape noise spectra might lead to a better shape for the bandwidth-control-signal separation filter of Fig. 7. The separation filter selector could be ganged with the pre-filter cutoff switch to eliminate one control knob. Perhaps a noticeable improvement could be realized by experimenting with the shape of the bandwidth control characteristic, Fig. 9. The attack time constant could be shortened by using a more elaborate filter at the precision detector output; this would improve the response to occasionally encountered wide-band transients.

An obviously desirable change would be to replace the FET bandwidth-control filter with one of the voltage-controlled state-variable type. This would eliminate the need for FET selection, but would increase the cost severalfold. It therefore appears that the original goal of high performance per dollar has been achieved, yielding a practical design which is within reach of the hobbyist.

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4. Scott, H.H., "Dynamic Noise Suppressor," *Electronics,* PP. 96-101, Dec., 1947.
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Parts List

Qty	Part	Mfg. Number	Source
1	Power Xformer*	UTC FT13 or Allied 6K48HF	Newark #2F702 Allied #705-0065
1	Bridge Rect (B)	Allied VE08	Allied #976-3021
4	250 μF, 25V (C)	Allied N-G-500	Allied #710-1356
6	Op Amp (A1-A6)	Fairchild U5B77-41393	Allied #569-2100
1	Op Amp (A7)	Motorola MLM301AP1	Newark (no # req'd)
10	FET (Q1 and Q2)	Motorola 2N4220	Newark (no # req'd)
4	Diode	Fairchild 1N4153	Allied #551-4153
1	VU Meter	Micronta 22-019	Allied #910-4519
3	SPDT Toggle Sw	Cutler-Hammer SF1SBX191	Newark #29F2274
1	Selector Sw (S1)	Mallory 3226J	Newark #22FO56
1	Selector Sw (S2)	Mallory 3229J	Newark #22FO61
1	Selector Sw (S3)	Mallory 3223J	Newark #22FO55
1	Pot, 100K Audio Taper (Level Set)	Mallory U39	Newark #9F221
1	Pwr. Sw. for above	Mallory US26	Newark #9F246
1	Pot, 10K Linear (Base Freq.)	Mallory U20	Newark #9FO89
1	Pot, 10K Audio Taper (Suppr Gain)	Mallory U18	Newark #9FO87

*Allied transformer is larger than UTC, but costs less.

These parts should be available at any industrial electronics supply store. Some electronics parts distributors will also carry. For mail order use Newark Electronics, 500 N. Pulaski Rd., Chicago, Ill. 60624 or Allied Electronics, 401 E. 8th St., Fort Worth, Texas, 76102.

ONLY TWO OR THREE years ago, the pundits were prophesying the early demise of the open-reel tape recorder, but (as with Mark Twain's famous remark) the reports of its death were highly exaggerated. New models are seen at every hi-fi show, ranging from inexpensive stereo models to semi-professional and quadraphonic machines with such features as logic tape-motion controls, multi-track sync facilities, line and mike mixing, plug-in head assemblies, and much more.

In the same period, however, cassette recorders have been improved almost beyond belief, and the first-line models—those in the $300 to $500 bracket—cheerfully invite comparison with most any open-reel machine. Recent innovations include monitor heads, mixing circuits, variable speed controls, greater choice of bias and equalization, and easier head adjustments. And there are even some luxury machines in the $500 to $1200 range! So, let's see how the cassette deck really stacks up, whether it will kill off the open-reel recorder in home applications.

Before we try to sum up the pros and cons, it would be well to set down the various requirements of a tape recorder, and here are some—not necessarily in order of importance:

1. Wide frequency response;
2. Low distortion;
3. High signal-to-noise ratio;
4. Low wow and flutter;
5. Ease of operation;
6. Monitoring facility;
7. Editing facility, and
8. Additional features, such as peak limiter, mixing, sound-on-sound, track sync, etc.

There are, of course, other factors to consider when actually buying a recorder such as styling, guarantees, servicing, and price, but for the moment we'll ignore them.

Basic Parameters

Frequency response, distortion, and noise are all closely related so they will be considered together. As is well known, the performance of any tape recorder is greatly determined by the tape area and speed. Thus, a cassette recorder, with narrower tapes and working at the slow speed of 1⅞ ips, is basically handicapped relative to the open-reel recorder. Examining the track configuration of cassette tape compared with the larger tracks used by open-reel tape and taking into consideration the speeds, the tape area (or the number of magnetized particles, if you will) for the cassette is only 15 percent of that used by the open-reel machine working at 7½ ips. Thus, it follows that the inherent problems of tape hiss and random particle noise are more difficult for the designer of cassette machines.

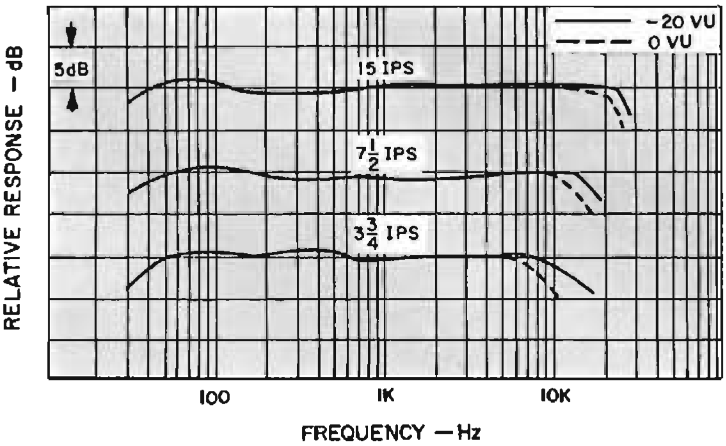


Fig. 1—Frequency response of a moderately priced, three-speed, open-reel tape recorder at 0 (dashed line) and -20 VU (solid line).

Assuming that the signal output is in direct proportion to the track area, it is easy to see that if the cassette output is amplified to the open-reel level, the noise level is, almost certainly, going to be a lot higher. Figure 1 shows the frequency response of a typical quarter-track, open-reel machine costing around $350.00 and using a low-noise tape such as Maxell UD, Scotch HO/LN, or BASF LH Super. Figure 2 shows the frequency response of a typical top-quality cassette deck, and it will be seen that its high-frequency response is somewhat reduced. Note also the difference between the Normal, Low Noise, and CrO_2 tapes. High-frequency response might be extended to some extent with the new ferric-chrome, dual-layer hybrid tapes, but not all machines can use them to advantage at present. Thus, it would seem that 18 kHz is about the top limit for cassette machines at present.

But this is not the whole story. One of the limiting factors presently is the head itself, as most cassette decks use a single head for recording and playback. But the requirements of the different functions are conflicting, as the gap on the recording head needs to be large for efficient signal transfer, and the playback head gap must be small to give a good output at high frequencies (or put another way, the gap must be small compared with the recorded wavelength). A limit is reached when the recorded wavelength is equal to the gap length because the two oppositely-magnetized half wavelengths will cancel and the induced voltage is zero. The wavelength of a 20-kHz signal at 7½ ips is about 8.5 microns, but reducing the tape speed to 1⅞ ips brings down the wavelength of the same frequency signal to about 2.2 microns! Although the effective magnetic gap is a little larger than the actual physical gap, if we want a cassette deck to have a linear response to 20 kHz (without too much high-frequency boosting), then the head gap should be less than half wavelength or about 1 micron. But in this case, recording efficiency would be low. Therefore,

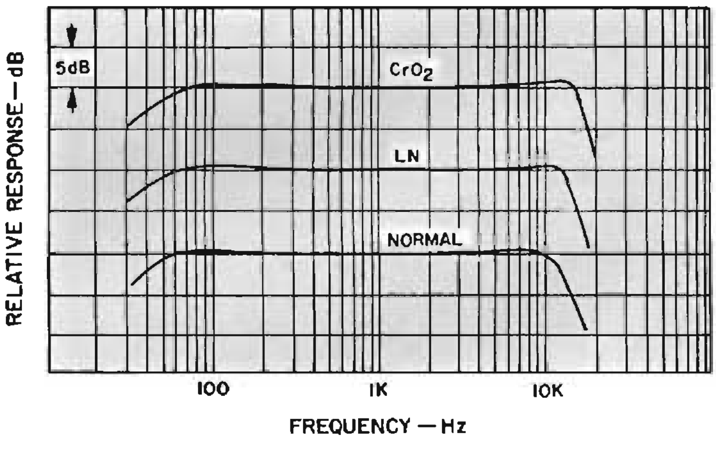


Fig. 2—Frequency response of a typical top-quality cassette deck with three different tapes at -20 VU.

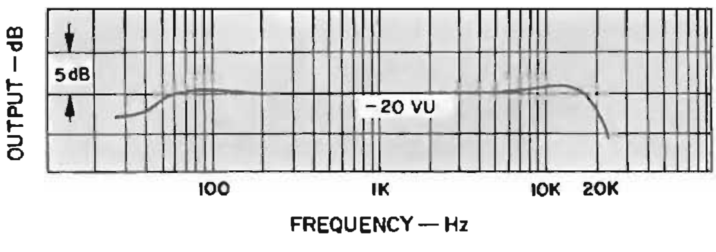


Fig. 3—Frequency response of a Nakamichi 1000 at -20 VU with CrO_2 tape.

machines using combined record-playback heads usually settle for a compromise gap between 2 and 4 microns.

Recently, several three-head recorders have become available, and they do have a rather better high frequency performance. Figure 3 shows the frequency response of a Nakamichi 1000, which uses a 5-micron recording head with a playback head having a remarkably small 0.07 micron gap. The response extends to over 20 kHz, and results are apparently only limited by the tape medium itself.

Frequency response is usually measured at low levels, either at -20 VU or sometimes at -30. The dashed line in Fig. 1 shows the frequency response of our open-reel recorder at 0 VU, and it will be seen that the response at high frequencies falls relative to the response at -20 VU (shown as a solid line). This phenomenon gets progressively worse as the tape speed decreases, and it is called tape saturation because all the magnetic particles have been affected and further signal increases can do nothing except erase what is there already! At 15 ips, the effect is quite small, but it becomes significant at 3¾ips, and, as you might expect, it is even more important with cassette decks working at 1⅞ ips. What does this mean in practice? Simply that care must be taken to keep that VU meter pointer well below the 0 VU mark when making recordings of music having large transient peaks or the sound will lack definition and brilliance. But, there is a snag; if the overall level is reduced by 10 VU as shown on the meter, then the overall signal-to-noise ratio will also suffer. Some cassette machines have peak limiters which do help, but special noise reduction systems, such as the Philips DNL, JVC's ANRS, and Dolby, provide the real answers.

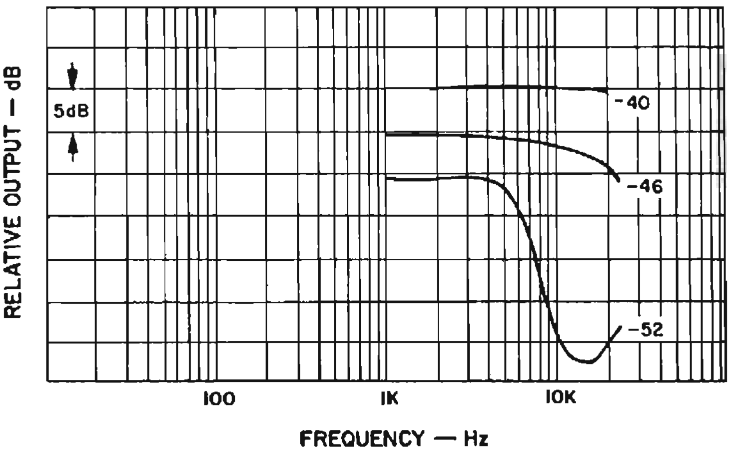


Fig. 4—Action of the DNL system. Note that it is inactive above levels of -40 dB.

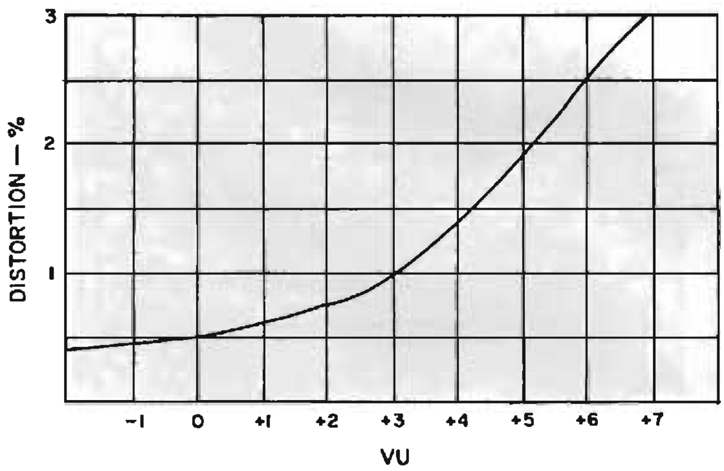


Fig. 5—Distortion versus recording level at 1 kHz for a typical open-reel machine costing about $350.

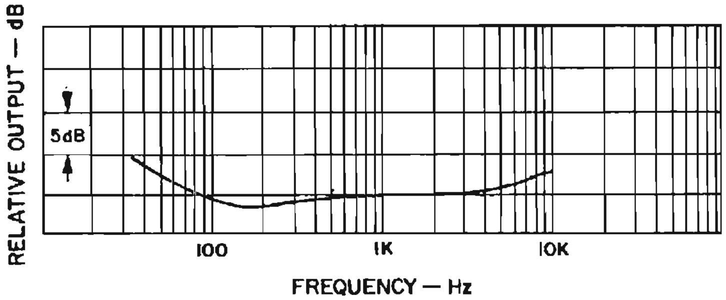


Fig. 6—Distortion versus frequency at 0 VU for the same typical open-reel deck.

How do we score the two sorts of tape recorders on these basic parameters? Well, top marks have to go to the very best open-reel machines, but we also must note that many of the first-line cassette recorders offer a quality of performance superior to the less-sophisticated open reelers. It's a case of paying your money and taking your choice.

Noise Reduction

Most readers are at least moderately familiar with noise-reduction systems, so that a long and involved technical explanation would be inappropriate here. In brief, however, the Dolby and ANRS systems work by increasing the signal level of frequencies above 500 Hz during recording, if they are low, and then reversing the procedure for playback, thus reducing the high frequency signal level to the amplitude of the original and reducing the noise by a like amount. There are two Dolby systems, A and B; the latter is the one used in domestic machines. The amount of high frequency lift is determined by both amplitude and frequency, and the circuitry is quite complex. Although the ANRS also operates above 500 Hz, it is not a dynamic, continously-controlled system, and so recordings made with the two are not directly compatible. The Philips DNL system is not a two-way type, as it functions on playback only to attenuate signals as shown in Fig. 4. Note that the high level signals are not affected and the maximum effect is in the region of 10 to 12 kHz.

A cassette deck using a Dolby system can show an increase of up to 10 dB in signal-to-noise ratio over a non-Dolby recorder, which means that recordings can be made with a lower level and you would still have a low background noise with less danger of tape saturation. A typical high quality cassette recorder thus would have a signal-to-noise ratio of 50 dB without Dolby and 60 dB with (weighted). On the other hand, an open-reel recorder would probably have another 6 to 8 dB at least, more if it also had a built-in Dolby system. But, of course, the cassette's 60 dB does represent a very good signal-to-noise ratio indeed, quite adequate for most purposes. The user, then, must decide for himself whether he has to have that few extra dB greater signal-to-noise ratio. Incidently, a S/N has to be related to a reference point, and unfortunately there is no fully-accepted standard. Some manufacturers use 0 VU, some prefer the 1 percent distortion point, while still others opt for 3 percent which makes the figures look better. Therefore, care must be taken when making comparisons as there might well be as much as 4 dB difference between the cassette machine figures and even more for open-reel machines. It might be thought that 0 VU seems pretty conclusive as a standard, but it isn't. What is 0 VU on one deck might be +2 VU on another, and further confusion is caused by some manufacturers who try to avoid tape saturation by tailoring the VU meter response so it reads higher above 5 kHz or so! All this is because volume units (VUs) only measure relative levels and not absolute levels.

Distortion

Figure 5 shows the distortion (THD) with a typical open-reel recorder, and it will be seen that the 3 percent level is not reached until the signal gets to a level of +7 VU, while at 0 VU the distortion is about 0.5 percent. In terms of distortion vs. frequency, there would be a slight rise at each end of the frequency range (see Fig. 6). The unfortunate cassette deck designer does not have the same leeway to work with, and many cassette decks have a distortion characteristic like that shown in Figs. 7 and 8. In other words, the distortion is higher and the headroom lower, once again emphasizing the importance of a noise reduction system in effectively giving a wider dynamic range. So, in terms of frequency response, distortion, and signal-to-noise ratio, an open-reel recorder working at 3¾ ips can have a 3 to 4 dB better signal-to-noise ratio, an extra 2 or 3 kHz in frequency range, and a bit lower distortion than an average cassette deck fitted with a Dolby system. At 7½ ips, the disparity is greater, and if the open-reel machine also has a built-in Dolby system, then there is scarcely any contest. But we are here talking about a machine that would probably cost at least half again as much as the cassette deck!

In actual practice, though, a cassette machine can make tapes from discs or FM that sound identical in an A-B comparison to tapes made by an open-reel recorder using 3¾ ips or even 7½ ips, but greater care is necessary when using the cassette deck. When it comes to making direct recordings with top-quality microphones, the open-reel recorder will win, especially if the 15 ips speed is used and an ultra-wide dynamic range required.

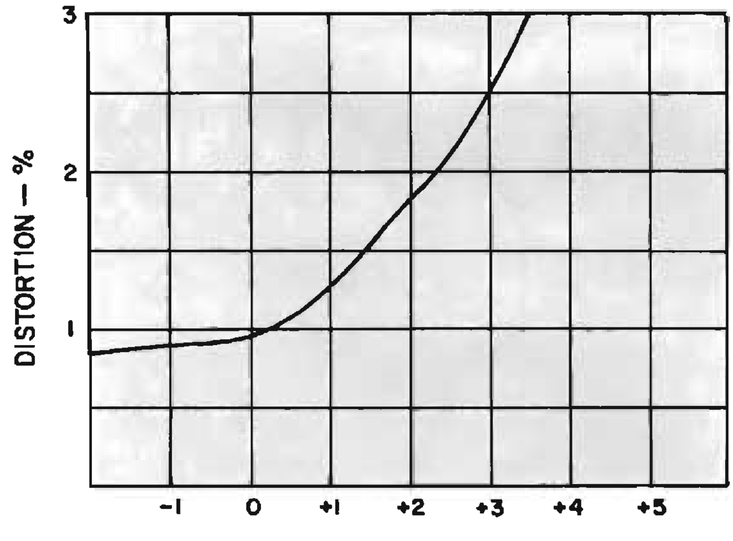


Fig. 7—Distortion versus recording level at 1 kHz for a typical cassete deck.

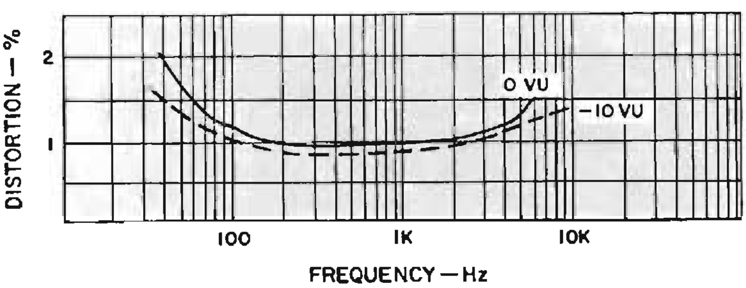


Fig. 8— Distortion versus frequency at two recording levels for a typical cassette deck.

Wow and Flutter

Wow can be defined as a slow variation in speed below 5 or 10 Hz, while flutter is a faster variation. Measurements are now usually made using the DIN or IEEE standard, which give a combined and weighted figure. Wow is frequently caused by capstan shaft eccentricity or by an unstable drive-tension system, and flutter can be produced by erratic tape motion as it passes over heads and guides or between various constricting portions of the cassette. Poorly designed or produced motors can also cause speed variations, but these days both open-reel and cassette decks use hysteresis-synchronous motors which are frequency controlled from the power supply. Both use servo-controlled motors in the higher-priced models, with the speed electronically controlled by feedback circuits which automatically correct for any fluctuations. The big difference between the recorders, cassette and open reel, is the way the tape is held steady so it passes smoothly over the heads. Some open-reel recorders use two motors, others have three, though at least three European recorders use a single heavy-duty motor with a tension clutch arrangement. All use a kind of back torque system to keep the tape under tension. Few cassette decks use anything but the simple pressure pads of the cassette to maintain smooth head-to-tape contact. However, if properly designed, this relatively simple method can work very nicely at the low speed involved. The cassettes themselves often present problems, since unlike tape reels where both tape and reel turn together, the cassette tape moves but the plastic case does not. While the various sorts of liner sheets, steel pin rollers, etc. do help with the friction, it and other variables are still not fully under the control of the deck designer.

Not only will an unsteady tape movement cause annoying flutter, but worn pressure pads or dirt on the heads can produce a poor contact with the tape resulting in a loss of high frequencies. It can also cause modulation noise—a kind of IM distortion. Further, it is not generally realized just how serious this sort of spacing loss can be at the slow speeds. A speck of dust only one-eighth of a mil will produce an attenuation of 54.5 dB or about 99 percent. Furthermore, the loss is compounded when the same head is used for recording. In practice, it is unlikely that the spacing will be that large, but it doesn't take much oxide build-up to produce a loss of 6 dB. The inference is obvious; cassette recorders need a lot more care, TLC, to get the best results, and the heads must be kept clean, *really clean.* And for truly top-grade recordings, only the best cassettes should be fed to the recorder. The cassettes should also be stored in their protective plastic boxes, just as with records, to prevent the accumulation of dust either on the tape itself or in the sprockets and hubs.

A look at the measured wow and flutter figures for open-reel recorders shows results from 0.04 to 0.09 percent, while cassettes run (no pun intended) between 0.04 and 0.17 percent, depending on price. It should be noted that these figures for cassettes are a good more variable than they are for open-reel machines. Even figures for the very same test cassette and recorder combination will vary from day to day. The main point to remember, however, is that the performance by cassette machines in this area make them worthy of serious consideration by the enthusiast.

Ease of Operation

Now here is where the cassette deck really scores! Anyone—even my Aunt Agatha—can load a cassette into a machine without getting into a tangled mess! And the decks are smaller and more portable. Some, such as the Yamaha TC-800GL, the Nakamichi 550 or Sony/Superscope 152 SD, will work on batteries too, and most have easy-to-use push-

(Continued on page 50)

(Continued from page 42)
button controls. The more recent front-loading models, such as the Pioneer CT-7171, Sony/Superscope TC-177 SD, and Technics RS-676 US, are not only simple to operate but they provide facilities previously found only on open-reel decks. For example, the Technics recorder has a memory circuit that can get the tape rewound back to a predetermined point before switching the machine back into the play mode. It also has a Dolby 25 and standard 75 μS deemphasis network switch for recording from tuners with either time constant, a peak or normal VU reading switch, and provision for CrO_2 low noise, and ferric-chrome tapes. The bar switches have built-in indicator lights, a useful refinement for the non technical user.

Monitoring

In my opinion, this is an essential feature for the truly serious recording enthusiast, and most open-reel machines over the $300 mark do have the three heads necessary. Among the cassette decks with this facility are the Nakamichi 1000 and 700, AKAI GXC-352D, Sony/Superscope TC-177 SD, and the Technics RS-279 US. Curiously, the Technics model uses a separate head for monitoring, while retaining a combined record-replay head. A word of warning; because a deck has three heads, it doesn't always mean that one is available for monitoring. For example, the third head in the Toshiba PT-490 is a second erase head for the reverse tape direction.

Editing, S-O-S, etc.

I stated earlier that it is difficult to edit cassette tapes, and so it is. While it can be done, the process does require a great deal of patience. The 3M Company makes a repair kit, and Editall, among others, makes the right size blocks and tabs for splicing. None of the special effects like sound-on-sound, multitrack sync, and echo are possible with cassette decks, although certain effects are possible with the aid of external units.

The Future

We will undoubtedly see more cassette decks with monitoring facilities. In fact, several were introduced at the recent Consumer Electronics Show in Chicago. Many of these had provision for ferri-chrome tapes and servo motor control. Automatic bias and equalization adjustment for CrO_2 tapes, actuated through sensing an indentation on the cassette, is now standard practice. More and more sophisticated cassette decks will appear with such features as variable speed control, but the biggest advance will be in the tape itself.

The originator of the cassette medium, Philips, recently introduced a super fine-grain iron oxide formulation which they say gives an improvement of 10 to 12 dB in the signal-to-noise ratio. Both TDK and Maxell have even more recently introduced cassette tapes making use of cobalt for improved performance. TDK's is called Super Avilyn (SA) and uses a cobalt ion added to an extremely fine ferric oxide particle by absorbtion. Maxell's UDXL uses cobalt ferrite epitaxially grown on extremely fine ferric oxide particles. Bias and equalization characteristics of these two tapes are similar to CrO_2, so we will be spared a proliferation of bias/eq switches. However, I don't see cassette machines challenging open-reel units very much more seriously than they do now because both benefit from technological advances.

So, then, back to our opening question; how do the cassette recorders of today stack up? And the answer, obviously at this point, is very, very well!

Understanding The NAB EQ Standard

Herman Burstein

THE CURRENT National Association of Broadcasters (NAB) Standard for Magnetic Tape Recording and Reproducing (Reel-to-Reel) appeared in April 1965. Although more than 10 years have passed, the nature of standard tape equalization tends to remain obscure and imperfectly understood. I gather this both from audiophiles questions and statements that are made in some of the popular periodicals devoted to audio.

Such misunderstanding is partly due to the complexity of the subject, which entails the velocity characteristic of the playback head; gap loss, electrical losses, resonance effect, and contour effect of the playback head; electrical losses of the record head; magnetic losses of the tape; surface induction of the tape; magnetic flux entering the core of the playback head; maximization of signal-to-noise ratio; minimization of distortion, and achievement of flat record-playback response.

Misunderstanding is also due to the indirect and piecemeal manner in which the equalization standards are presented. The playback curve in the 1965 Standard is visually much different than the playback curve ordinarily shown in the popular audio periodicals, although one is translatable into the other. The 1965 Standard shows a playback curve with *apparent* treble boost and bass cut, whereas the curve popularly shown has treble cut and bass boost. To put the entire NAB equalization standard together, one has to hunt through various sections of the 1965 Standard, *including* footnotes. One finds only hints, rather than a straightforward statement, that record-playback response should be flat (within certain tolerances).

Presumably an understanding of standard tape equalization is desirable to the aduiophile, perhaps as knowledge in itself and perhaps to assist him in the use, modification, and even construction of tape equipment. To provide this understanding, we shall assume operation at a standard speed of 7½ ips. In principle, what is said about 7½ ips applies to other speeds, except for differences in amount of frequency losses and therefore in equalization required. We shall also assume operation at "normal bias" for a given recording tape—the tape recommended by the tape deck manufacturer and/or chosen by the user. Normal bias is approximately that which minimizes distortion in recording.

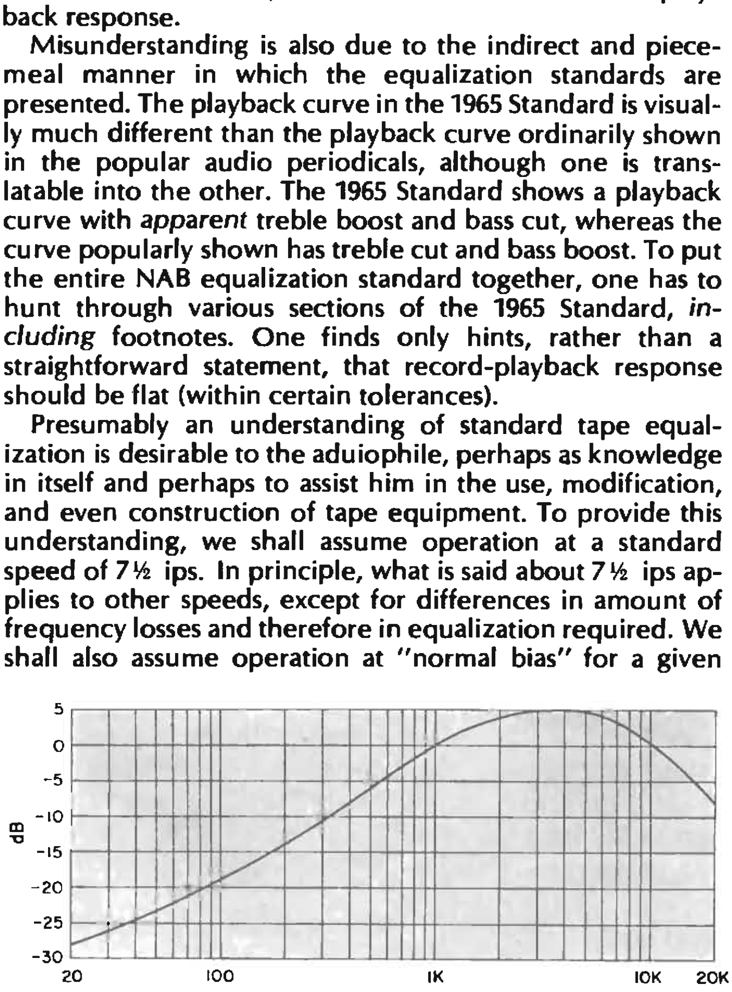


Fig. 1—Typical unequalized record-playback response of a tape deck at 7½ ips.

Unequalized Record-Playback Response

First we require a clear understanding why equalization is needed. Therefore, Fig. 1 shows the typical *unequalized* record-playback response of a high-quality tape deck at 7½ ips, using good tape, with normal bias for this tape. Although input to the deck is flat—constant amplitude for all frequencies in the audio range—output is anything but flat. Output rises steadily until about 4,000 Hz and soon after drops quite abruptly.

Ideally, record-playback response should be flat (or nearly so), represented by a horizontal line throughout the audio range. Clearly, bass boost and treble boost are needed to restore flat response.

Departure of an unequalized tape system from flat response is *largely* explained by two factors: (1) rise in output of the playback head as frequency increases; (2) serious treble losses on the tape owing to magnetic phenomena.

These and other factors are examined in the next two sections, which respectively deal with playback "losses" and record "losses." We put the term losses in quotes to draw attention to the fact that, like the NAB Standard, we will use it in both a positive and negative sense: Losses include gains as well as declines in frequency response. In other words, losses designate both upward and downward deviations in response.

Playback Losses

In playback, the chief deviation from flat response is due to the playback head being a velocity device. As such, it produces steadily rising output voltage in response to a flat input signal (constant magnetic flux in the core of the head). This rise, illustrated by the solid line in Fig. 2, amounts to 60 dB over the audio range of 20 to 20,000 Hz.

A velocity device is one whose output voltage is proportional to the number of changes per second in a magnetic field. For a playback head, the changing magnetic field is due to the signal recorded on the tape. In short, head output is proportional to signal frequency (assuming all frequencies are recorded so as to produce equal flux in the core of the head). Thus, if frequency doubles, voltage output of the head doubles. One octave represents a doubling of frequency, while 6 dB represents a doubling of voltage. Therefore, as in Fig. 2, playback head output tends to steadily rise 6 dB per octave. (More accurately, output rises 6.0206 dB per octave, or exactly 20 dB per decade, on 10-fold rise in frequency such as from 100 to 1,000 Hz.)

A steady 6-dB-per-octave rise characterizes an *ideal* playback head—one with no deviations from Fig. 2. However, a *practical* head displays some irregularities. For a high-quality head operating at 7½ ips, the irregularities are *quite minor* and are as follows.

1. *Treble Loss Due To Gap Width.* Modern playback heads have gaps as narrow as 4 microns (.000160 in.) or less, sometimes approaching 1 micron (.000040 in.). A useful formula for approximating playback head response before gap loss becomes appreciable is $f = 0.85S \div 2G$, where f is frequency in Hz, S is tape speed in ips, and G is gap width in inches. Substituting 7.5 for S and 0.000160 (4 microns) for G, we find that f is about 20,000 Hz. If the gap is appreciably narrower than .000160 in., significant treble loss owing to the gap does not occur until well above 20,000 Hz. Therefore, in the case of a high-quality playback head operating at 7½ ips, gap width accounts for negligible deviation from Fig. 2. (The horizontal dimension of the gap is usually called gap *width* in the popular literature and gap *length* in the technical literature, such as the NAB Standard.)

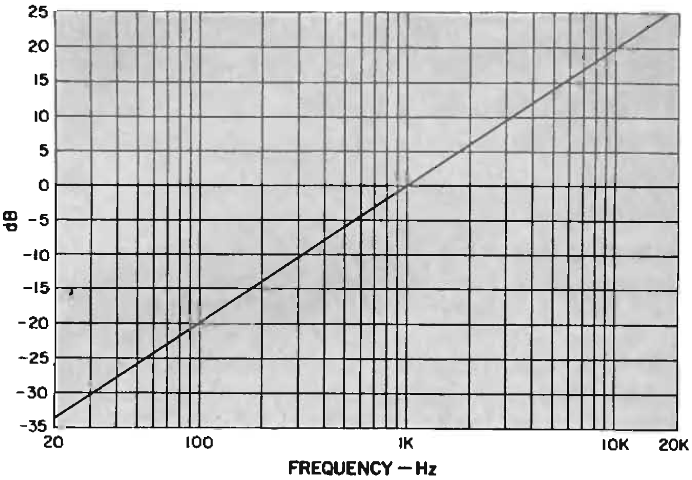


Fig. 2—Response of an ideal playback head with constant magnetic flux in its core.

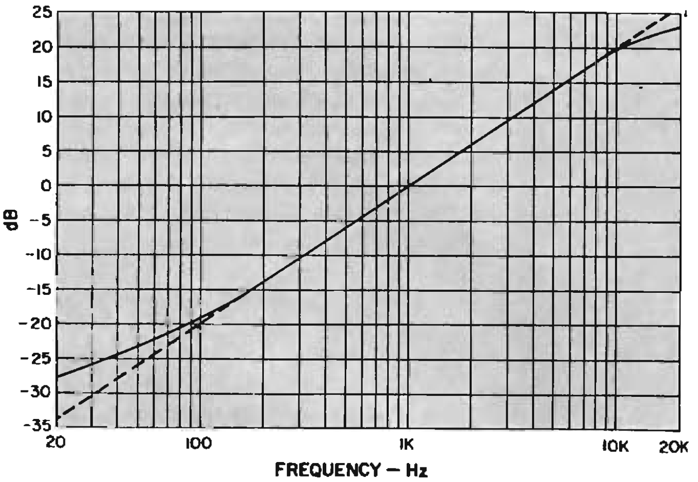


Fig. 3—Typical response of a practical playback head with constant magnetic flux in its core.

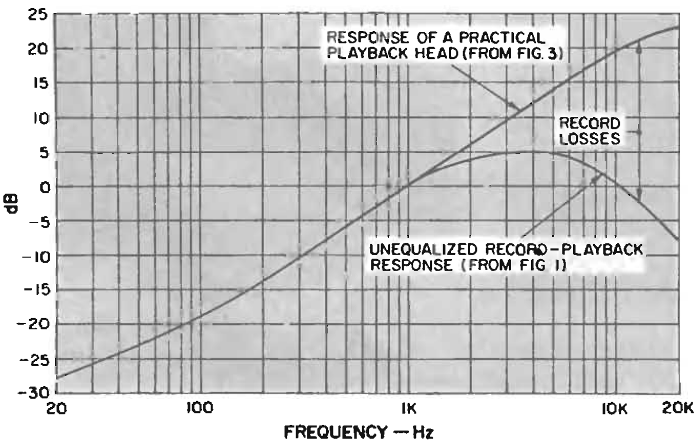


Fig. 4—Typical record losses at 7½ ips.

2. *Electrical Treble Losses.* These are largely due to hysteresis and eddy currents in the core of the playback head, and to winding capacitance of the head. Partially offsetting these is a resonance effect due to the head capacitance in series with load capacitance. For a well-made playback head in a well-designed circuit, the net electrical loss tends to be very little within the audio range, perhaps in the vicinity of 1 to 3 dB at 20,000 Hz.

3. *Bass Rise Due To Contour Effect.* As frequency declines, wavelength of the signal recorded on the tape grows longer. (Wave length is tape speed divided by signal frequency, or inches of tape per audio cycle.) As recorded wavelength increases, the entire playback head (not only the gap) reacts to the tape's magnetic field, augmenting response. Thus, the output of the playback head tends to rise in the bass region relative to the theoretical, or ideal, response slope of 6 dB per octave. The nature and extent of this relative rise in bass depends in part upon the angle at which the tape approaches and leaves the head. Sometimes the effect of this angle is separately identified as the "wrap effect." However, we can conveniently combine all these phenomena under the single term "contour effect." For a well-designed playback head in a well-designed transport, the contour effect tends to be moderate—resulting in something like 3 dB relative bass boost at 50 Hz.

Altogether, for a flat signal input, the irregularities in output of a high-quality playback head cause it to deviate very little from the ideal response of Fig. 2. In other words, for constant magnetic flux in the core of the head, output would typically be about that of Fig. 3. Here the response rises 6 dB per octave through most of the audio range, but at a slightly slower rate in the low bass and in the high treble owing to the contour effect, gap loss, and electrical losses.

Record Losses

Figure 4 compares the response of a practical playback head with unequalized record-playback response. The difference between the two curves is due to losses in recording. Playback head response assumes constant magnetic flux in the core of the head owing to a flat signal on the tape. Unequalized record-playback response reflects the actual signal on the tape—one that embodies vast treble losses.

Depending upon the kind of tape employed and the amount of bias, the treble losses in recording may vary somewhat from those shown in Fig. 4. Typically, however, they total around 30 dB at 20,000 Hz at 7½ ips. (They are appreciably smaller at 15 ips and appreciably larger at speeds below 7½ ips.) In order of importance, following are the factors that cause record losses.

1. *Self-Demagnetization.* Frequencies recorded on tape are in effect a series of bar magnets oriented lengthwise on the tape; each bar magnet corresponds to a half-cycle. The opposing north and south poles of a bar magnet tend to cancel as the bar becomes shorter. With increasing frequency, the recorded wavelength (tape speed divided by frequency) becomes shorter, thus the bar magnets become

shorter. Hence, with increasing frequency, there is greater tendency for the poles of each magnet to cancel. This is called self-demagnetization. The upshot is that, for a constant magnetic field applied to the tape at all frequencies, the strength of the magnetic field recorded on the tape declines as frequency rises.

2. *Bias Erase.* In recording, a high-frequency current (in the range of 75,000 to 200,000 Hz) is applied to the tape in order to minimize distortion and maximize the amount of signal recorded on the tape (thereby maximizing signal-to-noise ratio). Bias is fed to the record head along with the audio signal, or it is separately fed to a cross-field head mounted opposite the record head on the other side of the tape. Unfortunately, bias current has an erasing effect, which increases with frequency. Tape magnetization does not penetrate as deeply into the magnetic oxide at high frequencies as it does at low ones. Therefore, the higher frequencies are more vulnerable to erasure. (Note that bias current is also fed to the erase head, but in much larger amount, in order to totally erase the tape.) In sum, bias current produces treble loss.

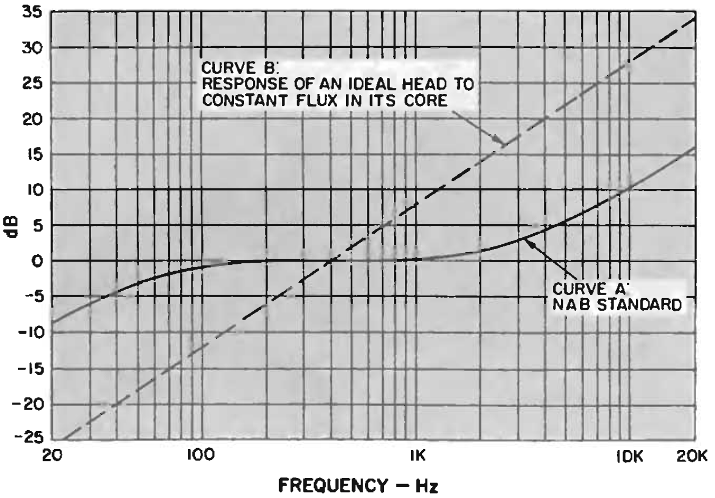


Fig. 5—Actual NAB standard playback equalization at 7½ips (playback amplifier output for constant flux in the core of an ideal playback head).

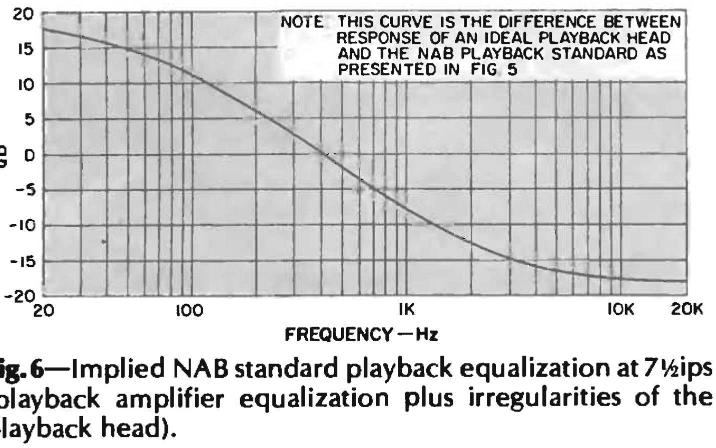


Fig. 6—Implied NAB standard playback equalization at 7½ips (playback amplifier equalization plus irregularities of the playback head).

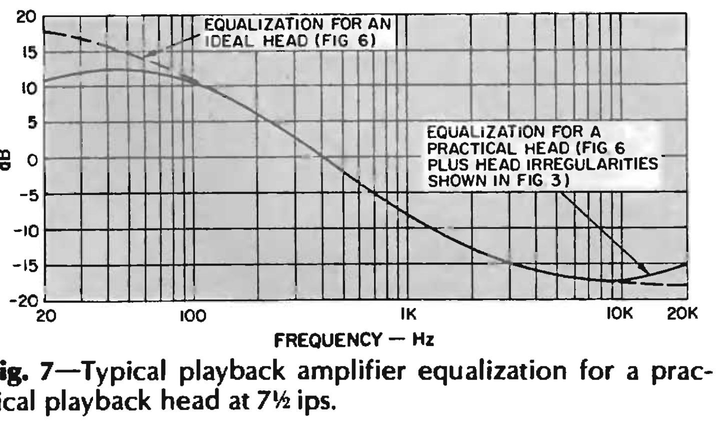


Fig. 7—Typical playback amplifier equalization for a practical playback head at 7½ ips.

3. *Electrical losses.* As in the case of the playback head, there are minor treble losses due to hysteresis and eddy currents. Ordinarily there is little or no loss due to winding capacitance, and no resonance or contour effect. And there is no loss due to gap width; recording takes place at the trailing edge of the gap (the edge last encountered by any given point on the moving tape), and a relatively wide gap is employed for optimum results.

All told, at 7½ips, self-demagnetization loss might typically amount to about 16 dB at 20,000 Hz, bias erase to about 12 dB, and electrical losses to about 1 or 2 dB.

NAP Playback Equalization

Figure 5 is NAB's way of showing standard playback equalization; this corresponds to Curve A in Fig. 5 of the NAB standard. It assumes an ideal playback head and flat recorded signal on the tape (more exactly, constant magnetic flux in the core of the head for all recorded frequencies). Under these conditions, Curve A shows the response of the playback head *in conjunction with the playback amplifier.* Curve A may be viewed as the standard response of an ideal reproducing system (head plus amplifier).

However, we know that the ideal head *alone* produces a voltage output that climbs 6 dB per octave, namely Curve B in Fig. 5. Therefore, the NAB playback characteristic (Curve A) implies that, *for an ideal head,* the playback amplifier must supply equalization corresponding to the difference between Curves A and B. By plotting the difference between these curves, we obtain the playback equalization curve of Fig. 6. The playback equalization curve commonly shown in the popular audio literature is that of Fig. 6.

Playback equalization at 7½ ips is thus seen to consist of a very substantial amount of bass boost, commencing (3 dB up) at 3,180 Hz and ending (3 dB below maximum) at 50 Hz. Total bass boost is 36 dB. By and large, it offsets the bass decline of unequalized record-playback response, shown in Fig. 1.

Until now we have assumed an ideal playback head, however, as shown in Fig. 3, there is a difference—usually modest—between the response of an ideal head and that of a practical one. Therefore, a further requirement of the NAB Standard is that the playback amplifier provide equalization to compensate for this difference. Footnote 24 of the NAB Standard states, "...it is...necessary to modify the amplifier response to compensate for practical reproduce head losses...."

Accordingly, the playback amplifier must modify the curve of Fig. 6 to provide slightly more treble and slightly less bass, thus compensating for the irregularities of a practical head. Figure 7 shows the typical modified equalization of a playback amplifier.

At the risk of repetition, it must be noted that it is erroneous to regard Fig. 6, the implied NAB standard for playback equalization, as consisting solely of playback amplifier equalization. *Figure 6 is the sum of amplifier equalization plus irregularities in response of the playback head.* To the extent that head irregularities are pronouned, the error in question becomes serious.

One may wonder why NAB chose to present standard playback equalization in the oblique manner of Fig. 5, rather than in the more direct fashion of Fig. 6. According to one person who helped formulate the NAB Standard, this was done to avoid the error just cited—mistaking Fig. 6 as applying solely to playback amplifier equalization.

NAB Record Equalization

Returning to Fig. 5, let us carefully note what Curve A tells us: If the magnetic flux in the core of an ideal playback head were constant at all frequencies, the output of the playback amplifier would show a moderate drop in bass and a pronounced rise in treble.

However, we naturally expect output to be flat over the audio range. (Curiously, the NAB Standard does not directly say that record-playback response after equalization shall be flat. The closest it comes to suggesting this is in Section 4.03.03, which deals with Standard Test Tapes. Here we find: "The signal frequencies are recorded on these tapes in such a manner that they would supply a constant output level when reproduced on an Ideal Reproducing System." From this, we infer that record-playback response for program material shall be flat.)

To offset the drop in bass and rise in treble of Curve A, there must be a complementary frequency characteristic in recording. Specifically, the recording characteristic and therefore the magnetic flux in the core of the playback head must conform to the difference between Curve A and flat response. This difference is plotted as Fig. 8. It shows that the magnetic flux must exhibit a bass rise and a treble drop—the complement of Curve A in Fig. 5. Together, the playback characteristic of Curve A and the recording characteristic of Fig. 6 produce flat response.

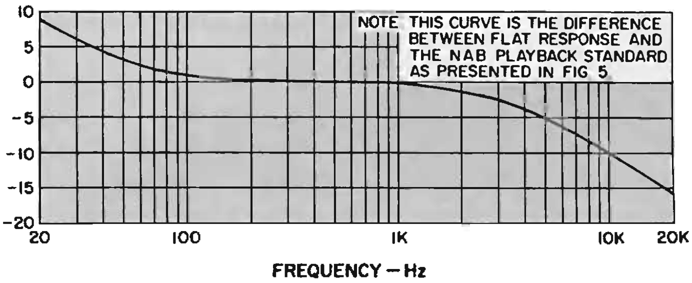


Fig. 8—Implied NAB standard recorded induction characteristic (approximately equivalent to magnetic flux to be developed in the core of the playback head).

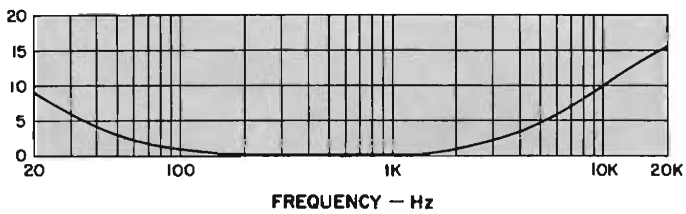


Fig. 9—Typical record amplifier equalization to achieve NAB standard recorded induction as in Fig. 8.

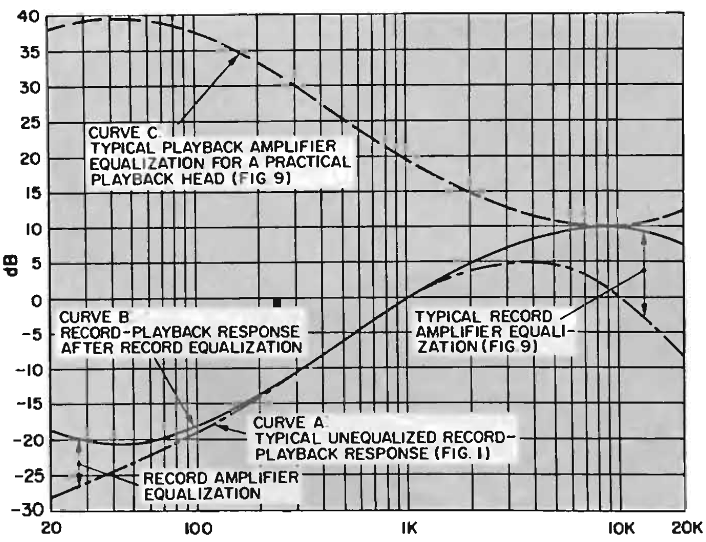


Fig. 10—Typical amplifier equalization at 7½ ips in accordance with the NAB standard.

The frequency characteristic of the signal recorded on the tape is essentially that of Fig. 8. We may refer to the recorded signal as recorded induction. (More accurately, we should use the term surface induction, which is the density of the magnetic flux at right angle to the surface of the tape.) The recorded induction produces magnetic flux in the core of the playback head. The amount of magnetic flux in the core does not correspond exactly to the recorded induction, which is altered by the presence of the head. However, for a head with a gap narrow enough to avoid treble loss, the alteration tends to be approximately the same over the audio range. Therefore, the frequency characteristic of recorded induction is virtually that of Fig. 8.

In sum, we may view Fig. 8 as the implied NAB standard curve for recorded induction.

The treble decline in recorded induction is chiefly the result of the record losses previously described (due to self-demagnetization, bias erase, and electrical effects). However, comparison of Fig. 4 with Fig. 8 shows that these losses exceed the required treble decline. To compensate, treble boost is needed in the record amplifier. The appropriate amount of treble boost is that which, in conjunction with record losses, will produce the standard recorded induction curve of Fig. 8. Figure 9 show typical treble boost provided by the record amplifier at 7½ ips.

The bass rise in recorded induction is achieved by equalization in the record amplifier, also indicated in Fig. 9.

A Total View Of Amplifier Equalization

Figure 10 presents a total view of record and playback equalization typically provided by the tape deck amplifiers in accordance with the NAB Standard at 7½ ips. Curve A repeats the typical unequalized response of Fig. 1 for record-playback. Curve B shows the typical response after record equalization (but not playback equalization) has been provided. The distance between Curves A and B represents treble and bass boost typically supplied by the record amplifier to meet the NAB Standard. To achieve flat response, the playback amplifier must supply a curve complementary to Curve B. Therefore, the playback amplifier typically supplies equalization corresponding to Curve C. Together, Curves B and C achieve flat response on a record-playback basis.

Checking Conformance To Standard Equalization

It would be a very sophisticated and difficult procedure to *directly* check a tape deck's conformance to NAB equalization standards. Fortunately, the task can be vastly simplified by means of a standard test tape with recorded induction conforming to Fig. 8. At the time this article was written, NAB had not yet issued its own standard test tape, as visualized in the 1965 Standard. However, other standard test tapes, such as those made by Ampex, are available.

The standard test tape contains a series of tones with a frequency characteristic essentially that of Fig. 8. In other words, the recorded induction of the tape is close to the curve of Fig. 8, and therefore the magnetic flux produced in the core of the playback head is that of Fig. 8.

To check playback equalization, it is simply necessary to play the test tape and measure output of the tape deck with a suitable meter. The output should be virtually flat (within ± 1 dB between 100 and 10,000 Hz at 7½ ips; down no more than 3 dB at 30 Hz and 15,000 Hz). To the extent that output deviates from flat, the tape deck's equalization deviates from standard. Such deviations may be eliminated by appropriate changes in the equalization of the playback amplifier. Some tape decks contain controls for this purpose.

To check record equalization requires a signal generator and a recording tape in addition to the test tape and meter. The recording tape should be compatible with the tape deck under test; that is, the deck should be able to provide "normal" bias for this tape. The procedure is as follows.

(1) Measure and graph the output of the deck when playing the test tape. (2) Using the recording tape, record at constant input level the same frequencies as on the test tape. (3) Play the recorded tape; measure and graph its output. (4) If the deck produces standard recorded induction, the graphed output of the recorded tape will be the same as that of the test tape. Any difference between the graphed outputs represents the deck's deviation from standard record induction (for the particular tape used in recording).

At 7½ ips, the NAB Standard permits ± 1 dB deviation from standard recorded induction between 30 and 10,000 Hz; and no more than −2 dB deviation at 15,000 Hz.

Deviation from standard recorded induction may be corrected by appropriate changes in the equalization of the record amplifier. *The nature of such changes depends, of course, on the tape used for recording.* Footnote 9 of the NAB Standard states: "The recording equalization of a recorder/reproducer should be adjusted for an overall response which matches as nearly as possible the response of the reproducer from the NAB Standard Test Tape."

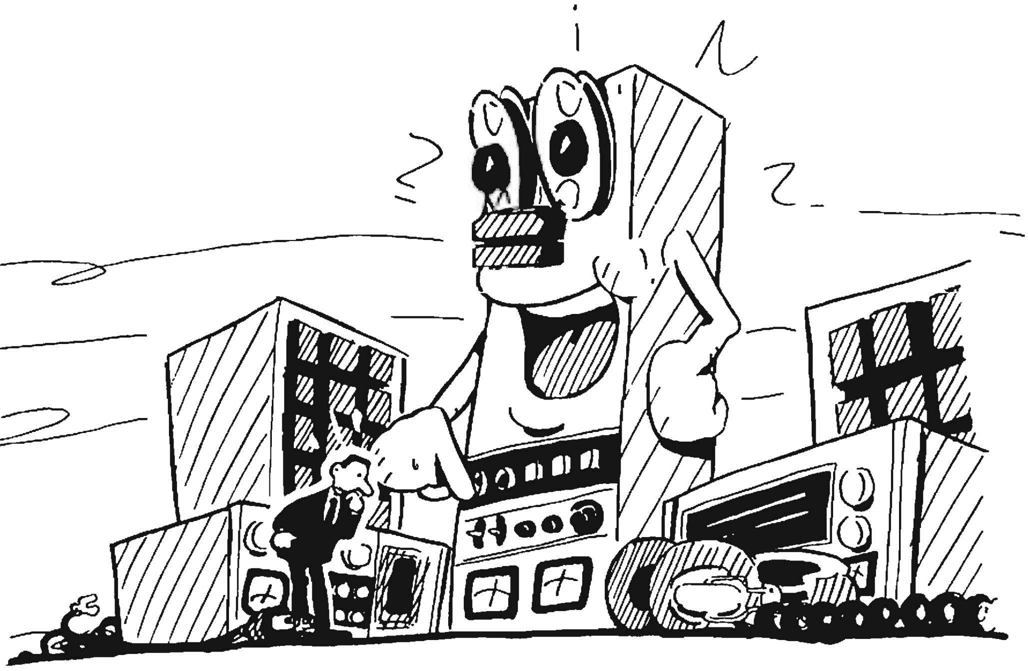
That is to say, record equalization should *not* generally be adjusted for flat record-playback response. It should be adjusted so that playback of a recording tape has the same frequency response as playback of a standard test tape. If playback of a standard test tape yields flat response, only then should record equalization be adjusted for flat record-playback response.

Location of Equalization Circuits

Altogether, as suggested by Fig. 1, the tape amplifiers must provide a substantial amount of bass boost and a fair amount of treble boost. Our discussion has shown that bass boost takes place largely in playback (Fig. 7), and treble boost largely in recording (Fig. 9). (A minor amount of bass boost occurs in recording to achieve the bass rise in recorded induction shown in Fig. 8, and a minor amount of treble boost may occur in playback to offset losses of the playback head.)

Why is bass boost confined mainly to playback, and treble boost mainly to recording? (In effect, we are asking why the NAB equalization standards are as they are.) Other combinations of equalization than those shown in Figs. 7 and 8 could also restore flat response. The answer has to do with minimizing distortion and maximizing signal-to-noise ratio. If much bass boost were employed in recording, this would tend to overload the tape, causing distortion. If much treble boost were used in playback, this would accentuate playback noise, which is dominant in the treble range. Inasmuch as most audio sources have less intensity at high frequencies than at mid-frequencies, it is possible to use moderate amounts of treble boost in recording without overloading the tape.

Reading VU Meters



C. E. Moule

The VU meter was introduced in 1939 as a standard meter to avoid the confusion that existed in the broadcasting industry at that time. Audio level meters up 'til then had a bewildering array of reference levels, various speeds of movement, slow, medium or fast, and some people even chose to ignore the fact that they were dealing with a non-symmetrical wave shape. The meters still do vary on cheap equipment where they were meant to solve these problems. However, they can do very well, on the technical specifications[1,2]. But now there seems to be more confusion, because of the different ways of reading the meter and also expressing what is read. The problem has recently become more acute, now that the use of the meter is not confined to broadcasting, but has found ready acceptance in sound recording, both for commercial and private uses. Even a novice to the hi-fi market looks for a VU meter on each channel of his new equipment.

But why have a VU meter? Perhaps its need should be justified before detailing the reasons for confused reading. The use of one piece of equipment confined to one person would seem to be the occasion for least use of a level indicator—for the low level passages, listening to a speaker will let you know when the level is getting too low and into noise, and the speaker can still be used to get a rough assessment of "when it's too loud, it distorts." Even so, with the best of equipment, where no compression of volume is required, a starting point for setting the volume control is difficult without the aid of a level meter. Then the meter has a further use, to enable a repeat of the same maximum volume level which has been found to be satisfactory before—the hearing of a speaker won't be accurate enough after only an hour or so. The advantages are more obvious when a change of personnel is involved, or when there is a change of tapes or programs.

A Standard Meter

Not the least of the claims for the use of a VU meter, compared with other level indicators, is that its dynamic characteristics are rigidly specified and controlled, such that the speed of movement happens to correspond very closely with the effect of music and speech on the ear, i.e. the loudness effect. Combined with the background color on the scale, the meter is easy to read and yet easy on the eyes. The same cannot be said for some "peak" reading meters; they need extra equipment to drive them, they look odd with their wild upswing and slow return and still never quite make the peaks.

Let's have the "loudness" type meter as long as provision is made for the peaks that are there above the VU reading. Most VU meter users allow 8 or 10 dB above the +4 VU or +8VU, or whatever standard they use, but peaks of 14 dB above the meter reading have been observed on unusual program material. Also it is rare to find an operator that can aim and maintain the pointer deflection just up to the "0" mark most of the time, without getting, say, 2dB over. If 2dB is allowed for the human error and 10 dB for the peaks, then 12 dB seems to be the appropriate allowance to make for overload and distortion tests, e.g. +20 dBm for a +8 VU circuit. This could be the first of the confused ways of reading a VU meter. How many tape recorders are tested for distortion, with a continuous tone only deflecting the meter to the "0" mark? The VU meter will stand 14 dB above this mark continuously without any ill effect, so should the rest of the equipment. Why not specify and check it at the right level (not 5 dB above,[3])?

Even if the VU meter had no other advantages, it would still be worth using, just because it is standard, and so valid comparisons can be made between various pieces of equipment and between different locations in a network.

Origin

It was the network folk who probably helped to conceive the idea of a standard meter with rigid specifications[4], but the advice on how to read was then slanted towards network operators, with observations being made at a transmitter, repeater or a switching center. This was good at the time because each location was staffed and was presumed to have a complete VU meter, i.e. a meter with an attenuator in 2 dB steps, with the recommendation that the attenuator be adjusted until the majority of peaks reached within + or − 1 dB of the "0" mark on the meter. Then the measurement was expressed as the sum of the meter reading and the attenuator—easy, if the meter deflects to the "0" mark, just read off the attenuator setting and there it is, so many VU.

But the position has changed, there are fewer transmitter, repeater and switching locations that are staffed, and at the places where programs originate, and certainly on most tape recorders, there is no attenuator visible, and no marks on the meter to say +4, +8, +10, or whatever standard is used. Furthermore, the instructions as to where the needle should deflect are either too brief, such as "up to zero," or too ponderous, "the reading is determined by the greatest deflections occurring in a period of about a minute for program waves, or a shorter period (e.g. 5 to 10 seconds) for message telephone speech waves, excluding not more than one or two occasional deflections of unusual amplitude."[5] There appears to be 4 different interpretations given to these instructions:

1. Keep the needle above the "0" mark, most of the time.
2. Never let the needle get above the "0" mark.
3. Look for some mythical average of the deflections and adjust until this average is at the "0" mark.
4. There must be a correct way. "In average program material (speech and light music), peaks occurring at the rate of 6 per minute are regarded as the most significant, and should deflect to the zero mark, allowing for an occasional overswing; classical music will need to be observed over a longer period."

Perhaps this last interpretation is more significant when it is seen how some people have twisted it around, and said that you should see a peak every 10 seconds. This is quite wrong, because music and speech are not performed with such regularity; it would be terribly monotonous if it were so. But it does raise the importance of elevating the monitoring speakers or phones to equal status with the VU meter—how can you make a valid observation without ears to tell you what type of program material is present? Even without a knowledge of music, most people can tell from the way the music is being played whether it should be loud or soft.

Using a VU Meter

The instructions should be simplified further and included with each piece of equipment using a VU meter. The advice to novices, just starting with their first tape or cassette recorder could be: get some prerecorded material, either news type speech or light music with a rhythm, i.e. something with only small variations in volume and, while it is playing, adjust the volume control until it is easily seen that the majority of peak readings come to the same point on the scale, adjust the volume control again until these peaks reach the 100% mark, then you have correct level. A little more time is needed to get correct level on classical music or unusual program material, but what to call it?

There is neither the time or the need to call it anything in some situations[6]. If the concern is only with your own tapes, then it needs only to be called "correct level," and the figures on the meter are superfluous. The manufacturers of the equipment may tell you in the instruction book that "0" on the meter is +4 VU, +8 VU, or whatever they choose, but they have also allowed a margin of 8 dB to 14 dB above the VU meter reading for peaks. The "medium-fi" market is probably best served by what some of the equipment is now using; i.e. a meter with a black arc up to about 70% of the scale, then the arc continues with red—no figures or letters to confuse the user; just the direction that most of the peaks should come up to the division between black and red.

Something better is needed for the professional and broadcast equipment, but not the present "A" scale with its confusing -20 to +3 VU scale. Certainly this scale has its merit when used in a system that requires line-up tone tests, but for the majority of users, concerned with the origin of a program, and with no meter attenuator in sight, perhaps the best solution would be to have the meter marked +4 or +8 at the present "0" mark. The marking chosen would depend on the reference being used, and then as the meter is only going to be read between the present -1 and +1 VU marks, all that is needed in addition is +3 and +5 markings for +4 VU equipment. Similarly, for +8 VU equipment, the scale would read +7, +8, +9 VU under the arc, and on top, retain the 0 to 100 to emphasize the idea of 100% utilization of the equipment. This could be called a "C" scale to distinguish it from the present "A" and "B" scales. Now the operators and technicians could abandon such phrases as "it's zero on plus eight," "its zero level" (this was supposed to be buried long ago) or "it's peaking at minus five." This last expression is the most confusing of all: is it -5 VU, -1 VU, +3 VU or -5 on the scale or what? Wouldn't it be better to be able to say "this program is +4 VU (or whatever the standard is in use)? No more is required, but it would imply three things:

1. An approved VU meter, correctly calibrated, is being used.
2. Program material is being observed, not tone or peaks.
3. The progress has been observed for a long enough time to make a reliable statement.

As starters for a new scale meter, how about only 3 markings for VU and the 0 to 100% in 20% steps on top? For those interested in a composite program of speech and music, the percentage scale is very useful for setting the level for the less important parts of the program, e.g. themes to introduce a speech session or bridging music between two scenes in a play. Most broadcasting organizations look for this refinement in the presentation aspect of operating, and issue instructions worded something like "keep the less important portions of the program down to -5 on the scale, or down 4 dB or down to 60%. The % mark is easier to see and is another reason for retaining the 0 to 100% markings.

It is worth noting that most motion pictures produced over the last 15 years appear to be using this refinement, combined with the use of VU meters. The effect on the ears is certainly better than what is still evident on older films, where peak type meters were used and the music intervals always sounded 5 to 10 dB louder than the speech. The intelligent use of a VU meter can assist in making a very well presented program.

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Understanding S/N Ratios

Herman Burstein

Signal-to-noise ratios for tape decks tend to be more puzzling than ever due to the multiplicity and variation of factors involved in measuring S/N. The maximum signal which can be put on the tape is defined in various ways. One of these ways is the signal which causes the tape deck's meters to read 0 VU, but such a reading may represent a variety of signal levels. S/N also varies with the type and brand of tape used, and S/N varies, of course, with inclusion of a noise reduction system, such as Dolby.

Because of these and still other factors, the S/N rating for a given model tape deck may vary over a considerable range—possibly 10 dB or even more. The purpose of this article is to explain why S/N ratings for the same deck can vary so much, but first it will be useful to briefly review a bit of history concerning the subject, and to explore the meaning of S/N for tape decks.

A Bit of History

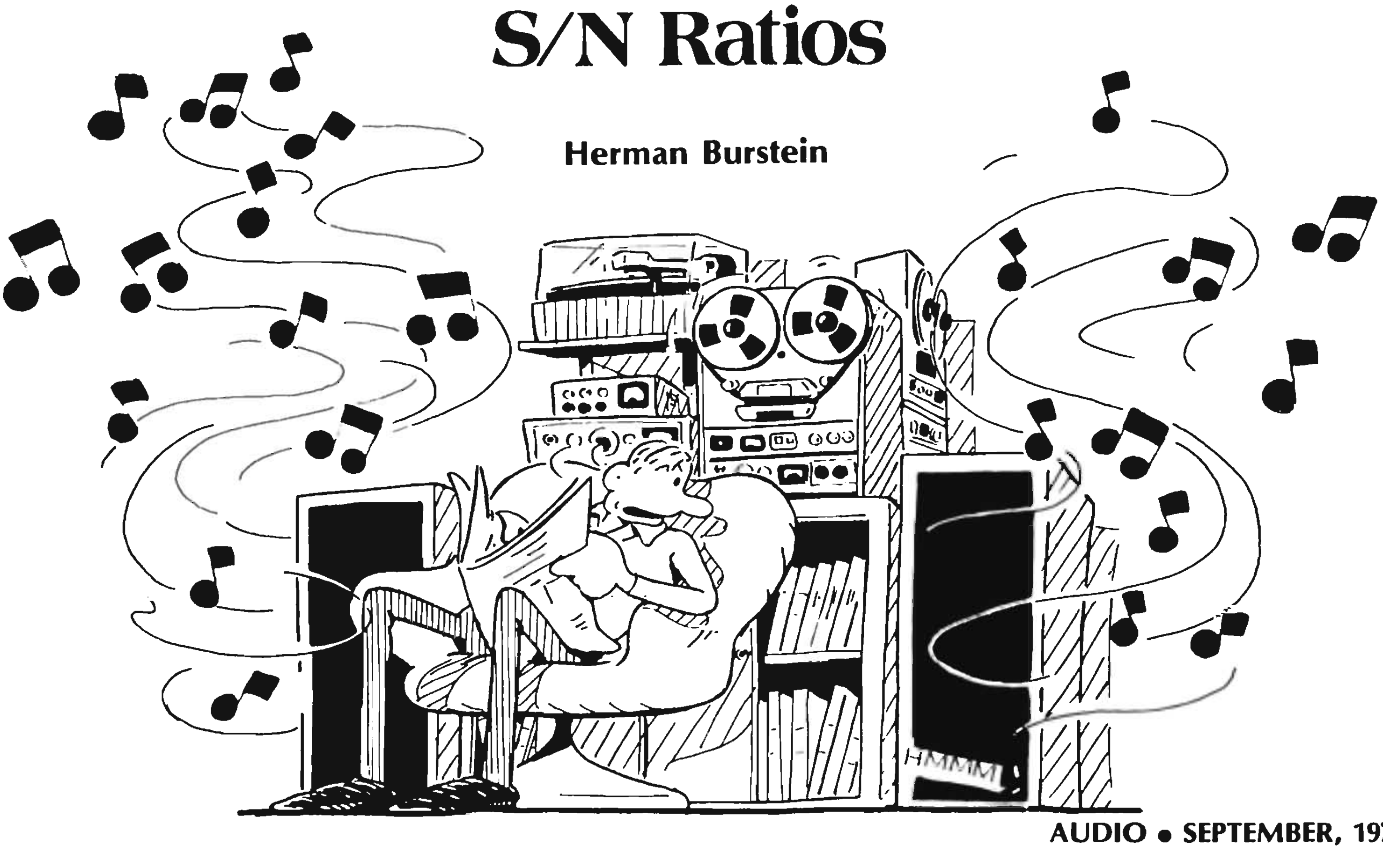
S/N of 55 dB is about the minimum for high fidelity reproduction. Ten years ago, many home tape machines achieved no better than S/N of 45 dB or so. As recently as about two years ago, a home tape deck was considered to be doing excellently if it attained 55 dB S/N. However, improvements in tape electronics, tape heads, and tape, plus the advent of noise-reducing devices such as Dolby, dbx, and JVC's ANRS, have made possible home machines with S/N ratios on the order of 65 dB and even close to 70 dB. Now that really good S/N ratios are within reach at reasonable cost, there is a good deal more candor in the tape industry about the prime importance of a high S/N ratio.

Meaning of S/N for Tape Decks

In a general manner of speaking, S/N refers to the ratio between the *desired* audio signal (program material) and the *undesired* audio signal (noise). This ratio is expressed in decibels (dB), as further discussed in the next paragraph. More specifically, the numerator of the S/N ratio refers to the maximum permissible audio signal which can be put on the tape; and the denominator refers to the noise in the tape system (produced by the record and playback electronics and the tape). Maximum permissible audio signal is generally accepted, at least for home machines, to be that which at 400 Hz results in 3% harmonic distortion on the tape. Therefore, measurement of S/N involves recording a tone of 400 Hz (or similar frequency) at a level that produces 3% distortion on the tape; measuring the level of this tone in playback; rewinding the tape and again putting it through the recording process but this time *without an input tone of 400 Hz*; measuring the playback output, now consisting entirely of noise; and expressing the ratio between the first and second playback measurements in terms of dB.

S/N of 55 dB signifies a maximum power ratio of 316,000:1. That is, assuming a fairly flat audio system, the loudspeaker delivers about 316,000 times more power for the desired program material than for the undesired noise. A S/N of 60 dB signifies a maximum power ratio of 1,000-000:1; 65 dB, 3,160,000:1; 70 dB, 10,000,000:1.

Note carefully in the preceding paragraph our reference to *maximum* power ratio. Much of the time the audio signal is well below its peak (maximum) level—often 20 dB, 30 dB, or still lower. Then the S/N ratio drops the same number of dB. During a very quiet passage, the audio level of program material may drop as much as 45 dB (perhaps more on a disc or tape with wide dynamic range), so that the S/N ratio drops 45 dB. To illustrate, assume a tape system has a S/N ratio of



55 dB based on maximum permissible recording level. But if the audio level drops 45 dB during a quiet passage, the S/N drops 45 dB to only 10 dB; the level of the program material is now only 10 dB above the noise level.

Thus one may realize how desirable it is to attain S/N above 55 dB, particularly when dealing with program material having a wide dynamic range (ratio between the loudest and softest sounds). For example, a tape deck with 65 dB S/N would assure us that the audio signal is *at least* 20 dB above the noise, assuming a dynamic range of 45 dB. That is, the S/N ratio would be 20 dB on the quietest passages, more than that on the louder passages, and 65 dB on the loudest passages.

Reasons for Discrepancies

There are at least eight factors that explain why a given model of a given brand of tape deck can receive different S/N ratings.

1. *Reference Level for Measuring S/N.* As previously stated, S/N for home machines is usually based on a recording level that produces 3% harmonic distortion on the tape. This may be called the 3% reference level. However, two other reference levels—which usually tend to be about the same as each other—are also used. One of these alternative reference levels is that which produces 1% harmonic distortion. The 1% reference level tends to be a recording level about 7 or 8 dB below the 3% reference level. Therefore, when the 1% reference level is employed for measuring S/N (often the case for professional tape decks), the S/N rating drops 7 or 8 dB. For example, a 65 dB S/N rating based on 3% distortion drops to a 57 or 58 dB S/N rating based on 1% distortion.

The second alternative reference level is the recording level which causes the VU meter to read 0 VU. Typically, VU meters in high quality machines are calibrated to read 0 VU when the recording level is that which at 400 Hz produces 1% harmonic distortion on the tape. In such cases, the 0 VU reference level produces the same S/N rating as does the 1% reference level. Sometimes, however, a 0 VU reading will correspond to less than or more than 1% distortion. Correspondingly, the S/N rating goes down or goes up. We should take particular note of tape decks with peak-reading VU meters, which (unlike the standard VU meter) indicate peak level rather than average level of the audio signal. Peak-reading meters tend to be calibrated so that 0 VU corresponds to the 3% reference level. Then the S/N rating based on 0 VU tends to be the same as the S/N rating based on the 3% reference level.

2. *Noise Weighting in Playback.* While the hearing range of humans extends between approximately 20 and 20,000 Hz, we do not hear all frequencies equally well, assuming that all frequencies are presented to our ears with equal acoustic power. Our hearing tends to be less sensitive at low frequencies and at high ones. Therefore noise at low and high frequencies is less audible than noise at middle frequencies. To allow for this phenomenon, the measurement of noise in playback (as described earlier) is *sometimes* weighted in accord with what is believed to be the typical human change in hearing sensitivity as frequency changes. That is, the noise produced by the tape deck in playback is put through an electrical filtering device which gradually reduces the amount of noise energy at low frequencies and at high ones. *Then* the noise is measured. The result of this process is to reduce the amount of measured noise.

Various "weighting curves" are employed, that is, various amounts of reduction of noise energy at low and high frequencies. An example is the ASA Standard C16.5-1961 weighting curve, adopted by NAB (National Association of Broadcasters) in its 1965 standards for tape recording. Noise reduction becomes significant (reduced 3 dB or more) below approximately 800 Hz and above approximately 7500 Hz. Noise reduction is greatest for the low frequencies, being about 32 dB at 50 Hz and still more at lower frequencies.

All in all, use of weighting in measurement of S/N results in an increase in S/N rating. It is difficult to say how much the S/N is increased. The amount of increase depends upon which weighting curve is employed. It also depends upon the particular tape deck being measured. For example, if a deck has particularly strong 60 Hz hum, it can benefit more from a weighted S/N measurement than a deck with very little hum. As a rough guess, weighting can improve the S/N rating by about 6 to 10 dB. (Thus, the NAB standard stipulates S/N ratios 10 dB higher on a weighted basis than on an unweighted basis.) A tape deck with an unimpressive 50 dB S/N rating on an unweighted basis might achieve an impressive 60 dB rating on a weighted basis.

3. *Use of Dolby and Other Noise Reduction Systems.* For a cassette deck to have a real claim to high fidelity, it must include a Dolby or similarly effective noise reduction system (NRS). While open-reel tape decks can achieve S/N of high fidelity caliber without NRS, there is a trend toward inclusion of such systems; earlier discussion has pointed out the advantages of exceeding the minimum high fidelity requirement, namely 55 dB S/N. Dolby and similar NRS can improve S/N by amounts typically varying from 6 dB to 10 dB, particularly at lower speeds (1-7/8 and 3-3/4 ips). The improvement tends to be somewhat less at higher speeds (7½ and 15 ips).

4. *Tape Used.* For a given amount of distortion, say 3%, some types of tape can deliver more audio signal than can others. Thus, high output tape delivers more signal than conventional tape. Further, low noise tape produces less tape noise than do some other kinds of tape, and low noise/high output tape has both advantages. Thus, there may be a few dB difference in S/N ratio depending on which type of tape is used. In addition, for a given type of tape (such as low noise/high output, 1 mil, ferric oxide) there may be a variation of about 1 to 3 dB among various brands of this tape.

5. *Chance.* Random variations in components used (transistors, resistors, capacitors, etc.) and in lead dress (wiring) may result in slight variations in S/N from one unit to another of the same model and brand of tape deck. Such variations might be in the vicinity of 1 or 2 dB.

6. *Manufacturer's Conservatism.* The conscientious manufacturer, wishing to attain and live up to a good reputation, will be conservative in stating specifications for his tape deck, including its S/N rating. He will make allowance for chance variation from one deck to another, as just described, for different tapes that may be used with his unit, and possibly for other factors. His may be a "worst condition" specification for S/N. Therefore, the typical purchaser of his tape deck may find that actual S/N performance exceeds rated S/N by several dB.

The reverse can also be true, particularly for tape decks of generally lower price and quality. Rated S/N may be a "best condition" specification; it may assume that a selected tape deck from a given model line is employed, and that it is used with the best of tapes. Thus, the typical purchaser may find actual S/N below specification.

And of course we can have the intermediate situation, where many purchasers find actual S/N above specification, and about an equal number find actual S/N below specification.

7. *Quality Control.* Control over the quality of tape decks reaching the purchaser may range from very rigorous to quite loose. Quality control is one of the unseen things (such as extensive research) that go into a costly machine. In the case of S/N, tight quality control helps insure that there is little variation from one tape deck to another of the same model and that none fall below specification. Loose quality control makes possible fairly extensive variation.

8. *Design Improvements.* During the lifetime of a given model of tape deck, which may be several years, the manufacturer typically makes changes, some of which may result in improved S/N. These changes are based on continued research by the manufacturer, new developments by others (such as the supplier of tape heads), and feedback from dealers, purchasers, equipment reviewers, and others. Therefore, a later version of a given model (higher serial number) may have a few dB better S/N ratio than an earlier version.

Conclusions

While the foregoing discussion hopefully is enlightening, it can hardly be satisfying. It cannot be satisfying to know that S/N ratings for tape decks vary not only because of inherent differences in quality of the decks but also because of differences in methods of measurement and differences in tapes used for measurement. How is the audiophile to compare the S/N of one tape deck with that of another? How is he to know the S/N of a given tape deck—simply just to know it or to compare it with the S/N of other audio equipment?

It seems this situation can be resolved by the audiophile asking two basic questions and by the industry—manufacturers, dealers, and others concerned—being prepared with the answers for each brand and model of tape deck. The two questions are as follows.

Based on (a) the tape recommended by the manufacturer for his tape deck,[1] (b) the 3% reference level, and (c) measurement on an unweighted basis:

1. What is the S/N ratio *without* use of NRS (noise reduction system, such as Dolby)?

2. If the tape deck includes NRS what is the S/N *with* use of NRS?

We have to be able to compare tape decks' S/N ratios on a common basis. Until the day when all tape decks include built-in NRS, the common basis for measuring S/N is necessarily without use of built-in NRS. Therefore, when a tape deck has built-in NRS, we need the answer to Question 2 as well as the answer to Question 1. (Of course, when a tape deck lacks built-in NRS, we can only raise Question 1.)

NRS are available today either built-in or as an external device made by several companies. Therefore, one may rightfully want to know the S/N of one's tape system including NRS. If NRS is built-in, this information is provided by the answer to Question 2. If not, we can get a reasonable approximation by adding the answer to Question 2 to the S/N improvement claimed by the maker of the external NRS. To illustrate, assume a tape deck without NRS has a rated S/N of 55 dB, based on the 3% reference level and unweighted measurement of noise. And assume that 8 dB S/N improvement is claimed for the NRS device. Then the S/N of the tape system, including the NRS device, amounts to about 63 dB.

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[1]It is unlikely that the manufacturer would recommend a tape chosen solely because it serves to maximize S/N. Other factors in his choice of tape will include its frequency response, availability, cost, physical characteristics (such as oxide shedding, tendency to cup or curl, accurate dimension, etc.)

Probably a better alternative to (a) would be an industry-accepted standard tape, issued under the auspices of NAB, which all tape deck manufacturers would agree to use for the purpose of specifying S/N. (A manufacturer could still recommend commercially available tape or tapes to be used with his deck.)

Fighting Distortion In Tape Recording

Wayne Saylor*

High performance tapes being introduced in today's audiophile market are characterized by such features as low distortion, high undistorted output, more output at high frequencies before saturation, more dynamic range, and more headroom. This discussion deals with how to take advantage of those benefits; to present an understanding of how to recognize and control distortion inherent in the tape recording process so that tape recordings truly have higher fidelity.

Understanding Distortion

Strictly speaking, distortion is the degree to which reproduced sound differs from original sound, and this definition, lack of fidelity, holds true regardless of whether the *distortion* is caused by equipment or by recording medium. Distortion includes amplifier overload and hum as well as tape "hiss" and any change in frequency response between input and output of the tape recorder.

The scope of this discussion will, however, focus on distortion as related to magnetic tape *and how to* optimize the record and playback levels to obtain minimum distortion. The particular types of distortion to be discussed are odd harmonics, compession and saturation, and intermodulation distortion.

These types of distortion are all forms of nonlinear distortion and occur when the output *and input* of a recorder depart from an ideal straight line relationship. The function of a tape machine's bias circuitry is to linearize the output-input relationship which otherwise would be highly non-linear (owing to magnetic hysteresis). Beyond having a unit's bias correctly set (usually done in the factory or service department), there are other things which must be done to record with minimum distortion, and these aspects of recording will be discussed here.

Odd Harmonics Of the Input Signal

Harmonics can be described, generally, as an addition to the fundamental tone of spurious tones having frequencies that are whole number multiples of the fundamental. Thus, third harmonic distortion is a combination of the fundamental frequency and a second frequency which is three times the fundamental. The percentage level of a harmonic almost always decreases with each higher harmonic order; the fifth is much lower than the third, etc. Consequently, only the third harmonic is generally measured in the case of odd harmonics distortion, because it usually has a substantially higher percentage level.

Since all musical instruments have rather significant amounts of their sound energy in harmonics, harmonic distortion is not, in itself, unpleasant to listen to. Third harmonic distortion is very convenient to measure, however, and is used to quantify distortion more because of its convenience and correlation with other types of distortion. Of course, third harmonic distortion, inherent in tape recording is only perceptible or measurable using fundamentals up to 1/3 of upper frequency limit of the recorder. The laboratory method of measuring third harmonic distortion involves simply recording a pure tone (single frequency) and measuring the relative level of the third harmonic of that tone.

Even harmonics are not *generally* characteristic of magnetic tape systems because the hysteresis curve is symmetrical, even though it is nonlinear, and asymmetry is required to generate even harmonics. However, even harmonics can occur in the recording system and are caused by any nonsymmetrical polarizing magnetization. Such even harmonics *are*, however, equipment related, rather than tape related, and are usually caused by either magnetized heads, a d.c. current component in the heads, one-sided amplifier overload, or asymmetrical distortion of the bias signal.

Compression and Saturation

Compression of the recorded signal occurs when a given amount of increase in input level does not result in as much increase in output level. *Compression occurs at all* frequencies, but is much more prevalent at high frequencies. Obviously, this alters frequency response. This happens principally at high input levels due to the inability of the oxide coating to be magnetized further and a phenomenon known as self-erasure occurring predominately at higher frequencies.

At the level compression begins, that is where *no increase* in output level results from increasing the input level, *saturation* has been reached. Beyond saturation, further increases in input record level result in decreases in output. Use of recording input levels *beyond* saturation alters *frequency* response much more adversely than compression *prior* to saturation. Compression and saturation result in non-true *level*, rather than the introduction of new or different frequencies.

Intermodulation Distortion

Intermodulation or IM distortion occurs throughout the spectrum of frequencies which the recorder is capable of reproducing and is the generation of new frequencies *not harmonically related* to the input frequency. This form of distortion results from the combination of two or more input frequencies at the same time; the output contains not only the original two frequencies, but also the *sum* and the *difference* of the original two frequencies. These are usually referred to as *beats* of the original frequencies. IM Distortion has no musical relationship to the original frequencies from which the distortion is derived, so IM Distortion causes the greatest subjective impairment to the music. The words harsh, dissonant, and "fuzzy" are used to describe the unpleasant subjective effects of IM distortion.

Signal-to-Noise Ratio

The main idea in making superb quality recordings is to use the highest record level possible without objectionable distortion. The higher the *Record* level, the more distortion, but a lower *Record* level results in less difference between the inherent tape noise and the level of music during *Playback*. Tape hiss gets louder as *Playback* level is increased, but does not get louder as *Record* level is increased. As *Record* level is increased, the recorded signal is increased, so

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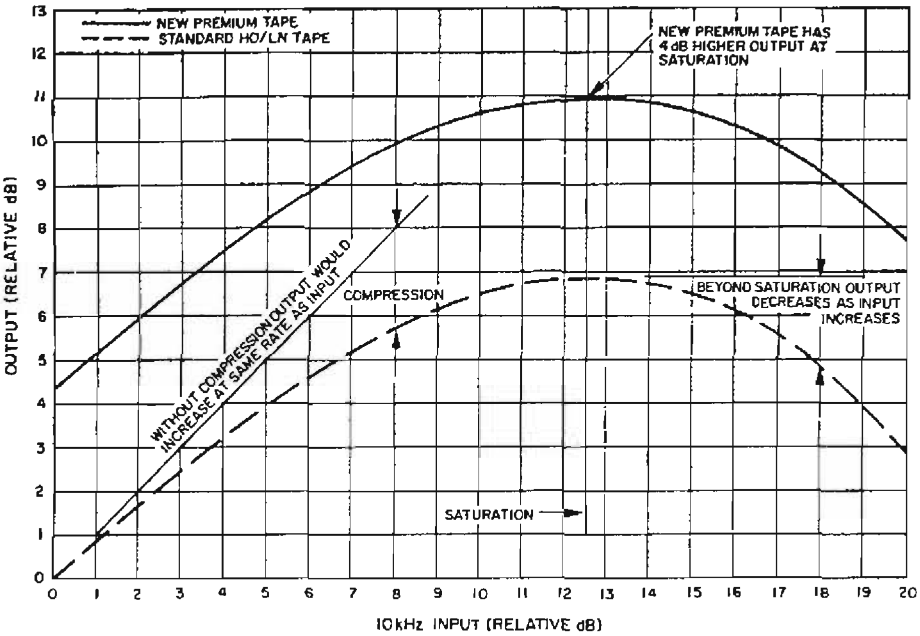


Fig. 1—Compression and saturation.

better signal-to-noise ratio is achieved by recording at the highest level possible.

Since today's superior performance tapes achieve their performance through high record level and output capability, it is necessary to know how to record at high levels and still control distortion in order to utilize the full performance capability of the tape.

Tape noise or tape "hiss" is present in all recording tape and stems from such factors as the size of the oxide particles, irregular distribution or dispersion within the coating, surface roughness, and poor orientation of the particles. Within basic categories of tape (such as high performance audiophile tape), the noise is usually quite similar. The predominant factor in determining the relative signal-to-noise ratio of competitive tapes is maximum output signal level rather than how low is the noise.

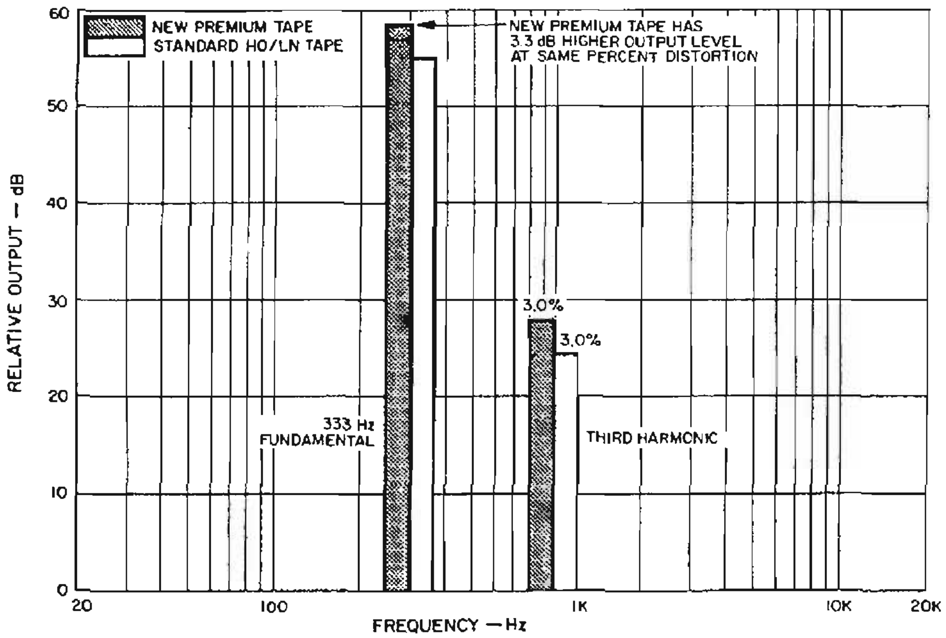
Optimum Recording Procedure

Optimum record level settings can be determined by the recorder user by utilizing some of the tape recorder's features to conduct an A-B comparison test as commonly used by audio engineers and other recording professionals. The easiest way to conduct an A-B test is to use a three-head tape machine with a *Source-Tape* switch. Most open-reel recorders sold today have separate erase, record and playback heads. Using the operating instructions for the tape deck, connect the recorder to a sound source such as a phonograph record player. While in *Record* mode you can then switch repeatedly from source to tape, back to source, etc. In this way, you can compare the source and the tape playback. This comparison should include not only careful listening, but observing and making note of the level indicator reading on the loudest passages. There will be a slight delay in music between the *Source* and *Tape*, but this fraction of a second does not detract significantly from the A-B comparison. The delay is dependent upon tape speed and the distance between the record and play heads. Note that in the *Record* mode on a three-head machine, the level indicator shows *Record level in Source* and *Playback* level in *Tape*. For each A-B test, the playback level should be adjusted to match the loudness when listening to *Tape* to the loudness in *Source*. This is important because a difference in loudness may mask a difference in distortion.

If your machine has only two heads (erase and record/play), you can still conduct an A-B test, but it is more difficult and time consuming. The difficulty stems from the fact that you can't listen to tape playback during the same pass as record, necessitating rewinding and playing later. Such a machine doesn't need a *Source-Tape* switch since the level indicator reads *Source* during the record pass and *Tape* during the playback pass. To A-B compare the source and tape in this case, the phonograph record source and tape recorder playback should be connected to their respective inputs of the stereo system preamplifier. The A-B comparison is then made by *switching the system pre-amp* between *Phono* and *Tape*. It is, however, necessary to closely synchronize the phonograph and the tape during playback.

Once your tape machine is set up in the A-B mode previously described, you are ready to experiment. Initially adjust your recording level so that the level indicator consistently reads far into the red zone so that the recording is purposely very distorted. Match playback levels and switch between the original source and the now distorted recording. Listen to the distorted recording and compare it with the original source. Now that you know how distortion sounds, start reducing the recording level (compensating for any drop in volume by increasing the playback level) until you reach a point where you no longer notice any distortion in the record-

Fig. 2—333 Hz maximum output level.



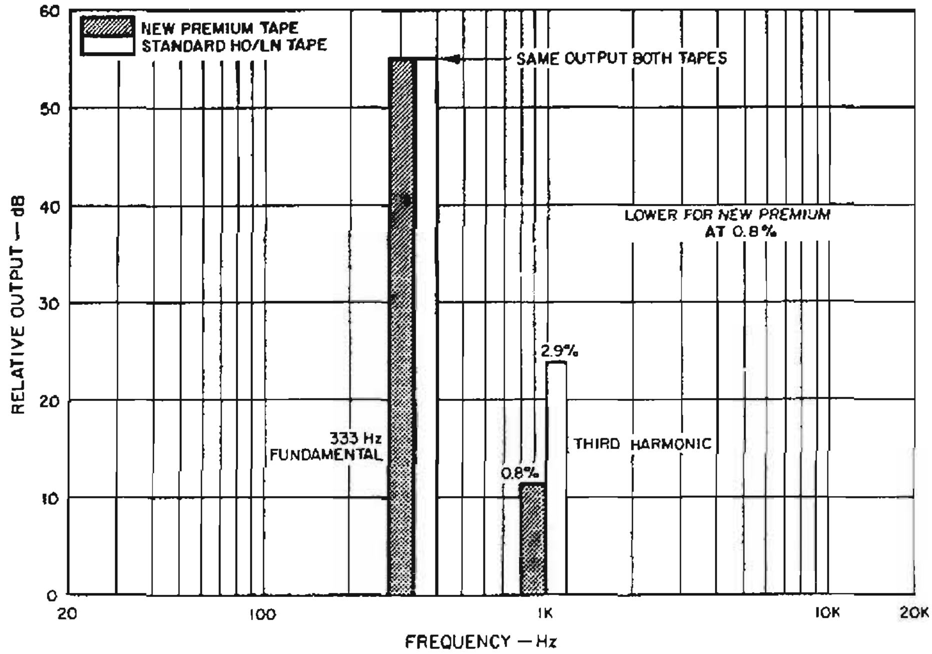


Fig. 3—Relative distortion.

ed program relative to the original program. By observing the level indicators at this point, you have identified the point on your meters where distortion becomes unnoticeable. Signals swinging further into the red zone on your meter will give noticeable distortion.

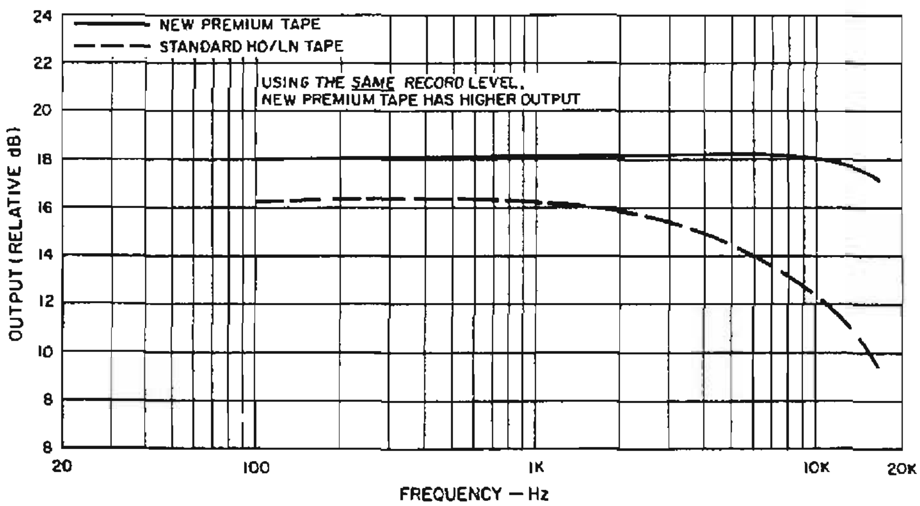
Ideally, you should adjust your record level to as high a level as possible without inducing distortion. And you should recognize that recordings from different program sources made at the same recording level setting may differ in distortion because they differ in dynamic range and average level. Thus, we should repeat the above experiment for various types of music. Remember that with some tape machines, you may be able to operate well in the red with no noticeable distortion. With others, you may have to keep the indication below the red even for sudden peaks. The single most valuable tool you have in controlling distortion is the level indicator. The recordist must learn from the equipment manufacturer and from his own experience how to use his particular indicator to control distortion.

There are great differences among level indicators of which there are several types. The most well known is the VU meter which averages the signal, ignoring sudden peaks. There are also bar-graph indicators, LEDs, and peak reading meters. One type of peak reading meter measures "average" loudness much the same as VU meters except that it has a fast-attack, slow-decay ballistic response so it responds to shorter transient bursts of sound. It measures the flat response input signal when metering source. The other type of peak metering has the same peak response but meters the *record* signal in *Source* AFTER record equalization so that the actual level applied to tape is indicated. (High frequencies are recorded higher relative to low and mid-frequencies in what is known as *Record* equalization or *Record* pre-emphasis.)

With some tapes you can *record* at very high levels, but with some you can't. Thus, you should determine what sounds best to you, and then operate on that basis. This test will also allow you to explore different brands of open-reel tape, giving you the opportunity to determine if one brand offers better performance on your machines than another. One brand of tape may have more "headroom" than another, and this might allow the level indicator to read further into the red zone. On the other hand, if the tape also had greater sensitivity, it would give a higher undistorted output for the same reading on the level indicator. Remember that high undistorted output is the key requirement in obtaining a stronger, more dynamic recording with less hiss. While you can, of course, play it safe and always avoid distortion by recording signals at a low level, you would be recording too close to the noise floor, so you would notice more "hiss" than the recording would have if the *Record* level were set to its highest "distortion-free" position.

Compression is most noticeable in the high frequency end of the spectrum. Therefore, certain percussion instruments are best for observing where compression takes place. Cymbals and some tambourine sounds are good examples of this. An example of records very good for experiencing what compression really is and how to control it is **I've Got The Music In Me** on Sheffield records, **Rough Trade—Live** on Umbrella (through Audio-Technica cartridge dealers), or **Direct Disco** by Crystal Clear Records. These records are particularly clear, free of distortion, and have great high level high frequencies. A record that has great dynamic range from very soft passages to very loud ones is **Saint Saen's Organ Symphony in C Minor,** Phila. Orchestra, Columbia MS - 6469. For A-B comparison, however, the most useful type of record music is one with lots of high frequencies at high level on a sustained or repetitive basis. Other than that, you should choose music of the kind you like most and listen to most frequently.

Fig. 4—Sensitivity.



Ways to Control Distortion

Although wise use of your level meter and experimentation is a primary way of controlling distortion, there are other techniques to be considered. They fall into two general categories, machine adjustments and tape selection.

As mentioned earlier, the sole purpose of bias in audio tape recording is to promote a linear relationship between reproduced output and recording input. Ideally, this should be done for each different type of tape to minimize distortion. Too little bias causes non-linear distortion and increases high frequency output. (This is due to bias field strength gradient throughout the oxide coating thickness, the strength being higher nearer the surface where high frequencies are recorded.) Too much bias effects the frequency response, decreasing the high frequencies more than the lows.

Very few tapes decks provide a convenient means of bias adjustment by the user. Unless your deck provides such a means, it is unwise to tamper with bias if you are not equipped with proper external test equipment to measure the degree of adjustment. If you purchase a new machine or change brands of tape, you might find a worthwhile improvement by having a service agent rebias your machine. Many new machines are not properly biased for the brand of tape being used. Some machines have a switch with more than one fixed bias setting, labeled *Standard/High, Normal /LN,* etc. You should use the position recommended by the recorder manufacturer or tape manufacturer.

The higher the tape speed, the easier it is to get good signal output versus noise, without getting into distortion. The slower speeds place the most demands upon the tape performance, and it is at these speeds where the differences among tapes are the most apparent.

Dirt on the record head separates the tape from intimate contact with the head and causes distortion because the separation effectively lowers the bias as well as the record signal. A dirty playback head can cause distortion mainly through recordist confusion if you erroneously increase record drive to make up for lower playback level caused by head to tape separation.

Magnetized heads cause even harmonic distortion and increased noise level, so periodic degaussing is recommended.

Tape Selection

For best results you must select the tape which will provide the greatest performance parameters with minimal distortion. Even among premium tapes, it is important to try out various brands and experiment with record levels in order to find out which offers the best performance on your tape deck. Generally speaking the important characteristics to consider include:

Distortion. If your tape has relatively low distortion, you will be able to record higher levels while still maintaining high quality reproduction.

Signal-to-noise ratio. The higher you can record above the tape noise, the less "hiss" you will hear even on soft, quiet music since the output level setting can be lower relative to the record level setting.

Saturation. A high output before onset of saturation results in a greater dynamic range and, most important, provides assurance against over recording high frequencies to the point where higher input results in lowered output.

Sensitivity. A tape with higher sensitivity is one which has *higher* output than another tape for the *same* record level. While the M.O.L. (Maximum Output Level, usually quoted relative to a certain percentage distortion) is the more important parameter, sensitivity is ordinarily a reliable indicator of relative M.O.L. among tapes of similar bias requirement. A more sensitive tape usually has higher M.O.L., especially at low or mid-frequencies. To check this out yourself, record some music which has a sustained level from a phonograph record at about -5 to -10 on your level meter. With the same record level settings, check which brand of tape plays back at the highest level. You may find 2 or 3 dB difference between major premium brands. More sensitive tape usually has lower relative distortion as well. This lets you capture all signals at a higher level relative to the noise.

Conclusion

In conclusion, it should be evident from this discussion that any recording enthusiast can control distortion if, first, he has a good basic understanding of the mechanics and limitations of his tape recorder and its meters; and secondly, if he understands the performance characteristics of the tape and how it interacts with the recorder. Both conditions necessitate a certain amount of recording experience and experimentation, but once this is acquired you have the tools you need to control distortion.

OPEN-REEL vs. CASSETTE

Herman Lia*

The enormous sale of cassette tape recorders in the last few years is proof enough that this product meets a demand. At the same time there is a danger that people will forget the open-reel tape recorder, which in many important respects is a much better basic concept. A cassette machine's advantages are that it is easy to operate, weighs very little, and that a broad range of prerecorded tapes are widely available. On the other hand, as this article will show, cassette machines will always be inferior to open-reel machines when the major performance characteristics, such as signal-to-noise ratio, are compared. At the same time, it must be admitted that for a large number of consumers, cassette machines are usually quite good enough. However, when the very best quality recording is required, an open-reel machine must be used. To be fair and realistic, we must also say that cassette machines have expanded the total market for tape recorders, as well as capturing a portion of the open-reel market. However, they will never take over *all* of the open-reel market because there are fundamental differences in the quality level obtainable with the two systems, differences which result from the internationally recognized standards governing each system.

General Considerations

The two most important characteristics that determine the performance of a tape recorder are signal-to-noise ratio and frequency response. In this context, frequency response means a response relative to a signal level that lies substantially below the saturation curve of the tape and substantially above the level of residual tape noise.

Let us consider a tape recorder as a black box where we connect a signal to the input and take out another signal from the output, as shown in Fig. 1.

Ideally, the only differences between the input and the output signal are time delay and possibly some scale or amplification factor, A. The lowest possible time delay is determined by the distance between the record and playback heads, together with tape speed. Unfortunately, real world tape recorders are not ideal, and we need to make some measurements to discover their characteristics. We can begin by measuring signal capacity. We do this by applying a single tone at a particular frequency to the input and then raising the input level voltage $e_i(t)$ until the signal at the output has a particular amount of distortion, e.g. 5 per cent harmonic distortion. This can be done for a number of frequencies, and the typical results are shown in Fig. 2 for one particular tape speed. Next we remove the input signal $e_i(t)$ and short circuit the input. There should, of course, be no signal at the output, but in practice there is a noise spectrum which is the sum of the residual tape noise and the noise from the record and playback electronics.

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In a well-designed tape recorder, the noise from the electronics is so low that the dominant noise component is the tape noise. The noise spectrum can be analyzed by means of one-third octave filters, and this is shown in Fig. 2 along with a saturation curve.

These measurements tell us quite a lot. They tell us that there are upper and lower limits of the signal a tape can accommodate with acceptable quality. If the input signal is too high, the distortion will be above the acceptable maximum, and if the input signal is too low it will get lost in the noise. The distance between the two curves in Fig. 2 at an individual frequency is therefore a measure of the signal capacity of the tape recorder at individual frequencies, while the total area between the two curves is a measure of the signal capacity over a chosen frequency range.

Using information theory, the signal capacity can be defined for a general transmission channel by the following integral:

$$SB = \int_B \log\left(\frac{S+N}{N}\right) df \qquad (1)$$

where SB is the signal bandwidth product, S is the signal, N is the noise, and B is the bandwidth. This is Shannon's definition of signal capacity, given in 1948.

Now, since $\log \frac{S+N}{N} = \log (S+N) - \log N$, equation (1) can therefore be rewritten as:

$$SB = \int_B \left\{\log (S+N) - \log N\right\} df \qquad (2)$$

This precisely defines the area between the signal and noise curves in Fig. 2, and therefore equation (2) gives us an opportunity to put forward a quantitative measure of a tape recorder's ability to accommodate signals. More exact theoretical considerations we have developed show that the SB product for a tape recorder is given by:

$$SB = \log\left\{\frac{B_s . d . \sqrt{b}}{N_{(f)}} \cdot \frac{1}{\frac{B_r}{H_c} + \left(1 + \frac{B_r}{H_c}\right)\sqrt{1 + \left(\frac{\omega_o}{v} d\right)^2}}\right\} f_o \qquad (3)$$

B_s = Induction in the tape caused by the signal (Gauss)
B_r = Maximum remanent induction (Gauss)
H_c = Coercivity (Orsteds)
v = Tape speed
b = Track width
d = Thickness of oxide coating
f_o = Highest frequency considered
N(f) = A characteristic function of tape noise.

The most important conclusion to be drawn from equation (3) is that the SB product is dependent on the physical properties of the system, such as the tape speed, track width, tape parameters, and so on, rather than the electronics, *as long as* we maintain the true dynamic range in the program which is to be recorded with no signal processing. We will see later that it is possible to process the signal so that the tape hiss becomes less audible to the listener.

Despite this conclusion, we find the frequency-dependent equalization in a tape recorder greatly affects the audible results. We should, therefore, take a closer look at the main requirements influencing the choice of these equalizations. These turn out to be maximum subjective signal-to-noise ratio and flat frequency response at low signal levels.

The measurements for Fig. 2 were made with one particular playback equalization (120 μS). If we choose another equalization, say 50 μS, and make additional measurements, we obtain the curves shown in Fig. 3. Note that the distance between the two curves is the same, but the shapes of the curves have changed. When we record a program, we are dealing with a complex signal with a particular power distribution over the frequency spectrum, and it should be obvious that we will obtain the best subjective signal-to-noise ratio if we can "pack the sound" as far as possible up under the tape's saturation curve.

Let us assume that we have a program with relatively little power in the high frequencies. We then set the input sensitivity of the system to fully load the tape at the middle and low frequencies. If we use the 120 μS playback equalization, the high frequencies will lie far under the tape's saturation curve and therefore near to the noise level. In this case, we could advantageously alter the equalization to 50 μS, say, and thereby drop the noise level away from the signal. If we change the equalization or time constant in this manner, to improve the signal-to-noise ratio, we must be consistent and change the input level to produce the flat test frequency response at low levels. On the other hand, if we now have a program with a lot of power in the high frequencies, a time constant that is too short will cause the high frequencies to overload the tape before the tape is saturated at the low frequencies, and low frequency noise can then become a problem. From this discussion, we can see that the SB product defined in equation (3) is an objective measure of the best signal-to-noise ratio that can be obtained.

Frequency-dependent equalizations are thus used to match the characteristics of the tape to practical conditions and produce the best signal-to-noise ratio, which means that the SB product is exploited to its maximum. At the same time, we have seen that the optimum playback equalization depends on the type of program the tape recorder must handle. We are therefore led to seek a dynamic equalization that automatically adjusts itself to the power-frequency curve of the program being recorded. This is exactly the concept behind complementary noise-reduction systems, such as Dolby, dbx, Burwen, etc. If we make the same measurements used in Fig. 2 with a Dolby circuit added, we obtain the curves shown in Fig. 4.

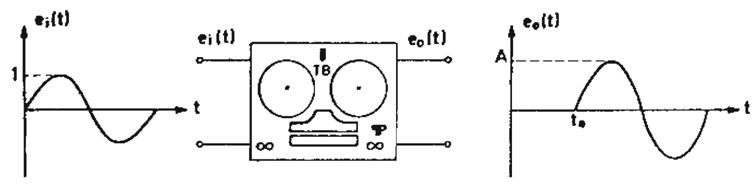


Fig. 1—Ideally the only differences between input and output of a tape recorder are time delay and possibly amplification of signal.

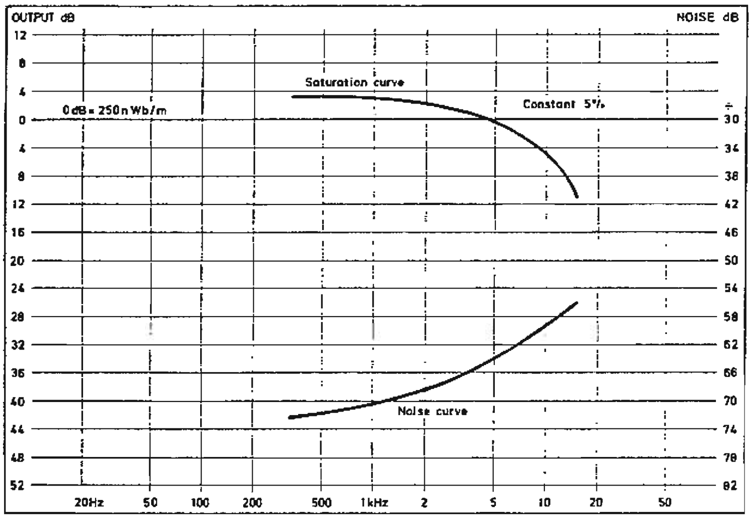


Fig. 2—Recording system performance, showing maximum output level versus frequency at a constant 5 per cent THD and residual noise level of the system.

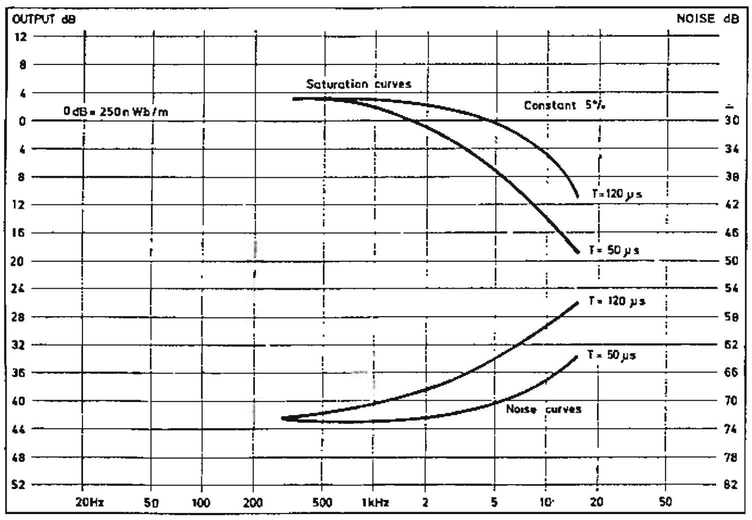


Fig. 3—System performance, as in Fig. 2, with two different equalization time constants, showing how the shapes of the curves change. Note that the area between the pairs of curves produced by each constant remains the same.

At the higher input levels, the signal is not processed and the tape recorder performs as if the Dolby circuit had not been included. When the signal falls, the higher frequencies receive extra amplification and are, therefore, recorded with a larger margin above the tape noise than normal. During playback the opposite process occurs, and the overall frequency response is therefore correct.

Increased amplification of the higher frequencies during recording requires reduced amplification of the same frequencies during playback (complementary system). Therefore, noise and other unwanted signals introduced in the process after encoding and before decoding are reduced. Tape noise is reduced by the same degree as the processing of the signal. Figure 4 shows typical output of a cassette machine with a time constant of 120 μS. At the higher levels, the signal swamps the noise, and the performance is acceptable. The corresponding noise level is given by curve B. At the lower levels, curve A is of no interest, but the signal processing in the Dolby circuits yields noise curve C which is equivalent to a time constant of 40 μS because it has the effect of reducing the noise at higher frequencies by about 10 dB. Accordingly, there is a dynamic change in the time constant from 120 μS to about 40 μS, depending on the amount of high frequency energy in the program.

Measuring the Signal-to-Noise Ratio

The signal-to-noise ratio is often measured according to the German DIN and IEC standards which defines it as the ratio between the signal at 333 Hz with 3 per cent distortion and the tape noise weighted and measured according to the ANSI A weighting curve. One important weakness in these measurement methods is that they only take account of the low frequency signal capacity, and large differences in the high frequency signal capacity can be missed by the measurements.

This point is brought out in Fig. 5; the two curves A and B will show the same signal-to-noise ratio, but without a doubt you will hear the difference on a recording. From the foregoing argument, we can see that the signal bandwidth product is a better measure of the dynamic range.

Let us now make a comparison between cassette machines and open-reel machines in the light of equation (3). Experience shows that maximum remanent induction and tape noise characteristics are exactly the same for cassette and open-reel tapes because the magnetic particles, size, and density of the particles are the same for the two types of tape. (While there are differences in tape formulations actually available to the consumer, any formulation can be applied to either system.)

Assuming the same bandwidth for open-reel and cassette machines, we can use equation (3) to find an expression that shows the difference in the signal-to-noise ratio for the two systems:

$$\frac{OR}{CC} = 20 \log \left\{ \frac{d_1}{d_2} \sqrt{\frac{b_1}{b_2}} \cdot \frac{-\frac{B_r}{H_c} + \left(1 + \frac{B_r}{H_c}\right) \sqrt{1 + \left(\frac{\omega^o}{v_2} d_2\right)^2}}{-\frac{B_r}{H_c} + \left(1 + \frac{B_r}{H_c}\right) \sqrt{1 + \left(\frac{\omega^o}{v_1} d_1\right)^2}} \right\} \qquad (4)$$

Where, for OR (open reel), $d_1 = 13\ \mu m$, $b_1 = 1.0$ mm, $v_1 = 1\frac{7}{8}, 3\frac{3}{4}, 7\frac{1}{2}, 15$ ips, and for CC (compact cassette), $d_2 = 5 \mu m$, $b_2 = 0.6$ mm, $v_2 = 1\frac{7}{8}$ ips. The numeric results of (4) for the four speeds are given in the accompanying table where $W_0 = 2\pi \cdot 20$ kHz. The results show that tape speed is the dominating factor in the SB product. For cassette machines, tape

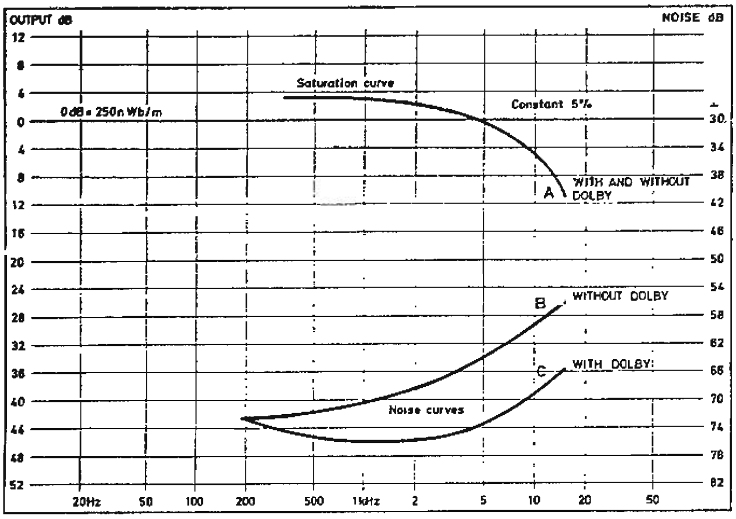


Fig. 4—System performance, as in Fig. 2, but with a Dolby NR circuit added to reduce system noise.

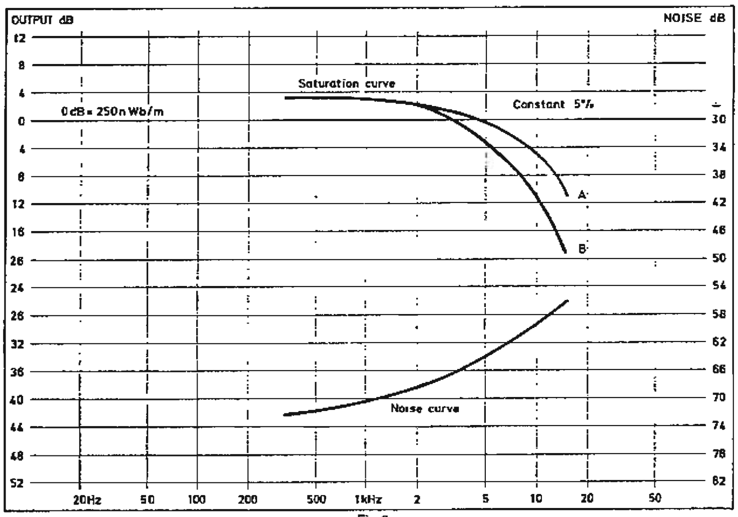


Fig. 6

Fig. 5—Comparison of two systems with the same noise floor, but different maximum high frequency output curves. While both will have the same signal-to-noise ratio, when measured via conventional standards, system A will sound better because of its extended high frequency response.

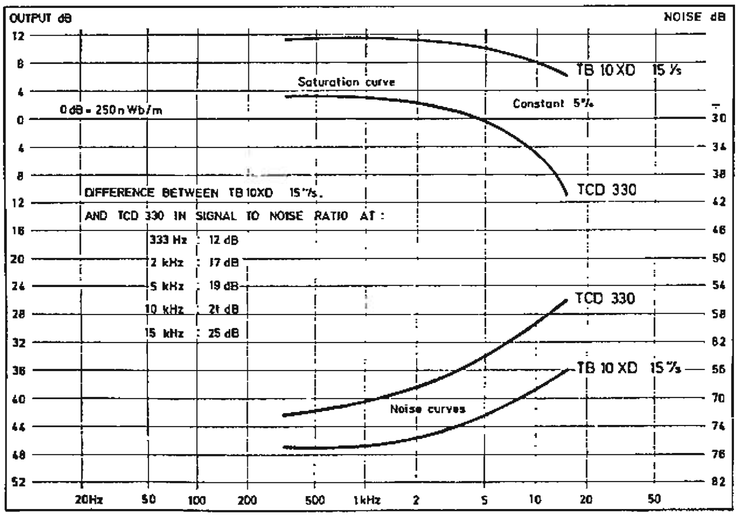


Fig. 6—Comparison of a Tandberg TB-10XB open-reel recorder with a TCD-330 cassette machine, as in Fig. 2.

speed has been prescribed at 1⅞ ips in the standard, but for open reel the tape speed can go up to 15 ips. In addition, there is an opportunity to choose the most suitable track width with open reel. The table shows that there is 8 dB difference between cassette and open reel at 3¾ ips, or that an open reel machines at 3¾ ips without Dolby is about as good as a cassette machine with Dolby. It can be argued that it is an advantage not to use any complementary noise

Table I

v (ips)	1 7/8	3 3/4	7 1/2	15
$\frac{OR}{CC}$ (dB)	2.0	8.0	15	22

reduction system since all forms bring with them undesirable turn-on transients caused by the positive time constants in the control circuits. Furthermore, the Dolby system does not provide any noise reduction at frequencies below 500 Hz, and this is frequently a region where we require an improvement. A poor signal-to-noise ratio at low frequencies causes reproduction to sound impure and damages the quality substantially.

Let us examine equation (4) for frequencies lower than 500 Hz. This is the case where $\frac{w}{v}d \ll 1$ and equation (4) becomes:

$$\frac{OR}{CC} = 20 \log \left\{ \frac{d_1}{d_2} \sqrt{\frac{b_1}{b_2}} \right\} \qquad (5)$$

A new comparison of cassette with open reel at 3¾ ips gives a difference of 10.5 dB, below 500 Hz.

Practical User Qualities

In addition to the considerations concerning recording qualities, it is important to note the differences in user qualities between the two tape recorder systems. Here is a list of points where we feel the open reel machine is superior to the cassette machine:

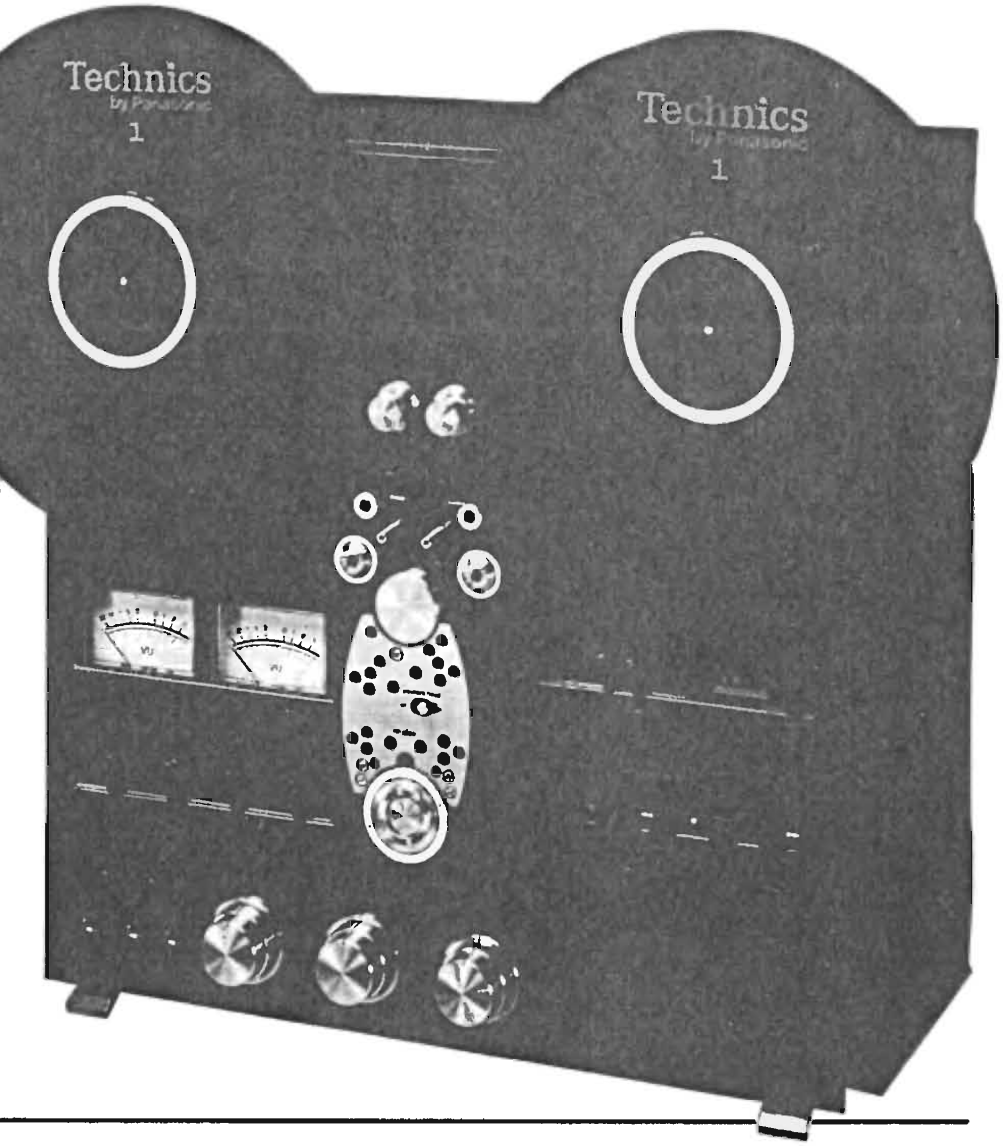
1) Better signal-to-noise ratio.
2) Acceptable signal-to-noise ratio without noise reduction system.
3) Longer playing time.
4) Greater tape reliability.
5) Better editing facilities.
6) Sound-on-sound recording facility.
7) Better copying facilities.
8) A-B test without adjusting the azimuth when the tape is changed.
9) Less wow and flutter.
10) Better channel separation (track-to-track).

The most important advantages of the cassette system are:

1) Easy to operate.
2) Large choice of pre-recorded tapes.
3) Also found in the low price category.

We conclude by presenting measurements made on our TCD 330 cassette machine and the TB 10XD open-reel machine (Fig. 6). The curves tell their own story and are a fitting conclusion to this article.

Technics Model RS-1500US Stereo Open-Reel Tape Recorder



MANUFACTURER'S SPECIFICATIONS
Frequency Response: 30 Hz to 30 kHz @ 15 ips, 30 Hz to 25 kHz @ 7½ ips.
Harmonic Distortion: 0.8 per cent.
S/N Ratio: 60 dB.
Separation: 50 dB.
Input Sensitivity: Mike, 0.25 mV; Line, 60 mV at 150 kilohms.
Output Level: Line, 420 mV; headphone, 60 mV @ 8 ohms.
Wow and Flutter: 0.018 per cent W rms at 15 ips; 0.03 per cent W rms at 7½ ips.
Speed: Accuracy ±0.1 per cent; fluctuation, 0.05 per cent max.
Fast Forward & Rewind: 150 seconds for 2500 feet.
Dimensions: 18 in. (45.7 cm) W x 17½ in. (44.5 cm) H x 10 1/8 in. (25.7 cm) D.
Weight: 51 lbs. (23.1 kg).
Price: $1500.00.

The Technics RS-1500US open-reel tape deck provides features and a level of performance that should be of interest to serious audiophiles and budding professionals. The front panel of the deck, which can be operated either vertically or horizontally, is dark brown with gold lettering. The unit can handle up to 10½-inch reels and operates at 3¾, 7½ and 15 ips. At the top of the closed-loop drive system are the two air-damped tension rollers. Immediately below are the tape-position markers, and then two tape guides. Of particular interest is the single, large-diameter, 1.3 inch (33 mm), capstan with pinch rollers on each side. This direct-drive capstan rotates at a very low rate, just 3.6 revolutions per second for 15 ips tape speed. The capstan motor and its speed-dependent frequency generator are part of a phase-locked servo loop. The basic time/speed reference is a quartz crystal operating at about 4 MHz, which is divided down for the capstan speed reference. The phase locking provides tight, responsive control, and the crystal ensures accurate speed. With the tape being metered in and out of the head assembly by the single capstan, there is substantially complete isolation from external tape motion effects. The tape first passes over the ¼-track playback head and then the ½-track erase head. The turn-around idler at the bottom incorporates helpful editing marks and an inner movable reference to facilitate moving the tape from the center of either playback head exactly to the nearest tape-position marker. After the idler, the tape passes the ½-track record and playback heads before being fed out by the capstan. A switch on the face of the head assembly selects ¼- or ½-track playback. The open construction permits very easy cleaning and demagnetization when needed. To the right is the cue control which can be used for editing or monitoring during high-speed winding, and the real-time minute-second counter, which is scaled for 15 ips.

Below are the tape-motion switches, featuring a light touch, and LED indicators for *Play*, *Record* and *Pause*. The IC logic control permits switching between any desired functions with the exception that the unit will not go into *Record* from *FFWD* or *RWD* unless *Record* and *Play* are held down as the tape comes to a stop. *Pause* will stop the tape in *Play*, but sensibly its status light will not go on unless the pause is made in *Record*. In that case, recording will resume simply by pushing *Play*. Flying-start recording is easily accomplished by holding in *Play* and pushing *Record*. Below the good-sized VU meters on the left are the power switch, the pitch control which is pulled out to operate, the speed selector, and the timer-start switch. In the bottommost section to the left is a meter scale to select either +3 (normal) or +6 VU for the maximum reading. The meters have the +6 VU scale in smaller print below the regular markings. Just to the right is the mike attenuator switch which can insert a very useful 20 dB reduction of the output from high-level mikes. The phone jack mike inputs are below next to the headphone jack, which has its level controlled by the output pot.

From left to right are the mike, line input and line output dual-section pots which are friction clutched to permit channel level adjustments individually or simultaneously as desired, a worthwhile feature. The mike and line inputs, which can be mixed, have helpful settable marker rings. The output level control has a reference marker dot for a zero VU indication for a tape flux density of 185 n Wb/m (nanoWebers per meter). Lever switches to the right provide source or tape monitor selection for each channel, three settings each of record equalizations and bias, and record mode preset for each channel.

End pieces attached to the metal frame are made of particle board and the back is hardboard. Line input and output connections are made with phono jacks, all nicely paralleled. Also on the back are an a.c. convenience outlet and sockets for a remote control and for 24 V d.c. power, permitting operation from storage batteries, which could be very handy at a remote location. A lock plate prevents switching between power modes accidentally. Removal of the back cover revealed the large-diameter construction of the low-speed direct-drive capstan motor. Circuit cards had components neatly identified, and board soldering was excellent. Access to adjustments appeared to be somewhat difficult, but further disassembly was not attempted to verify this assessment.

Performance

Playback responses at both 7 ½ and 15 ips were within 1.5 dB with slightly greater deviation at the lowest frequencies. The meter indications to the specified 185 nWb/m flux level were within 0.2 VU for both channels with the output pots set to the marker dot, within possible errors in the test tape itself. A pink-noise source and a third-octave real-time analyzer were used to check the machine's basic performance with a number of tapes. While observing the playback, the EQ (equalization) and bias switches were operated to find the best combination for all three tape speeds. Record/playback responses were excellent for all tapes tried, and Scotch 206, TDK Audua, Memorex Quantum, Ampex 456 Grand Master, and Maxell UD were used for more detailed testing. At 15 ips, the response was within 3 dB from 28 Hz to 45 kHz or wider for both Scotch 206 and Memorex Quantum at levels up to 0 VU! Response with the Scotch 206 at +10 VU extended from 30 Hz to 24 kHz, a good demonstration of the excellent headroom (Fig. 1). Responses were plotted with the Scotch tape with the three EQ settings and with the Memorex tape with the three bias positions. Responses were also taken at 7 ½ and 3 ¾ ips with the Scotch 206 tape (Figs. 2 & 3). At 0 VU, results were 18 Hz to 17 kHz and 16 Hz to 9 kHz, respectively. At -20 VU, the high-frequency end shifted out notably to 35 kHz for 7 ½ and to 19 kHz for 3 ¾ ips.

Measurements of the third harmonic distortion generated in the record/playback process used a 1-kHz signal with a record level from -10 to +10 VU for 15 and 7 ½ ips with Scotch 206, TDK Audua and Maxell UD (Fig. 5). At 15 ips, the distortion was 0.18 per cent or less at 0 VU, and 0.8 per cent (the specification limit) or less at +6 VU. The distortion level was less than 2.2 per cent at +10 VU, and was down to 0.025 per cent or less at -8 VU. At 7 ½ and 3 ¾ ips the distortion figures were, respectively, 1.4 and 2.8 times the 15 ips results at most record levels. The 3 ¾ ips data was obtained for Scotch 206 only. Distortion was also measured with test frequencies from 20 Hz to 15 kHz at record levels of 0 and -10 VU. As the distortion levels were too low at -10 VU to obtain valid data at all frequencies and test speeds, the data reported here is from 0-VU tests only. Because of the unit's very wide frequency response, it was possible to measure third harmonics with test frequencies as high as 15 kHz for 15 and 7 ½ ips and 12 kHz for 3 ¾ ips (See Fig. 6). In all of the distortion tests made, there was very little evidence of other harmonics in the output.

The A-weighted signal-to-noise ratio for two tapes and 7 ½ and 15 ips ranged from 63.2 to 66.4 dBA with the manufacturer's specified 185 nWb/m +6 reference. Even at 3 ¾ ips the figure obtained with Scotch 206 was 2.7 dBA above

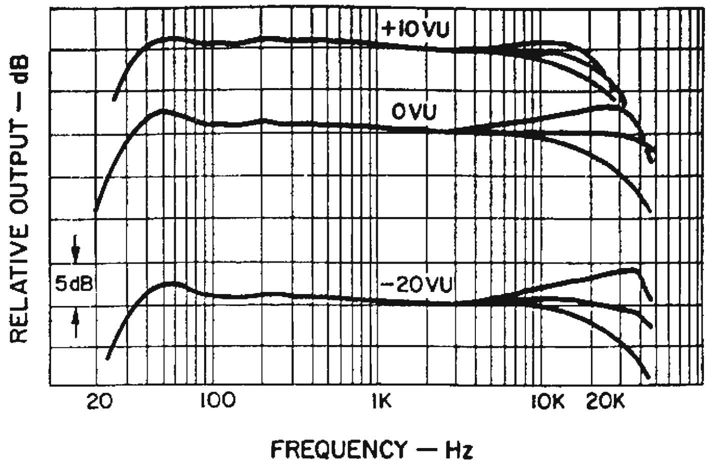


Fig. 1—Record/playback response @ 15 ips with minimum bias and three equalization settings.

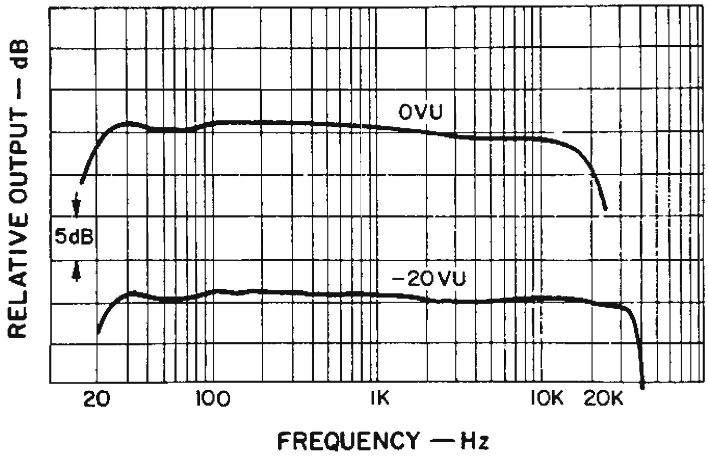


Fig. 2—Record/playback response @ 7 ½ ips with minimum bias and minimum equalization boost.

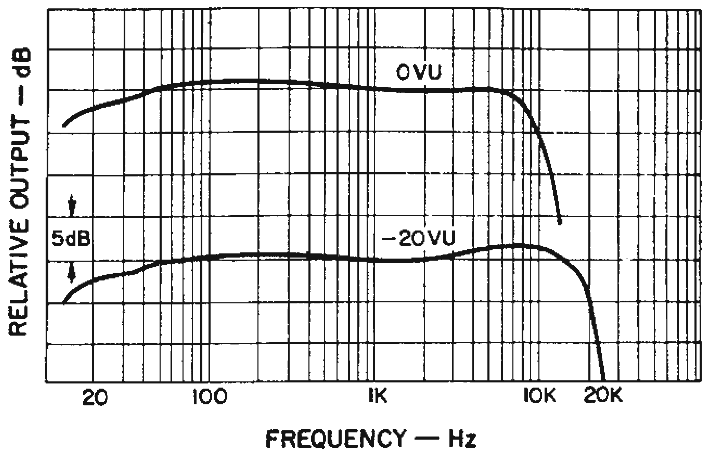
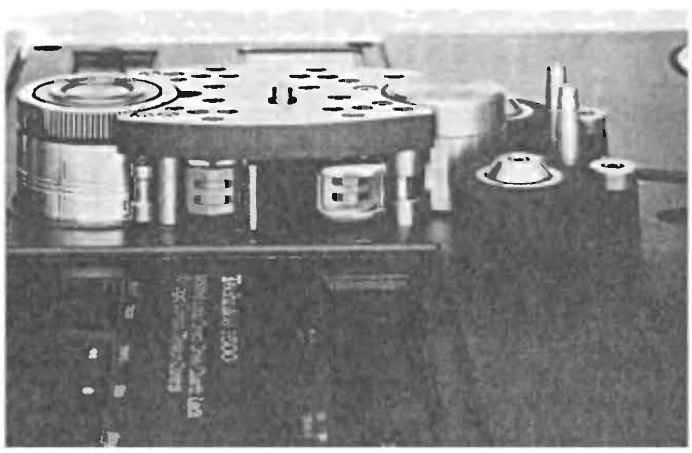


Fig. 3—Record/playback response @ 3 ¾ ips with minimum bias and medium equalization boost.



the specified 60 dBA. A two per cent distortion level seems proper for the VU meters used, which would provide ratios of 66.8 to 70.0 dBA for 15 and 7 ½ ips, and 62.0 dBA for 3 ¾ ips. Separation between tracks was at least 58 dB with a +6 VU test signal, easily better than the specified 50 dB. Erasure of the same signal was 63 to 67 dB down, depending on tape speed. Input sensitivities were 0.23 mV for mike and 59 mV for line, both slightly better than the specifications.

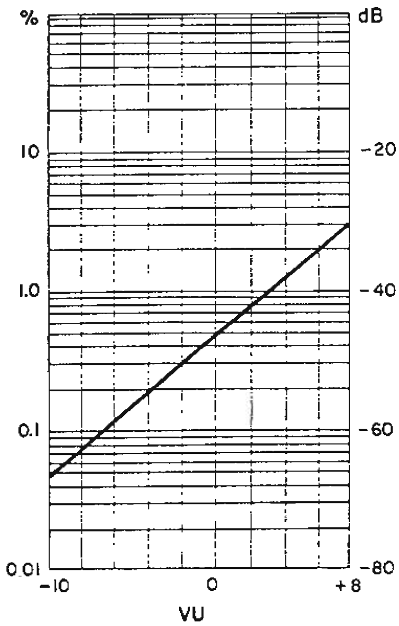
Playback of 0 VU record was exactly to spec on the right channel, 0.8 dB low on the left channel. Maximum output levels were 690 mV (L) and 735 mV (R) for the same record level. The headphone drive across 8 ohms was 67 mV, greater than specified. The meter response to a 300-millisecond burst was to VU standards, and the frequency response was down 3 dB at 27 Hz and 56 kHz. Meter scale readings were within 0.2 dB at all levels, and tracking of the channel pots was excellent.

To get a good collection of data on various speed characteristics of this excellent deck, a very stable 3160 Hz tone was recorded the length of reels of Scotch 206 and Ampex 456. The reels were then flipped and played back in opposite direction to maximize any possible errors caused by varying tape tension. The meter terminals of the flutter and drift meter were fed to a calibrated strip-chart recorder. At 15 ips the flutter was nominally 0.025 per cent DIN weighted peak throughout the entire tape run, with values as low as 0.005 per cent and occasional maximums of 0.04 per cent. At 7 ½ ips, typical figures were 0.04 per cent Wtd peak, with lows of 0.01 per cent and occasional maximums around 0.05 per cent. The manufacturer's figures of 0.018 per cent Wrms for 15 ips and 0.03 per Wrms for 7 ½ appear to be completely justified. Two other open-reel machines were measured for flutter at 7 ½ ips for comparison. One had demonstrated superior low-flutter performance in the past, but it did not

The Measurement of Tape Recorder Distortion

A plot of percentage distortion vs. output level for a typical amplifier can be flat over a considerable range with perhaps some rise at the lowest levels, and at maximum output the distortion figures increase sharply. Many plots of tape recorder distortion in the past have looked generally similar, but were in error at lowest levels because of noise effects. The fundamental-rejection type of harmonic distortion meter needs and has a wide bandwidth to measure the energy from all harmonics. This approach works well with amplifiers which have low noise. Tape recorders, relatively speaking, have high noise, and the result is that the measured "distortion" at lower levels is determined by the noise, not the harmonics.

To get accurate distortion figures in such a case, it is necessary to reduce the effect of the noise so that the discretes stand out. It might be noted here that your ears are regularly detecting signals below the overall, broadband sound level. Since April, 1975, *Audio* has been reporting recorder distortion figures that have been obtained with a spectrum analyzer. By settings its i.f. bandwidth to 10 Hz and scanning slowly, it is possible to display harmonics at levels considerably below 0 VU. Many had come to believe that recorders had distortion that was a minimum of perhaps 0.5 to 1.5

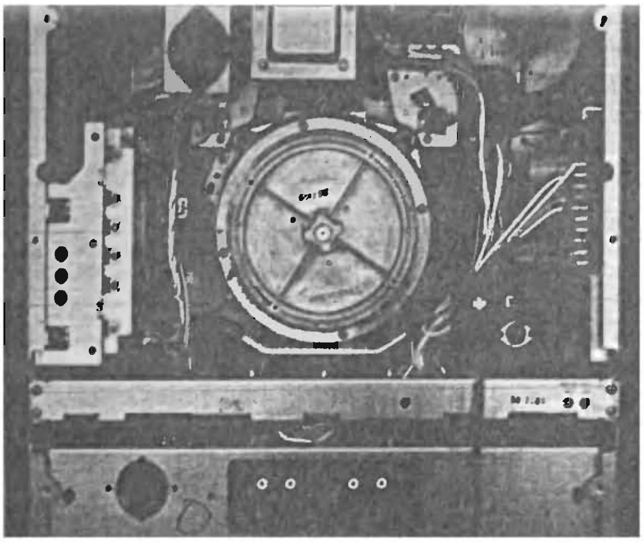


per cent. The experience of making such tests for two years has shown that the distortion can be as low as 0.1 per cent for recording at 0 VU, and possibly less than 0.01 percent at -10 VU. Actually, the distortion level is more directly related to the resultant flux level on the tape, but these figures give an idea of what might be expected from a high-performance recorder/tape combination.

The distortion products of the magnetic record/playback process normally consist almost exclusively of third harmonics. Any second harmonic is probably the result of some sort of maintenance problem: a magnetized head, a leaky coupling capacitor, or other defect. Higher odd harmonics, particularly fifth, may show up at higher record levels, but are usually of low amplitude compared to that of third. For the majority of operating conditions, the level of the third harmonic is essentially equivalent to the total harmonic distortion level. If the percentage distortion figures are plotted on a logarithmic scale against a linear scale for record (or flux) level in VU (or relative dB), the resultant curve is a straight line. The percentage distortion usually changes about ten to one, or 20 dB, for each 10 dB change in record level. This two-to-one relationship holds all the way from the noise limit at the lower end to perhaps 10 dB or more above 0 VU. Deviations from the nominal two-to-one slope can be caused by the contributions from lower-level fifth-order distortion products.

In the accompanying figure of a possible recorder/tape characteristic, the third harmonic distortion is about 0.5 per cent at 0 VU and reaches 3 per cent at +8 VU. At -10 VU, distortion is down to 0.05 per cent, just one-tenth that at 0 VU. Probable figures for distortion at levels below -10 VU can be obtained by extending the straight line and filling in the grid. To date all of the evidence indicates that the function remains straight down to the lowest levels. There is less data on very high levels, due to a general reluctance to burn out meters. It can be stated, however, that the measurements at least suggest that the straight-line character is maintained at least part way into saturation and self erasure. The evidence to date also supports a tentative conclusion that the slope determined for a test frequency such as 1.0 kHz applies to other frequencies for the same recorder/tape combination. In other words, the distortion figures for another frequency such as 200 Hz can be determined over a range of levels by applying the slope from the 1.0 kHz level data to the distortion level shown for 200 Hz on the distortion vs. frequency plot.

Howard A. Roberson



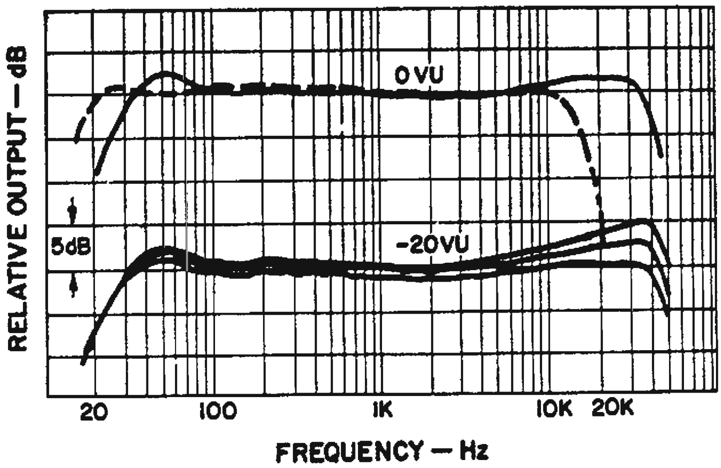


Fig. 4—Record/playback response with Memorex Quantum tape with the three bias settings at -20 VU @ 15 ips; upper trace @ 7 ½ at 0 VU at the medium bias setting.

quite match the Technics deck. The second unit was obviously inferior in this comparison, although its specifications would usually be given a very high rating. To check modulation noise, a 1.0 kHz signal was recorded at 0 VU, and the narrow-band spectrum of the playback was plotted from 500 to 1500 Hz (Fig. 7). There was an evident reduction of sideband levels with Scotch 206 as tape speed was increased. On the low-flutter "other" machine, the modulation noise was generally comparable. Then, Ampex 456 was used with the Technics at 7 ½ ips (See Fig. 8), and the reduction in modulation noise with this mastering tape was quite obvious.

Speed fluctuations were measured by playing the 3160-Hz tapes and feeding the strip-chart recorder as described above. The counter reading on playback was exactly 3160 Hz throughout the entire length of tape, and the plotted drift showed minute variations around a constant speed (frequency). The conclusion was drawn that the unit was exhibiting outstanding tape tension control and speed stability. At both 15 and 7 ½ ips, the speed variations were within 0.05 per cent of a nominal reference, with 15 ips slightly better (Fig. 9).There was no measurable variation in tape speed with line power changes from 90 to 130 V. There was no external method refined enough to prove the exact tape speed, but according to the built-in strobe lamp and the marks on the edge of the turn-around idler, the tape was running a miniscule 0.015 per cent fast. With the variable pitch control pulled out, all three tape speeds could be varied from 6.1 per cent slow to 7.3 per cent fast. Recorder starts were tested by using the drift circuit as a speed detector, with its output fed to an X-Y recorder. Overshoot in the drift circuit itself from the suddenly applied signal obscured the earlier conditions, but the plots for 15 and 7 ½ showed the relatively short time required to obtain very steady tape motion. Improved instrumentation would be required to secure more complete veritification of the recorder's excellent speed characteristics. Rewind for a 2500-foot 10 ½-inch reel was 116 seconds, well within the specification. Readings taken of the counter at timed intervals during fast wind proved that with either 7- or 10 ½-inch reels the tape speed was constant, showing excellent control of winding tension. The noise levels from the machine measured at one foot were 31 dBA at 15 ips, 28 dBA at 7 ½ ips and 25 dBA at 3 ¾ ips. At a few feet in a reasonably dead room, all readings would be less than 20 dBA; excellent low-noise operation.

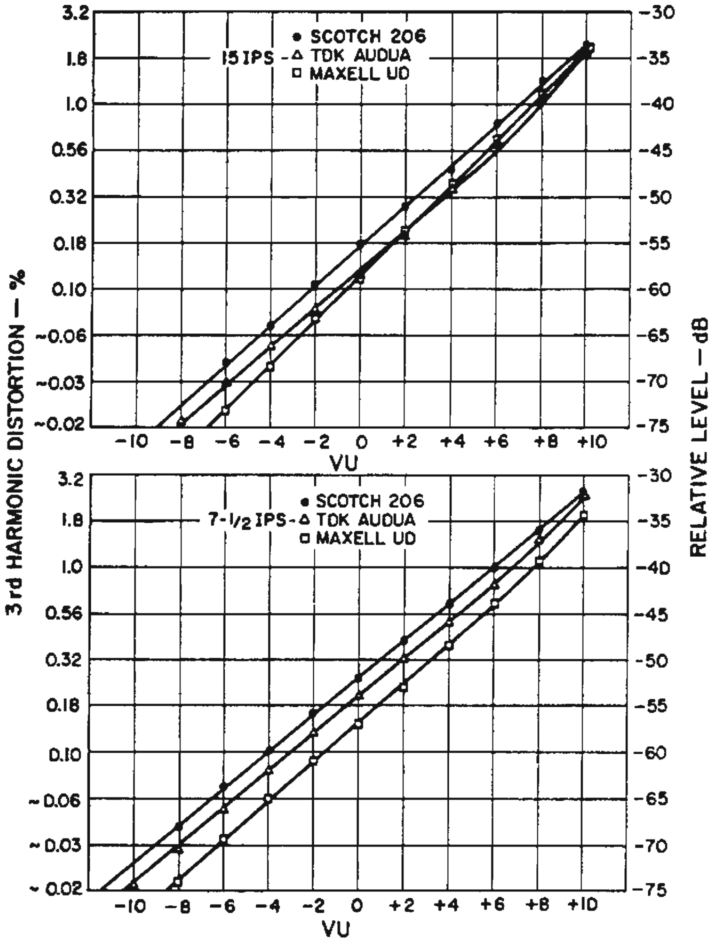
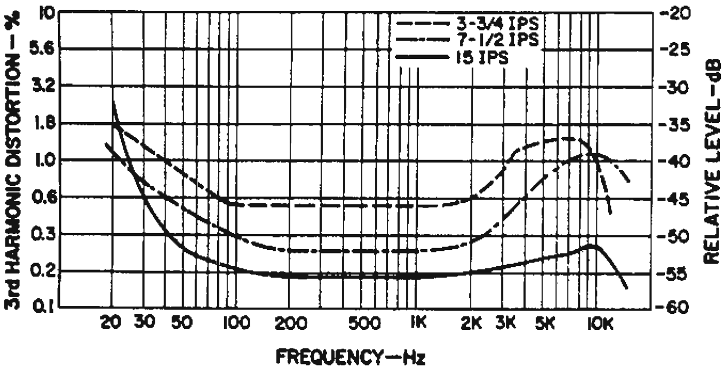


Fig. 5—Third harmonic distortion vs. record level @ 15 ips (top) and 7 ½ ips (lower) with Scotch 206, TDK Audua, and Maxell UD tapes.

Fig. 6—Third harmonic distortion vs. frequency at 0 VU with Scotch tape @ 3 ¾, 7 ½, and 15 ips.



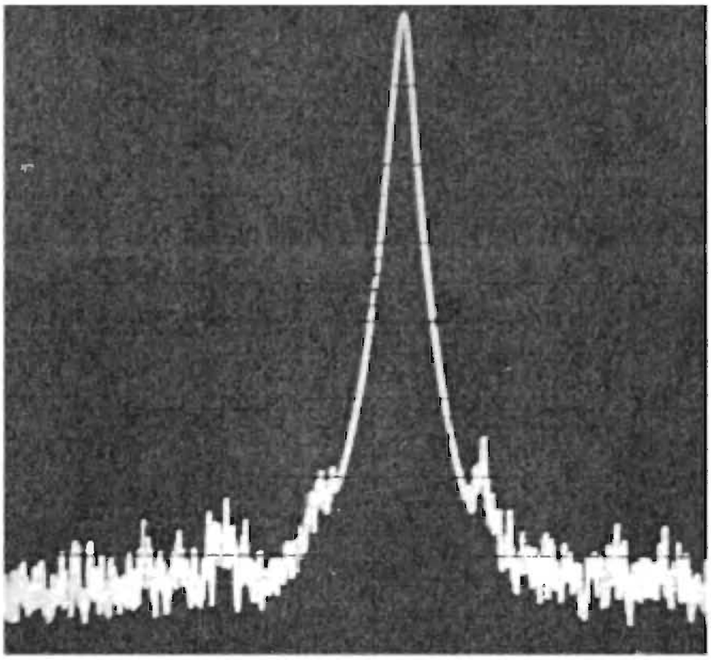


Fig. 7—Modulation noise with the Scotch 206 tape @ 15 ips on a 1 kHz signal recorded at 0 VU. The scope settings are: horizontal, 100 Hz per division to 1 kHz; vertical, 10 dB per division to 80 dB.

In-Use Tests

Threading the machine was much easier than might be judged looking at the collection of rollers and idlers, as the tape loop falls naturally around the head assembly. The feather-touch tape motion switches and the associated logic performed without error, and a thrown loop or stretched tape was never achieved in many attempts. Longer spindles on the reel turntables would aid putting on reels, particularly when the machine is vertical. The 10 ½-inch-reel adapters caused a bit of fumbling as they fit into the outside of the reels, and then this assembly had to be placed on the spindle. All of the pot knobs are of good size and have an attractive appearance. The friction clutching worked very well in keeping sections together or slipping when desired, and the marker rings would benefit from a little more knurling. The meters have good visibility, and it was easy to set levels on both channels. Although the unit has excellent headroom, particularly at 15 ips, some form of peak indication would have been helpful. The machine was operated for many hours at all three speeds, switching back and forth from source to tape monitoring. Minor modifications to the sound could be detected only at 3 ¾ ips. Superior speed characteristics, especially the low flutter, and the extended response at higher levels were judged to be major factors in providing the excellent sonic results. Switching between source and tape did not generate clicks in the recording that could be detected. *Record* on/off clicks could be heard by listening carefully, but the actual level was down to that of tape noise. A brief check was made of the operation of the ¼-track playback head.

The EQ switch provided maximum boost when it was down in position 1, minimum boost when it was up in position 3. The bias switch provided minimum bias when it was down, maximum bias when it was up. It seems that it would be more logical to most operators if lifting the levers up would cause the high frequency response to go up, the reverse of what happens with the present scheme. Comments cannot be made on the instruction book coverage in this or other areas, as it is still in process. The unit operated well in timer start, in *Record* if the record preset switches were on.

The performance of the Technics RS-1500US deck leaves

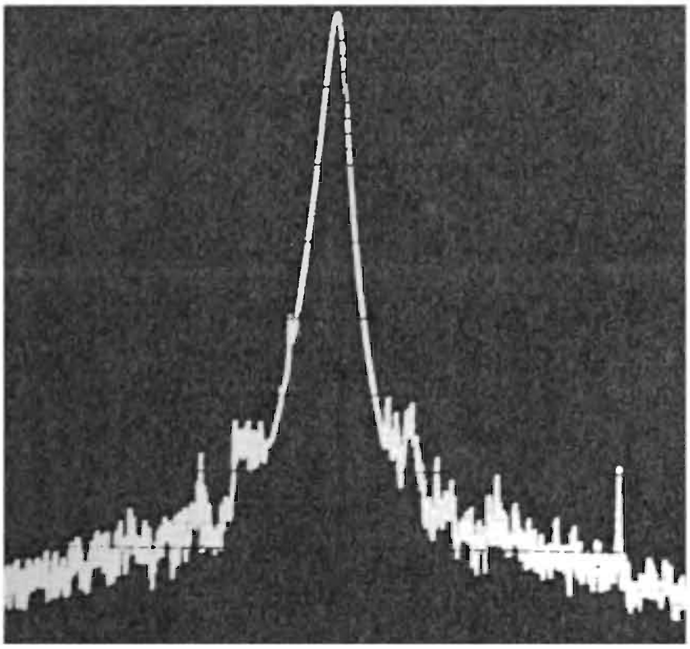


Fig. 8—Modulation noise with the Ampex 456 tape @ 7 ½ ips on a 1 kHz signal recorderd at 0 VU, with horizontal readings to 1 kHz in 100 Hz per division; vertical to 80 dB at 10 dB per division.

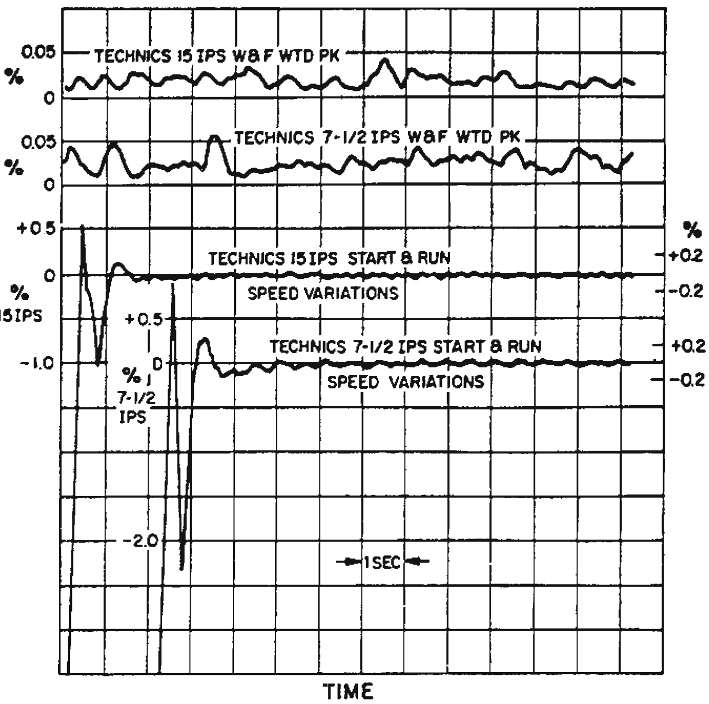


Fig. 9—Speed variations and wow & flutter at 7 ½ and 15 ips with the Technics RS-1500US reel to reel tape deck.

substantially nothing to be desired by serious audiophiles. The unit is even capable of recording the carrier channel of CD-4 systems right out to 45 kHz. It must be noted that the excellent signal-to-noise ratios are achieved without Dolby circuitry. For the professional, the major limitations would actually be in the area of input/output interfacing. Balanced line in/out could be added at relatively small expense, however, so budding professionals should consider this recorder quite competitive to other somewhat more expensive machines. Hopefully, Technics will add peak indicators to the benefit of all users. *H. A. Roberson*

Behind the scenes

Bert Whyte

A little over 10 years ago, Philips introduced their compact cassette system, which stripped to essentials, was a tape recording system utilizing tape slightly over 1/7th inch in width, which operated at a speed of 1-7/8 ips, and was enclosed in a plastic shell which could be easily inserted and withdrawn from the record/playback mechanism, thus eliminating tape threading. The cassette was intended to be an inexpensive, convenient, portable speech recorder, suitable for dictation purposes. Within the Philips company, I'm sure that even the most wildly optimistic advocate of the cassette system never envisioned it becoming a high-fidelity stereo recording medium; in terms of prerecorded cassettes and in the opinion of many people the cassette is now a viable alternative to the phonograph disc. The advent of high-energy oxide formulations, new types of high-efficiency magnetic heads, Dolby-B noise reduction, advanced solid-state record/playback circuitry ... all have contributed to the establishment of the cassette as a truly high quality means of magnetic tape recording.

The high quality, versatility, and relatively modest cost of the cassette resulted in a veritable sales explosion, with literally hundreds of models of cassette recorders on the market. While the cassette drove "low-end" open-reel magnetic tape recorders off the market, open-reel afficionados with the wherewithal supported the "high-end" recorder market and pointed out the technical shortcomings of the compact cassette. They said that cassette tape was lacking in headroom ... it couldn't handle a wide enough dynamic range ... the tape went into saturation too fast and caused distortion ... tape motion wasn't stable, because guidance was a function of the shell, and the shells were inconsistent in construction. While acknowledging all this, there were many tape enthusiasts who idly speculated on what a good thing it would be if someone could combine the best features of both systems.

Japanese Ingenuity

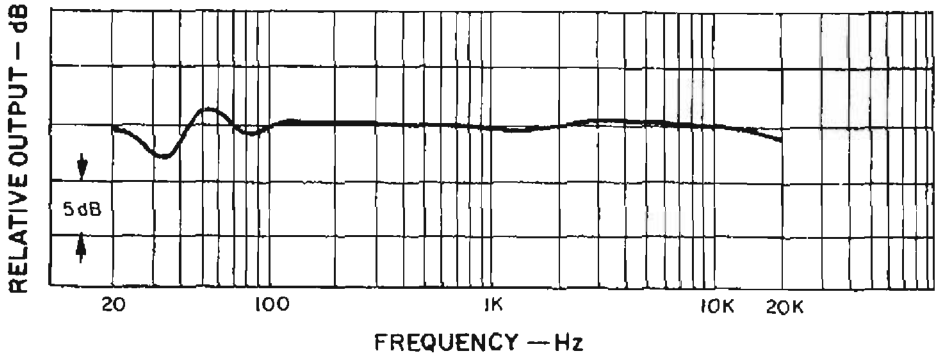
The ever-industrious, ever-ingenious Japanese evidently had just this sort of thing in mind when they announced the Elcaset tape recording system early last year. In essence, the Elcaset is a scaled-up version of the compact cassette, some 2½ times larger in size, and it uses standard quarter-inch magnetic tape, operating at 3¾ ips ... double the speed of the cassette. Besides these obvious points, the Elcaset is considerably more sophisticated than the regular cassette. For instance, in the Elcaset the tape is pulled out of the plastic shell and onto the tape heads, eliminating the guidance problems inherent in the cassette system. The Elcaset shell was *designed* from the first to accept three heads, so true "off-the-tape" monitoring is possible. Sensing notches are moulded into the Elcaset to automatically program the recorder for such things as Dolby-B noise reduction and proper bias and equalization for the three specially formulated tapes designated Type One ... a low noise/high output gamma ferric oxide, Type Two ... a ferrichrome tape, and Type 3 ... chromium dioxide tape. While the Elcaset comes in C-60 and C-90 lengths and is stereo/mono compatible, a third "pilot" track between the stereo tracks can be used for such things as slide show synchronization and pre-set program selection. With the quarter-inch wide tape, 3¾ ips speed, and special tapes, the Elcaset is capable of a much wider dynamic range, wider frequency response, less distortion, and a better signal-to-noise ratio than the cassette.

The Elcaset was a joint development of Sony, Matsushita, and Teac, and they were soon joined by JVC and Akai. There were some previews of the system for the audio press corps, who learned there were models of the Elcaset to be shown at the 1976 summer CES. As it turned out, to a limited extent, this did indeed happen. I was "button-holed" by the public relations minions of one company, who gave me very positive promises that an Elcaset unit would be sent to me in short order for my evaluation. Well, tempus fugited, I fidgited, and no Elcaset appeared. During this period, it seemed that some internecine unpleasantries were going on within the Elcaset consortium.

One of the things the group had to contend with was a certain amount of apathy and indifference to the Elcaset concept on the part of some audio dealers and some of the press. There were the usual expressions of "who needs it?"—common to many new developments in audio. These nay-sayers are entitled to their opinions, but to voice such ideas before they ever got to see or hear an Elcaset is patently unfair. If the ideas of these people prevailed, we would still be sharpening cactus needles and playing our music through "morning glory" horns!

Licensing Lethargy

Eventually, the reason for the reduced activity and nonappearance of Elcaset models from the various companies became apparent. It appears that Sony, probably the prime-mover in the Elcaset project and sole manufacturer of the special Elcaset tapes, wanted other tape companies like TDK and 3M, to pay a fee for a license



to manufacture these tapes. They are, of course, perfectly within their rights in making such a request, and one cannot castigate them for so doing. However, might we gently suggest they take a leaf from Philips' book and offer their technology free of any license fees? There is little doubt that this Philips gesture was of inestimable value in the rapid proliferation and establishment of the cassette format. As of now, negotiations are still going on, and one hopes that the issue will soon be resolved. One hopes that there may be some movement in this direction from the announcement by Teac that their Elcaset deck will soon be available. Superscope has begun to market the Sony decks, and the Sony decks are on the market in England, and have, in fact, been glowingly reviewed in the prestigious *Hi-Fi News and Record Review*. Technics demonstrated their Model RS-7500US Elcaset deck at the New York AES convention and at the recent Washington and Philadelphia Hi-Fi Shows. In December of last year, I was with a group of audio writers at the Technics plant

in Osaka, where we got a thorough rundown on this Technics Elcaset unit. Now I finally have a RS-7500US Elcaset unit and a supply of Type One and Type Two blank Elcasets. It seems that the chromium dioxide Type Three Elcasets are not yet available.

The RS-7500US Elcaset recorder is a very rugged-looking unit, in the "rack mount with large handles" configuration and the currently popular "black look." It is a front-loading unit, and there is an ingenious mechanism which automatically pulls the tape out of the Elcaset when the shell is inserted and locked in place. There is a double-gap ferrite erase head and separate permalloy record and playback heads, permitting off-the-tape monitoring. The tape drive is via a frequency generator servo-controlled d.c. motor, with a connecting belt driving the take-up reel. Wow and flutter is claimed to be 0.06 per cent rms, and while I did not test it with a flutter bridge, sustained piano chords (piano is a fixed-pitch instrument) sounded quite clean and stable. Tape motion and record/play functions are controlled by the usual mechanical "finger" leverage system. The unit has two good-sized VU meters (peak indication with an LED would be helpful), tape-monitor switch, memory rewind, and separate pots for mike/line mixing. As noted previously, bias and equalization are automatically set by the sensing notches on the Elcaset shell, and front panel lights indicate what type of tape is being used. A panel covering the three heads unscrews for easy cleaning. There is no Dolby-B noise reduction furnished in this $599.00 Elcaset deck, and this brings up a marketing point .... The Sony EL5 Elcaset deck is the same price as the Technics and has Dolby B noise reduction ... but it only has two heads and does not permit off-the-tape monitoring! Which is the more important? I personally feel the answer is self-evident ... both machines should have the combination of three heads and Dolby B noise reduction. Both Sony and Technics do have higher priced Elcaset units with both facilities. Frequency response of the Technics RS-7500US is rated at 25-18,000 Hz ±3 dB with Type One tape and 25-20,000 Hz with Type Two or Three tape.

Testing Tells ...

I have a great new device for measuring overall record/playback response with three-head tape machines. This is the United Recording Equipment Industries (UREI) Model 2000 automatic X/Y response plotter. This sweeps from 20 to 20 kHz, and a pen recorder plots the curve on log audio paper. I will be bringing you a detailed report on this unit in an upcoming column ... I am awaiting a new UREI module 2010, which can plot amplitude and frequency from test tapes and records, etc. so I can give a full report. Anyway, the curve I obtained speaks for itself!

I hooked up a Dolby 505 Type B noise reduction unit to the Technics Elcaset, and at 9-10 dB better than the claimed 63 dB S/N ratio, hiss was no problem. I played a 15-ips master with Dolby-A noise reduction through an Ampex 440C and recorded it on Type One Elcaset tape. On A/B comparison, it was almost impossible to consistently tell one from the other. The Elcaset handled the wide dynamics of Prokofiev's *Lt. Kije* suite with no strain, and the S/N was quite good. I played quite a few Elcasets I had recorded from master copies, and without exception, people who have heard them have been singularly impressed, not the least of which were some ladies, who loved the simplicity of the loading. It is early in the game, but on the basis of my experience and the fine reviews from England, the Elcaset deserves a hearing (no pun intended).

THINKING ABOUT PRINT-THROUGH

William A. Manly*

Some years ago, when Joan Sutherland had recently burst upon the musical scene and Grand Opera was new to television, I had just settled back to watch and hear a TV network broadcast of an opera starring the aforementioned lady. The credits were run and the opera was started—whereupon my enjoyment was rudely marred by an unusual occurrence: each time just before Miss Sutherland began, her actual line was *preceded* by an echo, fainter but very clear, of what she was about to sing. After she stopped, the echo was also to be heard, but it was much less audible and almost unobtrusive. Some of the instruments had audible echoes, but not all of them, and the male singers were not appreciably affected.

My professional curiosity temporarily overcame my desire to enjoy the opera, and I moved closer to the set to examine the picture in detail. No disturbance in the stability of the picture could be seen, indicating that the TV synchronization signals were not affected. Also, there was no trace of "ghosting" at all, indicating that the video was totally unaffected. The tape, however, had about the worst case of audio print-through which I had ever observed in a professional setting, but since the whole tape had gone through the same conditions, why were certain parts of the signals disturbed and not the others?

Print-Through or the Printing Effect

All these things are not as mysterious as they may seem at first. In this article I hope to relieve some of the mystery about this effect, and show how tapes can be stored so as to keep the problem below the annoyance level of audibility.

Contact transfer of signals from layer to layer occurs in wound magnetic tapes mostly as a result of temperature cycling or exposure to external magnetic fields. The transferred signal is a function of the original signal wavelength and strength, the temperatures and magnetic fields to which the tape is exposed, the time of exposure, and the time since exposure. It happens to all tapes in storage, and since the storage conditions can be controlled, the amount of printing can be controlled as well. Printed signals are an annoyance in audio tapes, but contact printing is the basis of a growing video tape duplication business.

Printed signals can be partly erased without disturbing the original signals to any great extent. Two companies showed how this could be done with an altered tape machine[1,2]. At least one company made a print-through eraser for sale[3], and another had such a device designed in the laboratory[4]. None of them were in existence very long. One problem was that recordists—both professionals and amateurs—were wary of doing anything which had even a slight possibility of erasing even a tiny part of their valuable recordings! Another was that tape materials underwent improvement and better storage conditions were employed, thus relieving the problem to some extent. Even with the improvements, it can certainly still occur, and the magnitude of annoyance depends strongly on how your tapes are stored.

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Two Kinds of Printed Signals

There is a decided difference between the printed signals created by thermal effects and those caused by the impression of external magnetic fields (or other causes). The thermally-caused print is unstable with time and is easily erased, while the magnetically-caused print is almost as stable as a recording made on a machine and does not erase much more easily. It is thus much more important to keep magnetic fields away from your tapes than to keep them thermally comfortable. It requires only a fraction of the magnetic field to cause magnetic transfer that it does to erase signals already on the tape, so preventing magnetic transfer of signals will also ensure that you will have no other problems caused by stray magnetic fields.

Time and Temperature are the Enemy

Figure 1 is a plot of the print-through vs. time of a tape at constant ambient temperature (solid line). The dotted lines are the time plot of the printed signal amplitude after the original signal (master) is removed from the vicinity of the part of the tape having the signal printed on it (slave). The information for this figure is taken from several of the first papers on print-through [5, 6, 7]—the tapes were of 1950 vintage, and the numbers may be somewhat exaggerated for the present day. In addition, different tapes will have different slopes and positions of the lines, so the numbers should be taken as indicative of the effect and not to apply to any particular tape or situation.

Print is Log-Linear vs. Time

In Fig. 1 and all other plots of the printed signal, the printed signal level is referred to the master signal level in dB. The reason that we can do this without stating the master signal level is that print-through is dB-linear, i.e., it doesn't

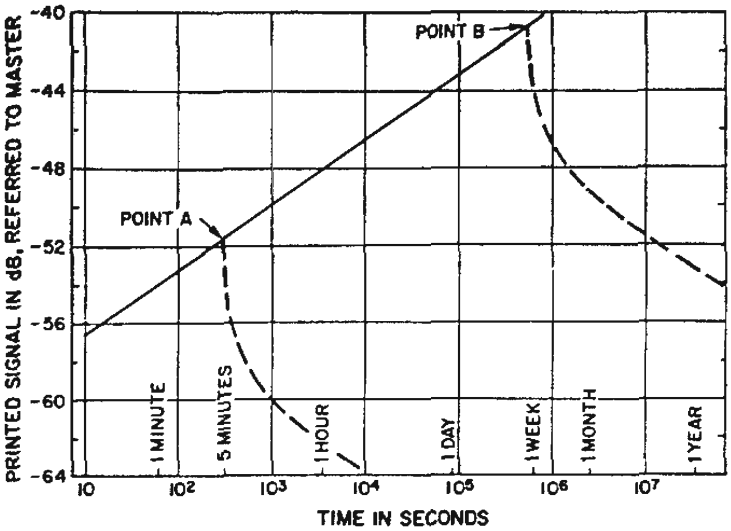


Fig. 1—Print-through vs. time at a constant ambient temperature (solid line). Dotted lines show the time decay of the printed signal after contact is lost between slave and master tapes. (Data from Lippert[5], Daniel and Axon[6], and Westmijze[7].)

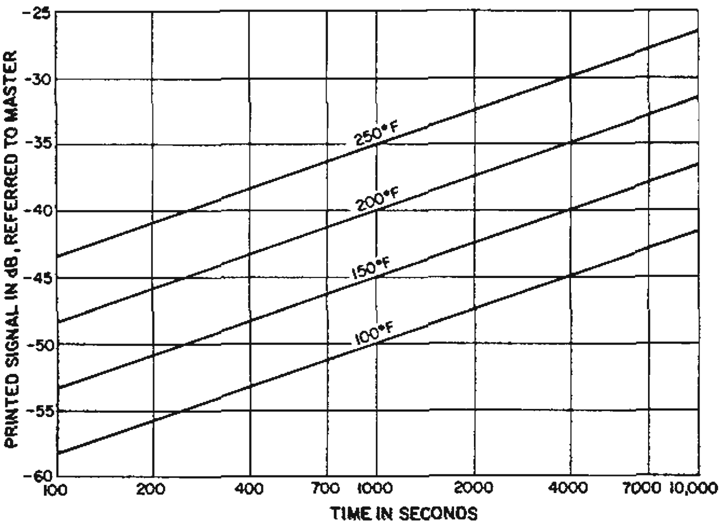


Fig. 2—Increase of printed signal with time, with curves shown for four different temperatures. Data from Daniel and Axon[6], Westmijze[7], and Johnson[9].

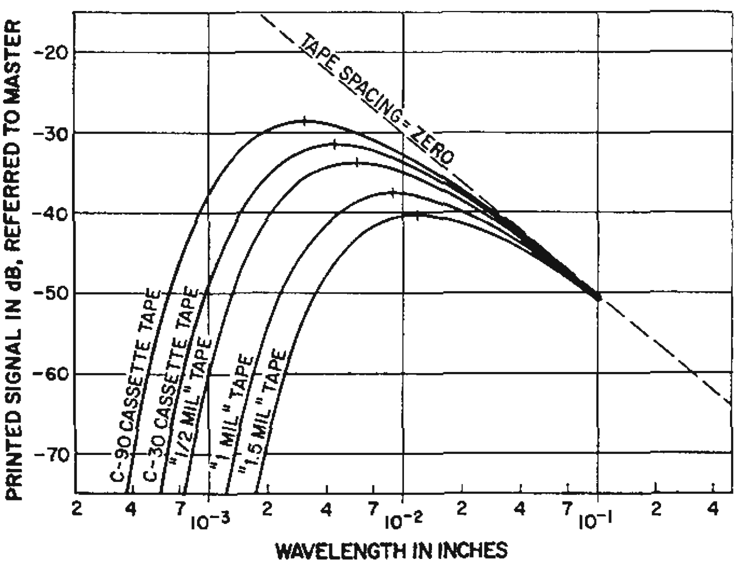


Fig. 3—Printed signal vs. wavelength for a number of different tape thicknesses. Data from Daniel and Axon[6].

matter what the master signal level is, under the same conditions the printed signal level will be the same number of dB below it. A dB (decibel) is a logarithm, and you will note that the time is also plotted logarithmically. Such a *log-log plot* is a very powerful method of simplifying things so that laws or rules of behavior can be perceived in a glance. One must use caution in interpreting such a plot, however, since there is a strong temptation to simplify things too far.

Masters and Slaves

Two tapes are placed in contact. The master has a signal on it, while the slave has been erased. The master could simply be one turn of tape wound on a reel, and the slave is the adjacent turn (either outside or inside). After they have been in contact for five minutes, they are taken out of contact (point "A" on Fig. 1). The printed signal is then measured at intervals, and the curve dropping down from "A" is plotted. Note that the time is not restarted at zero when the slave is removed from the master, but it keeps increasing. If the experiment is repeated, but printing process is allowed to go for a week before it is interrupted, the change is made at point "B." The shape of the resulting decay curve is nearly the same, but there is a large practical difference. In each case, there is an immediate drop of 2 dB or so, and then the rate slows. In the decay curve starting at "A," there is a 6-dB drop in about three minutes, but it requires nearly a week for the decay curve starting at "B" to drop by 6 dB. The reason for this is that time is changing about 2000 times as fast at "A" as it is at "B." If those decay curves are replotted starting at zero (or near zero) when the slave is removed from the master, the shape of the curves is so different that they hardly resemble each other. In each case, the final rate of decay (after separation) is about the negative of the rate of print signal increase (before separation).

There is some advice[8] about print-through on tapes which says that one should rewind tapes about 10 to 15 minutes before they are to be played, the purpose being to let the print signals decay previous to playing. This has an element of truth to it, but only for printed signals which are just a few hours old. If printing has been in progress a long time, decay also takes a long time. If tapes are to be in extended storage, they should be rewound once every few months (or once a year) to allow the printed signals to decay. It is better to wind just once, so that the old *inside* end is now the *outside* end. This prevents the tapes from being wound back into exactly the same geometrical arrangement as before (actually, there is not much chance of this when both ends are free from attachment to the reel). When rewinding, fast wind is satisfactory insofar as print-through prevention is concerned, but the pack will be smoother, with less distortion to the tape, if the wind is done under normal playing conditions.

Log-Linear vs. Temperature

Figure 2 illustrates the simple relationship of printed signals vs. time, with temperature as a parameter[6,7,9]. For any given set of temporal conditions, the logarithm of the printed signal is a linear function of the temperature. This particular tape has print-through which increases about 5 dB for every 50° F. The slope vs. time is somewhat different from the tape whose characteristics are shown in Fig. 1. Different tapes have different slopes and also different spacing between the isothermal lines (lines of constant temperature), but most are not too far away from the one whose characteristics are shown.

Thus, another rule: Letting your tapes get hot is a definite detriment. The practice of letting cartridges and cassettes stay in a car in the hot summer sun is especially bad. I keep mine in a carry-case and remove them when I leave the car for any length of time during hot days. The print-through caused by exposure to heat remains when the tape is brought down to room temperature, and it is considerably more stable than print obtained near room temperature.

Geometry

There are some geometrical effects which are quite important. It seems logical that the printed signal would be strongest in the adjacent layer of tape, and this is true. However, not obvious at all and difficult to explain is the fact that the printed signal which comes off the reel in the layer *before* the master signal (pre-print) is stronger than the printed signal which comes off the layer *after* the master signal (post-print). Since the printed signal is strongest in the closest layers, one might also guess that a thin tape would have more print-through than a thick tape (since the nearest slaves are closer), and this is true. Lastly, since *everything* concerned with tape recording seems to be a function of the wavelength of the signal on the tape, we are not really surprised to find a strong functional relationship of the print to the wavelength. The signal wavelength (λ) and the head-to-tape speed (S) are related by the signal frequency (f):

$$\lambda = S \div f \qquad (1)$$

so, given any two of the three, we can find the other.

Figure 3 shows the wavelength effects on the printed signal for a number of different tapes[6]. The position of the curve on the plot is controlled by the overall thickness of the tape, which is taken as follows:

Tape	Overall Thickness
"1 ½ mil"	1.9 mils
"1 mil"	1.4 mils
"½ mil"	0.9 mils
C-30 cassette	0.7 mils
C-90 cassette	0.5 mils

Individual tapes will differ somewhat from these values, but this will not alter the conclusions to be drawn. The peak of each of these curves occurs at a wavelength of:

$$\lambda = 2\pi d \qquad (2)$$

where λ is the wavelength, as before, and d is the overall thickness. The tick on each of the curves indicates this peak value.

Ear, Ear

As far as the tapes are concerned, this is about the whole story on the wavelength response of the printed signal, but as far as the ear is concerned, we're not finished. The ear hears frequencies, not wavelengths. Using equation (1), we can construct a table of frequencies where the maximum print-through occurs as a function of the various tape speeds used (Table I). The ear normally hears print at a moderately low level. At such a level, the loudness response of the ear is approximately ± 3 dB from 500 Hz to 6000 Hz, and the most sensitive at about 3000 Hz. With this in mind, one can look at Table I and see just where the problems are. Obviously, no one should ever use "½ mil" tape at 15 ips. Equally obvious is the fact that "1 ½ mil" tape at 1.875 ips will not cause trouble. Commercially recorded tapes are usually on 1 mil base, with reel-to-reel tapes usually running at 7.5 ips and cartridges at 3.75 ips. Note that all but one (2632 Hz) of the frequencies listed are within the fundamental range of the singing voice and also many of the orchestral instruments.

Table I—Maximum print-through frequencies as a function of tape speed.

Tape	Head-to-Tape Speed				
	1.875	3.75	7.5	15	ips
"1½ mil"	158	315	630	1261	Hz
"1 mil"	213	426	852	1705	Hz
"½ mil"	329	658	1316	2362	Hz
C-30 cassette	426				Hz
C-90 cassette	605				Hz

Working from the 40 phon loudness contour of the Fletcher-Munson curves, I have calculated the actual loudness level of the peak printed signal for each of the cases in Table I, and they are listed in Table II. The least objectionable case is used as a reference, so the larger the number, the worse the print-through (for any set condition of the time and temperature cycling). With the levels in dB, they can be compared to each other by subtraction. For instance, a "1 mil" tape cartridge at 3.75 ips is 4 dB better than a "1 mil" recorded tape at 7.5 ips.

Pre-Print and Post-Print

Some old reel-to-reel tape machines wind their tapes with the magnetic material to the outside of the reel. This is known as an "A" wind. Almost all modern machines wind their tapes with the magnetic material to the inside of the reel, and this is known as a "B" wind. All cassettes and cartridges use an "A" wind so as to expose the magnetic material to the heads (which are on the outside). All of the material here concerning pre-print and post-print assumes a "B" wind. The effect of having an "A" wind is to exchange everything said about the pre-print with that said about the post-print.

Figure 4-A shows some of the magnetic field lines coming from a bar magnet which is fairly wide (in the direction perpendicular to the page). This bar magnet resembles a long wavelength signal on a piece of magnetic tape ("long" is with respect to the coating thickness of the tape). The two arrows, labeled "H" and "V," designate the "horizontal" and "vertical" directions. In general, each field line at any point, consists of both horizontal and vertical components. The horizontal and vertical components of the magnetic field lines are shown separately, and this separation is crucial to understanding the pre-print and post-print differences. Note that the lines nearest to the center of the magnet are mostly horizontal, while those nearest the ends of the magnet tend to have stronger vertical components.

In Figure 4-B, three of the outer layers of tape wound on a reel are shown. The middle layer (the master) is the only one containing a recorded signal. It has the bar magnets (long wavelength signal) laid end-to-end, and this accentuates the effect of the vertical field lines being near the ends, and the horizontal field lines being near the magnet centers. Only the strongest parts of the vertical lines are shown. If the con-

Table II—Loudness levels for the cases in Table I, calculated from the 40-phon loudness contour of Fletcher & Munson.

Tape	Head-to-Tape Speed				
	1.875	3.75	7.5	15	ips
"1½ mil"	0	7	14	15	dB
"1 mil"	6	14	18	18	dB
"½ mil"	13	20	21	24	dB
C-30 cassette	20				dB
C-90 cassette					dB

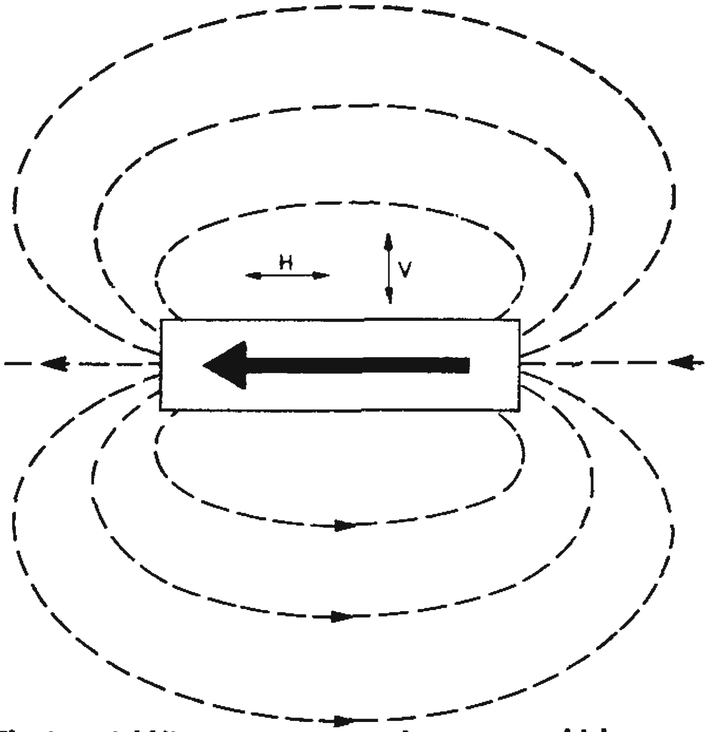


Fig. 4A—Field lines coming from a bar magnet which is wide in the direction perpendicular to the page.

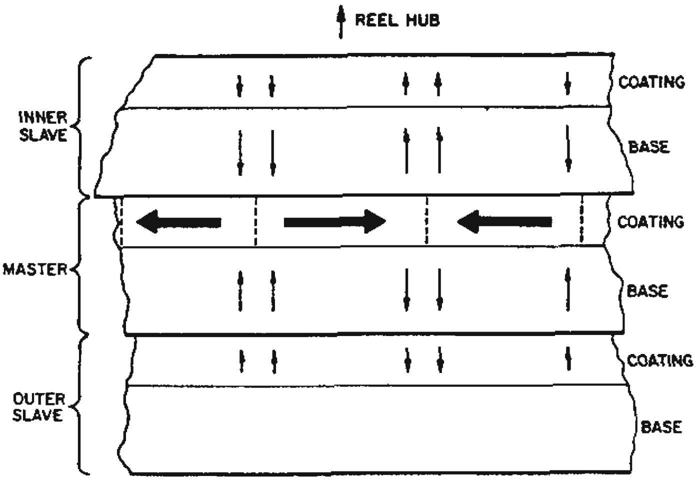


Fig. 4B—Master tape with long-wavelength signal, showing vertical field components in two adjacent slave tapes. After Daniel[11].

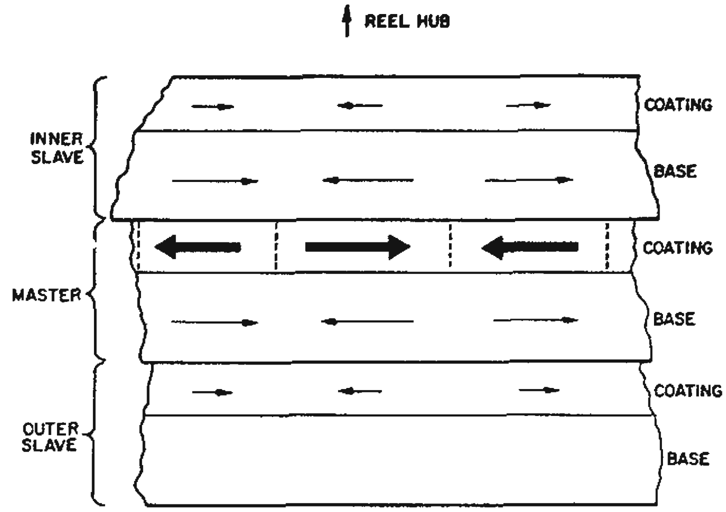


Fig. 4C—Master tape with long-wavelength signal, showing horizontal field components in two adjacent slave tapes. After Daniel[11].

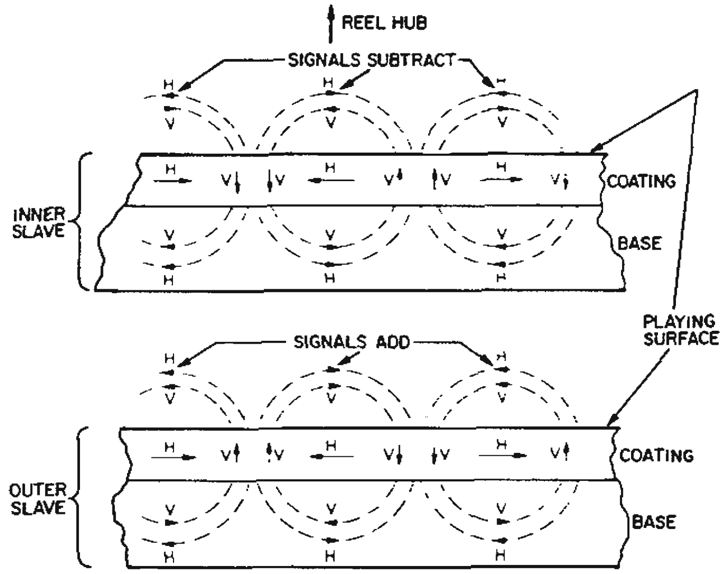


Fig. 4D—Slave tapes with master removed. Printed vertical and horizontal signals are shown, with field lines from the vertical and horizontal signals shown separately. Note how signals add on the sides closest to the master and subtract on the sides away from the master. After Daniel[11].

ditions are such as to cause printing, note that the front or playing side of the outer slave is printed from the back side of the master, while the back side of the inner slave is printed from the front of the master. Since the front side of the master was nearer the head when the signal was recorded, and thus has a stronger signal, a difference in the two printed signals is introduced, which should depend on the shape of the field from the record head. This difference turns out to be small, on the order of a dB[10, 12] and is obviously not the cause of the difference between the preprint and the post-print, which sometimes is as large as 12 dB[11, 12].

Figure 4-C shows the same situation, but with the strongest part of the horizontal field lines shown. Again, the field lines in the slaves can be caused to print a signal there.

We assume now that the signals have been printed on the slaves, and in 4-D we remove the master and show the printed signals, along with the field lines *they* produce. The field lines from the horizontal printed signals are shown separately from the field lines coming from the vertical printed signals. Actually, at any point the field lines add (or subtract) vectorially. The shape of the field lines from both H and V signals are similar. Mostly subtraction occurs near the playing side of the inner slave, and mostly addition occurs near the playing side of the outer slave. Thus, the pre-print, which comes off the outer slave, is stronger than the post-print.

The above is modified by several things. For one, some tapes are made so that vertical signals are printed poorly, if at all. In this case, the two printed signals are nearly equal. If the printing can be done equally in both H and V directions, the post-print nearly disappears, and the pre-print increases by about 6 dB. Another is that the actual master signal is not as simple as drawn—real recorded signals change direction and amplitude along their length. A third goes back to something previously stated: In case the tape is wound with an "A" wind, everything is reversed, and the post-print is stronger than the pre-print.

I have used primarily the explanation given by Daniel[11] for this pre-print/post-print explanation. Greiner[10] was substantially correct in his explanation, but his diagrams were quite complicated. Apparently, neither Daniel nor Rhodes[12] were aware of Greiner's somewhat obscure paper when they wrote their reports.

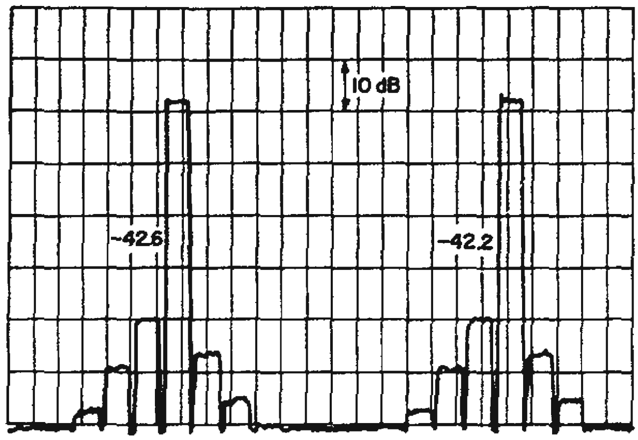


Fig. 5—Typical chart recording of a print-through measurement. Vertical scale is in dB, and horizontal scale is time.

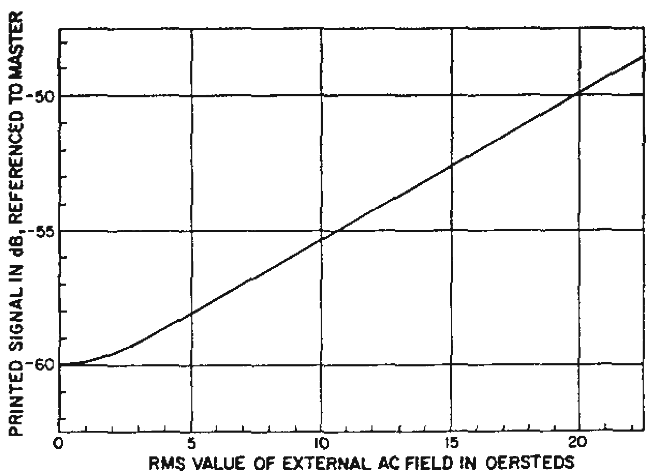


Fig. 6—Printed signal due to a superimposed a.c. field. (Data from Daniel and Axon[6].)

Winding Tension

One might wonder whether the tape winding tension has any effect. Daniel and Axon[6] reported that the tension increased the short-term print-through, but when the printing took place as long as overnight, there was essentially no difference between high and low tension. The time decrease of the printed signal after the slave was removed from the master was also affected, with the higher tension winds taking longer to fade, and fading away stops at a higher level, thus showing more stability. This tells us that print-through caused by strain is different from that caused by thermal effects. Rhodes was able to cause print-through by pressing master and slave between rollers brought together by an hydraulic press. The time of contact was a small fraction of a second. Essentially, high tension or high pressure in the tape pack will cause a small increment of print-through, but this is overwhelmed by thermally-caused print in a matter of a few hours. Nevertheless, the strain-caused printed signal is more stable than the thermally-caused print, and some remnant of it can be observed as the thermally-caused signal fades away.

Standard Print-Through Measurements

Print-through measurement is normally done on a few feet of tape wound on a 4 ½ in. diameter precision (metal) hub. The tape is thoroughly erased before starting. The "worst case" frequency is calculated by:

$$f = S \div 2\pi d \qquad (3)$$

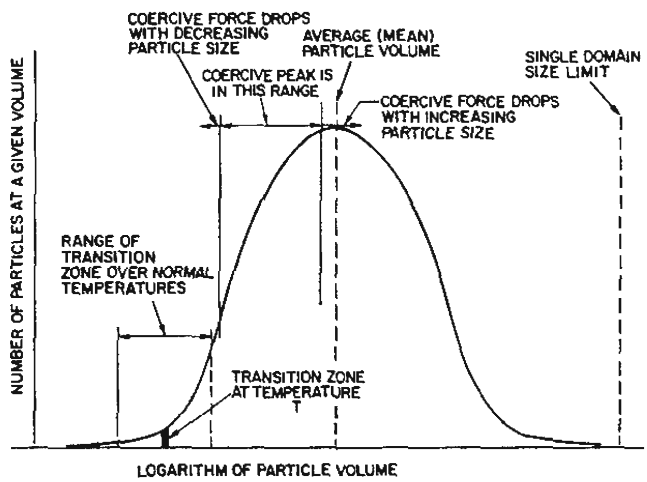
which is a combination of equations (1) and (2); the letters represent the same quantities as before. The tape speed S is normally kept high (7 ½ or 15 ips), so that the decay time before measurement is kept to a minimum. A few turns of tape are wound on the hub, then a signal of a frequency given by (3) is applied for a time slightly less than that required for the hub to turn one revolution. About 15 or 20 more turns of tape are put on, and the procedure is repeated once or twice more. The tape is then cut, and the piece on the hub carefully wound onto another hub, so that the tape coming off will be going in the same direction as when it was originally recorded. The signal is normally at the 1 per cent (nominal) distortion level, and it is a sine wave. The standard heat treatment is for four hours at a temperature of 150° F, and the treated hub is cooled to room temperature before measurement. The signal from the tape is run through a bandpass filter to improve the signal-to-noise ratio, and the tape is played back at the original recording speed. The output from the filter, which is set at the proper frequency for the tape under test, is sent to a chart recorder with a logarithmic scale. The chart recorder normally has a dynamic range of at least 80 dB, so that all signals, including the master signal and three or four post-print and pre-print signals, can be seen on the same scale. Repeatability of measurements is normally about ± 1 dB. Figure 5 shows a typical measurement from my lab. The pre-print, which is the one usually quantified, is a mediocre −42.4 dB (average for two measurements) for this tape, and the post-print is some 6 dB less than that. The tape noise caused that wiggly line just below the bottom line of the chart.

Printing by External Magnetic Fields

Exposure to magnetic fields can cause some relatively enormous print-through. Fortunately, the fields required to produce such printed signals are large compared to the fields in which a reel of tape is likely to be immersed. In contrast to the temperature-caused print-through, which uses a mode of magnetization called *thermo-remanent magnetization*, the print-through caused by a.c. magnetic fields uses a type of magnetization called *anhysteretic magnetization*. This is also the type of magnetization used by most audio recorders. There are a number of descriptions of this type of magnetization in the literature, but one written specifically for hobbyists was published in 1976[13]. Figure 6 shows how print is influenced by the presence of an a.c. magnetic field. The fig-

continued on page 81

Fig. 7—Typical size distribution curve of fine particles as used in magnetic recording tape, showing position of transition zone for normal operating temperatures.



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ure is redrawn from the paper by Daniel and Axon[6]. They stored the tape for a total of five minutes, during two minutes of which it was rotated in an a.c. field of the value shown. The numbers are the rms values, the peak fields being higher by a factor of 2. After a field of about 5 oersted is reached, the level of the printed signal in dB is nearly a straight-line function of the applied a.c. field, with a fairly high slope. The field required to produce audible print-through is greater than will normally be found in the home, and can easily be avoided. Table III gives typical fields from various sources, and shows that a little care is all that is necessary to avoid problems from magnetic fields.

D.c. fields can give print-through too, but are not nearly so effective as a.c. fields. The earth's field is about half an oersted in intensity, negligible where magnetic recording tape is concerned. In giving values for magnetic fields, gauss and oersted are interchangeable if the field is in air or vacuum. Gauss is the name for the unit of magnetic flux density, and oersted is the name for the unit of magnetic field strength. Both these terms are outmoded by the newer international units, but both are still in commercial use in the U. S.

Unpuzzling the Print Effect

What causes print-through? This was a considerable puzzle for a while. It was well-known that a magnetic material could be magnetized by heating it above its *Curie temperature* (the temperature above which it is not ferro-magnetic) and cooling it in a magnetic field. The only trouble was that the Curie temperature (T_C) of iron oxide was known to be so high that the tape would have been destroyed. A number of magnetic "after-effects" had been described for bulk materials, but they didn't seem applicable to the fine particles used in tapes. Then, in 1959, two researchers from the General Electric Research Laboratory wrote a paper about fine magnetic particles which gave the essential clues to what was going on[14]. They were working on permanent magnet materials, and did not mention magnetic tape print-through in the paper, but gradually all the magneticians working in magnetic tape recording realized that the answer was here.

The explanation goes something like the following. All bulk ferromagnetic materials have their magnetism arranged in small volumes called *domains,* all about the me size. If the material is completely magnetized, all the magnetic vectors from all the domains point in the same direction and add together. If the material is demagnetized, the magnetism is not removed, but the directions of the domains are randomized, so that they add up vectorially to zero for the whole piece of material. In fine particle magnetics, the particles are so small that there can be only one domain per particle. These particles are known, interestingly enough, as *single-domain* particles. A single-domain (S.D.) particle cannot be demagnetized, though a collection of them can. The collection is demagnetized by randomizing the magnetic vector directions (the directions can be changed, or "flipped," in the particles). Single-domain particles are normally used in magnetic recording tapes.

The temperature scale for this discussion is the *Kelvin* or absolute temperature scale. A degree Kelvin is the same size as a Celsius degree, but the zero point is at absolute zero, which places the freezing point of water at about 273° K, and room temperature at about 293° K. There is a quantum of energy associated with the Kelvin temperature by the following equation:

$$E = kT \qquad (4)$$

where E is the energy quantum, T is the Kelvin temperature, and k is Boltzmann's constant ($k = 1.38 \times 10^{-16}$ erg per degree). An erg is a very small amount of energy—it takes about 2000 ergs to lift a penny one centimeter! Equation (4) defines an amount of energy which is the average amount of energy possessed by a single gas molecule when the gas is at temperature T (in degrees K). In a solid, the energy causes vibrations in the crystal lattices; in a suspension of colloidal particles in a liquid, it causes Brownian motion of the particles. In a resistor in an electronic circuit, the noise voltage ("Johnson Noise") is related to kT in a simple way. As you can see, kT is a very fundamental quantity of energy.

Getting back to our S.D. particles, each one of the particles has a magnetic energy associated with it which is easily calculated and which depends on the material and the volume of the particle. As the particles get smaller, the magnetic energy gets smaller, going down with the volume. At some point where the magnetic energy is in the same range as the thermal energy, the particle becomes magnetically unstable, and its magnetic vector begins to wander around randomly. Bean and Livingston[14] say that this instability transition occurs at about the point where the magnetic energy is about 25 times the thermal energy. If the particles are much larger than the transition size, they are stable and will not respond to small applied fields. If they are much smaller than transition size, the magnetic vector is moving at random and a small applied field will only influence them in a statistical manner. In the transition size range, most will follow a small applied field. If the temperature then drops, some of the particles will become stable with their magnetization vectors pointing in the same direction as any applied field.

This last is what happens in thermally-caused print-through. At any average temperature, there are microscopic random temperature variations. Some of the particles are always moving into the transition range, and some are always moving out of it. With a constant applied field, as would occur in a signal-containing tape wound on a reel, more and more of the particles become aligned with the field as time goes on. Most of them be-

Table III—Measured fields from various magnetic and electronic equipment.

Equipment	Magnetic Field, Oe	Distance from Unit, Inches
B-H meter	5	24
	2	30
Large 6000 Oe tape eraser	2	12
Tape head demagnetizer	2	4
Tape eraser (Ampex)	5	9
	2	12
Isolation transformer (instrument)	1	3
Vacuum cleaner (shop)	1	2
Fluorescent lights (overhead)	1	6
Large 12 lb. magnet (Edmund Scientific)	5 (d.c.)	9
	2 (d.c.)	12
Fluorescent light transformer (desk)	1	6
Electric drill (1/3 hp)	2	3
Lightning (250,000 amperes calculated)	656 (transient)	30

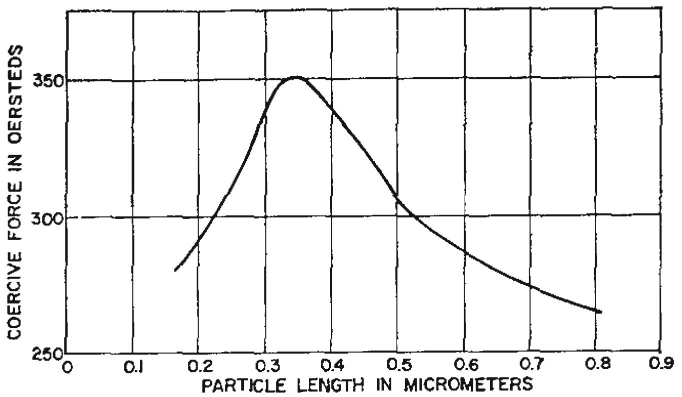

Fig. 8A—Coercive force of gamma ferric oxide particles as a function of average particle length. (Data from Tochihara, et al[15].)

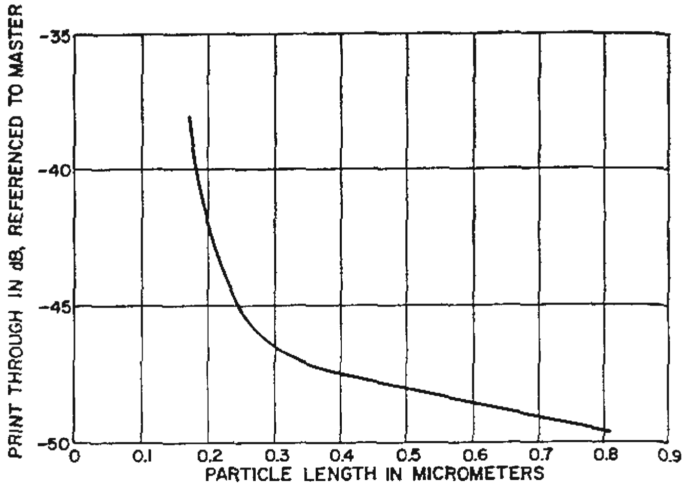

Fig. 8B—Printed signal of gamma ferric oxide particle magnetic recording tapes, as a function of the average particle length. (Data from Tochihara, et al[15].)

come aligned at first, then fewer and fewer as time goes on. The ones that become aligned later result from extreme temperature fluctuations and are thus farther into the stable range than those aligned at first. Thus, the initial print is easier to get rid of than the later print, and this is exactly what is observed and plotted in Fig. 1. In print-through testing, the average temperature is deliberately raised, and many more particles are passed through the transition range. As the temperature drops, the moving transition range leaves behind a trail of particles magnetized according to the applied field.

Please note that in this explanation I have used several words which give a picture of physical movement. Very little actual movement takes place. Only the magnetization vectors rotate—not the particles, and the transition range is moving through the particle size distribution—not the volume of the tape coating.

Nature's Own Tape Recorder

It is interesting that (with thermo-remanent magnetization) a record of the earth's magnetic field over the ages has been left in rocks as they cooled down after being formed from the melt. In the center of the Atlantic Ocean, the rocks are formed in a trench from molten slag coming up from the center of the earth, and they spread out in both directions (east and west). There is a recorded signal left on the ocean floor extending hundreds of miles, showing that the earth's field has reversed itself many times. Nature actually invented the tape recorder!

Printed Signal Amplitude

Now let's look at why the signals are the size they are. Figure 7 shows a typical size distribution curve of particles used in magnetic tape. This type of distribution is called *log-normal*, and most small particle distributions are like this, no matter how they are made. The particle length corresponding to the mean particle volume is, for most oxides, from 10 millionths of an inch to 40 millionths of an inch. The actual width of the transition zone is very small, and only a few particles are involved at any given time. The transition zone moves over a larger area during normal temperature variations, but for most materials, the total volume of particles involved (the area covered by the moving transition zone) is only about 1% or less of the total particle volume (the total area under the curve). A -40 dB print-through would indicate that 1% of the particle volumes were involved, and a -60 dB print would indicate that 0.1% of the particle volumes were involved. It is now obvious how to make a low-print tape: (1) Use a material with a narrow particle size distribution; (2) Remove most of the small particles; (3) Use a material with a transition zone located far to the left; (4) Use a material with a larger mean particle size, or (5) any combination of these. As with any engineering decision, there are tradeoffs to make in any case chosen.

Coercing Particles to Flip

The coercive force of a magnetic material is a measure of the size of the applied field that it takes to cause the material to change its magnetization. For a particle, the coercive force is the field just required to cause the particle to "flip" its magnetization vector from one direction to another. All else being equal (such as the particle shape), the coercive force goes up as the size of the particle goes down, until just before the transition zone is reached when the coercive force goes through a peak and drops to zero inside the transition zone. Some of the tiny particles near transition-zone size are influenced by small fields, as is apparent from Fig. 6 (fields so much smaller than the material coercive force theoretically should not change the magnetization of the material). A small number of larger particles may be influenced by their neighbors in the coating so that they are just about ready to flip, and all it takes is an additional push from a small field to make them go. If this happens, the magnetically caused printed signal will be difficult to erase, because the larger particle has a coercive force near the average of the material, and a small field now has no help in pushing it back the other way. Because of this effect, and the fact that the magnetic fields are flipping particles farther and farther away from the transition zone with increasing applied a.c. field strength, one should not expect to find strong fading of the magnetically printed signal after contact with the master is lost, and this is just what Daniel and Axon[6] reported.

Figure 8-A shows how the coercive force of iron oxide particles go through a peak as the particle size is changed, and Figure 8-B shows the rapid increase in print-through as the average particle size goes below the value for peak coercive force[15]. The length given is the average length, and the particle size distribution extends down into the transition zone in each case. These results are more recent than those previously given. The particles were acicular (needle-like), with a ratio of length to width of about 7:1. The 0.3 micrometer (a micro-meter, or micron, is about 40 microinches) particles are about the size used for modern low-noise tapes. It is a triumph of the particle manufacturer's art to be able to make

gamma ferric oxide particles this small without unduly compromising the print-through characteristics. The reason that "smaller is better" for audio tapes is that the background noise ("tape hiss") is less the smaller the particles.

Print is Noise Too

Wideband noise, on a good audio tape recorder, is generally 45 dB or better below the 1 per cent distortion level. When a tape is stored properly, the printed signals are likely to be down 55 to 60 dB, and thus inaudible under most circumstances. Even so, a loud transient signal could be recorded as much as 10 dB above the nominal 1 per cent distortion level, and its pre-print signal might become audible when no other signal was present. It is wise to record at such a level that these heavy transients do not approach the saturation level of the tape (which is 10 to 15 dB above the minimal 1 per cent distortion level). A compromise must be sought in the recording level of the tape, since too low a level will get rid of the print, but bring in a lot of hiss. Cassette machines usually have about 10 dB less wideband S/N than do open-reel units, but a good cassette deck will have Dolby, and this equalizes the situation to a large extent and makes them comparable. In fact, Dolby should improve the print-through situation for cassettes by about the same amount that it improves the S/N ratio.

Storage for Low Print

One might think that audiophiles should rent space in a meat locker to store their tapes. This would probably avoid print-through, but would cause other problems. The wound tape pack does not change dimension with temperature change at the same rate as does the reel or hub the tape is wound on, so tapes that have changed temperature drastically will probably not be wound well, and the pack will slip, possibly damaging the tape.

Another possibility is that moisture could condense on a cold tape brought into a warm room. Modern tapes are not really very sensitive to moisture, but it would make a mess and likely bring on an attack of fungus. Best storage conditions are about 60°F to 75°F and about 35 per cent to 50 per cent RH (Relative humidity). Too dry an atmosphere will give rise to static electricity problems, especially if the tape is wound or unwound. Temperature cycling during storage is especially bad. Tapes should be periodically rewound.

Ambient Magnetic Fields

There shouldn't be much of a problem storing the tapes in a magnetic-field-free environment. Remembering that a.c. fields less than about 5 oersteds really don't cause much trouble, and that d.c. fields are less harmful than a.c. fields, look at Table III. Several instruments and devices found in my lab and at home were measured for their stray fields. All were 60 Hz a.c. except as labeled differently. The most offensive item was the B-H Meter (magnetic hysteresis loop tracer) in my lab. This instrument, used for making measurements on magnetic materials, has an unshielded coil of some 6000 turns of #12 wire. The coil weighs over 100 pounds and has an inductance of 1 Henry. The field is very nearly a pure dipole field (more about this later). This beast will disturb electronic instruments and tape recorder circuits several feet away, and I make no other measurements when it is being used. Still, my valuable standard alignment tapes are stored only about 10 feet away, and I have had no problems with them. The large tape eraser is also unshielded and consists of two coils about 14 inches in diameter, connected with a C-shaped core, 4x4-inches in cross-section, which weighs 650 pounds. The field in the gap goes to 6000 oersteds. The tape head eraser is a standard type, and the Ampex tape eraser is also a standard unit.

Compared to the B-H meter and the tape erasers, the other things are not very impressive. The point is, it is very difficult to obtain the field magnitudes which would cause any trouble with print-through. All the old tales everyone has heard about tape problems with floor polishers and vacuum cleaners are just so much nonsense. I can say this without measuring every appliance around, and I can also state that no conceivable state of disrepair could cause large fields to emanate from such appliances. The reason is that any magnetic field can be described as a combination of dipole fields, quadrupole fields, octupole fields, etc. The higher the order of the field, the faster the field falls off with increasing distance from the source. Only a simple geometry, such as a solenoid coil, can produce a dipole field. Other sources produce higher order fields and thus have fields which don't extend very far. An exception to this is the field produced by a wire extending to infinity in both directions. This produces a cylindrical field around the wire, but the model is unreal. Close to a long straight wire, the cylindrical approximation holds quite well. Two wires with currents

Printing effect has many names. Here is a list of the major ones:

Accidental Printing—*
Anhysteretic Contact Duplication—Deliberate duplication of tapes using magnetic fields.
Anhysteretic Contact Printing—See last definition
Contact Duplication—Either type of deliberate duplication of tapes.
Contact Printing—See last definition
Cross-Talk, Magnetic Tape—* (This is not a good definition, as "cross-talk" usually refers to interference from the adjacent channel).
Kopiereffekt (copy-effect)—*
Layer-to-layer Signal Transfer—*
Magnetic Transfer—*
Nachecho (Post-echo)—Synonymous with "Post-print"
Print—*
Print Effect—*
Print-Through, Magnetic—All with an asterisk (*) are synonymous with this.
Post-Print—The echo after the signal.
Pre-Print—The echo before the signal.
Spurious Printing—*
Thermal Contact Duplication—Deliberate duplication of tapes by thermal methods.
Thermal Contact Printing—See definition above.
Thermal Transfer—Usually used to mean deliberate duplication.
Thermomagnetic Recording—Deliberate duplication.
Thermo-remanent Magnetization—* (Usually used with deliberate duplication).
Vorecho (Pre-echo)—Synonymous with "pre-print."

*Synonymous with print-through.

traveling in opposite directions, such as power leads, produce fields which cancel out except in between the wires.

The last entry in Table III is calculated, not measured, and it uses the cylindrical field approximation. The current is taken from the article on lightning in my encyclopedia. Two and a half feet away is extremely close to a lightning bolt of that maximum size, and the field is quite large; it would erase most audio tapes made from iron oxide, and the field is only falling off as the inverse of the distance (twice the distance gives half the field, etc.). Dipole fields fall off as the inverse cube of the distance, and quadrupole fields as the inverse distance to the fifth power! Such an occurrence as this king-size lightning bolt would be a disaster, but it's very unlikely. Yet ... perhaps it might be a mistake to run the ground wire from your antenna mast down the other side of the wall from your tape storage shelves!

Differences in Materials

There are distinct differences in the various types of particles which are used to make magnetic tapes. The worst particle for print-through effects in commercial use is cobalt-doped gamma ferric oxide. Tapes made with magnetite or ferrite materials are somewhat better. Standard gamma ferric oxide tapes span a range from only fair to very good. Most chromium dioxide tapes are very good due to their narrow particle-size distribution. Metal-particle tapes, not yet available to the audio recordist, are generally excellent, because their transition zone is located so far down in the tail of the particle-size distribution. All the commercial tapes which I have tested that are labeled "Low Print" seem to be just that (these are all from well-known and reputable manufacturers).

Contact Recording

It is possible to use both the themomagnetic and the anhysteretic methods to duplicate magnetic tapes by contact recording. Sugaya and Kobayashi[16] have published a comprehensive review which points out that these methods are old, though their first commercial use is recent.

Camras[17] and Herr[18] were the first to suggest commercial tape duplication (of audio tapes) using contact duplication. Both their processes used the anhysteretic copy method invented by Muller-Ernesti[19]. Both processes used a high-frequency a.c. field applied when master and slave tapes were in contact. Both failed to be commercialized, for several reasons. The transfer efficiency was poor (the slave signal was quite low, compared to the master signal), the long and short wavelengths rolled off badly (similar to Fig. 3), and the signal-to-noise ratio was not very good. Also, the market for duplicated tapes was not developed at that time[16]. The difficulty with the efficiency was that both tapes were of the same type. Any field that tended to magnetize the slave would tend to demagnetize the master. What was needed was a high coercivity master and a much lower coercivity slave.

The same problem occurs in thermo-remanent magnetization. If the tapes are of the same type, the efficiency will be poor. With gamma ferric oxide, temperatures to get usable efficiency will be high enough to destroy the tape. Even if high temperature tapes were used, bringing the transition zone up into the large part of the normal curve (see Fig. 7) would tend to destroy the magnetization of the master. In 1959 the idea arose[20] that tapes made of chromium dioxide be used as slaves, since the Curie temperature of this material is only slightly above the boiling point of water, a temperature which most modern tape materials can stand. If the master (made of iron oxide) is placed in contact with the slave (made of chromium dioxide), the two are heated above the T_C of the chromium dioxide, then cooled to room temperature while in contact, the signal from the master is impressed on the slave. The transfer efficiencies of this process can exceed 100 per cent; the slave coming out with more signal than the master.

With the availability of chromium dioxide tapes in the 1960s, the thermo-remanent process became commercially feasible, and many companies began work on it. At about the same time, high coercive force tapes made of cobalt-doped gamma ferric oxide also became available, and the anhysteretic method is also now used commercially. Neither of these contact recording methods is used for duplication of audio tapes. Audio tapes are duplicated using a master tape player driving a bank of satellite recorders, all running at 120 ips, and the industry is not likely to scrap all that machinery to replace it with a process which would cost more and give poorer results. All the contact duplication equipment is now used for duplication of various types of video recordings. Video recordings, surprisingly enough, do not require such a wide wavelength bandwidth as do audio tapes! The standard audio band of 20 to 20,000 Hz is four decades of frequency (or wavelength), while most high quality video recorders get by with only one decade of frequency (or wavelength). Thus, even though there is a loss at both short and long wavelengths, it is easy to fit the video signal into a region of little loss, but almost impossible to do this with audio. The control (or servo) signals and the audio track on video tapes are recorded in the usual way, with only the video signals being transferred by contact duplication. The reason that this expensive and complicated process is so attractive to video tape duplicators is that it is even more expensive and time-consuming to copy video tapes machine to machine. Video tape machines must always be operated in "real time" (at standard speeds) and cannot be run at many times normal speeds as are the audio tape duplicators.

Resolving the Mysteries

Now to pick up some loose ends. One should explain why there are a few complaints of printing on 8-track cartridges, which lie in automobiles under the hot sun for hours; or on cassettes, which use the thinnest of base films and are also used in automobiles. Part of the answer is in the type of tape pack. Both use a very loose wind, and the cartridge pack is designed to slip. The worst heat exposure is in automobiles, which also have a lot of vibration. When a tape undergoing print-through slips its relative slave-to-master position, the printing process has to start all over again, as the fields are changed. As the tapes ride around in the automobile, this slipping is constantly happening, and the result is the same as if the tapes were being constantly rewound. Cassettes or cartridges used in the home are not normally exposed to such heat as those in automobiles. There may not be much of a problem, but I think I'll still take mine out when my car is sitting under this hot Texas sun!

There are some other mitigating circumstances for cassettes and cartridges. I've mentioned the suppressing effect that Dolby should have on the printed signals, since it should treat them like the rest of the noise. In addition, the dynamic range in a car is very limited, and printed signals probably get lost in the background noise. Also, because of the "A" wind, the post-print is worse in cassettes and cartridges. Lastly, a post-echo is less irritating than a pre-echo and may not be heard in many cases.

We can now explain all of the effects which occurred on that operatic broadcast. Video tape of that era was all about 1.4 mils thick and ran at a speed of 15 ips. This is still a widely-used standard. Miss Sutherland's high notes were well within the print-through peak range at this tape speed, and the pre-print is worse, so her lines would always have the pre-echo. The instruments in the same pitch range had the same problem. The bass instruments and male singers were out of it on the low end and did not show any print effects. Since this was normal face-to-back print through, and not face-to-face as it is in deliberate copying, the video signals were of too short a wavelength to print at all, and the FM video system would not have been sensitive to them anyway. The synchronization signals were of very long wavelength and did not print. It wasn't much of a mystery after all, especially when you understand what's going on!

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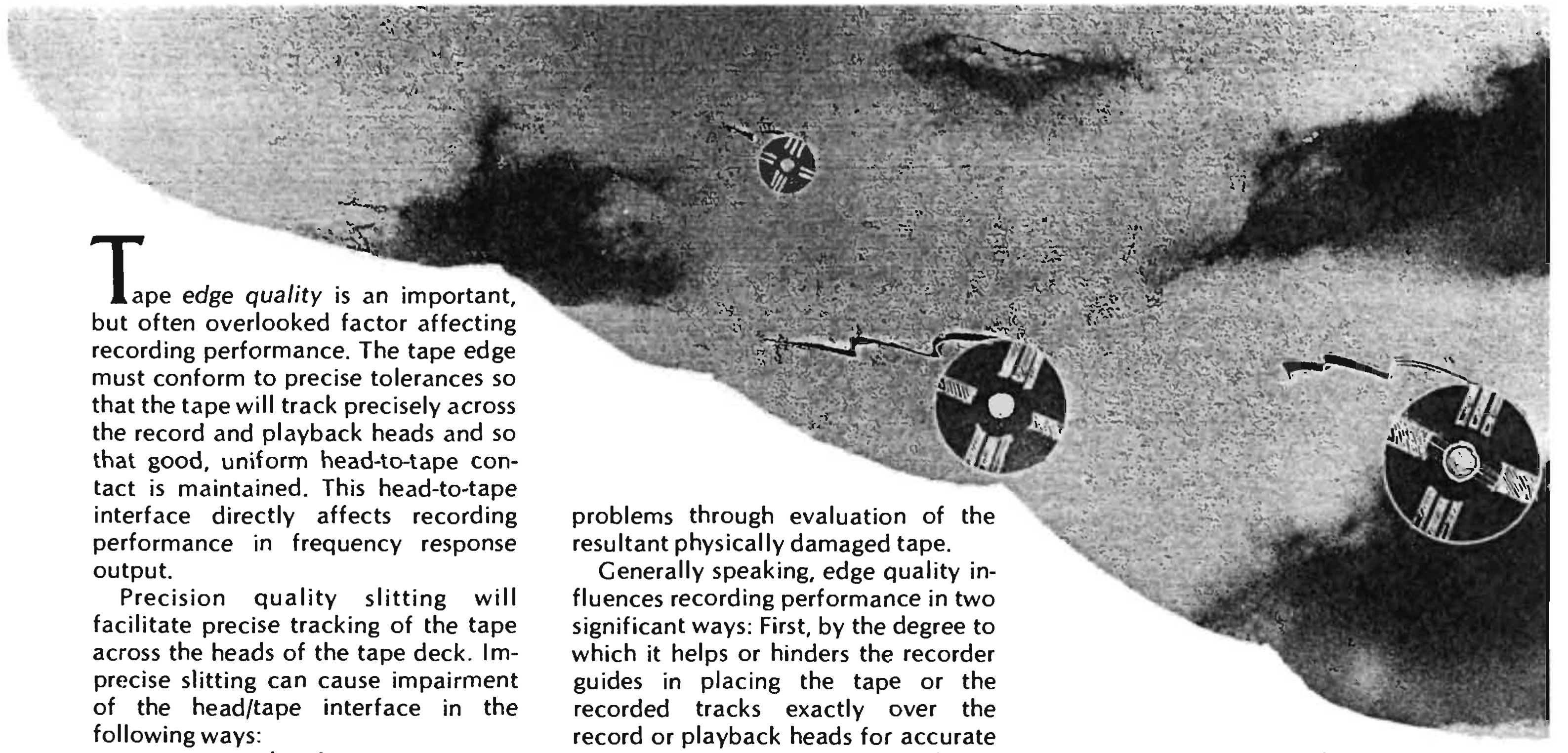


Anthony Weitz

Tape *edge quality* is an important, but often overlooked factor affecting recording performance. The tape edge must conform to precise tolerances so that the tape will track precisely across the record and playback heads and so that good, uniform head-to-tape contact is maintained. This head-to-tape interface directly affects recording performance in frequency response output.

Precision quality slitting will facilitate precise tracking of the tape across the heads of the tape deck. Imprecise slitting can cause impairment of the head/tape interface in the following ways:

1. Loss of head-to-tape contact caused by debris from poor slitting results in loss of output, which is greater at higher frequencies.

2. Loss of head-to-tape contact caused by curvature, resulting in poor contact or head-to-tape separation on one edge of the tape relative to the other edge. This results in a different frequency response and output between the two stereo channels. The edge track (left channel) will usually have more output fluctuation.

3. Tape travel relative to the head in other than a purely longitudinal direction. This causes the recorded tracks to not entirely traverse the full width of the playback head gaps.

4. Azimuth errors due to curvature resulting in loss of output proportional to frequency (the higher the frequency, the greater the loss).

Effects of Slitting Deficiencies

Before discussing the specific ways in which edge quality affects recording performance, it must be noted that the symptoms (especially the physical symptoms) of poor edge quality are similar, if not identical, to many of the symptoms of consumer tape deck guiding, transport, and tape handling problems. The following discussion is made assuming the tape is being played on a machine with ideal tape handling (i.e., heads are properly aligned, guides are not worn, and tensions and brakes are in good condition).

For an in-depth discussion of tape deck transport problems, refer to "Cassette Transport Problems" in the September, 1974, issue of *Audio* magazine. The article, written by the Memorex Audio Tape Plant Manager, describes how to recognize transport problems through evaluation of the resultant physically damaged tape.

Generally speaking, edge quality influences recording performance in two significant ways: First, by the degree to which it helps or hinders the recorder guides in placing the tape or the recorded tracks exactly over the record or playback heads for accurate signal performance, and, secondly, by the amount of debris produced. Poorly slit tape contains oxide residue from the slit edge when the tape is new and will generate more debris as the fractured, rather than sheared, edge deteriorates (see Fig. 1). During playback, this debris will speed up residue accumulations on the head and transport mechanisms, causing signal loss due to separation of tape-to-head contact.

Reason Tape Must Be Slit

Magnetic recording tape is coated much wider than the ¼-inch width (0.150 inch for cassette) used on consumer tape decks and therefore has to be slit to the appropriate width, depending on the application. Slitting is the tape manufacturing step that determines *edge quality*.

Magnetic tape consists of a layer of highly specialized oxide coated onto a clear polyester base film. The base film used in the coating process is usually either 12- or 24-inches wide, depending on the width of the coating machine being used. The length of the base film is at least 7,500 feet; sometimes multiples of 7,500-foot lengths are used. A roll of base film 7,500 feet in length by 12 or 24 inches in width coated with magnetic oxide is called a *web*.

The oxide coating consists of the oxide powder together with the polymeric system which binds the oxide particles together and to the base film. During the coating process, the oxide mixture is liquified with solvent along with such additives as lubricants, plasticizers, and dispersants or wetting agents.

Immediately following the coating process and oven drying, the coated web is "calendered." Calendering is a surface treatment and oxide compaction process used mainly to obtain a smooth, highly polished tape surface which is necessary for good output, especially at the higher frequencies. Following calendering, the web must be slit into individual tape widths so that the tape can be loaded either into cassettes or onto reels.

The Slitting Process

Strictly speaking, "slitting" refers to the single manufacturing step of shearing the coated web into exact 0.248- or 0.150-inch widths. The slitting process should, however, be considered as three inter-related steps forming a continuous process (see Fig. 2):

1. The *unwinding* of coated web and in-feed to the slitter blades.

2. The *slitting* of the tape with the slitter blades.

3. The *winding* of slit tape onto take-up hubs.

Imprecise performance in the first or third step can also have negative effects on slitting quality. Quality tape manufacturers constantly monitor the slitting process and inspect the tape slit to insure that all three slitting steps are being performed correctly within close tolerances (see Fig. 3). Typical width tolerances for cassette and open reel tape are +0.000 inch and −0.002 inch. Within the slitting process, there are three critical factors that determine tape edge quality. They are the *tracking* of the tape in and out of the blades, *blade geometry,* and *control of tensions* and speed.

Tape Tracking

The entire slitting process, from the unwinding of the web through the

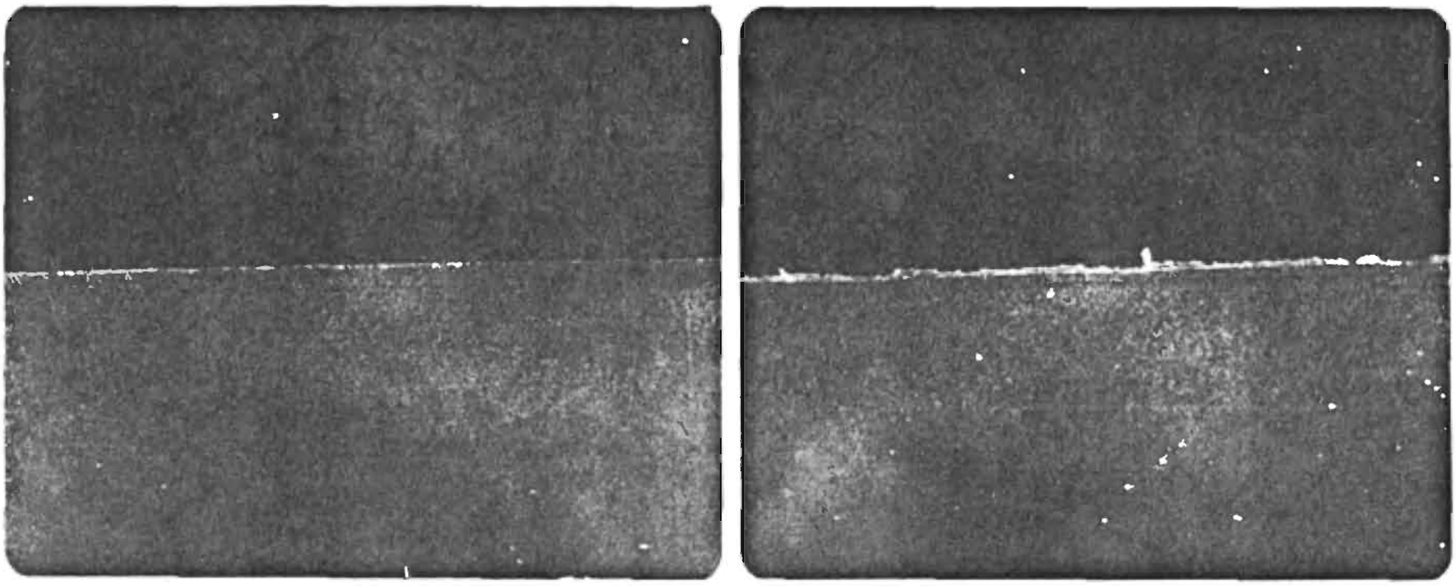


Fig. 1—Tape edges compared. Precisely slit edge (left) contrasts with poorly slit edge (right) when each edge is magnified 200×.

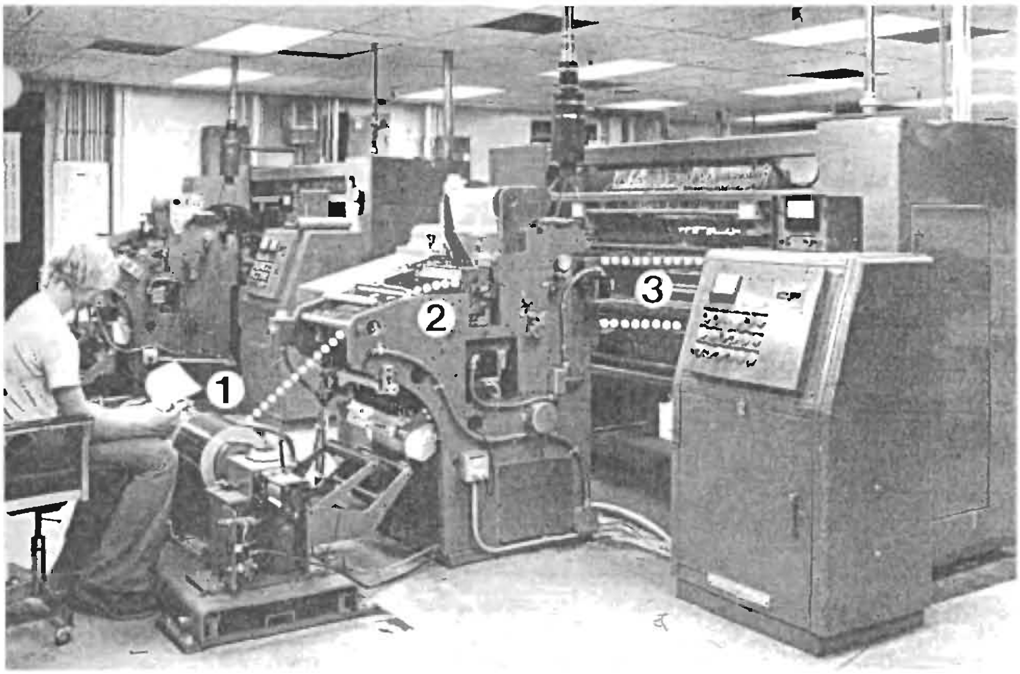


Fig. 2—In slitter operation, tape is unwound from the web (1), transported through blades (2), with slit tape drawn past guides (3), and wound onto pancakes on the far side of the machine. An operator monitors the web during this operation.

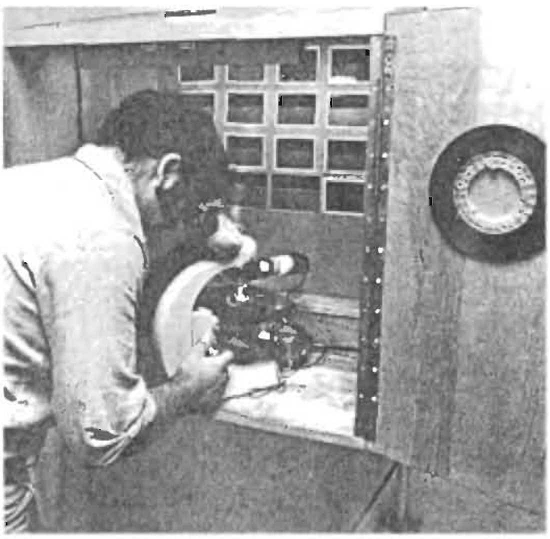


Fig. 3—Memorex maintains a quality control station next to the slitters where operators regularly check edge quality with a microscope. The photographs posted over the microscope illustrate acceptable and unacceptable degrees of edge quality to aid operator in determining quality.

take-up of slit tape onto hubs must be performed as a single, straight, smooth, and even-tensioned motion. Fluctuations in tape movement result in tape which is not cut straight or which is torn or stretched, each of which will significantly affect the record and playback performance.

Blade Geometry

The slitter is a shearing mechanism utilizing pairs of circular knives for continuous, rotary-shear cutting. To slit a 12-inch web into 80 widths of cassette tape or 48 widths of open-reel or 8-track tape, 81 or 49 pairs of blades are used respectively (see Fig. 4). The blade pairs cut the tape in a manner similar to the way scissors cut. The fact that they are circular and rotate merely allows the two blades of each pair to maintain positive edge contact near the periphery continuously, while slitting the entire length of the web at high speed. The shearing or cutting is done where the two blades of each pair intersect on the periphery of the blades (see Fig. 5). The sides of the two blades of each pair not only actually touch during slitting, but additionally, a pressure is applied in the direction of the axis of rotation to insure that the two blades of each of the 81 or 49 pairs actually make positive contact at all times. This is referred to as *blade side loading*. Because of critical cumulative and mechanical dimension tolerances in the assembly (consisting of the pairs of blades), tape is slit from 12-inch, rather than 24-inch width, webs. The blades, made of hardened tool steel, must be ground to exact dimensions and must be precisely assembled on the shafts of the blade set so that contact between all pairs is uniform. Cleaner shear is obtained with lower

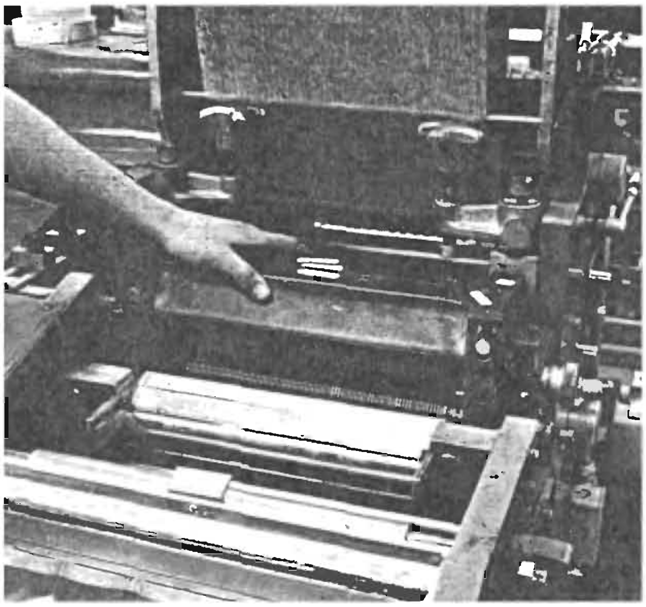


Fig. 4—The 81 pairs of slitter blades used for cassette slitting are precisely mounted on two parallel shafts.

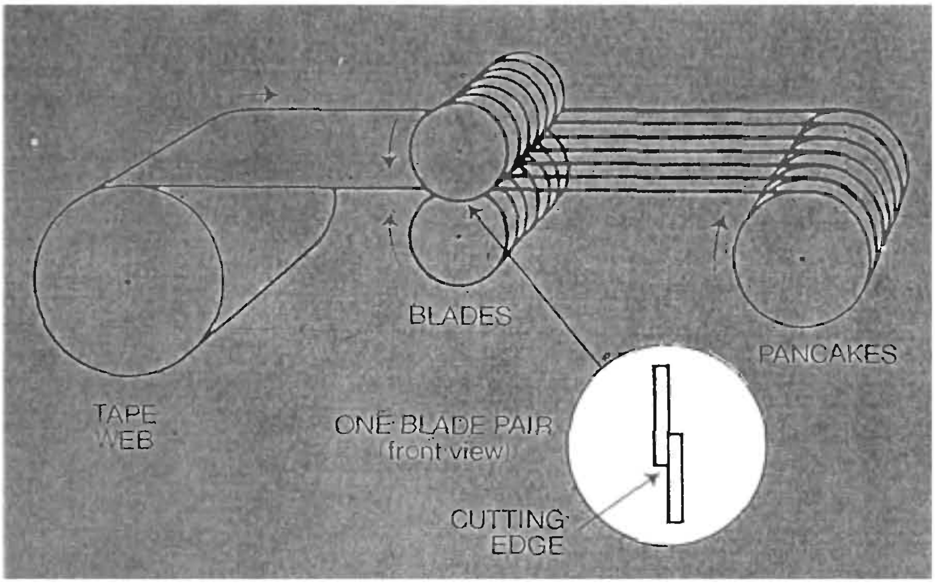


Fig. 5—Diagram of the slitting machinery.

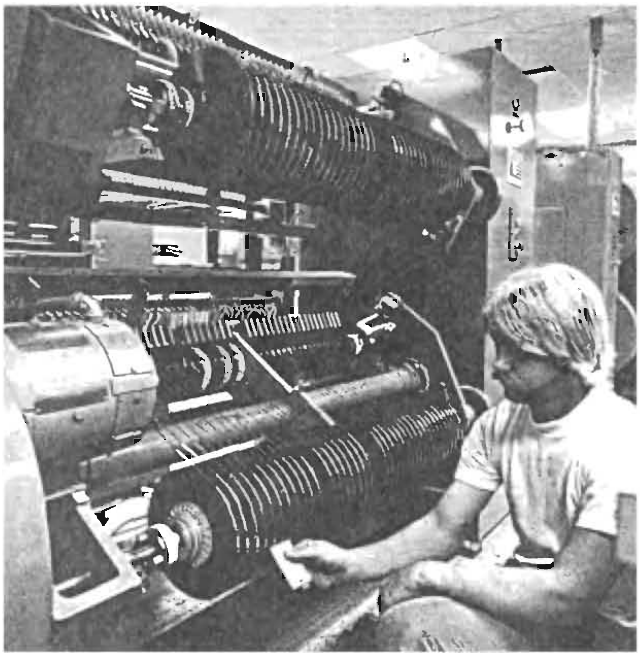


Fig. 6—Pancakes of cassette tape are shown at the take-up end of the slitting operation. Each pancake contains 7500 feet of tape.

side loading and engagement. The more exact the blade dimensions, the lower the engagement and side loading that can be afforded.

Control of Tensions and Speed

The rotating blades move approximately 3 per cent faster than the web speed. The coated web is unwound and transported through the slitter blades at speeds ranging from 200 to 2,000 feet per minute. Speed differences reflect such factors as oxide thickness, coating formula types, and slitting machine precision and general sophistication. Varying tensions have the effect of varying the relative blade/tape speed. This can cause uneven shear quality and result in poor slitting. The slitting machine must be capable of stopping, starting, accelerating, and decelerating in exactly co-ordinated procedures. This is especially true in slitting open-reel tape since the leader at the beginning and end of each reel length must be spliced into the web and slit with the tape.

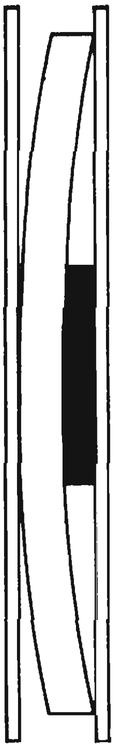
The tape which has been slit and wound onto a hub is called a "pancake" (see Fig. 6). The pancake is the configuration of slit tape prior to other steps within the manufacturing process such as cassette and cartridge loading and winding onto reels.

The tape in many layers wound onto the hub or on a reel is referred to as the "pack." A reel or pancake is said to have a good pack if it is not wound either too loosely or too tightly and if all wraps of the tape are exactly on top of the previous layer laterally so that the slit edges of the tape on the pack are parallel with the side of the hub. For each slit edge, all wraps should be in the same plane, which itself should be perpendicular to the axis of the hub.

Slitting Technology

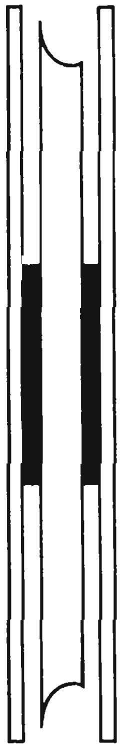
A contributor to the vast improvements made in open-reel quality during the last decade has been the technology developed to permit precision high-speed slitting of the more delicate cassette tape. Yet another contributor is the even more stringent requirements of some video helical scan formats.

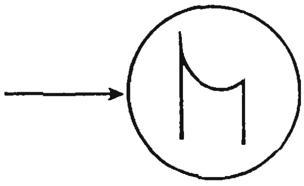
Cassette tape is more difficult to slit than open-reel tape for several reasons. Cassette tape is narrower (0.150 inch versus 0.248 inch). It is not as thick; a typical cassette tape is about half the thickness of open-reel tape. The thinnest base film used in open-reel is the same as the thickest base



DISHING

Fig. 7





RIDGING

Fig. 8

film used in cassette tape (half mil); the thinnest base film used for cassette tape is one quarter mil. Open-reel tape is typically one mil, but some open reel is still coated on 1½-mil base film. Coating thickness for cassette tape is also only about half the coating thickness of open reel (200 versus 400 microinches). Because cassette tape is thinner and narrower, it is more delicate. The slitting is therefore more difficult because the process is more susceptible to tension differences and much lower tensions must be used in slitting cassette tape. Another aspect of cassette versus open-reel slitting is the fact that cumulative tolerance requirements are more stringent for the 81 pairs of blades than for the 49 pairs of blades for ¼ inch open reel. The track width for cassette is only half the typical width for open reel, making more demands on straightness, and the recorded wavelength in the cassette is half that of open reel, thus making any head/tape separation problems, due to debris, curvature, etc., twice as severe with the cassette format. Video recording utilizes even narrower tracks than cassette and the symptoms of poor slitting are not only audible but visible, yielding a greater subjective impairment overall.

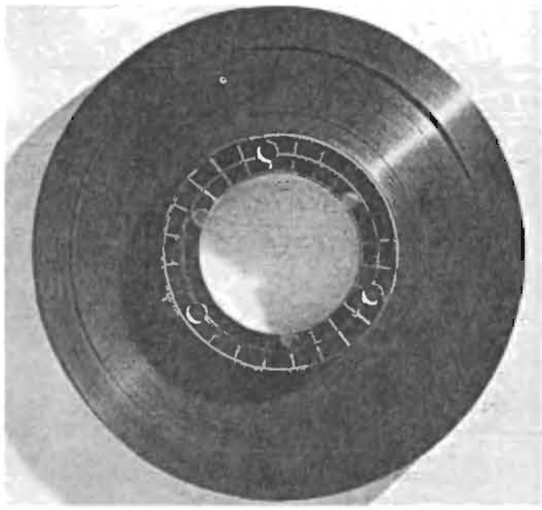
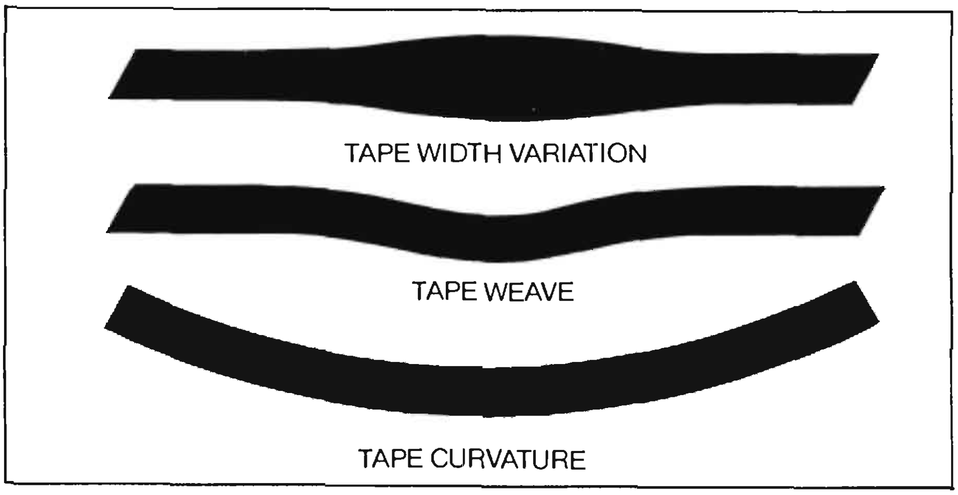


Fig. 9—An example of a blowout, layers of tape protruding outside the tape pack.

Fig. 10—Examples of width variation, weave, and curvature.



Recognizing Slitting Problems

Evidence of poor edge quality (poor slitting) can be detected by visual inspection. Since, as noted, the symptoms of poor edge quality can also be the result of machine transport problems, the audiophile should make it a practice to visually inspect each new reel of tape while it is new before ever playing it. Inspect for the following:

Dishing is caused by misalignment in *the take-up or winding of slit tape, by a* stretched edge, or slit curvature. Dishing occurs when successive layers of tape on the pack are gradually packed toward one side of the hub. Thus, the side of the pack will not be perfectly flat but will be concave shaped—somewhat resembling a dish. Tape which has been slit with curvature tends to "curve" toward one edge as it is being wound onto the hub (see Fig. 7). A stretched edge results in much the same pack characteristics and can also be caused by the user's tape deck. It is important to inspect the pack of new tape as it comes from the manufacturer in order to pinpoint whether a bad pack is due to the tape or the tape deck.

Ridging describes the condition in which one edge of the tape pack is higher than the other when the tape is viewed looking at the periphery of the pancake (or reel) with a line-of-sight parallel to the pack edge (see Fig. 8).

Ridging indicates that one side of the tape has been stretched or fractured but that guiding while winding up the slit tape has forced the tape to wind onto the pancake flat rather than dished. Ridging can be caused by poorly sheared slitting often due to incorrect blade geometry where the edge of the tape is microscopically deformed. This results in a greater thickness at the edge, thus causing the "ridge."

Blowouts are individual layers or a group of several layers of tape protruding outside the tape pack (see Fig. 9). When the blowout occurs, these layers are wrapped onto the pancake offset to one side. Subsequent layers may be wound onto the pancake in the correct lateral location relative to the hub. Blowouts are caused by improper winding tension or lack of precision guiding. Blowouts tend to be aggravated by curvature. Reel flanges often crush the protruding blowout, causing servere damage to the tape edge.

Cinching is apparent in either of two ways, as an accordion-like layering in the tape pack and as a loop of tape in the pack wound back on itself. The cause is usually loose winding followed by higher tension, rapid rotation, or stopping suddenly. It will affect recording performance by presenting a folded or distorted surface to the head causing signal loss. Cinching can occur easily on many tape decks if winding tensions are too low, especially in "fast" modes. A low-tension wind followed by abrupt stops and starts causes cinching. It is easy to cause cinching while threading if the tape needed for threading is pulled from the pack with brakes on. Tape needed for threading should instead be unwrapped from the pack.

Debris will occur even in the highest quality tape, which will have some residue from the slitting process, but quality manufacturers use fastidiously maintained slitting equipment and stringent process quality control monitoring to insure that slitting debris are virtually nil. One way to check for debris on a new reel of tape which has never been played is to examine the sides of the pack for evidence of oxide dust. Debris due to poor slitting will usually cause one side of the pack to be a slightly different color than the other side. The most common symptom of slitting debris with new tape is the accumulation of oxide on and around tape deck heads, guides, tape lifters, etc. Debris can clog heads when they cause a separation of the tape from the heads. Of course, such tape deck problems as worn tape-edge guides or burrs in the tape path on any tape-contacting mechanism can generate debris on any tape.

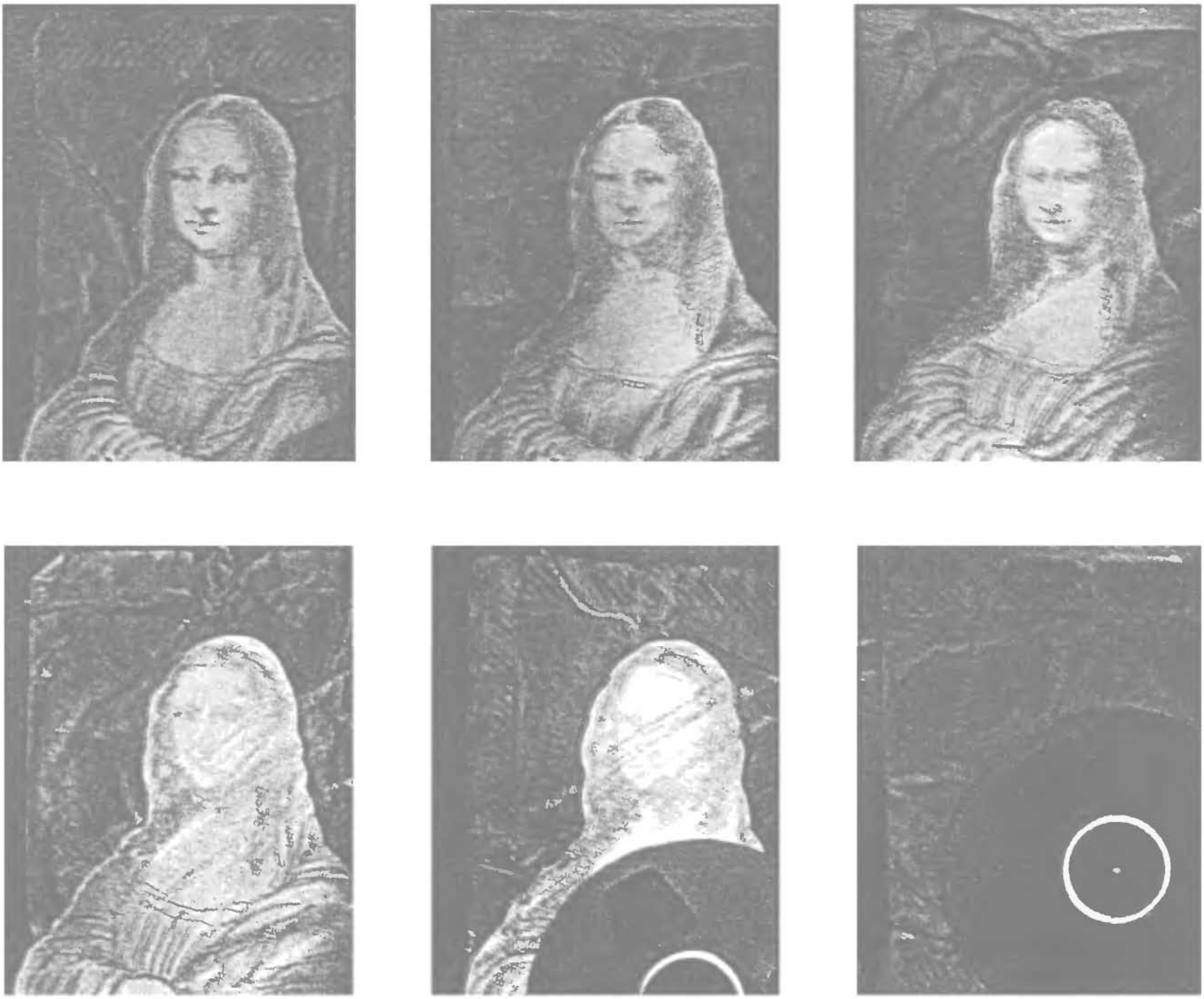
Three other, more subtle, tape edge-related problems are *width, weave,* and *curvature* (see Fig. 10). Tape which has width variations usually has some weave and vice versa. In the purest sense of the word, tape could have weave and the measured distance between the edges would always be the same (parallel edges), but the tape, if laid out on a flat surface, would exhibit a cyclic serpentine "weave" along its length. Blade wobble causes width variations and weave.

Tape exhibiting width variations sometimes alternates from overwidth to underwidth, but not necessarily symmetrically. Width variations are more specifically related to wobble of the blades, while weave can also be due to lateral instability and side-to-side differences in web tension as the web enters the slitter blades.

Tape is slit with curvature when the web tension at the slitter in-feed is greater on one side than on the other side of the web. Slitter alignment and web tensions are crucial to slitting without curvature. Curvature is measured by laying a length of slit tape on a flat, smooth surface such as plate glass. When the tape is completely flat on the surface, the curvature measure is the distance of the middle of the length of tape from a line between the two ends of the length of tape.

Summary

In summary, there are many ways in which tape edge quality affects recording performance mostly through factors relating to head-to-tape interface. Slitting deficiencies are indicated when dishing, blowouts, ridging, cinching, or debris is visible in the tape pack. Tape edge damage caused by the tape deck can also be identified if these edge-related parameters are, after several passes, compared with the condition of the tape edge when it was new. By careful visual inspection of each new reel, the discerning audiophile can assess the slitting quality of the tape he buys to be sure that maximum recording performance is achieved.



Anthony Weitz

Open-Reel Renaissance

Gary R. Gruber, Ph. D.*

Two years ago, after more than a decade of production of pre-recorded, 7½ ips, open-reel tapes, the Ampex Corp. bowed out of the business. Ironically, however, the last set of pre-recorded reels Ampex produced, Solti's *Beethoven Nine Symphonies,* were sonically and interpretively the best they had ever produced.

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Mill Valley, Calif. 94941

During the last years of production Ampex even developed a national tape club but, in spite of the efforts of Ampex's Russ Fields and Mike Ayers, the operation was closed down. That was over two years ago, and with Angel, Columbia, and DGG out of the open-reel tape business, the only company left in the production of prerecorded open-reel tapes was Stereotape, which at the time wasn't producing any Dolby B tapes and was gradually offering fewer selections on the open market. For a time it appeared as though the medium of prerecorded open-reel tapes was coming to a dismal end.

Reel Renaissance

At the same time, Barclay-Crocker on the East Coast was contemplating entry into the open-reel market. Sonar, also on the East Coast, was still producing some open-reel tapes, and Stereotape, on the West Coast, was planning a revival of open-reel production.

About a year ago, Barclay-Crocker produced their first open-reel tapes and, at the same time, Stereotape also resumed production. Barclay-Crocker, with the Musical Heritage Society, Vanguard, Unicorn, Desmar, and Halcyon labels, is geared towards the more esoteric tastes, while Stereotape with London, DGG, RCA, etc. attracted the standard classical and popular tastes. Both companies Dolbyize their tapes so the noise level is quite low.

Some tape enthusiasts used to complain that they had trouble guessing where the program on the first side of some reels began and, occasionally, they had to *Fast Forward* the tape on the first side in order to reach the first

selection. That was because the second side of the tape contained a longer program. Barclay-Crocker has virtually eliminated this problem with what they call the "sonic sentry." The way this works is, when the second side of the tape has a longer program than the first side, you are instructed to put your tape deck in the *Fast Forward* mode and listen for a high frequency oscillating tone. When you hear the tone, stop the tape and start your machine in the normal playback mode.

As far as the samples I've reviewed, Stereotape always seems to start their programs at the beginning of the tape, making it unnecessary to employ a process similar to Barclay-Crocker's "sonic sentry." Both companies let their program on the first side end where the program on the second side begins. If you do not have a reversing mechanism on your tape deck, all you need to do is flip the reels over to continue with the second side.

All Barclay-Crocker and Stereotape reels are Dolby B encoded and if your deck is equipped with Dolby, just put the switch on. However, if it is not, you may reduce the treble controls on your amplifier slightly.

Open Reel vs. Cassette

Many readers may ask the following pertinent question: With the technical advances recently taken with the pre-

Table 1

Here are some outstanding new reels produced by Barclay Crocker (B and by Stereotape (S). I have rated each of these tapes by E* for super-excellent, E for excellent, G for good, and F for fair concerning interpretative performance (P) sound quality (Q) and signal-to-noise ratio (N)

	P	Q	N
Saint-Saens. *Caprice and Woodwind Sonatas*, Minneapolis C iamber Ensemble-MHS-3224 (B)	F	G	G
Kodaly *Duo Sonata* Ofe sky (cello), Posner (violin) MHS-3047 (B)	G	E	G
Mahler *Symphony # 4* Abravanel/Davrath/Utah SO Van-10042 (B)	E*	E	E
Haydn: *Symphonies Nos. 90 & 91* Blum/Estherhazy Orch. Van-10044 (B)	E*	E	E
Leroy Anderson: *Fiddle Faddle and 14 other Anderson Favorites* Van 10016 (B)	E	E	E
18th Century Concerti o Harps and Strings MHS 3320 (B)	E	E	E
James Taylor s Greatest Hits—WST 2979 (S)	E*	E	E
Copland: *Clarinet Concerto;* Crusell: *Grand Concerto* Unc 0314 (B)	E	E	E
J.S. Bach: *Cantatas Nos. 11 & 80* Van 71193 (B)	E	E	E
Goldmark *Rustic Wedding Symphony*, Unc 2142 (B)	G	E	E
Verdi *Il Vespri Sicilliani:* RCA E0P4 0370 (S)	E	E	E
Beethoven-*Nine Symphonies:* Solti/CSO Lon CSA09 (S)	E*	G	G
Dvorak. *Dumky Trio*· Smetana. *Piano Trio* DG 2530594 (S)	E	E	G
Gould *Latin American Symphonette* Gottschalk: *Nigh in the Tropics* Van 721218 (B)	E	E*	E
Rachmaninoff *Third Symphony*, Stokowski, Des 1007 (B)	E	E*	E*
Mahler *Symphony No. 3* Levine RCA 1757 (S)	E	E	E
Neilson *Symphonies Nos. 6 and 3*, Unc 0326 (B)	G	E	E
Prokofieff-*Five Piano Concerti*-Ashkenazy Lon 2314 (S)	E	E	G
Brahms: *Violin Concerto*, Milstein, DG-2530592 (S)	E	E	G
Tchaikovsky: *First Piano Concerto* Berman DG 2530677 (S)	E	E*	E
Monteverdi *Vesperae Della Beate Vergine*-Ar-2710017 (S)	E	G	G
Music of the Gothic Era, Munrow AR 2710019 (S)	E	E	G
Gershwin *Porgy and Bess*, Lon OSAO 13116 (S)	E	E	E
Saint-Saens: *Third Symphony* DG 250619 (S)	E	E	G

The above recordings are well worth the investment. Some record outlets carry the Stereotape reels. However it would be best to write to these addresses for ordering information and catalogs:

For Stereotape Releases: The Reel Society, 8125 Lankershim Blvd., North Hollywood, Cal. 91605

For Barclay-Crocker Releases: Barclay-Crocker 11 Broadway, Room 857 New York, N.Y 10004

Table 2

"The Reel Revolution" as Russ Fields of the Reel Society put it, "rolls on. Here are some interesting tapes that will be available in the next few months. Again, (B) denotes Barclay Crocker, and (S) denotes Stereotape (Although Barclay-Crocker is producing new tape releases steadily, I did not get an advance release sheet at the time of writing this article thus accounting for many more Stereotape selections here).

Vaughan Williams: *Fantasy on a Theme of Tallis* Dvorak: *Serenade*, Stokowski, Desmar 1011 (B)
The Best of the Grateful Dead, WST-2764 (S)
This is Duke Ellington, EPP2-6042 (S)
Dvorak: *Slavonic Dances* (Complete), ST 801 (S)
Chopin: *24 Preludes Polonaises* (Pollini), ST 803 (S)
Bernard Herrman conducts great British Film Scores Psycho and other film scores ST 902 (S)
Mahler: *Symphony # 2*, Mehta/VPO, CSAO 2242 (S)
Tchaikovsky: *Symphony # 5* Solti/CSO, CSO 6983 (S)
Rimsky-Korsakov. *May Night* (Complete) DG 3652104 (S)
Wagner: *Die Meistersinger* Lon OSAO 1512 (S)

recorded cassette tapes, is the sound really that much better on the new open reels? To answer this question honestly, I must say that some cassettes, particularly Polydor's DGG, London, Advent, and Angel, elicit some startling sonics. But the new open reels produced by Stereotape and Barclay-Crocker do show the differences between the new open reel and the current cassette. The sound on the open reel is fuller with a greater dynamic range and the distortion at high volume levels is noticeably less. As an example, the Advent cassette recording of the Rachmaninoff *Third Symphony* with Leopold Stokowski (E-1046) is technically superb, while the Barclay-Crocker open-reel tape version (D-1007) of the same Desmar recording is technically ultra-superb! The Tchaikovsky *First Piano Concerto* with Berman and von Karajan on the DGG Polydor cassette (3300-677) is big sounding and quite breathtaking, however, the Stereotape open-reel version of the same DGG recording (DG-2530-677) is even more so. Physically, we should expect this difference since the 7½ ips open-reel tape travels past the playback head four times as fast as its cassette counterpart. Also, the width of the open reel is greater than the width of the cassette tape, providing for retention of better and more signal. The structure of the cassette invites more wow and flutter—many times you will need a cassette deck that will adjust for azimuth angle of the tape head for each tape played on the machine. Finally, the tolerances on an open-reel deck are less critical than those on a cassette deck. This is the reason why some cassettes jam, squeal, or have much more drop-out of signal than an open-reel tape does. Of course, if you want convenience, you can't beat the cassette; in this regard, the cassette is better than either the record or the open reel. But, if you want to come closest to realistic sound, then the open-reel format is best, especially when you are listening to large orchestral pieces or operatic works.

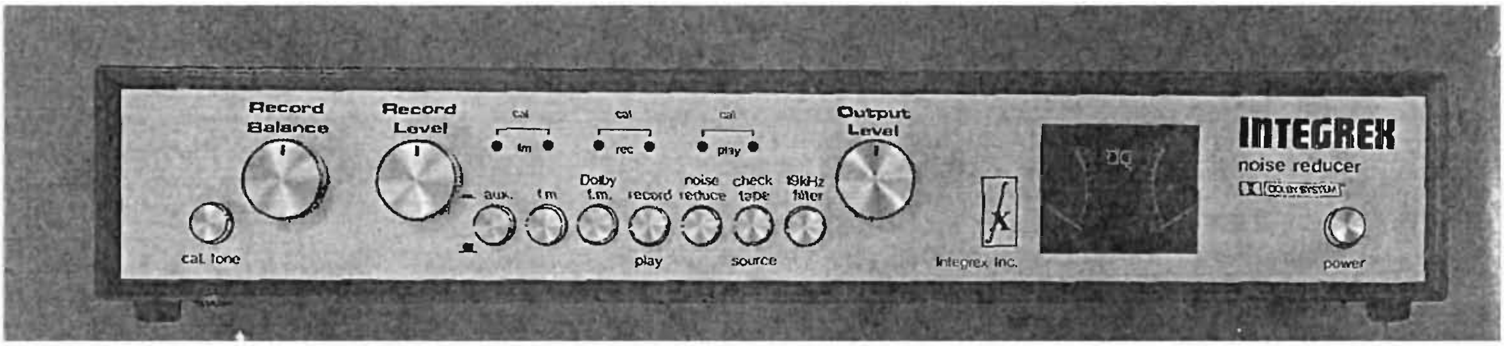
Epilogue

It is now apparent that the open reel pre-recorded tape format is here to stay. Hopefully, in the near future, Stereotape and Barclay-Crocker will produce tapes by the companies of Angel, Philips, Argo, Das Alte Werk, Telefunken, Orion, and Columbia. In any event, what is now being produced is quite satisfactory and should indeed quench the ever increasing thirst of the demanding audiophile and music lover. A

PART I—Theory of Operation

Build a Dolby Noise Reducer

Geoffrey Shorter*



This is the first portion of a three-part series on building a Dolby-B noise reduction unit, and we would like to thank the British magazine Wireless World *for their pioneering work on the kit and its back-up construction article, which appeared beginning in May, 1975. In particular, we would like to thank Geoffrey Shorter,* Wireless World's *Technical Editor, for his work in putting together the technical material presented here. We should note that parts two and three, as presented in* Audio, *will differ from those of the original article, inasmuch as the present version is based on a second-generation, high-performance chip, a National 1011A, rather than being discrete.*

For audiophiles interested in audio developments in Europe, we commend them to Wireless World's *subscription department, which is located at:*

Dorset House, Stamford St., London SE1 9LU, England.

A word of caution to potential builders; we do not feel this should be attempted as a first project by the novice builder, since a good set of tools, familiarity with building techniques, and the resistor and capacitor color code, etc. will be needed. For those with some experience, we believe the project will be both fun and worthwhile. A kit of parts for this project is available from Integrex, Inc., whose advertisement appears towards the end of this article.—Editor.

***Technical Editor**
Wireless World Magazine

In audio systems dynamic range can be defined as the ratio of the largest to the smallest program signal. Dynamic range is typically limited at the high-level end by tape saturation or amplifier signal handling problems; there is usually a fairly well-defined level beyond which compression occurs and distortion rises at a rapid rate. At the other extreme there is a limit on the lowest signal that can be handled, set typically by the noise level of electronic circuits, tape noise, surface noise on discs, or granularity on optical soundtracks.

In concerts, dynamic range can be as high as 90 to 100 dB, but once such program material has been recorded, dynamic range is reduced to 60 or 70 dB. (When broadcast the range can be as low as 20 to 40 dB.) In this situation there are three options—lose that part of the program below noise level, distort the peaks, or distort the range by compression either manually or automatically. None of these options is altogether acceptable in itself, all distort the original in some way. What is needed is a way of getting around this limitation of dynamic range without the distortion of overmodulation, without losing program in noise and without distortion of range. Before discussing various techniques that have been proposed and tried, we will be more specific about what is required.

As well as not introducing any perceptible non-linear or dynamic range distortion of both steady-state and transient signals, any proposed technique for high-quality use should not perceptibly alter the signal in respect to frequency and transient response. Any signal processor must be able to operate to the normal constraints of audio channels, i.e. operation should not depend on freedom from phase and amplitude versus frequency errors or changes, nor on a linear phase-frequency response; channel overload characteristics should not be worsened. In addition to compatibility with transmission channels, there must be compatibility between processors to the extent that recordings can be interchanged. In reducing perceptibility of noise, there should be no noticeable noise modulation effect and ideally all noises should be reduced by a similar amount, otherwise reducing one kind might unmask another.

Noise-Reducing Techniques

"Static" methods The most well-established methods of avoiding the constraints imposed by high noise levels are "static" ones. Examples are the high-frequency pre-emphasis, and subsequent de-emphasis, applied to FM broadcasts and gramophone records and the low-frequency pre-emphasis used in tapes. They are static because the amount of emphasis given is fixed and does not take account of the signal in any way. At some frequencies, there is thus an intrusion into the possible range of levels that signals can occupy which may mean that

some lower than normal limit must be placed on the program level.

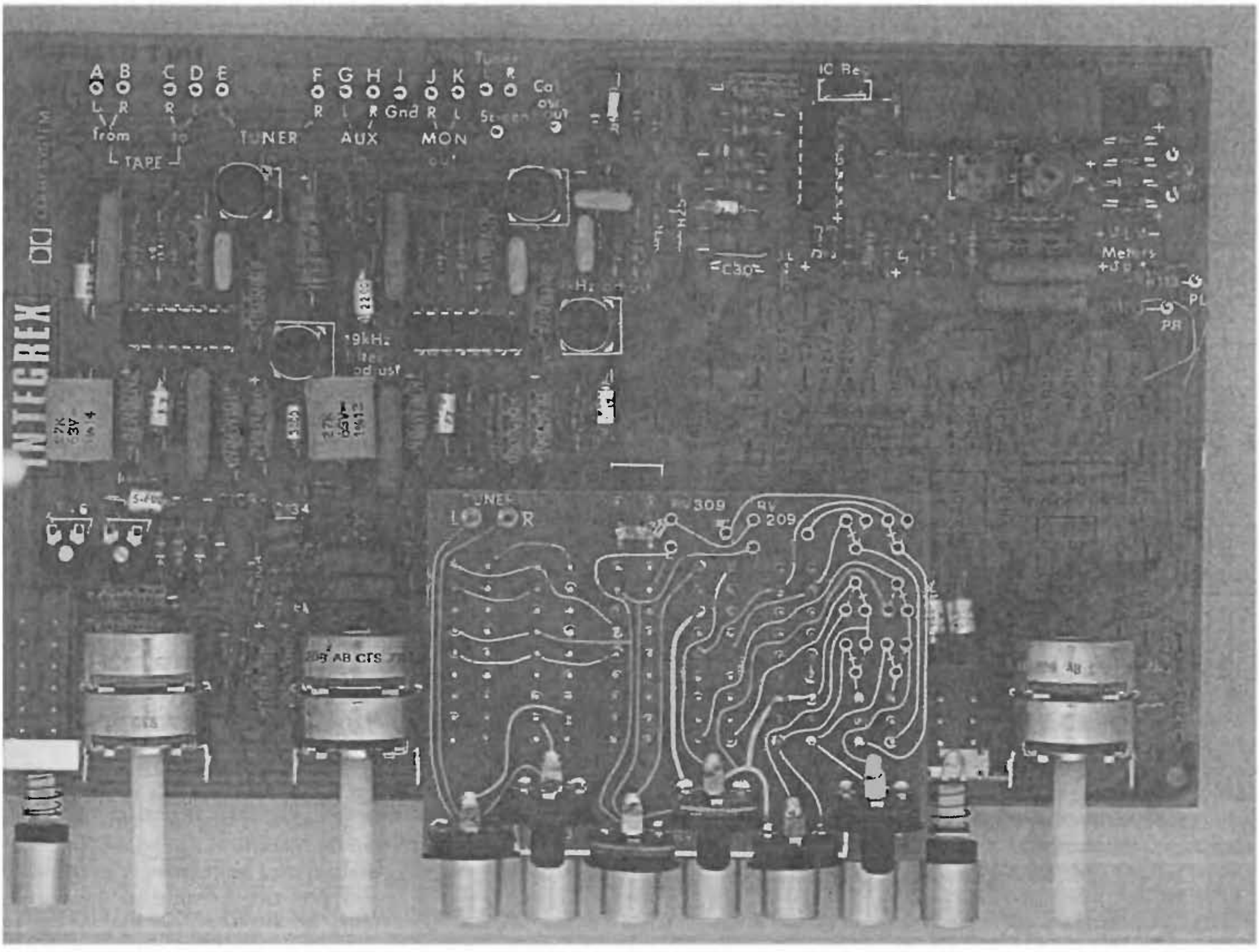
Single-ended methods An alternative approach is the dynamic one of altering the level of a signal by an amount that depends on the signal level, at either the sending/recording end or at the listening end. In examining such dynamic techniques, it is expedient to look at the possibilities from a steady-state signal level point of view, with the thinking that frequency- and time-dependent variations can be seen as special categories within a level classification. In practice, however, the success of each kind will undoubtedly depend on how well complicated time-varying, multi-frequency signal patterns are responded to by the processing circuitry, and to whatever psychoacoustic, or perceptual, effects such as auditory masking can be discovered and made use of.

The simplest kind of device, within our terms of reference, is the low-level oise gate, depicted graphically in Fig. 1(a), which eliminates signals below a certain threshold level. More useful is a stepped noise gate, where signals and noise below a certain threshold are attenuated by a finite amount rather than an infinite amount—Fig. 1(b). There are a host of variants on this theme, Fig. 1(d) showing another possibility.

A number of commercially available expanders have used the general approach of Fig. 1(b), including H. H. Scott's "dynamic noise suppressor" and R. Burwen's "dynamic noise filter," operating only at low and high frequencies and with a passband that varies according to signal level. The Philips "dynamic noise limiter" is another example, though its operation is restricted to high frequencies. With these devices, the bandwidth restriction at low signal levels can cause some loss of program. Further, any reduction of noise level that can be achieved is likely to be modulated by intermittent mid-frequency signal components, giving rise to what is called breathing. Because they are "single-ended," these techniques can result in a distortion of dynamic range. Thus, you can either have the original dynamic range plus unreduced noise, or a distorted dynamic range and loss of some low-level information with a reduced noise level—but not both at the same time.

Besides altering the level of low-amplitude signals, a similar expansion can be achieved by expanding high-amplitude signals, Fig. 2(c), but as well as exhibiting the two major disadvantages already mentioned, this would suffer a third. By having a variable-gain element operating at a high level, there are obviously greater risks of generating intrusive, unwanted signals as a result of overshooting, high non-linear distortion, and a high circuit noise level.

Dynamic processing is often carried out prior to recording or transmission. The low-level compression characteristics of Figs. 1(c) and (e) and the high-level characteristic of Figs. 2(a) and (b) both enable the average signal level to be increased relative to the noise level. But in themselves they suffer from the same disadvantage as do the ex-



panders. Clearly, single-ended methods are difficult to adopt to normal high-quality reproducing systems.

Complementary methods One way of avoiding the difficulty of alteration to dynamic range is by the complementary method—the dynamic equivalent of static "equalization." In complementary systems, signal processing before transmission and recording, normally compression, is followed by an equal degree of complementary processing, normally expansion, prior to audition so that the original dynamic range is restored. Noise added by the medium after compression is reduced by the degree of expansion used. In the expander of Fig. 1(b) the complementary compressor characteristic would be (c) and the complement of (d) would be (e). Likewise, the transfer characteristics of Figs. 2(b) and (c) form another compander system.

Another kind of diagram makes it easier to visualize what happens so far as levels are concerned. Figure 3(a) is a typical high-level compander characteristic, showing both the compression and expansion curves. Its equivalent level diagram of Figure 3(b) shows the reduced dynamic range (indicated by arrows) where the maximum level to be handled by the interposing medium is assumed to be the same—the region marked "overload margin" giving an increased margin against overload and thus lower distortion. Figure 3(c) shows the same reduced dynamic range produced by the characteristic of Figure 3(a), but with the intermediate gain shifted so that the low signal levels can be increased in relation to the noise level.

Figure 4(a) shows low-level compander characteristics, with the level diagram of Fig. 4(b) illustrating the use of the compressed dynamic range to bring up the low-level signals relative to the noise. Figure 4(c) shows how, by reducing the levels by a constant amount, increased overload margin can be obtained. Notice the similarity between Figs. 3(b) and 4(c) and between Figs. 3(c) and 4(b), the difference being the locating of the region of "linear" operation at either a high level or a low level. Despite the immediate visual contrast between Figs. 3(a) and 4(a) there is clearly a close resemblance between curves.

In practice the characteristic curves do not have the discontinuities shown, corners being rounded to prevent objectionable noise modulation. The curves should be capable of easy realization, be readily reproducible, and the two complementary curves must be matched to within the required tolerance.

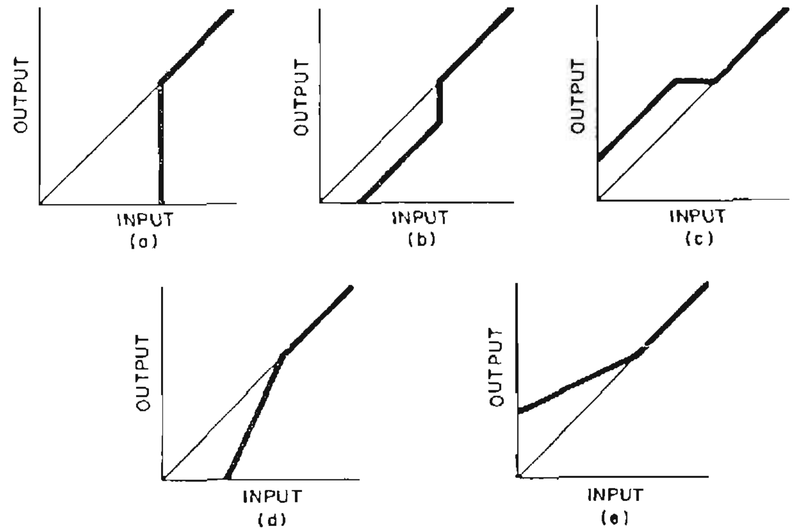


Fig. 1—Low level noise gate (a) simply loses both signal and noise below a certain threshold level. Finite attenuation of low-level signals is achieved with the expansion of the transfer characteristics of (b) and (d) Such "single-ended" expanders often reduce noise at the expense of distorting dynamic range. Compressors at the signal-source end can raise low-level signal above noise levels, but can similarly distort range (c) and (e).

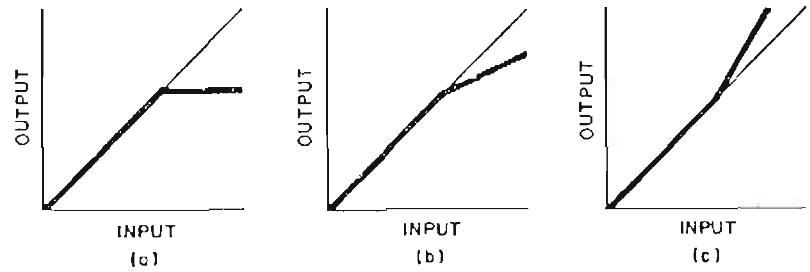


Fig. 2—High-level limiter and compressors (a) and (b) and expanders (c) suffer a possible disadvantage because of processing at a level where distortions would be more obvious.

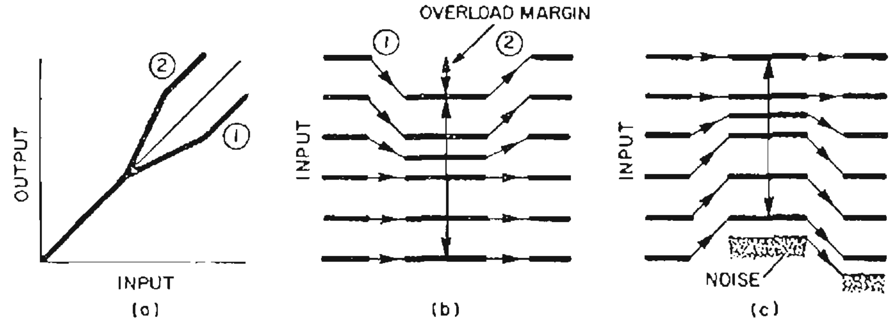


Fig. 3—Complementary high-level system (a) is able to reproduce original dynamic range while either reducing maximum level to give more overload margin (b), reducing noise (c), or giving a combination of both.

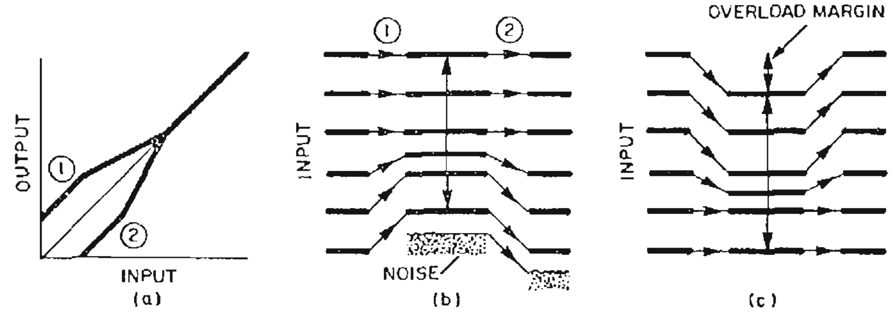
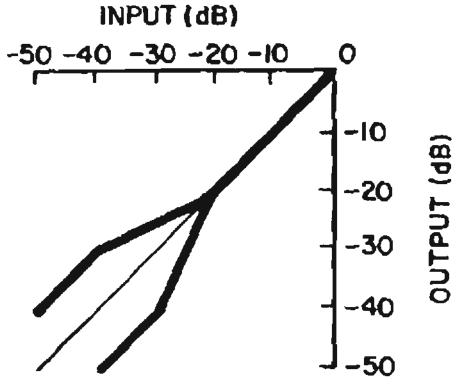


Fig. 4—Low-level complementary system has advantage that any distortion products are at a low level where they are less likely to be audible.

Two recently introduced studio companders use the general approach of Fig. 2(b) and (c), but with a threshold that is much lower than indicated. The dbx compander uses a square-law curve above a certain threshold (−60 dBm), which in logarithmic terms is a 2:1 compression ratio. The Burwen "noise eliminator" uses a cubic law (logarithmically, a 3:1 compression ratio) above a certain threshold. (A fixed h.f. pre-emphasis and a level-independent bandwidth are also features of these systems.)

In general, such high level companding techniques suffer from a number of drawbacks: Poor tracking between the two processors, high sensitivity to errors in gain between processors, overshooting, and a risk of overmodulation, both of which could lead to compression in the transmission medium that would go uncorrected on expansion, noise modulation by signals, modulation-product formation as a result of rapid gain changes, all of which are undesirable in a high



In most electronic signal-processing systems, there is usually some maximum level beyond which the signal must not be allowed to go and to which levels are frequently referred. Transfer characteristics are therefore usually given in the quadrant shown in which the point of reference is made to some arbitrary maximum level, rather than the zero signal level of Figs. 1 to 4. (A zero at the axes intersection would represent 0 dB, and not an origin as in Cartesian coordinates.) In practice such curves are not discontinuous but are smoothly connected to prevent unwanted modulation and to permit easy realization and matching.

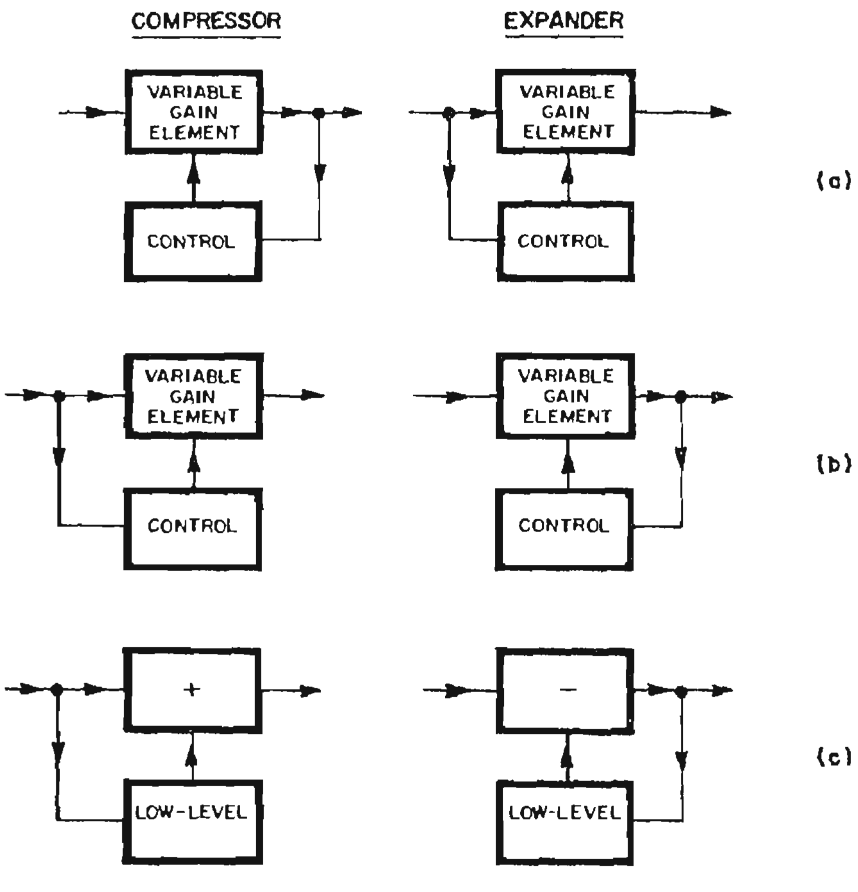


Fig. 5—Conventional companders use the equivalent complementary systems of (a) or (b) whereas the Dolby system (A&B) uses an additive to method (c), enabling processing circuitry to be separated from the main signal path.

quality link. High level companders can be very useful, however, in telephone circuits for example and the British Post Office's Lincompex scheme is an example of a compander in which dynamic range is reduced to zero. (Subsequent expansion would not be possible were it not for the fact that information on signal amplitudes is contained in separate pilot or control channel.)

The low-level method (Fig. 4) has a high tolerance of channel gain errors, produces modulation distortion at low signal levels rather than high levels, and there is less risk of overloading the medium. It seems a good idea anyway because one might expect the ear to be less sensitive to low-amplitude effects than to the same effects at high level. This then is the basic companding technique used in the Dolby system.

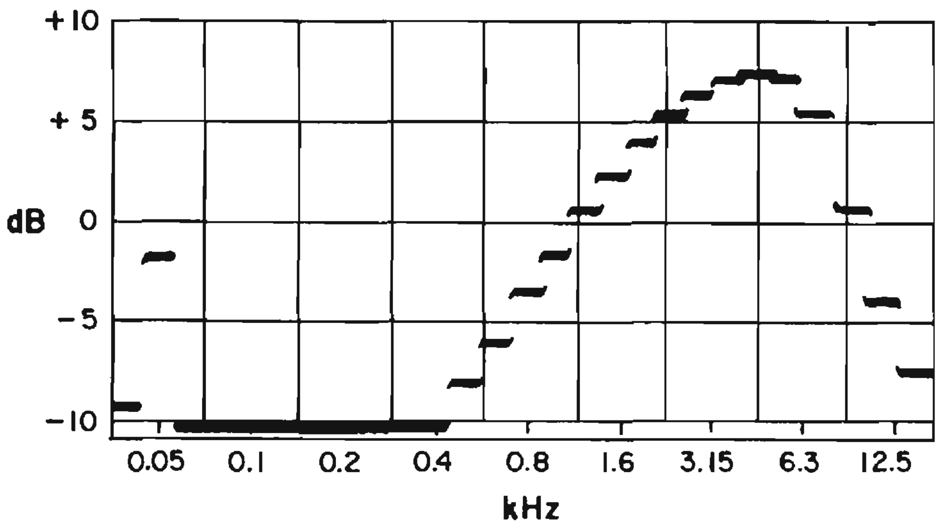


Fig. 6—Noise spectrum of low-noise ferric-oxide tape shows the problem is a mid- to high-frequency one, rather than a broadband one for which the Dolby A system was developed.

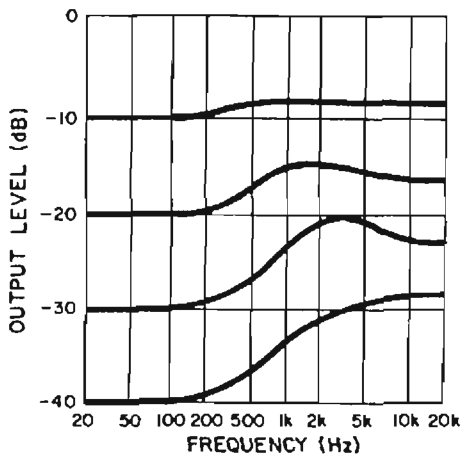


Fig. 7—In the Dolby B system, low-level high-frequency signals are boosted during encoding by 10 dB at low levels, the amount of boost decreasing as input level increases. Characteristics shown are amplitude-frequency response curves, with input level as a parameter, for the encoding process.

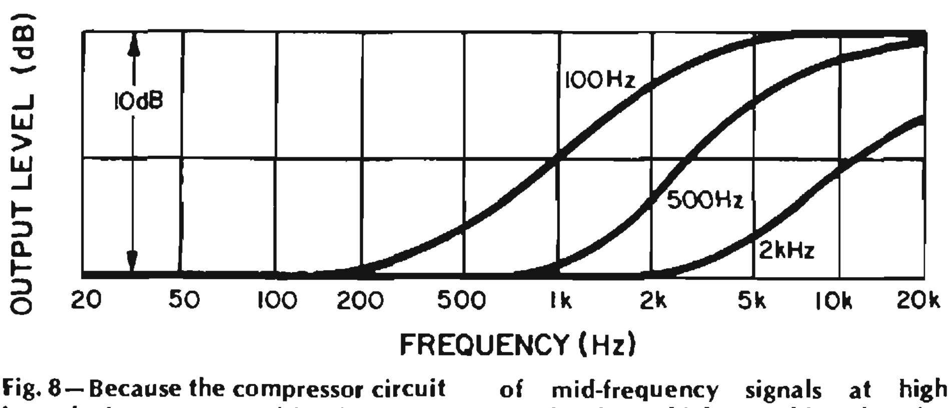


Fig. 8—Because the compressor circuit is made frequency sensitive in B-type processors, frequency at which boost, and hence noise reduction, occurs rises with increasing input level. Thus noise reduction is preserved in the presence of mid-frequency signals at high amplitude, which would otherwise reduce or prevent noise reduction. Curves show response below threshold level in presence of 0-dB tones.

Dolby Low-level Compander

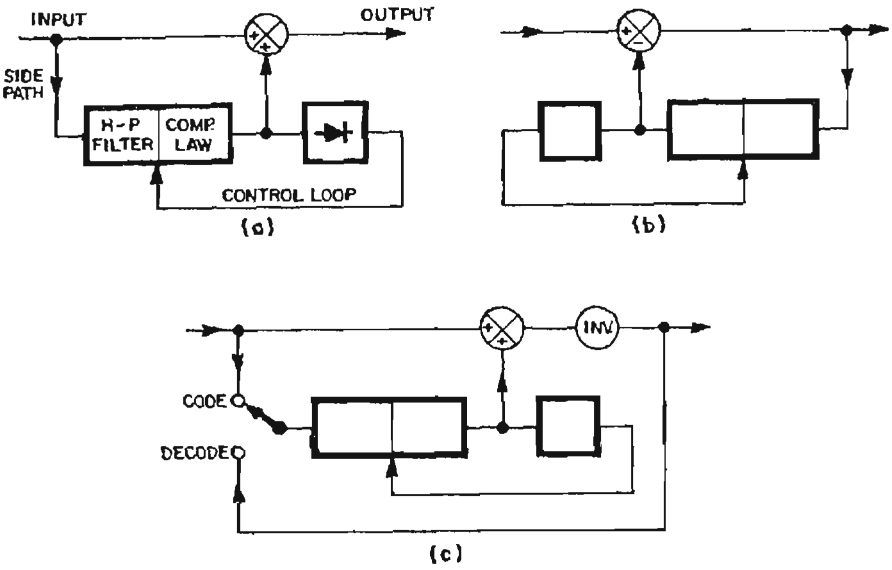
In conventional companding systems, there are two equivalent ways of achieving compression and expansion. One is to derive a control signal, after subjecting the input signal to a variable-gain element (compressor); expansion or "decoding" would then be achieved by the converse process—the control signal being derived prior to a variable-gain element (expander), Fig. 5(a). The equivalent, alternative way is to derive the control in the compressor part before the variable-gain element and to subsequently expand by using a control obtained after the variable-gain device, Fig. 5(b).

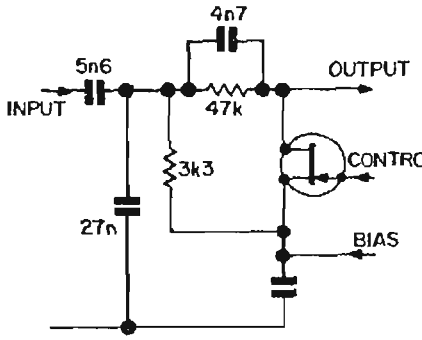
The Dolby technique makes use of a different approach—with an important difference; compression is achieved by deriving a special low-level signal that is added to the main signal, and expansion is obtained by subtracting a low-level signal from the main one, Fig. 5(c). (Within the low-level processor block, compression is achieved with method (a).)

Of course, the required compander characteristics could have been derived in the normal way, i.e. by direct action of a compressing circuit on the main signal path Figs. 5(a) and (b); but in the low-level approach the whole range need not be subjected to processing. It is obviously in the interests of quality that low-level signals be processed separately, leaving the main signal to a linear path whose quality is not restricted by that of the variable-gain path.

Tracking at high levels becomes easier using this low-level approach, and a tracking error due to channel gain variation would occur at an unobtrusively low level. Additionally with this technique, it is found that sufficiently accurate tracking can be maintained using a control derived from peak and average signal values. Thus, the elaboration of an rms-derived control, which would strictly be necessary for channels having a non-linear phase-frequency response, is avoided.

Notice that in the subtractive part of Fig. 5(c), a negative feedback loop is effectively formed in the low-level "contribution" to the main path. Advantage of this is taken in the Dolby system (and in the JVC a.n.r.s. system) in that an identical network to that used to produce the additive low-level signal at the encoder, can be used in forming the subtractive component at the decoder, merely by inserting the network in the negative feedback loop of a main path amplifier. Among other things, this means a single processor can be used for both encode and de-





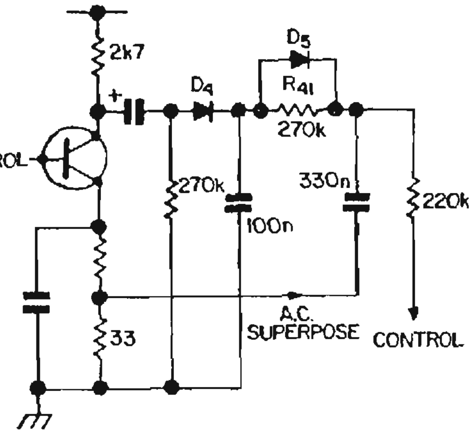


Fig. 9—Characteristics of Fig. 8 are realized by a voltage-controlled filter and compressor which adds up to 10 dB of subsidiary signal to the main path during encoding (a). In decoding, a similar network is used to subtract from the main path (b), the network forming part of a negative-feedback loop. This loop means that identical networks can be used for encoding and decoding. By placing the phase inversion in the main signal path, as shown (c), it can be left permanently in- circuit, simplifying encode/decode switching.

Fig. 10—Output of high-pass filter decreases after the compression threshold, set by gate bias, has been exceeded by the control signal. Response curve of combined fixed and variable filter sharpens when the two turnover frequencies coincide.

Fig. 11—Control-loop integrator has variable attack and decay times depending on speed and amplitude of signal changes. Large transients cause D_5 to conduct, shortening loop response time. Super-position of a.c. signal on control loop is to allow FET to operate symmetrically, thus keeping second harmonic distortion to a low level.

code functions by a suitable switching arrangement.

In a wideband compander of this kind having the kind of characteristic of Fig. 4, a low-amplitude signal below the operating threshold would result in the maximum amount of low-level boost being applied, and on decoding the noise level will be appropriately reduced; a high-amplitude signal would result in no noise reduction. Thus an intermittent high-amplitude signal could modulate the noise level, producing breathing (unless high-level signals were present in the same frequency band as the noise. This breathing can occur in any kind of wideband compander, of course).

In the Dolby A system, this effect is overcome by splitting the audio band into sections in the additive signal path, each section having its own compression and control circuitry. A high-amplitude signal in one band will not then prevent noise reduction being obtained in bands above and below. Within each band, the presence of a high-amplitude signal is relied on to mask, that is reduce the perceptibility of, noise components close to that signal. Studies of auditory masking show a shift in the hearing threshold in the presence of a (masking) tone, which effect can extend upward in frequency to a considerable extent; downward to a much lesser extent, the amount depending on the level of the masking tone.

When the economics of band splitting are judged against the extent of this masking effect, the amount of noise reduction required, and the value of threshold level in relation to the benefits of the additive technique, it turns out that four bands give a satisfactory compromise of cost versus performance. Splitting the band with 12 dB per octave filters in the ranges 80 Hz low pass, 80 Hz to 3 kHz band pass, 3 to 9 kHz band pass, and 9 kHz high pass would give a uniform 10 dB boost (and hence noise reduction) to low-level signals, as determined by setting compression threshold at 40 dB below peak operating level. By making the 3- to 9-kHz band-pass filter into a high-pass filter, an additional boost is obtained, gradually increasing from about 5 kHz to a maximum at 15 kHz. The lowest band provides reduction in the hum and rumble range, the second reduces mainly broad-band noise, tape print-through and cross-talk, while the upper bands reduce hiss.

Dolby B-type System

The cost and complexity of the A system is not really appropriate to consumer products. Moreover, for slow-speed tape machines, in particular, the noise spectrum has a different distribution from that occurring in the studio situation, on account of the slower tape speed and thin oxide layers used in tape cassettes. Figure 6 gives a typical DIN-weighted noise spectrum taken from a low-noise ferric oxide tape cassette, showing the noise problem is mainly a mid- to high-frequency one. Noise reduction in the B-type system is therefore limited to this frequency range and Fig. 7 shows the amount of boost (hence noise reduction) applied at various input levels; a fixed high- pass filter in the subsidiary signal path, as is done with the Dolby system, would achieve this end. The Dolby system filter *also* has a variable frequency characteristic, but what about the noise modulation which in the A system is greatly reduced by the multiband feature?

In the B system, such a filter prevents high-level, low-frequency tones from activating the compression circuit, so there is no noise modulation by l.f. components. But there could still be modulation by high-level signals close in frequency to the filter cut-off. The trick used to avoid this in the Dolby B circuit is to move the filter pass band higher in frequency, so that the high-level signal would then be below the filter pass band. The curves of Fig. 8 show the effect of the variable-frequency filter under the influence of a high-level tone at three different frequencies; the lowest-frequency curve representing the lower limit of the combined filter's translation in frequency. As the figure shows, with a high-amplitude tone of 500 Hz applied, there is some 8 or 9 dB of noise reduction at 10 kHz; even with a tone at 2 kHz there is still some noise reduction obtained. Had the filter pass nd remained fixed, these high-level nes would have caused the variable-gain element to operate, resulting in reduced or zero contribution from the subsidiary path, and hence little or no noise reduction.

Figure 9 shows a simplified block diagram of B-type processors, the encoder at (a), and the decoder at (b) with the same filter and compressor circuitry now in a negative feedback loop. In (b) a phase inversion is clearly required, which in (a) it is not. A simple dodge, that leads to a simplified encode/decode switching arrangement, is to relocate this phase inverter in the main signal path after the summing amplifier. The inverter can now remain in-circuit permanently, forming part of the feedback loop only during decode, Fig. 8(c).

Circuit operation The way in which the voltage-variable filter and com- ssor operates is interesting. A fixed high-pass filter, formed by the parallel combination of the 5.6- and 27-nF capacitors (fed from a low-impedance source, they are effectively in parallel) and the 3.3 kilohm resistor determines a turnover frequency of 1.5 kHz (Fig. 10). Imagine that a simple compressor then follows, i.e. a variable attenuator formed by a fixed resistor and the FET voltage-variable resistor (ignoring the 4.7-nF capacitor). The FET is to be controlled by a direct voltage obtained after rectification of the signal passed by the filter FET combination. Without any direct voltage applied to the FET gate, as would be the case for inputs of any level below the filter passband and for low-level inputs within the passband, the FET resistance is nominally infinite. The filter circuit would thus give minimum attenuation of h.f. signals and pass them to the main path, allowing h.f. noise reduction to be obtained. When an h.f. input is of sufficiently high level for the control signal to overcome the FET bias (this determining the compression threshold), the direct voltage to the gate would cause the FET resistance to fall, attenuating the signal, and reducing the amount passed to the main path. As the h.f. signal increased, a progressively smaller amount would be returned to the main path. Operation of this principle is shown by the curves in Fig. 7(a), which in fact apply to the Dolby B and a.n.r.s. circuits.

By replacing the fixed resistor with a capacitor (4.7 nF) in series with the FET resistance a second, variable, high-pass filter is formed. With increasing FET gate voltage, actioned by an increasing signal frequency and/or level, the filter characteristic rises in frequency, "overtaking" the fixed filter curve to largely determine a new, higher, pass band (after equilibrium between signal level control and filter is reached). Thus the frequency at which a significant signal is returned to the main path is raised, as depicted in Fig. 8, preserving some h.f. noise reduction in the presence of mid-frequency signals. In the region where the two filter curves are close, the combined filter shape is sharpened to around 10 dB/octave, so the effect of the filter action is heightened in this region, and the immunity of the circuit to noise modulation therefore improved.

Dynamic Operation

To avoid modulation products being generated by rapid changes of gain in the compressor, which may or may not be canceled in the complementary expansion process, a long attack time is desirable in the rectifier circuit providing the FET control voltage. On the other hand, a short attack time is needed to minimize the effect of overshoots, which could have an amplitude equal to the amount of compression.

The elegant solution chosen is to use a time constant that depends on the rate of change of signal. Referring to Fig. 11, the 2.7-kilohm collector resistor and the 100-nF capacitor allow rapid following of a slowly changing input signal. But the time constant of the 270-kilohm (R_{41}) and 330-nF component gives an attack time for the control signal of 100 mS—long enough to prevent audible modulation products being formed. Diode D_5 is not brought into conduction because the voltage drop across it is never large enough (the discharge time of the 100-nF path being shorter than through the 330-nF capacitor). For large transient changes of input signal, the potential across the 100-nF rises faster than that at the 330-nF capacitor so D_5 conducts, reducing attack time to around 2 mS or less. Between these two extremes, charging of the 330-nF capacitor is shared by D_5 and R_{41}, as determined by the potential difference across them.

While the effects of transients are limited by the variable attack time, high amplitude transients require more rigorous treatment. Overshoots, as a result of the control loop not operating quickly enough, are limited to a maximum amplitude of 2 dB by two silicon clipper diodes. When added back to the main path the clipped subsidiary signal can result in a momentary distortion of between 2 and 4 per cent, lasting for around 1 or 2 mS. This distortion is momentary, occuring when the causal program transients mask the distortion and the ear is least susceptible to it.

As with attack time, recovery time is as much a problem—it must be so short that noise reduction immediately following a high amplitude signal is restored within the time the ear takes to recover its normal hearing threshold, but not so short that low-frequency or modulation distortion results. The Dolby A system insures a 100 mS decay time normally, but for large sharp reductions in signal level this value is reduced. The variable decay time is not a feature of the Dolby B system.

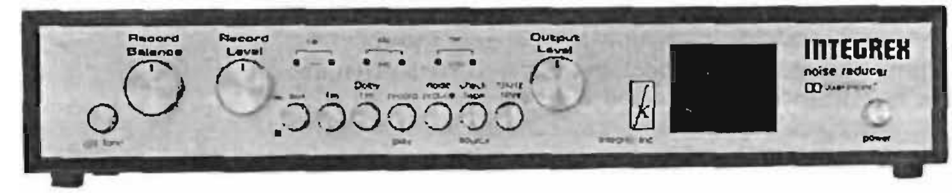
In Fig. 11 there is a proportion of a.c. signal from the emitter resistors superimposed onto the direct control voltage. This is to maintain symmetry of operation in the FET and thus keep second harmonic distortion to a low level by ensuring that $v_{gd}=v_{gs}$. Therefore an a.c. signal is applied to the gate that is half the value of that at the drain. By this means, and by keeping the signal voltage at the FET low by the capacitance divider prior to the FET, distortion is reduced from a peak of 0.5 per cent to 0.05 per cent (at 1.5 kHz and −15 dB).

This simplified introduction to noise-reducing systems should help in understanding operation of the B-type circuit, for which kit building instructions will be given in next month's issue.

Acknowledgement

We wish to thank Dolby Laboratories Inc. for their cooperation in developing this design and particularly Ian Hardcastle for his valuable assistance.

Build a Dolby Noise Reducer



Part II— Kit Building Instructions

This is the second portion of a three-part series on building a Dolby B-type noise reducer and deals with construction of a stereo pair of channels used for alternate encoding and decoding. The next portion of the series will deal with building an additional pair of channels for simultaneous encoding and decoding as for three-head tape machines. While the project was found to be both fun and worthwhile by the editor who built the kit, we do not feel it should be attempted as a first project by the novice. Though no test gear is required, a good set of tools, including a low-wattage, fine-tip soldering iron and a small pair of cutters is essential. Most helpful will be prior experience with kits and familiarity with the resistor and capacitor color codes. A kit of parts is available from Integrex, Inc., whose advertisement appears near the end of this article.—Editor.

Kit Assembly Instructions

In this portion of the article, instructions are given for building two Dolby processing channels for alternate stereo encoding or decoding. Ignore the component locations marked in black, i.e. all component numbers above 200. These are for the version intended for three-head tape decks and provides simultaneous encoding for the *Record* head and decoding of the output of the *Playback* head. Instruction for this section will be given next month.

Main PC Board

A number of PC board pins are supplied with the kit; fit them by inserting them from the foil-track side of the main board, tap down lightly with a small hammer or push in with the flat of a screwdriver so that the shoulder spline is firmly seated into the board. Solder them into place, making certain that **ALL 23** are soldered. The pin positions are as follows:

—Two for the transformer input marked "Vin."
—Four for right and left meter-drive outputs marked "Meters ± L and ± R."
—One for the calibration oscillator marked "Cal. Osc. Out."
—Eleven for the inputs/outputs and common marked "A" through "K."
—Three for the tuner leads marked "Tuner R," "L," and "Screen."
—Two at the right-hand end positions of resistors R13, 113 marked "PR" and "PL."

Close-tolerance components are packed separately. Mount and solder the close-tolerance components first, pushing all components close to the board before soldering. The close-tolerance components to be mounted are: R7, 107 (1%); R34 (2%); R35 (2%); C10, 110 (1%); C11, 111 (1%), C14, 114 (1%); C15, 115 (5%), and C4, 104 (5%).

There are five jumper links to be inserted on the main PC board. Use wire cut off from resistors for these at the positions marked "JL."

Fit the remaining resistors and capacitors EXCEPT R13, 113. Make certain that the electrolytic capacitors are inserted the correct way—the grooved end goes to the plus marking on the board.

Fit the small surface-mount trim pots RV6, 106, RV7, and 107. Fit coils L1, 101. **DO NOT ADJUST L2, 102.**

Fit the transistors, diodes, and ICs. Note the metal insert on the IC regulator and the round indentation on the ICs. The banded end of the diode goes to the plus mark on the board. The two transistors have a flattened "D" shape; the larger flat face, with the sharp corners of the "D" goes toward the two large, gray-bodied capacitors, C4, 104. See the outline drawing to check the locations of the collector, base, and emitter. Note the view is from beneath.

Before fitting the push-button switches, it is advisable to check that they function correctly as they are difficult to remove once soldered. Check mechanical interdependence of AUX, FM, and FM Dolby. Check with battery and bulb or ohmmeter for electrical operation; the switch positions are ganged in sets of three. Take care to push the switches fully into the board and insure that they fit squarely, using the front panel as an alignment guide; **any skew will result in misalignment with the front panel. Solder.**

Insert the "Cal. Tone" switch, taking care that the brass spring is up. Align it using the front panel as a template so that it is in line with the main bank of switches. Solder.

Solder the ends of R13, 113 away from the pin position. Leave the other end, the one toward the pin, standing loose, away from the board, and unsoldered.

Sub PC Board

Components are fitted on the **FOIL** side of the smaller PC board.

Insert the large, up-right trim pots RV3, 103, RV4, 104, and RV5, 105 into the board and solder the back leg. Attach the plastic adjustment inserts into RV3, RV4, and RV5. Adjust all of the trim pots so that they align with the "Cal." holes in the front panel and are square with the front of the sub PC board. Solder and trim remaining legs. The sub PC board should be spaced about 0.09 inch away from the top of the main switch bank to ensure that the "Cal." trim pot centers line up with the front panel holes. Wooden kitchen matches are a convenient spacer for this; the metal shoulders of the adjustment screwdriver are about this distance, but the driver should **not** be used while soldering the sub board in position.

PARTS LIST — INTEGREX NOISE REDUCER
Two-channel alternate encode/decode

RESISTORS		CAPACITORS	
R 1, 101	470K	C 1, 101	10μ
R 2, 102	150K	C 2, 102	10μ
R 3, 103	3.9K	C 3, 103	0.33μ
R 4, 104	4.7K	C 4, 104	10 nF 5% Gray body
R 5, 105	4.7K	C 5, 105	3900 pF Styrene 5%
R 6, 106	2.2K	C 6, 106	3000 pF Styrene, Installed 5%
R 7, 107	3.3K 1%	C 7, 107	2200 pF Styrene 5%
R 8, 108	47K	C 8, 108	10μ
R 9, 109	180	C 9, 109	10μ
R10, 110	270K	C10 110	5600 pF Styrene 1%
R11, 111	560K	C11, 111	4700 pF Styrene 1%
R12, 112	270K	C12, 112	10μ
R13, 113	330K	C13, 113	10μ
R14, 114	330K	C14, 114	27 nF 1% Styrene
R15, 115	150K	C15, 115	47 nF Mylar
R16, 116	150K	C16, 116	10μ
R17, 117	560	C17, 117	0.1μ
R18	3.9M	C18, 118	0.33μ
R19	3.9M	C19, 119	0.33μ
R20	1.2M	C20, 120	10μ
R21	18K	C21	10μ
R22	680K	C22	1000μ 25V
R23	680K	C23	0.1μ
˜24	680K	C24	1 nF Styrene
25	680K	C25	1 nF Styrene
R26	330K	C26	150 pF
R27	1M	C27	0.047μ
R28	3.9M	C28	0.047μ
R29	10K	C29	220μ 10V
R30	10K	C30	0.1μ
R31	100K	C31	47 nF Square, red plate ceramic
R32, 132	220		
R33, 133	220	C32	47 nF Square, red plate ceramic
R34	180K 2%		
R35	15K 2%	C33, 133	1 nF Disc
R36	82	C34, 134	1 nF Disc

SEMICONDUCTORS

TR1,	101	ZTX109C
IC1,	101	LM1011A
IC2		LM3900
IC Reg.		1415,131

COILS

L1,	101	30569
L2,	102	30568 Installed, **DO NOT ADJUST THESE COILS** for the 19-kHz filter.

POTENTIOMETERS

RV1	50K Log/Reverse Log Dual Control Pot
RV2	50k Log/Log Dual Control Pot
RV3, 103	50K Log Cal. Pot—Large, Upright
RV4, 104	50K Log Cal. Pot—Large, Upright
RV5, 105	5K Log Cal. Pot—Large, Upright
RV6, 106	47K Linear Trim Pot—Small, Surface-Mount
RV7, 107	1K Linear Trim Pot—Small, Surface-Mount
RV8	5k Log/Log Dual Control Pot

MISCELLANEOUS

SW1, 2, 4, 5, 6	7-position switch bank
SW3	4-pole on-off cal -tone switch
SW7	Main power switch, DPST
Large PC Board	
Small PC Board	
Phono socket assembly	
23 PC Board Pins (minimum)	
D1, 2, 3, 4	4001, 4002, 4003
D5, 105	1N914, 1N4148, 1S44
D6, 106	1N914, 1N4148, 1S44

Chassis; wood case; front panel with angle mounting brackets; knobs; shield for power supply area; line cord; strain relief; transformer; fuse and holder; meter; self-adhesive foam strip to cushion meter; bulb; 4-lug terminal strip; long sheet-metal screws with stand-offs for power switch; short sheet-metal screws for angle brackets to hold front panel; pan-head screws for main PC board; cadmium-plated screws with nuts for front panel and phono socket board; flat-head screws with nuts for transformer, fuse holder, and meter bracket; long flat-head screw with large washer to secure wood case to chassis; screened wire for connection of main and sub boards; meter and terminal strip connection wire; card for beneath main PC board; plastic inserts for trim pots RV103, 104, and 105 with adjustment screwdriver. Note: Extra hardware, e.g. screws, nuts, PC board pins, will be included.

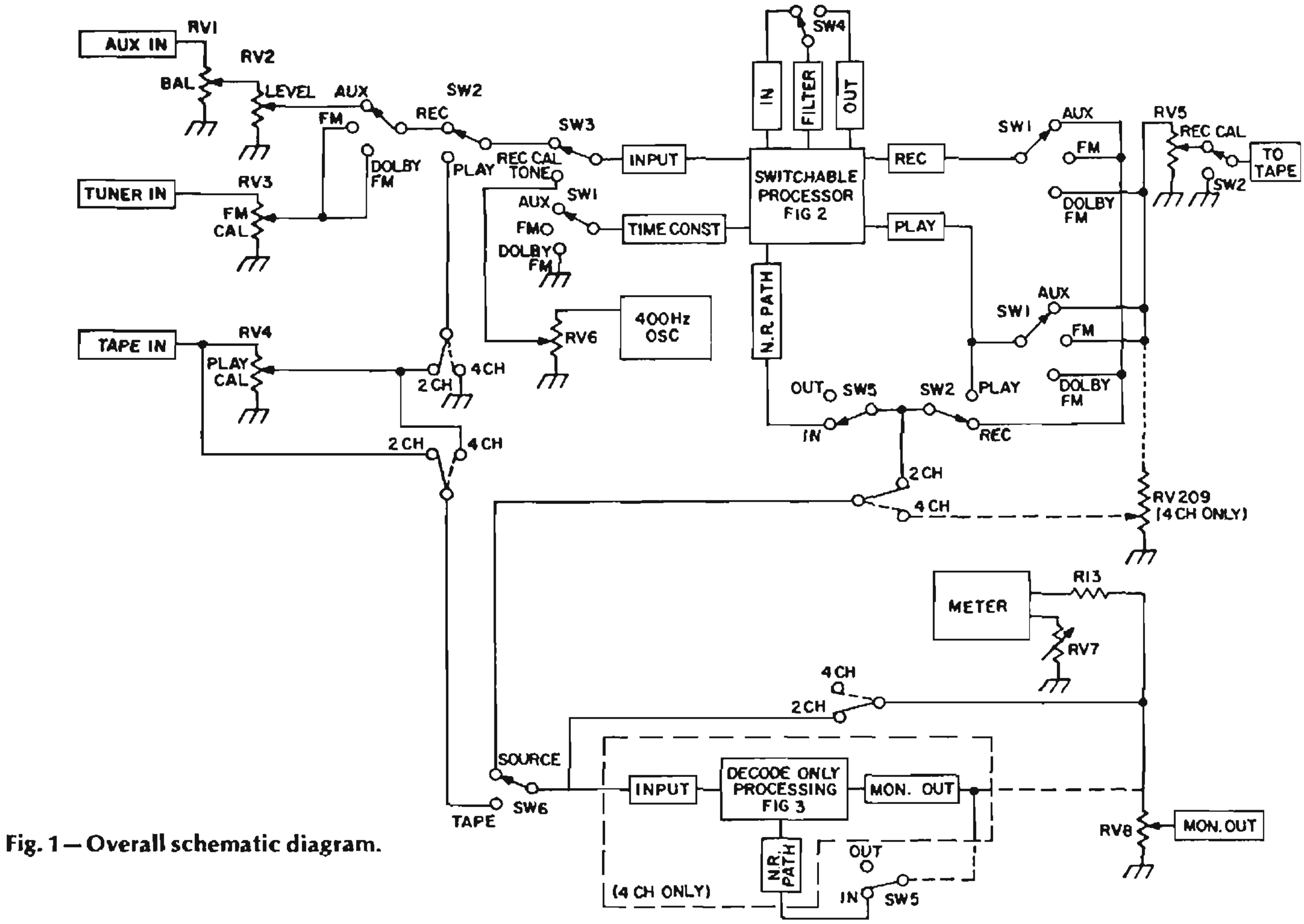


Fig. 1—Overall schematic diagram.

Position the sub board on the pins of the main bank of switches, check alignment using the front panel as a template and sighting through the "Cal." holes, and solder into position.

Join the areas of the sub board marked "Tuner L and R" to the corresponding points on the main board using the screened twin-lead cable supplied. Ground the **shield only** at the pin marked "screen." The other end floats; do not attach it to the sub board.

Returning to the main board, RV1, RV2, and RV8 can be fitted, using the front panel as a template to align the control pot spindles with the push buttons. Solder the pots with the front panel in position.

Check both boards for solder shorts and/or dry joints.

Crop all leads to avoid touching chassis.

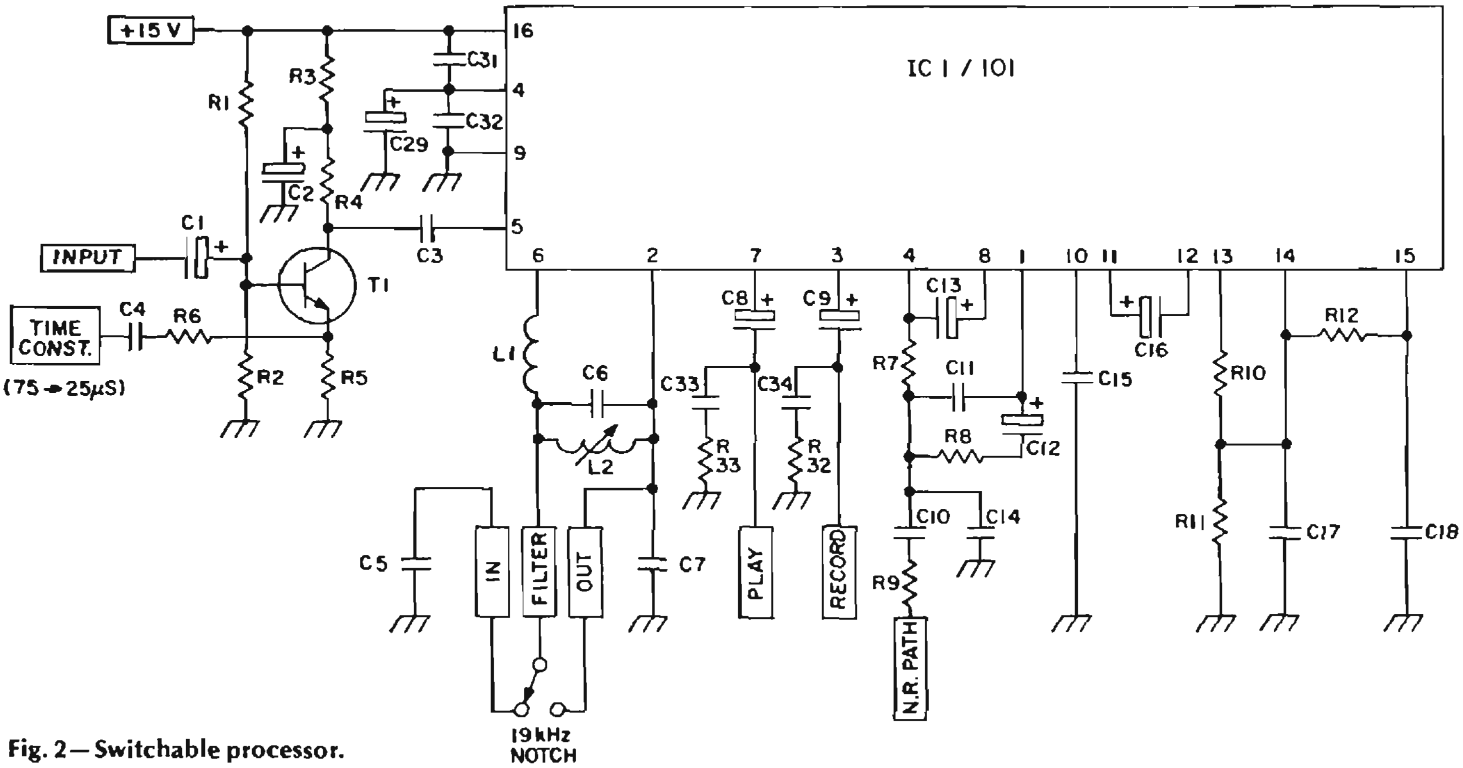


Fig. 2—Switchable processor.

Insert a thin piece of card between the main board and the chassis, and fix the board in position using the pan-head screws.

Phono Socket Board

Place the self-adhesive label on the phono socket board.

Fit the phono socket board onto the back panel from the inside using the cadmium-plated screws and nuts.

Cut a 5½-inch piece of the solid bare wire.

Thread it through the holes of all the **OUTER RINGS** and to pin "I" on the main PC board, which is marked "Gnd."

Connections between the **CENTERS** of the phono jacks are as follows. Looking at the phono socket board from the PC board side, connections between the centers and the pins on the board are, from left, **UPPER** level, A, D, E, G, and K. Similarly, the connections for the lower-level centers are, from left, B, C, F, H, and J.

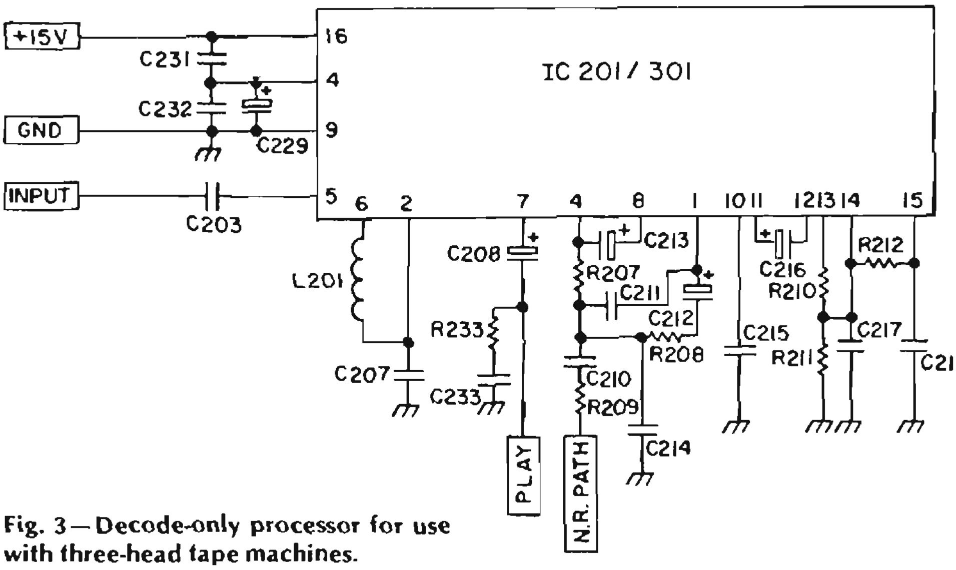


Fig. 3—Decode-only processor for use with three-head tape machines.

Off-Board Assembly

Note the exploded diagram of this area of the kit. Fit in position, using the flat-head screws from the bottom of the chassis:

—Transformer; black leads are the primary leads.

—Fuse holder, noting positioning peg.

—Main power switch, using long sheet-metal screws and stand-offs between meter/switch bracket and switch.

—Meter/switch bracket and terminal strip with the terminal attached via hole behind meter opening. It is convenient to hold the nut in position through the meter opening while tightening the screw.

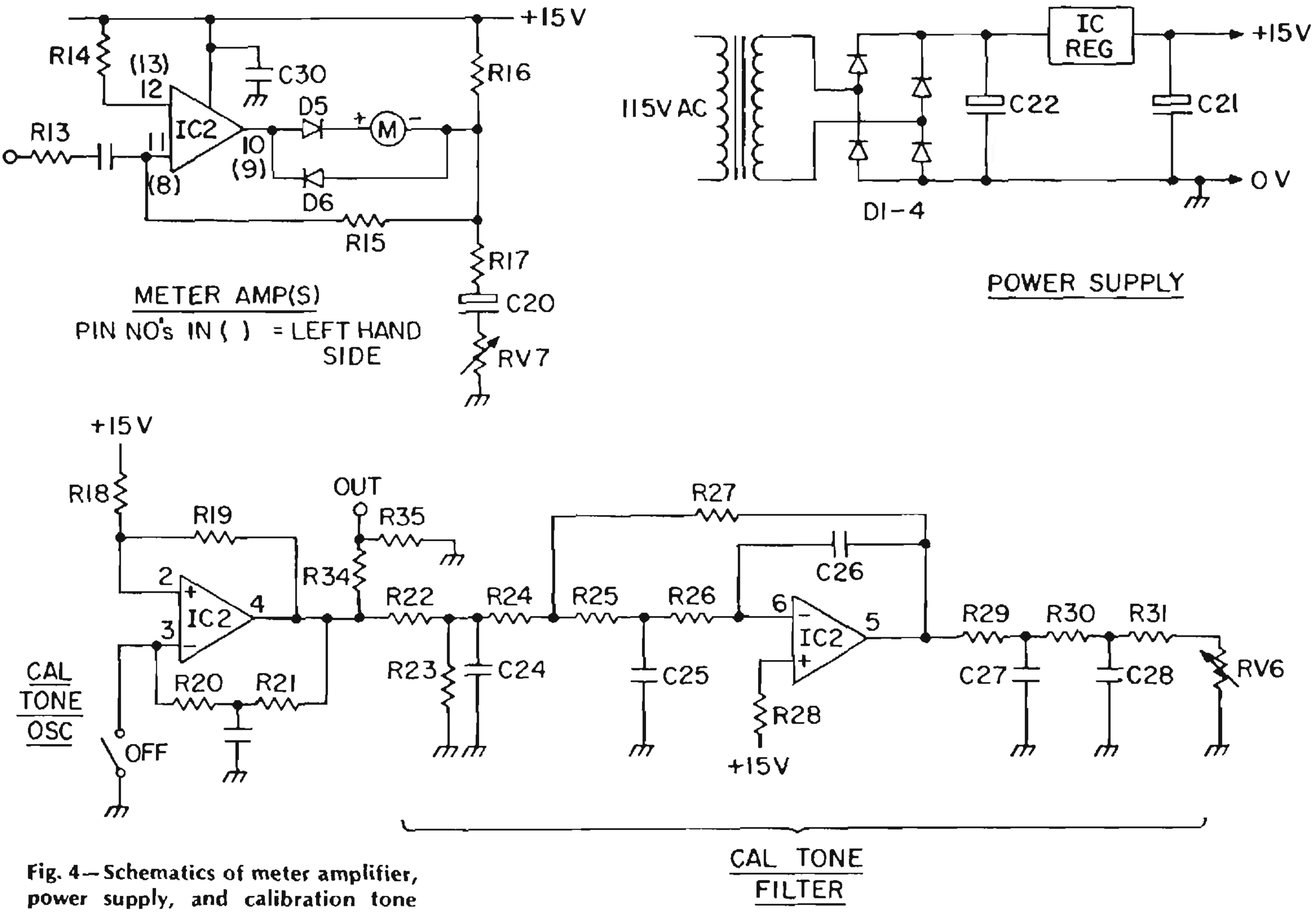


Fig. 4—Schematics of meter amplifier, power supply, and calibration tone oscillator and filter.

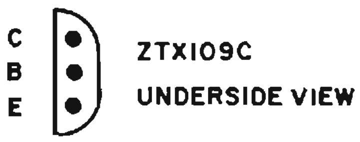


Fig. 5—Identification of leads of ZTX-109C. NOTE: THIS IS THE VIEW FROM UNDERNEATH.

Tape the meter to the front of the bracket using the self-adhesive foam between the meter and the bracket. The foam goes at both top and bottom. Normally the meter will be held in place by the front panel.

Feed the white secondary leads from the transformer forward to the meter/switch bracket and then to the left toward the main PC board. Fix the transformer screen in position, being careful not to nick the secondary leads. Cut the white leads just long enough to be attached to lugs No. 1 and 4 on the terminal strip. Strip and crimp the white leads to lugs nos. 1 and 4 but **DO NOT** solder. Strip both ends of the remaining two white wires and crimp one end of each to terminals nos. 1 and 4. **DO NOT SOLDER.** Do not use lug no. 2, as it is grounded.

Connect and solder the loose ends of the white wire to the Vin points near the upper right-hand corner of the PC board (when viewed from the front).

Cut R36 leads to the proper length and crimp them to lugs nos. 1 and 3.

Tack solder the leads of the grain-of-wheat bulb to lugs nos. 3 and 4. Position the bulb behind the meter.

Solder the three leads at lug no. 1. Solder the two leads at lug no. 3. Solder the three leads to lug no. 4.

Connect and solder the meter terminals to the four meter-drive pins (±M.R. and ±M.L.) near the Vin pins. Note that the terminals on the meters are polarized and are reversed in polarity from side to side.

Remove the transformer screen.

Clip about an inch from the end of one conductor of the a.c. line cord. Strip both ends of this short piece and solder between the front end of the fuse holder and the front-left terminal of the power switch.

Feed the a.c. power cord through the hole in the back panel, strip and solder the shortened end to the back lug of the fuse holder, and strip and solder the longer end to the back-left terminal of the power switch.

Strip and solder the black primary leads of the transformer to the other two lugs of the power switch.

Put the strain relief around the line cord outside the chassis, pull the line cord so that it will be snug between the back of the chassis and internal connections, and insert the strain relief into the back of the chassis using a pair of pliers.

Install fuse into fuse holder.

Install transformer screen, using flathead screws and nuts; screws insert from bottom of chassis. Be careful not to damage transformer secondary wiring.

Place self-adhesive Dolby label on back panel and red warning label on top of transformer screen.

Meter Calibration

The amplifier section of IC2, based on pins 2, 3, and 4, is wired as an unstable multivibrator switching between the 15-volt supply rail and 0 volts with a mark-space ratio of approximately 1 to 1 and a frequency of about 400 Hz. The real voltage swing is slightly less due to saturation voltages but is highly repeatable from one sample to another.

Fig. 6—Wiring of terminal strip.

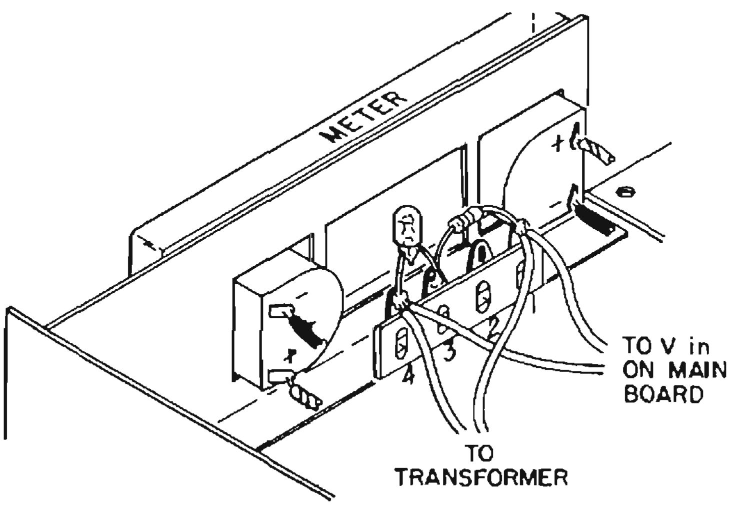
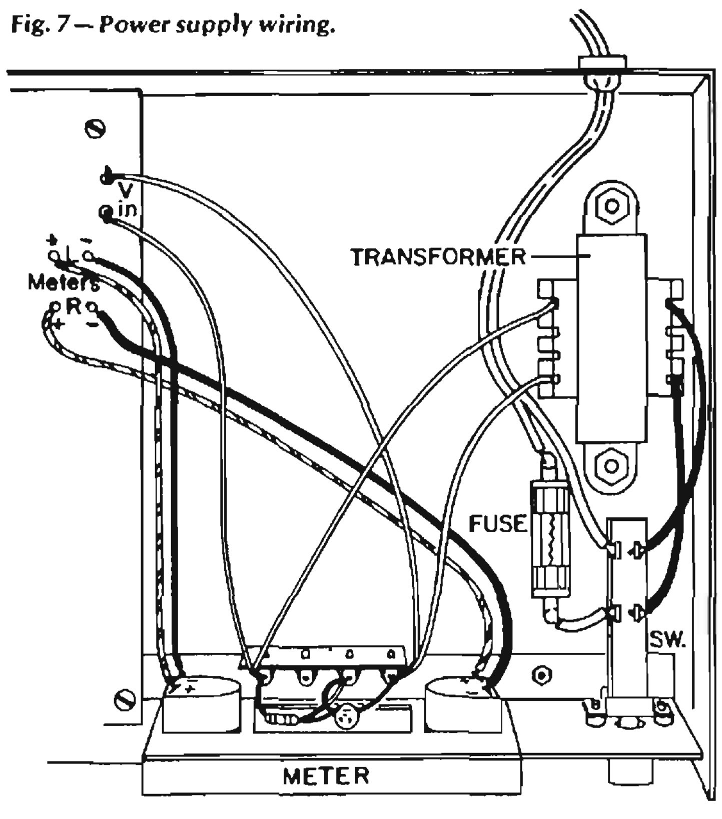


Fig. 7—Power supply wiring.



The calibration procedure is as follows:

Connect the "cal. osc. out" pin located in the back-middle portion of the main PC board to the end of R13 floating away from the board.

—Switch on the power.
—Push the "cal. tone" button in.
—Adjust RV7 for 0 dB on the right-hand meter.
—Switch the power off.
—Disconnect R113 and solder it to the pin PR, cut lead.

—Connect R113 to the "cal. osc. out" pin.

—Repeat steps above for the left channel using RV107.

—Disconnect R113 and solder it to the pin PL, cut lead.

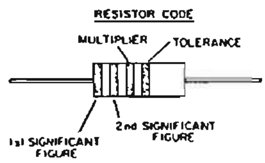
The meters are now calibrated for Dolby level, and they should be calibrated before the simultaneous encode/decode part of the kit is constructed.

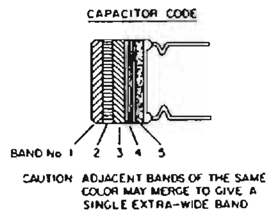
Oscillator Calibration

The square-wave output at pin 4 is low pass filtered by the active filter formed by the amplifier in IC 2 based on pins 1, 5, and 6 to produce a sine wave of less than 1 per cent distortion at 400 Hz. This signal is attenuated by RV 6 and 106 and injected into the circuit when the "cal. tone" button is pressed in.

To set the calibration oscillator ouput level, switch the unit on and push the "cal. tone" button in. DO NOT push Dolby FM or noise reduction while calibrating. Now adjust RV6 and 106 for 0 dB on the right-hand and left-hand meters respectively.

To be continued





RESISTOR COLOR CODE

COLOR	1st + 2nd SIGNIFICANT FIGURES	MULTIPLIER	TOLERANCE
Black	0	1	—
Brown	1	10	±1%
Red	2	100	±2%
Orange	3	1000	±3%
Yellow	4	10000	±4%
Green	5	100000	—
Blue	6	1000000	—
Violet	7	10000000	—
Gray	8	100000000	—
White	9	—	—
Gold	—	0.1	±5%
Silver	—	0.01	±10%
No Color	—	—	±20%

CAPACITOR CODES (CAPACITANCE GIVEN IN PF)

COLOR	DIGIT	MULTIPLIER	TOLERANCE 10 PF OR LESS	OVER 10 PF
BLACK	0	1	±2.0 pf	±20%
BROWN	1	10	±0.1 pf	±1%
RED	2	100		±2%
ORANGE	3	1000	±0.25 pf	±2.5%
YELLOW	4	10000		
GREEN	5		±0.5 pf	±5%
BLUE	6			
VIOLET	7			
GRAY	8	0.01	±0.25 pf	
WHITE	9	0.1	±1.0 pf	±10%
SILVER				
GOLD				

A New Recording System

Herman Lia*

This article describes a new recording system which has been developed by the Department of Magnetic Research & Development at Tandberg. As this system will be able to utilize the new generation of metal particle tapes, an overview of the merits of such tapes are also discussed in the context of future tape recording technology. This new recording system is called Actilinear, and patent applications have been filed and patents are pending.

The development of recording technology is being carried on in two separate and different groups: Manufacturers of magnetic tape represent one environment, and manufacturers of recording machines the other. Development in this connection is defined as the effort on the part of both parties to come up with better products for the consumers with regard to technical specifications, reliability, ease of operation, etc.

The development has traditionally been such that first the tape manufacturers bring new concepts to the market with properties that promise improvements over the existing state of the art. Then the machine manufacturers examine the nature of these improvements and how these can be used to advantage in the various types of recorders. Lately, however, a good collaboration has been established between these two groups, and this will naturally lead to better compatibility between the tape and the tape recorders. The greatest benefit of such a collaboration lies in the fact that the machine

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manufacturers will be able to include the advancement of new tape technology in an early phase of new product development.

If we look at the development of magnetic tape in the last 15 years, we find a clear trend towards higher saturation flux density, B_m, and higher coercivity, H_c. This has been a natural development based on a desire for a continuous improvement of the signal-to-noise ratio. In particular, it was an immediate requirement with the introduction of the compact cassette, since there was no opportunity to select track width and tape speed.

An increase of the B_m gives a better signal-to-noise ratio at lower and middle frequencies, whereas a higher H_c gives a better signal-to-noise ratio at higher frequencies. The first compact cassette that was introduced contained tape having H_c = 250 Oe. Then we had the so-called LH tape (low noise, high output) having H_c = 300 Oe. Later we got the CrO_2 level tapes with H_c = 550 Oe (e.g. TDK SA and Maxell UD-XL II). But this does not end the development. We know that experiments are going on today with types having H_c = 1000 Oe, and these are certain to come on the market in the near future.

In the midst of this development, a central question for the serious recorder manufacturing company is to what degree the present recorders will benefit from the new types of tape which will come on the market within a relatively short time. These are questions which gain momentum as new concepts from the magnetic tape industry are being marketed. We presently have recorders with selector arrangements which make it possible to choose between different types of tape, but only among the ones already on the market. Obviously, it is a hopeless task to design a recorder today which will give optimum performance with any type of tape five years from now. What can be done now, however, is to prepare the ground in the best possible way to allow for the possibility of adjusting the recorders to new types as they appear on the market.

Consider a tape with $H_c = 1000$ Oe which today is in the experimental stage, but which quite certainly will be commercially available in a year or less. Unless we are ready to take these new tapes into account now, we will end up in the same situation we had when the CrO_2 tapes came on the market, when there were no cassette recorders to take the full advantage of them. Of course, the recorders then were not adjusted to these types, but the fact that they did not have even a 3-dB margin in bias—and recording currents such that they could have been adjusted—is a clear testimony to a lack of a progressive design philosophy at that time.

One can learn from errors, however. The tendency today is towards a far greater ability to provide headroom and adjustments, and there is a desire to bring advanced design concepts into realization. This has, in fact, been the guiding spirit in the development of the recording amplifier chain in the new Tandberg tape recorders, for compact cassette as well as open-reel recorders.

Conventional Recording Systems

The conventional method of designing a recording amplifier is well known and will not be dealt with in detail. We will just note that the summation of recording current and bias current in the recording head is done through *passive* components, and this leads to compromise solutions which have their distinct and pronounced weaknesses.

The following difficulties should be mentioned 1) small headroom margin, 2) slew rate limitations for strong signals and high frequencies which results in intermodulation, 3) poor isolation between oscillator and recording amplifier which results in interference tones, and 4) too low a margin in bias and recording currents for readjustments to adopt to the new high coercivity tapes.

The New Recording Chain

With the development of the new recording system we have left behind and abandoned any form of compromise solution. The new design philosophy is based on the optimization of modules, that is, the whole chain is divided into natural functions, and each function by itself is realized as a module. Hence, a solution is obtained which is optimum on all points at the same time as a system is realized which is more amenable to adjustment to new types of magnetic tape. The new recording amplifier chain is shown in Fig. 1.

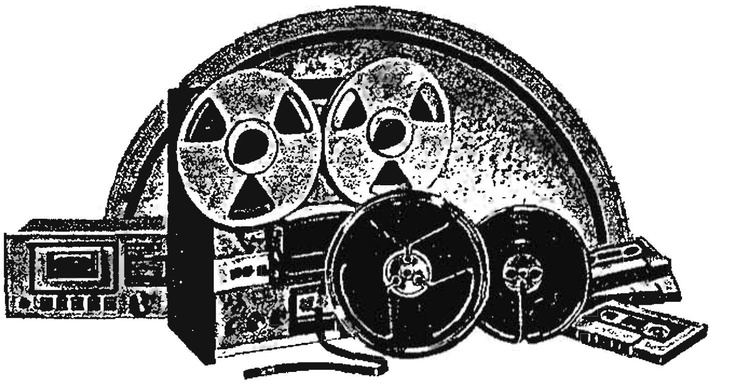
The Particular Functions

The equalizer module: This amplifier will give the recording chain the proper frequency equalization such that the overall frequency response of the recorder becomes as linear as possible. The C_2,R_4 network gives proper equalization at low frequencies, whereas R_2, R_3, C_1, L_1 gives the desired equalization from mid-frequencies and up.

Internal adjustment of recording sensitivity: This is simply the poteniometer R_5 and provides an internal sensitivity adjustment of the recording signal.

The transconductance module: This module has two main functions. It converts a voltage from the poteniometer R_5 to

Gains Using Metal Particle Tapes



As we have presented the new recording amplifier which is especially suitable in connection with high coercivity tape, we take the opportunity to carry out calculations for such a tape as an example. The most important figures to note are the S/N ratio at low and high frequencies and the total signal capacity integrated over the entire audible frequency range.

The S/N ratio at low frequencies is proportional to the maximum remanence flux density, B_r, in the tape and the coating thickness, d. At high frequencies, the S/N ratio increases proportional to the coercivity force, H_c.

To determine the signal capacity, we use Channon's definition:

$$SB = \int_B^{log} \left(1 + \frac{S}{N}\right) \Delta f,$$

where S is the maximum obtainable signal and N is the tape noise. For further details into this matter we refer to an earlier article in the *Audio* (April, 1977), where all relevant formulas for these calculations are stated.

To get an idea of the improvement with the new tape, the results are presented realtive to the Maxell UD which is one of the most popular types of tape used today. In the following table are listed the most important physical properties and the calculated figures for S/N ratio.

The improvements have been verified by measurements on a sample received from the 3M Company a few weeks ago. The measured values agree with the calculation with an accuracy better than 1 dB.

This new tape, as seen, is certain to present another dramatic improvement in tape recording performance levels, and particularly in the compact cassette format. Tandberg has plans to introduce recorders that include provisions for the usage of metal particle tapes as soon as they become commercially available.

Tape Qualities	UD	UDXL-II	Metal Particle
Retentivity B_r (Gauss)	1430	1540	3400
Coercivity H_c (Oersted)	360	545	1030
Coating Thickness (μm)	5.0	5.5	3.8
S/N Ratio at 333 Hz * (dB)	0	+1.5	+5.7
S/N Ratio at 10 kHz * (dB)	0	+3.6	+9.1
Signal Capacity * (dB)	0	+3.0	+8.5

*With reference to the Maxell UD.

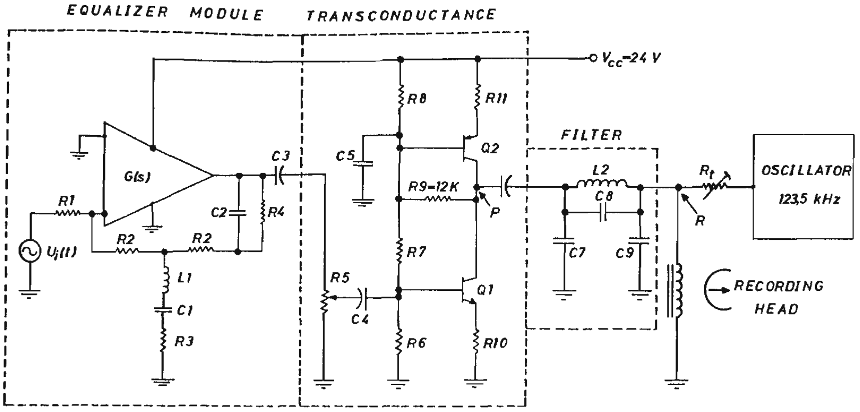


Fig. 1 — Block diagram-schematic of the Actilinear recording amplifier chain.

a current i_s which is the recording current. It shall also provide an electrical isolation between the oscillator and the recording amplifier such that interference tones are avoided and completely eliminated.

The circuit consists of the two transistors Q_1 and Q_2. Q_1 is used in a common-emitter configuration, and it has $R_g = 12$ kilohm and Q_2 as a collector load. Advantage has been taken of a special property of transistors in that the collector can appear as a low resistance to d.c., but a high impedance to a.c. signals. The two collectors are connected at point P. This point is put at 12 V d.c. and can swing between 2 V and 22 V, and thus has a maximal dynamic range available for driving current i_s through the head. The d.c. current through Q_1 and Q_2 is about 10 mA so that each single transistor represents the equivalent of a resistance of 12 V/10 mA = 1.2 kilohm at d.c. The output impedance for a.c. signals, however, is 1/hoe = 20 kilohm for each transistor and, hence, the total output impedance seen at point P is approximately 5 kilohms. Since the recording head impedance is substantially less than 5 kilohms (200 ohm at 20 kHz), the circuit acts as a current source, that is, a constant voltage at the input gives a constant current i_s through the recording head. This is also the justification for the name *Transconductance Amplifier.*

Any residual oscillator voltage at point P is being prevented from being fed back to the input by C_5. This way the circuit provides an electrical isolation between the equalizer amplifier and the oscillator.

Filter Module: The filter module prevents oscillator signals at point R from entering into the point P and interfering with the audio signal. At point R Uosc = 20V, but is reduced to about 50 mV at point P. The filter is of the low-pass type with a trap at 123.5 kHz.

Calculation of Headroom Margin in the Recording Amplifier

Since the recording amplifier is designed as a transconductance block, the limitation in headroom is determined by the maximum available current in the output stage. This is actually the quiescent current in the two transistors Q_1 and Q_2, and is set to $I_{QDC} = 10$ mA.
The maximum available a.c. current is then $I_{QDC} \div 2\sqrt{2}$. The headroom margin is the ratio between this current and that which is necessary to record the tape to maximum recording level i_{sm}. Therefore, we have:

$$\text{HRM}^* = 20 \log \left(\frac{I_{QDC}}{2\sqrt{2}\, i_{sm}} \right)$$

*Where HRM is an abbreviation for headroom margin. Numerical calculations with $I_{QDC} = 10$ mA and $i_{sm} = 0.4$ mA gives HRM = 19 dB.

Calculation of the Slew Rate

The slew rate of an electrical signal is defined as the time derivative of the amplitude and is determined in Volt/μS or Volt/mS.
Let: $e(t) = Em \sin \omega t$, be a general signal with angular frequency ω and amplitude value Em.

The slew rate is given then by

$$S = \frac{\Delta}{\Delta t} e(t) = \omega \, Em \cos \omega t$$

The maximum slew rate occurs when the term cos ωt has its maximum value which is 1. That gives the usual formula S = ω Em. As an example, we will carry out the numerical calculations for the TD 20A (the new series of Tandberg reel-to-reel tape decks).

The impedance of a recording head is almost pure inductive at audio frequencies and can be expressed by Z_H — ωL where L is the inductance of the head. The maximum signal voltage across the head occurs at the maximum recording level and is given by e_{HM} (L) = ωL i_{sm}, where i_{sm} is the maximum recording current. Let $i_{sm} = I_{sm} \sin \omega t$.

The slew rate of the signal voltage across the recording head is determined by the following calculations:

$$S_H = \frac{\Delta}{\Delta t} e_{HM}(t) = W^2 \, L \, I_{sm}$$

where, fo = 20 kHz, L = 5 mH and $I_{sm} = 0.5$ mA, then S_H = 4 V/mS.

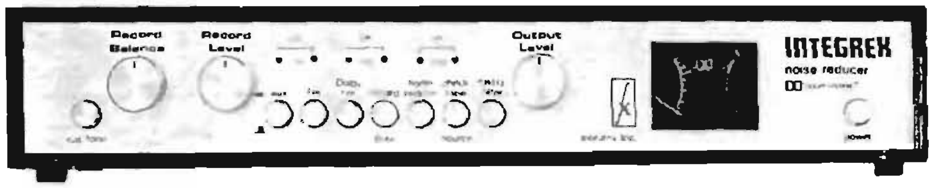
The maximum slew rate which the recording amplifier is able to handle is according to measurement equal to 400 V/mS. That gives a slew-rate margin of 40 dB which is a satisfactory figure.

Conclusion

A new recording amplifier chain has been designed which will be implemented in the new Tandberg cassette recorder, as well as open-reel machines. Improvements relative to conventional designs can be summarized as in the following:

1. More headroom in the recording amplifier, greater than 18 dB.
2. The recording circuitry operates at a lower voltage level and will, therefore, give less intermodulation because of slew-rate limitations.
3. An improved electrical separation between oscillator and recording amplifier which gives less interference with the oscillator.
4. Substantially greater possibilities of adjusting the recorder to new high coercivity tapes such as the new metal particle tapes.

Build a Dolby Noise Reducer



Part III— Encode-Decode Kit Instructions

This is the third and concluding portion of our series on building a Dolby B-type noise reducer and deals with the construction and adjustment of a pair of stereo decode channels. Paired with last month's stereo channels, the finished kit will simultaneously encode and decode a Dolby B-type program as for use with a three-head tape machine.

We should again emphasize that this is not a kit for the beginner and that a good set of tools is required, though no test gear is necessary. A kit of parts is available from Integrex, whose advertisement appears near the article.—Editor.

The "Check Tape" switch has become the "Tape/Source" switch. "Record" may now be permanently switched in.

Switch Functions

While the functions of the controls are, for the most part, self-explanatory, it is probably well to go through them in at least a brief fashion.

"Record Balance" changes the relative volume of the left and right channels of the "Aux." input.

"Record Level" changes the overall volume level of both left and right channels going to the recorder from "Aux." input.

The "fm," "rec.," and "play" trim pots change both the relative level of each of these sources and also the left and right channels, one versus anot

"Output Level" controls the volu level of both channels returning to the amplifier.

The "Cal. Tone" pushbutton, when pressed in, begins operation of the 400-Hz oscillator used for calibration of the meters.

The "Aux." pushbutton selects any external high-level source connected to the "Aux. in" terminals on the rear panel.

The "f.m." pushbutton selects standard broadcasts and does not use the Dolby processor.

To convert the alternate encode-decode processor to a simultaneous stereo encode-decode unit, very few alterations are necessary.

The black legend area, covering the 200 and 300 series of parts, is used for installation of the additional components as shown in the parts list.

The jumper links, JL, next to RV8, the output level control, and R333 are cut.

On the sub PC board, behind RV4, there are six tracks to be cut. This can be done with a small pocket knife or the point of a screwdriver. In the same area, six jumper links are to be installed and soldered. Check the diagram for these locations.

Install and solder trim pots RV209, 309 on the sub PC board.

The unit has now been converted to simultaneous encode-decode circuitry.

To adjust the new processors, switch the unit on, switch the "Cal. Tone", "Aux." and "Record" pushbuttons in. Leave "Check Tape" out. Adjust RV209, 309 to obtain 0 VU level on the meters.

PARTS LIST
Integrex simultaneous encode-decode kit

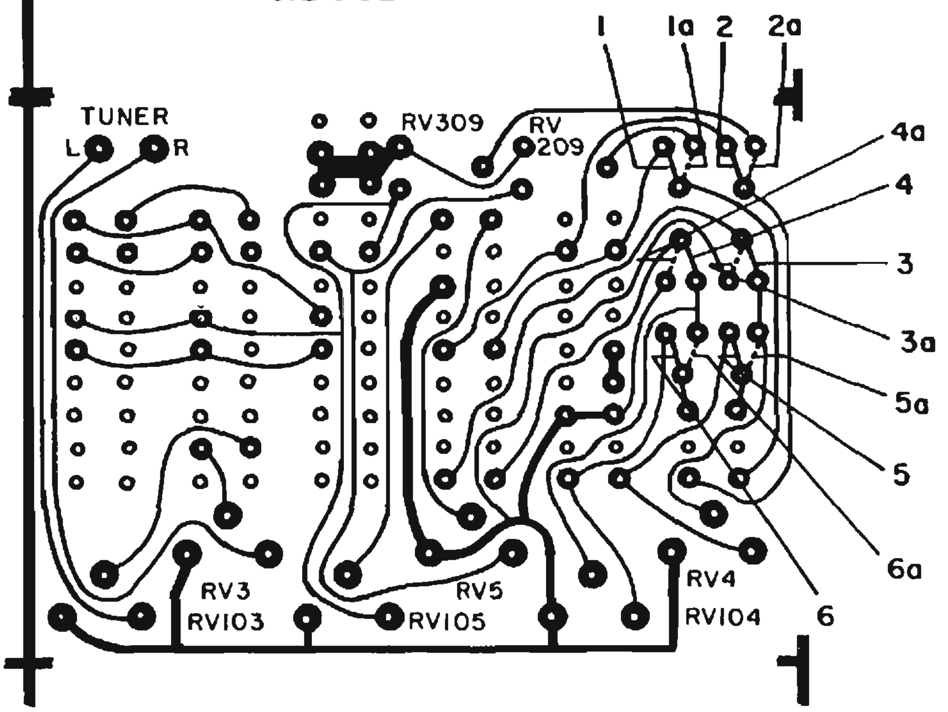
Resistors
R207, 307 3.3k 1%
R208, 308 47K
R209, 309 180
R210, 310 270K
R211, 311 560K
R212, 312 270K
R233, 333 220

Miscellaneous
IC201, 301 LM-1011A
L201, 301 30569 coil
RV209, 309 47K linear trim pot, small, surface-mount

Capacitors
C203, 303 0.33 μF, Mylar
C207, 307 2200 pF, Styrene
C208, 308 10 μF
C210, 310 5600 pF, Styrene
C211, 311 4700 pF, Styrene
C212, 312 10 μF
C213, 313 10 μF
C214, 314 27 nF, 1%
C215, 315 47 nF, Mylar
C216, 316 10 μF
C217, 317 0.1 μF, Mylar
C218, 318 0.33 μF, Mylar
C229 220 μF, 10V, Electrolytic
C231 47 nF, square red-plate ceramic
C232 47 nF, square red-plate ceramic
C233, 333 1 nF, disc

Note: It is no longer necessary that the meters be calibrated for Dolby level before these parts are installed.

CONVERSION SUB P C BOARD
FOIL SIDE UP



BREAK TRACKS IN SOLID LINE
MAKE TRACKS IN DOTTED LINE

The "Dolby f.m." pushbutton is for use with Dolby B FM broadcasts which are passed straight to the recorder and may be decoded for the monitor output.

The "Record/Play" button should only be switched when the alternate encode-decode unit is used, as it switches the circuitry into the proper mode.

The "Noise Reduce" pushbutton operates encoding or decoding depending on the choice of Record or Play. In the simultaneous unit, it operates encode in the record processor and decode in the replay processor. This must be selected whenever Dolby encode/decode is required.

The "Check Tape" pushbutton allows by-passing in the switchable unit and in the simultaneous encode/decode unit functions as a tape/source switch.

The "19-kHz Filter" pushbutton switches in a filter which keeps the 19-kHz FM pilot tone from fooling the Dolby circuitry when recording and/or when monitoring Dolby FM.

The "Power" switch we'll leave for you to figure out. Hope you enjoyed this one as much as we've enjoyed bringing it to you.

PERFORMANCE OF HIGH ENERGY MAGNETIC MATERIALS IN AUDIO CASSETTE RECORDING TAPES

Peter Vogelgesang *

The magnetic material which has been used almost exclusively in sound recording tapes for many years is the gamma form of iron oxide, or gamma Fe_2O_3. This material is a favorite because it is inexpensive, stable, and easily processed for tape manufacturing. Because early tape recording systems were limited in performance by electronic noise, mechanical instabilities, crude magnetic transducers, and by poor physical properties of magnetic tape, little attention was given to improving tapes by employing magnetic materials which had superior magnetic properties.

Today, solid-state electronic amplifiers produce only a small part of total system noise. Precision magnetic transducers can be manufactured of exotic ferrite materials or metallic alloys and have magnetic gaps controlled precisely to a few millionths of an inch. Magnetic tapes are slit to a width accuracy of one-thousandth of an inch, and tape surfaces have a mirror-like smoothness. Capstans, tape guides, and drive motors have been improved commensurately to utilize the improved physical properties of tape. And, a totally new improvement has been added; a preponderance of all magnetic tape manufactured today is enclosed in plastic cartridges or cassettes which protect it from both handling abuse and environmental contaminants.

As a consequence of these gradual improvements, the magnetic recording industry has seen a continued slowing of tape speed over the years. Full range audio recording once required a tape speed of 60 ips, but is now accomplished at the 1 ⅞ ips speed of the audio cassette. Limited bandwidth waveforms such as voice and background music are easily recorded at a speed of less than one inch per second.

With so many refinements occurring in related areas of magnetic recording, it was inevitable that improvement in the properties of the magnetic material used in tape would one day be required. Work directed toward this end was underway seriously in industrial laboratories in the early 60s.

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Today, consumers of audio cassettes have a choice of tapes using materials such as chromium dioxide (CrO_2), cobalt-modified iron oxide, and tapes made with multiple layers of these and other materials in combination. Manufacturers of cassette recorders have equipped their machines with multi-position switches to accommodate the new tapes by changing recording bias level and equalization.

The unique magnetic properties of chromium dioxide and cobalt-modified iron oxide used in these tapes are described by the generic term, "high energy." It is the purpose of this article to describe high energy materials and to show why and how they are used.

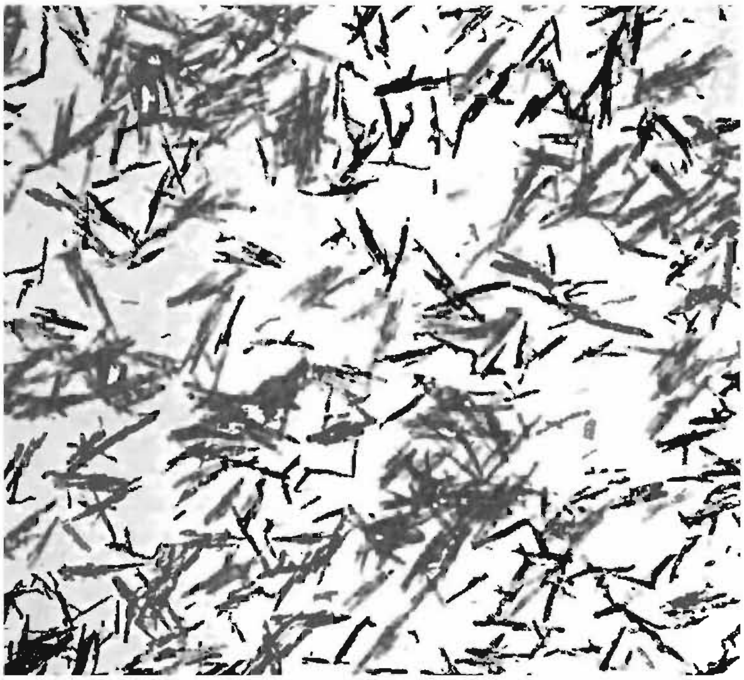


Fig. 1—An electron photomicrograph of a typical iron oxide magnetic material was made with a magnification of 45,000. The average length of these particles is about 0.4 microns, and the ratio of length to width is six to one.

Technical Background

Iron oxide magnetic material contained in recording tape consists of tiny, needle shaped or acicular particles which have an average length of 0.4 micron (16 millionths of an inch) See Fig. 1. These particles are mixed as a fine powder with a fluid binder, and the ingredients are milled using a process similar to the manufacturing of paint. The milled material, called a dispersion, is coated onto the surface of polyester film and then oven dried to make recording tape.

Each magnetic particle is an independent miniature permanent magnet. However, it behaves magnetically unlike a large magnetizable bar of iron with which we are all familiar. Because of the small size of the particles, each particle contains only a single magnetic domain. This domain cannot be demagnetized nor can it be made magnetically stronger or weaker. It is forever a permanent magnet of fixed strength, and the only magnetic change it will undergo is a polarity reversal of the magnetic field it produces. Fig. 2 shows how a single particle changes polarity and magnetization when immersed in a magnetic field which is reversing in a sinusoidal manner. A particle will behave in this manner when the direction of the applied field is parallel with the long axis of the particle. A very different characteristic is observed when the field is at right angles to the long axis. In magnetic tape, particles are aligned so that the long axis is parallel with the direction of the recorded tracks. Such orientation is achieved by immersing the coated tape in a strong magnetic field before the coating is dried, and while the particles are still mobile.

The time required for a single domain particle to reverse polarity, or switch, probably has never been measured (1) and may be considered instantaneous in any practical application of magnetic recording. In numerous laboratory investigations and computer simulations of the recording process, particle switching time has never surfaced as a limitation to the highest frequencies that can be recorded. Thus, designers of recording systems have built video recorders which record frequencies in the tens of megahertz without having to take into account a finite-response time of the magnetic tape.

Due to differences in size, shape, and chemical composition, particles within a bulk quantity of material will have a distribution of magnetic responses. In other words, although each particle within a batch switches at one critical intensity of applied field, the critical intensities of all particles are distributed in a manner which produces a familiar bell-shaped curve. Under the influence of a controlled applied field, a portion of the particles can be switched to one polarity, while the remaining particles stay in the original condition. Because the magnetic field produced by a bulk quantity of particles is the sum of all individual particle fields, the bulk quantity can be weakly or strongly magnetized, or it can be completely demagnetized. This effect is explained with the help of the magnetization curve of Fig. 3.

Coercivity of individual particles is largely responsible for coercivity of the bulk material. However, the density with which particles are packed together also influences the coercivity of a specific quantity of material (2). Coercivity of bulk material is reduced as packing density increases. Saturated remanent magnetization of bulk material also is dependent not only upon remanent magnetization of the particles, but on how closely particles are packed. Since the magnetic field produced by a quantity of material is the sum of the fields produced by individual particles, a greater number of particles in a given volume will produce greater remanent magnetization. Thus, remanent magnetization increases as packing density increases.

This fact has led to the necessity of distinguishing between the intrinsic remanent magnetization of particles and the magnetization of bulk quantities of particles. Intrinsic remanent magnetization essentially defines the remanence of isolated magnetic particles, and is determined by the chemical and physical makeup of the material. Remanent magnetization of a bulk quantity of material is dependent on this intrinsic property, but also upon how densely the particles are packed together. Remanence of a magnetic tape, then, is a function of 1) intrinsic remanence of the magnetic material, 2) density with which the material is packed, and 3) how thickly the material is coated on the tape.

Packing density in magnetic tape is limited by the percentage of polymer binder which must be used to hold particles together in a tough film. Attempts to increase magnetic material content beyond this point result in chalky coatings which will not withstand mechanical abrasion in recording machines. The "energy product" of a magnetic material is obtained by multiplying the values of M and H for a given point on the second quadrant of the major hysteresis curve. For the purpose of defining the performance of a particulate magnetic material used in recording tape, the "energy" of the material can be defined as the area enclosed by the second quadrant. Thus, energy can be increased by increasing either coercivity or remanent magnetization, and remanent magnetization, in turn, can be increased either by changing the intrinsic remanent magnetization of the particle or by increasing packing density of particles. For reasons which are the subject of the remainder of this article, increased remanent magnetization by either means should be accompanied by a corresponding increase in coercivity in order to obtain a maximum possible signal output of tapes used in audio cassette recording or any other kind of very short wavelength recording. The so-called "high energy" magnetic tapes available today in audio cassettes contain magnetic materials which have higher coercivity than gamma Fe_2O_3. The term "high energy" is a relative expression which uses iron oxide as the conventional standard.

The Effects of Demagnetization

The magnitude of a long wavelength electrical signal generated in a magnetic transducer during playback of a recorded tape is proportional to the remanent magnetization of the tape. A tape having twice the remanent magnetization of another will produce twice the electrical output. High transducer output is desired in order to obtain a high signal-to-noise ratio in the recording system.

As pointed out previously, remanent magnetization can be increased by 1) increasing the intrinsic remanent magnetization of the particles, 2) increasing the density of the magnetic material by packing more particles into a given volume, and 3) coating the magnetic material more thickly on the tape. At 1 ⅞ ips tape speed, none of these approaches to increasing remanent magnetization will produce a corresponding increase of high frequency output. A thicker coating will not increase high frequency output because short wavelengths are recorded only near the surface of a magnetic tape, and the thicker coating will create only an imbalance in the low and high frequency response of the recording system. The other two approaches to increasing remanent magnetization will not greatly improve high frequency output because of an effect known as "self-demagnetization."

Self-demagnetization in magnetic tape recording causes a loss of high frequency or short wavelength information at peak sound levels where the tape approaches magnetic saturation. Although a magnetic recording system may have an ideally flat frequency response at 20 dB below the maximum operating level (MOL) of the tape, the high frequency end of the audible spectrum becomes suppressed as the MOL is approached. Finally a point is reached where no amount of recording level increase will boost high frequency output, even though low frequencies retain a substantial headroom. Figure 4A shows that a cassette tape constructed with magnetic material having a coercivity of 340 oersteds is only 4 dB down at 19 kHz when the recording level is 20 dB below the MOL. At 0-dB recording level the output is 12 dB down at 10 kHz and is virtually non-existent at 19 kHz.

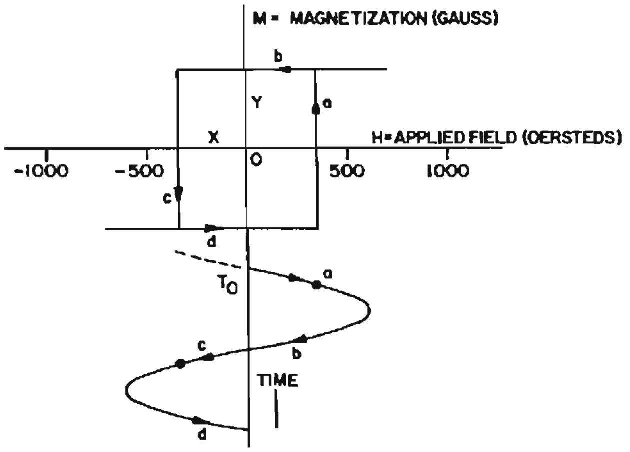


Fig. 2—The switching characteristics of a single-domain particle. The "X" axis represents the polarity and magnitude of a magnetic field, measured in oersteds, applied to a magnetic particle so that the direction of the field is parallel with the long axis of the particle. The sinusoidal waveform shows how the field changes with time. The "Y" axis represents the magnetization (M) of a single domain particle measured in gauss.

Starting at T_0, the applied field increases in a positive direction, passing through point A, which is equal to the coercivity (H_c) of the single particle, or about 340 oersteds for gamma Fe_2O_3. At this point, the applied field acts upon the particle to reverse its field virtually instantaneously. The new magnetization of the particle is exactly equal to the previous value, but polarity is reversed, producing a symmetrical figure.

The sinusoidal applied field returns to zero, but the particle remains magnetized in the positive direction. Not until the applied field reaches point C does the particle again switch to its original state. The pattern formed by these two variables is an ideal rectangle having sharp corners due to the instantaneous switching of the particle. This kind of figure is obtained with a single domain particle and will be considerably modified when particles having a wide distribution of coercivities are combined.

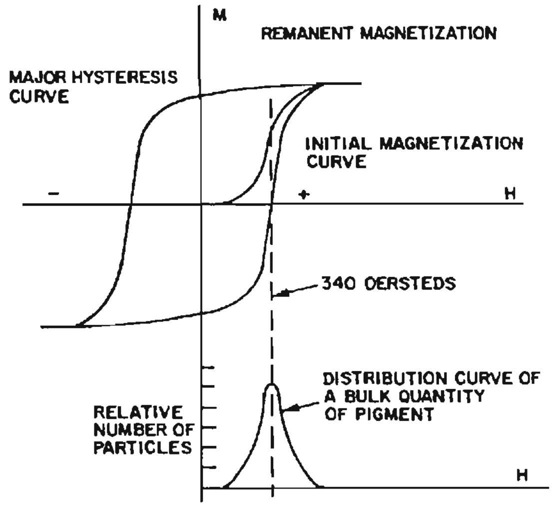


Fig. 3—The magnetization characteristics of a bulk quantity of iron-oxide particles which have a distribution curve in which a preponderance of particles switch at a coercivity of 340 oersteds. A small percentage have lower coercivity, and a similar percentage have higher coercivity, as shown in the figure. The distribution curve is essentially symmetrical. When initially mixed together, it can be assumed that the particles will be oriented in a random manner so that the net magnetic field produced by a bulk quantity is zero. In this state the bulk quantity can be considered demagnetized.

If an applied field is increased in the positive direction, magnetization of the bulk quantity follows an "initial magnetization curve" which is the integral of the distribution curve. The point of steepest slope of the initial magnetization curve coincides with the peak of the distribution curve, and this is the point where a maximum number of particles are switched. When the applied field is said to have reached saturation, and no further increase in applied field will increase magnetization.

Once saturated, the material then changes magnetic state in a manner described by the major hysterisis curve, and magnetization of the material in the absence of an applied field is termed remanent magnetization.

Although many people will not consider a loss of high frequency response at high levels to be a serious listening defect, Intermodulation distortion which results from this problem has a profound effect upon the quality of recorded music. With most types of music this intermodulation distortion may be more destructive to good listening quality than harmonic distortion, and it will occur in a magnetic recording system at levels well below the conventionally accepted maximum output level.

Figure 4B contrasts the high frequency performance of a tape which employs a magnetic material having a coercivity of 550 oersteds. The improvement in short wavelength response is attributable to a reduction of self-demagnetization.

Loss of high frequencies due to demagnetization occurs because of the very short wavelengths produced by a tape speed of only 1 ⅞ ips. The cassette is most vulnerable to this problem. Demagnetization losses at tape speeds of 15 and 7½ ips are negligible. Let us examine the causes of demagnetization using the familiar example of bar magnets.

Any permanent magnet has both an internal and an external magnetic field. The shape and intensity of both fields are dependent upon the physical shape of the magnet. The ratio of length to cross-section dimensions of a magnet is called the proportionality factor. That part of the internal field which passes through the body of the magnet has a demagnetizing effect which limits the magnitude of remanent magnetization, that is, it limits the strength of the magnet after the magnet has been withdrawn from a magnetizing field. As shown in Fig. 5, the pole ends of a long permanent magnet are far apart, and much of the internal field extends into air around the magnet. A magnet with this configuration will experience little demagnetization, and therefore it can be magnetized to a fully saturated state. The magnet in Fig. 6 is broad relative to the distance between pole ends. In this case a larger part of the internal magnetic field will exist in the body of the magnet, causing demagnetization. Even though these two magnets may be subjected to the same magnetizing field, the short magnet may have less remanent magnetization when withdrawn.

Demagnetization is resisted by maintaining a certain ratio between remanent magnetization and coercivity. A magnet having relatively low remanent magnetization will not have an intense internal field, and thus will remain saturated even though it may have a corresponding low coercivity. But a high remanence

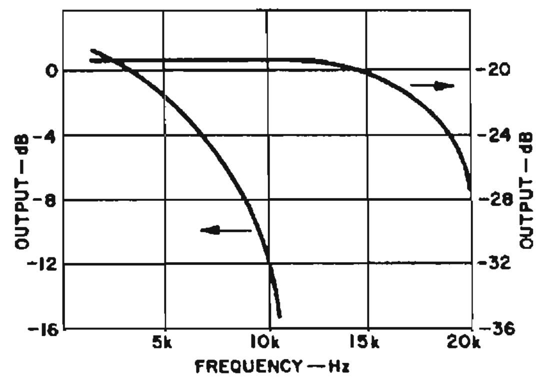
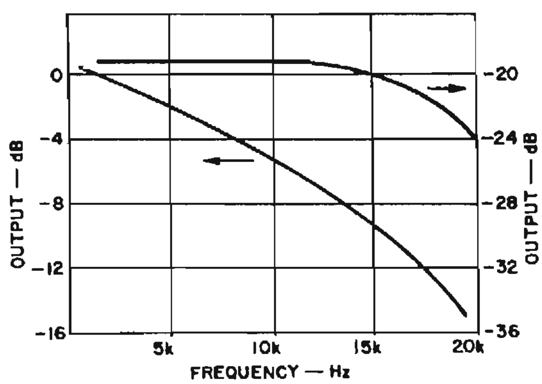


Fig. 4A—Frequency response of a gamma Fe_2O_3 tape at widely separated recording levels. Frequency response of a cassette tape is almost ideally flat when operated at a recording level 20 dB below the maximum operating level, which is generally defined as that level which produces 3 per cent third harmonic distortion in the reproduced signal at a frequency specified by a particular standard. High frequency output is greatly reduced when the tape is operated at the maximum output level, creating intermodulation distortion.

Fig. 4B—Less severe high frequency suppression occurs in a tape made with "high energy" magnetic material having a coercivity of 550 oersteds because of reduced self-demagnetization of the recorded signal.



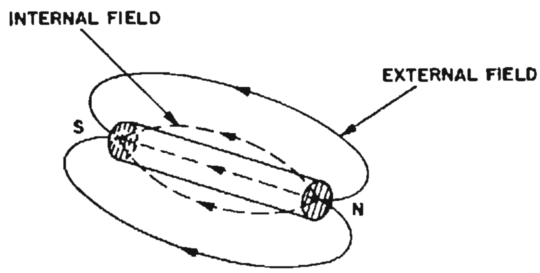
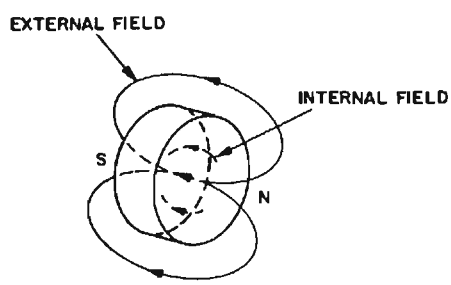


Fig. 5—The internal magnetic field of a long, thin permanent magnet exists largely in the air. This configuration will produce minimum self-demagnetization.

Fig. 6—Most of the internal magnetic field of a short, broad magnet passes through the body of the magnet, increasing self-demagnetization.



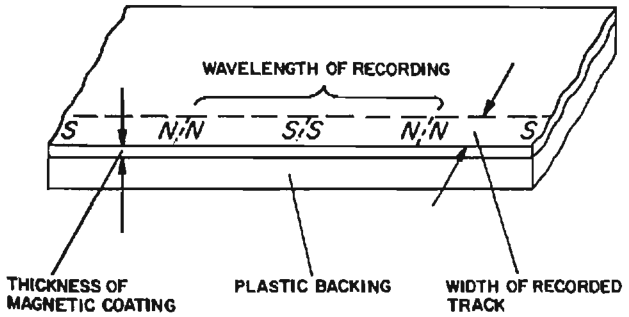


Fig. 7—The thickness of the magnetic coating and the wavelength of the recorded information establishes the proportionality factor of the "magnets" in a long wavelength recording. Even though the magnetic material is continuous, the magnets behave as though they were physically independent. The magnets grow shorter as the recorded frequency increases and will also become thinner at wavelengths shorter than one thousandth of an inch (about 2 kHz) because short wavelengths are recorded only near the surface of the tape.

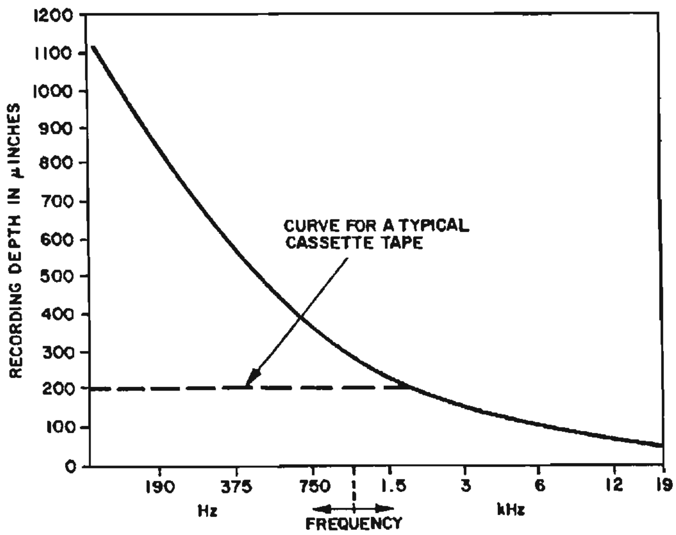


Fig. 8—In this recording depth vs. wavelength curve, the solid curve shows the recording depth that would occur in a thickly-coated magnetic tape moving at 1 ⅞ ips. Such a tape would produce far greater relative output at low frequencies than at high frequencies because more magnetic material would be utilized at the longer wavelengths. A typical cassette tape is coated only 200 microinches thick, so recording depth below 2 kHz is limited by the physical thickness of the coating.

magnet must also have a high value of coercivity, otherwise the internal field will spontaneously demagnetize the magnet, reducing remanence to a level which the coercivity will sustain.

Permanent magnets are used generally in magnetic "circuits" where magnetic fields are directed through soft iron poles. Examples of such uses are loudspeakers and permanent magnet motors. In these instances the soft iron circuits reduce the internal fields of the magnets, reducing susceptibility to demagnetization.

It is the self-demagnetizing effect that promotes the practice of placing a "keeper" across the pole ends of horseshoe shaped magnets to prevent demagnetization when not in use. The keeper is made of soft iron, and it short circuits the magnetic field to prevent the internal field from causing self-demagnetization.

Because the strength of the total field in and around a magnet is also dependent upon the level of magnetization, a magnetic material which is weakly magnetized will not demagnetize itself even though it may have a very poor proportionality factor. Self-demagnetization occurs primarily when an attempt is made to saturate the magnet or in other words to subject it to an intense magnetizing field which will leave it magnetized to the maximum level it will sustain. Summarizing, self-demagnetization in a saturated magnet is a function of 1) the dimensions of the magnet or proportionality factor, 2) the coercivity of the magnetic material, and 3) the remanent magnetization of the material.

Self-demagnetization induced by a poor proportionality factor is an important cause of loss of high frequency information in magnetic tape when high frequencies are recorded at short wavelengths (slow tape speed). If a square wave is recorded on magnetic tape, the recording signal will magnetize the magnetic layer in a series of end-to-end alternately polarized magnets, as illustrated in Fig. 7. The magnetic intensity of the magnets will be dependent upon the recording level and on the remanent magnetization of the tape. The length of the magnets is established by the frequency being recorded and the speed of the tape across the recording transducer. At low frequencies (long wavelengths), the proportionality factor of the recorded magnets, which is established by the thickness of the magnetic coating on the tape and the recorded wavelength, is favorable towards preventing self-demagnetization. Consequently, the tape will sustain a high level of magnetization at low frequencies. At very short wavelengths the proportionality factor is poor because the magnets are shorter. The recorded magnets will relax back to less than saturation at the instant they leave the magnetizing field of the recording transducer.

Recalling that self-demagnetization is greatest when an attempt is made to saturate a magnet, it can be seen that a low-level recording (low magnetization) will produce little demagnetization at either long or short wavelengths, and output of the tape is relatively uniform at all frequencies. But at high levels of magnetization, short wavelengths will spontaneously demagnetize as the recorded areas of tape move away from the recording transducer. Under this condition high frequencies will be suppressed relative to low frequencies.

The foregoing analysis of self-demagnetization versus recorded wavelength can be easily visualized in a physical sense. Although a typical magnetic tape may have a coating thickness of about 200 microinches, studies of the recording process (3) have shown that a thickness of only 50 microinches is utilized in recording a wavelength of 0.0001 inch (which is the wavelength generated by 19 kHz at 1 ⅞-ips tape speed) (see Fig. 8). Thus the cross-section dimension or thickness of the recorded magnets can be considered to be 50 microinches. The length of a recorded magnet is one-half the recorded wavelength, also about 50 microinches at 19 kHz frequency. The end-to-end magnets at this wavelength are only as long as they are thick, and this proportionality factor of one-to-one gives rise to substantial self-demagnetization.

The second quadrant of the M-H curve of a magnetic material, as shown in Fig. 9, can be analyzed to approximate the effect of self-demagnetization of tape for various wavelengths and values of coercivity and remanence. Remanent magnetization of a tape recorded at very long wavelengths is indicated by the point at which the second quadrant magnetization curve intersects the M axis. This point also corresponds to the saturated output level of the tape at low frequencies, and is determined by the intrinsic remanent magnetization and packing density of the material used in the magnetic coating.

The saturated output level of the tape at high frequency can be determined by using a "demagnetization loss" line drawn through the M-H curve. The angle of this line is dependent upon wavelength, and the angle will rotate counterclockwise as the wavelength becomes shorter (3). At the point where the loss line intersects the M-H curve, a second line is drawn perpendicular to the M axis. The point where this second line intersects the M axis represents the magnetization of the tape for a specified wavelength. Note that remanent magnetization (and output) at 20 kHz is substantially less than that of long wavelengths for a tape having a coercivity of 320 oersteds.

As mentioned earlier, the angle of a demagnetization loss line is a function primarily of wavelength, and will be the same for any tape tested. This angle was determined empirically by testing numerous tapes having widely separated magnetic properties and variable thicknesses. In every case, measured output of the tapes correlated closely to a demagnetization loss line of the same angle for a given wavelength (4).

Figure 10 shows second quadrant magnetization curves for eight tapes which have widely ranging ratios of remanent magnetization to coercivity. The angle of the demagnetization loss line drawn through the tape curves corresponds to 0.1 mil wavelength. If the intersections of this line with the magnetization curves ideally define magnetization of the tapes (and consequently output), then measured outputs of the tapes should all fall on the line of Fig. 11. Note that an excellent correlation is obtained.

This method of predicting demagnetization applies only to the near saturated case, and therefore applies to tape output where the recording level is near the maximum operating level of the tape.

It should be pointed out that, in addition to demagnetization of a recorded signal caused by poor proportionality factor of the recorded "magnets," another type of demagnetization occurs as the result of magnetic interactions of the tape and the recording transducer This second kind of demagnetization is termed "recording loss," and is included in the total demagnetization predicted by the demagnetization loss line. A definition of recording loss is beyond the scope of this paper.

Numerous conclusions can be drawn from the loss line method of predicting demagnetization losses. First, it can be observed from the curve of tape 2 in Fig. 10 that an increase in remanent magnetization without a corresponding increase in coercivity will not produce a proportional increase in 20-kHz output. The intersection of the 20-kHz demagnetization loss line occurs at a nearly vertical slope in the second quadrant curve, so increasing the height of the curve will produce only a small change in the height of the intersection point. In attempting to develop superior magnetic pigments for short wavelength audio recording, a mere increase in remanent magnetization is of limited value.

Second, remanent magnetization or output of a tape at 20 kHz can be substantially improved without any increase in the intrinsic remanent magnetization of the magnetic material. By increasing coercivity from 320 to 550 oersteds, as shown in Fig. 9, demagnetization at 20 kHz is reduced to half. This result clearly points to the direction of increasing coercivity to obtain improved short wavelength saturated output.

Third, the greatest improvement of short wavelength saturated output is obtained by simultaneously increasing both remanent magnetization and coercivity. High remanent magnetization is needed to produce magnetic fields of high intensity, and coercivity is needed to prevent spontaneous demagnetization that would otherwise occur.

Fourth, and finally, the shape of the magnetization curve can also have a significant effect upon demagnetization. If one imagines a second quadrant curve which has a sharp corner as opposed to a slanted curve, the point of intersection on the demagnetization loss line would be much higher on the M axis (5). Since the shape of the curve is a function of the distribution of coercivities of the magnetic particles, narrow distribution is a quality to be sought. Distribution is made narrow principally by maintaining uniformity in size and shape of the magnetic particles, and by obtaining chemical purity of the material.

The effect of reducing demagnetization by narrowing distribution can also be visualized as an elimination of those low coercivity particles of pigment which are most susceptible to demag-

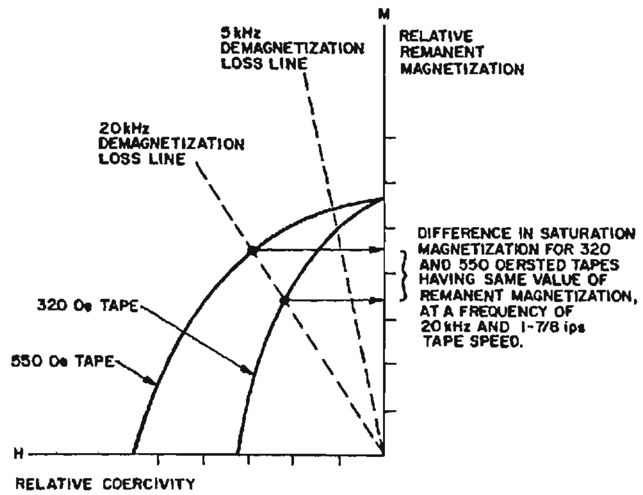
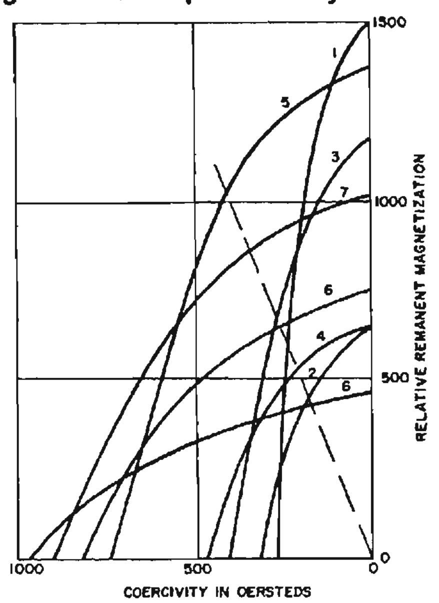


Fig. 9—This figure shows the second quadrants of the M-H curves of two tapes. Saturation magnetization at a given frequency, after self-demagnetization, is predicted by the points of intersection of the curves with the "demagnetization loss" line.

Fig. 10—Second quadrant analysis of eight experimental tapes shows the demagnetization loss line for a recording wavelength of 0.1 mil represented by the dotted line. Note that the predicted output of tape 1 is very much lower than might be anticipated by remanent magnetization alone. This tape provides an extreme example of self-demagnetization.



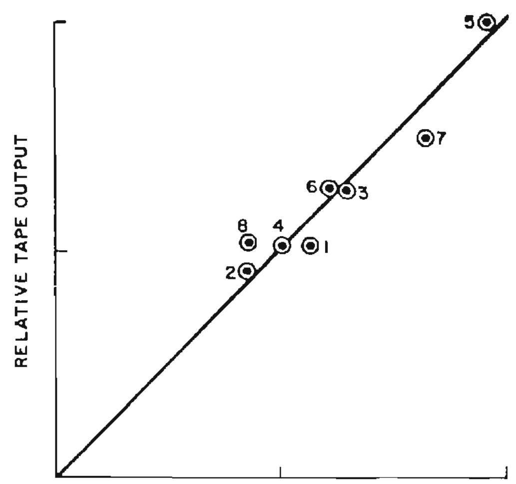


Fig. 11—Measured output of eight experimental tapes.The second quadrant analysis of self-demagnetization would predict that the output of the eight tapes of Fig. 10 would fall on the line.

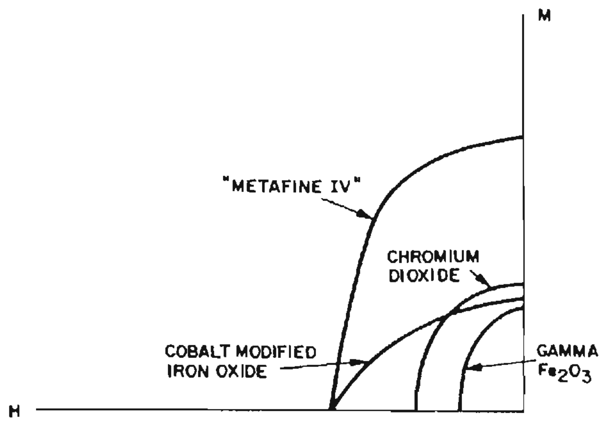


Fig. 12—The relative energies of four magnetic recording pigments. The area enclosed by the second quadrant of the hysteresis curve represents the "energy" of a magnetic material. Coercivity of cobalt-modified iron oxide can be adjusted in the manufacturing process to cover a very broad range, from that of Fe_2O_3 to well beyond the recording capabilities of a magnetic transducer. Metallic magnetic materials are probably the materials of the future since they combine an exceptionally high remanent magnetization with high coercivity.

netization. As remanent magnetization of a pigment is increased, the low coercivity particles will be the first to demagnetize. If these particles represent a large portion of the total material, then demagnetization will be appreciable, even though the value of coercivity represented by the center of the bell-shaped curve is high.

In certain types of short wavelength recording, such as FM video and high density digital recording, a very narrow distribution is highly desired. In fact, a magnetization curve similar to that of a single particle would be ideal. But audio recording is an analog recording process where the instantaneous magnetization of the magnetic material in the tape must be exactly proportional to the instantaneous value of the audio signal. If distribution is too narrow and if the magnetization curve is too steep, sensitivity of the tape becomes too critical to allow proper setting of recording levels. Fortunately this problem is not encountered in particulate magnetic materials because small variations in physical shape and dimensions, as well as variations in chemical composition of particles inherent in the manufacturing process, combine to produce a slope in the magnetization curve which is suitable for analog recording.

The foregoing analysis provides a quantitative means of translating several bulk magnetic properties of materials to performance of a tape, but more importantly, it clearly shows the direction that must be taken in the future development of magnetic materials. All areas of magnetic recording, including television, digital data, and instrumentation recording, are trending toward shorter wavelengths. Advanced materials having higher coercivity and remanence must be developed to meet future needs in all areas.

Material with increased coercivity is the most recent development to find a way into the marketplace. Chromium dioxide and cobalt-modified iron oxide became available in audio cassette tapes several years ago. The coercivity of chromium dioxide is nominally 550 oersteds, as compared to 340 oersteds of gamma Fe_2O_3. This value of coercivity is determined by the chemical nature of the material and by the size and acicularity of the magnetic particles. Approximately one-third of the coercivity is attributable to chemical composition and two-thirds to size and shape. A coercivity of 550 oersteds is a fortuitous characteristic of CrO_2 particles, since this value is ideally suited to compensate for demagnetization losses which occur at 1 7/8 ips tape speed. Saturated remanent magnetization of CrO_2 is slightly greater than that of gamma Fe_2O_3.

Unlike CrO_2, cobalt-modified iron oxide obtains a high coercivity by combining a small amount of cobalt with the iron oxide crystal. Coercivity can be controlled over a very wide range, up to 2,000 oersteds, by adjusting the amount of cobalt introduced into the iron oxide. This flexibility allows cobalt-modified materials to be adjusted precisely to the application.

Future Limits

If magnetic materials could be tailored to any magnetic properties desired, what are the limits to which remanent magnetization and coercivity could be taken? Certainly many different limits are imposed. Not the least significant is the magnitude of the magnetic field which can be generated by recording transducers. Materials which form the pole piece of transducers become magnetically saturated when attempts are made to record on tape having a coercivity of, say, 1500 oersteds. Even though a tape with this coercivity is highly resistant to self-damagnetization and will provide excellent short wavelength output, no practical method exists for placing a distortionless recording on the tape. Until significant improvements in head materials are made, a coercivity of 1200 oersteds is a practical limit for tape in most applications.

With this limit for coercivity established, it is a simple matter to show by analysis of the second quadrant curve what level of remanent magnetization can be employed before self-demagnetization at short wavelengths occurs. These values of coercivity and remanent magnetization become the goals of future pigments, established by the characteristics of magnetic recording transducers and firmly fixed until advances in transducer materials are forthcoming.

The limits of coercivity and remanence established by recording transducers are reached in a new generation of magnetic materials which utilize particles existing in a metallic state. Materials of this type are being developed in several laboratories around the world. One such material, identified by the brand name "Metafine IV," is supplied by the 3M Company. Such particles have an intrinsic remanent magnetization close to the theoretical limit for single domain particles. A comparison of the second quadrant area of the fine iron material and gamma Fe_2O_3 is shown in Fig. 12. As the figure suggests, the output of a tape made with this material is far greater than that of gamma iron oxide tape.

Magnetic materials having very high remanence are difficult to handle in the tape manufacturing process because of a tendency of the particles to cling together in clumps. Magnetic attraction and repulsion between particles in the fluid dispersion causes them to find stable physical relationships wherein opposite poles are locked together, end to end and side by side. This produces chains of particles which appear under a microscope much like the chains of iron filings used to demonstrate magnetic fields. Chained magnetic particles in tape will not behave as individual particles since the switching of one particle will have an influence on neighboring particles. The result of this infectious switching could be poor signal output from the tape.

Clumping or chaining is controlled by dispersing magnetic materials in fluid polymers and binders in which the chemical forces holding particles apart are greater than the magnetic forces pulling them together. The chemistry needed to achieve this equilibrium becomes more difficult as the remanent magnetization of particles is increased. Thus, chemical dispersion technology must be developed in concert with developments in magnetic materials, and advanced magnetic tapes are not made simply by substituting magnetic materials.

The development of advanced magnetic tapes involves much more than simply increasing the energy of magnetic materials. The magnetic component is only one ingredient in a complex chemical system, and the entire system must be redesigned whenever this component is substituted or modified.

Summary

This paper has explained the effects of demagnetization on short wavelength saturated recording, and has presented an empirically derived method of equating magnetic properties of particles to short wavelength performance of tapes. Although the examples used relate to cassette audio recording systems, the same principles apply to other short wavelength recording systems, and particularly to the whole new generation of video recorders now available to consumers. The trend toward shorter wavelengths, which has been seen in the past, will go on into the future, and refinements of all aspects of recording systems will continue. Development of magnetic materials having increased remanent magnetization, higher coercivity, and narrower distribution will be a large part of this effort.

The author wishes to acknowledge the work of Dr. T. J. Szczech, of the Data Recording Products Division Laboratory of the 3M Company, whose studies of the magnetic recording process are principally responsible for the information contained herein.

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2) A. H. Morrish and L.A.K. Watt, "Effect of the Interaction Between Magnetic Particles on the Critical Single-Domain Size," *Physical Review*, Vol. 105, No. 5, March, 1957.

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5) D. B. Richards and T. J. Szczech, "The Relationship of Switching Distribution to Tape Digital Output," Journal of Applied Physics, 1978.

Studer ReVox Model B77 Open-Reel Tape Recorder

MANUFACTURER'S SPECIFICATIONS

Frequency Response: 30 Hz to 16 kHz @ 3 ¾ ips, 30 Hz to 20 kHz @ 7½ ips.
Harmonic Distortion: 1 per cent @ 3 ¾ ips, and 0.6 per cent @ 7½ ips.
S/N Ratio: 59 dBA @ 3¾ ips, 62 dBA @ 7½ ips.
Crosstalk: -45 dB.
Erasure: 75 dB.
Input Sensitivity: Mike, Lo Z, 0.15 mV, Hi Z, 2.8 mV, Line 40 mV.
Overload Margin: 40 dB.
Output Level: Line, 1.55 V; Headphone, 5.6 V (open circuit) for 200- to 600-ohm phones.
Flutter: 0.1 per cent wtd. peak @ 3¾ ips, 0.08 per cent wtd. peak @ 7½ ips.
Speed Tolerance: 0.2 per cent.
Wind Times: 135 seconds for 3600 feet.
Dimensions: 17.8 in. (45.2 cm) W x 16.3 in. (41.4 cm) H x 8.14 in. (20.7 cm) D.
Weight: 37.4 lbs. (17 kg).
Price: $1295.00.

The Studer ReVox B77 open-reel recorder retains the advantages of the A77, eliminates limitations of the earlier models, and adds features of its own. (The A77 remains in the product line, at least for the time being.) The front-panel styling of this high-class audiophile recorder is similar to that of the professional A700 model. Interlocked push-button switches select speed, 3¾ or 7½ ips. Below are the large-handle toggle switches for power and monitor selection and paralleled jacks for headphones. This can be a handy feature at times, providing a front-panel output in addition to one for a set of phones. The level at these jacks is controlled by a dual-section, slip-clutch pot, which does not affect the regular line outputs on the back panel. The output selector switch, however, does affect all outputs including metering, and can be set for stereo, reverse stereo, left channel to both outputs, right channel to both outputs, and mono which combines the two channels and feeds the mix to all jacks. The choices provided with this switch facilitate a number of tasks without the need to change connections and use adapters. The smooth-acting input level pots have knobs of a good size, an improvement over those on the A77. Alongside are the record presets with a large status light above each switch, illuminated when the recording process starts. The large toggle handles are convenient, but were judged to be subject to inadvertent switching. The input selector switches have posi-

tions for both low- and high-impedance single-ended microphones, *Radio* (DIN), AUX, and for feeding in the output of the other channel for sound-on-sound recording. Mike inputs are standard phone jacks, and it might be noted that mike/line or other mixing is possible, if recording is done on just one channel.

The two level meters have even illumination and excellent legibility. The peak indicators are located in the meter faces which gives a continuous, no-effort display without diverting attention from the VU meters. The light-touch tape motion switches are logic controlled, and any order of commands will be followed, including *Pause* in a wind mode. Unfortunately, it does not latch in, and the button must be held for the duration of the pause. The logic also permits going into record from fast wind, as long as the record presets are in and *Play* is held until recording starts.

Behind a flip-down cover are the reel-size switch and the *Cue/Edit* slide switch, which puts the tape in contact with the heads from *Stop*. It is released by moving the pinch-roller arm or by going into play. When in *Cue/Edit*, fast-wind modes become momentary-contact controlled for any needed tape shuttling. Tape threading is quite straight-line, and there is notably more clearance than on the A77. There is easy access for cleaning and demagnetization and, with removal of the snap-on head assembly cover, alignment adjustments become immediately available. To the right is another aid to editing, a small splicing block with a built-in, shear-type cutter. The counter and its reset have a slide-sync function. The carrying handle on the top of the high-impact polypropylene cabinet allows one-hand transporting for that live recording, very possible with the admirably low weight of 37.4 lbs. After removal of the four retaining screws, the recorder assembly on its rigid, box-girder frame was pulled out for inspection.

There are 10 PCBs with high-quality components, and all had excellent soldering. The overlay on the cover for the plug-ins was marked to show the location of adjustments behind the panel, but the parts on the cards themselves were not identified. Plug-type connections were used for most purposes, with a minimum of interconnection soldering. Many items conveyed the impression of ruggedness for long-term reliability, but the turntable and capstan motors along with the brake and play solenoids were most notable. The capstan frequency pick-off was rigidly mounted adjacent to the slots in the rotor shell, fundamental elements in the ReVox drive system which are basically immune to variations in line voltage or frequency.

Performance

The playback response of the B77 was within 2 dB at both speeds with the exception of 50 Hz at 7½ ips. The pink-noise/RTA checks showed that the record/playback responses were very good for Ampex 456, Maxell UD, Memorex Quantum, and TKD Audua, as well as with Scotch 206/207 for which the unit had been adjusted. Meter indications for the reference levels on the test tapes were very close, with the zero indication within 0.5

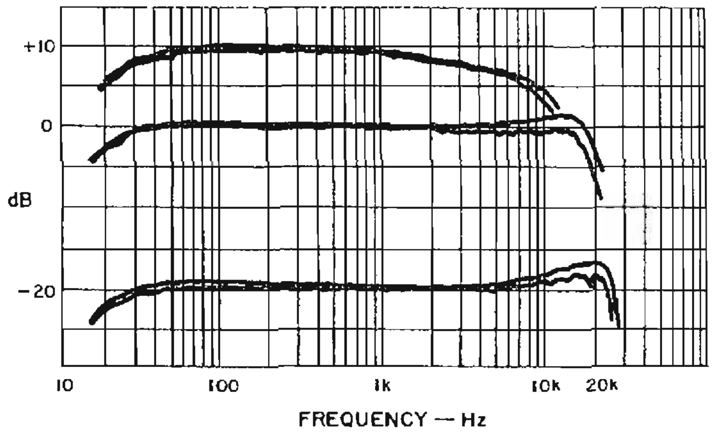


Fig. 1 — Record-playback response using Scotch 206 at 7½ ips. (Record 0 reference level is that for 200 nWb/m fluxivity at 1 kHz.)

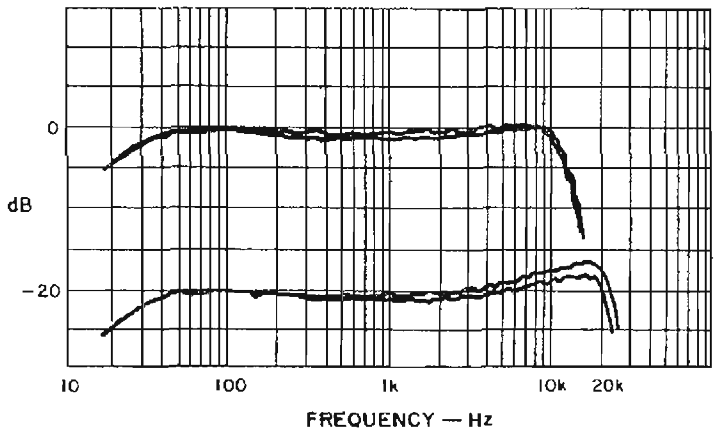


Fig. 2 — Record-playback response using Memorex Quantum at 3 ¾ ips. (Record 0 reference as above.)

are above the head assembly. The reel turntable shafts have spring-loaded retainers which work directly on small reels and will hold on the adapters required for large-hub 10½-inch reels.

The line-in/line-out phono jacks and their level-set pots on the back are clustered together with sockets for DIN-type systems and for optional remote controls for tape motion and capstan speed. The tape-motion control includes record and timer-start functions. The speed control, in conjunction with tape-speed selection, can provide speeds all the way from 2.5 to 11 ips. Just above are the voltage selector, a fuseholder, and the socket for the detachable power cord. With the addition of another plug-in module (not in the unit evaluated), the B77 can dB of the 257 nWb/m ReVox specification. The reference record level used was that for 200 nWb/m fluxivity at 1000 Hz, which was about −1.5 dB on the meters. (Note that flux levels at other frequencies are different even when the record/playback response is perfectly flat.) The response with Scotch 206 at reference level and 7½ ips (Fig. 1) was within 3 dB from 17 Hz to 18 kHz for the left channel, to 20 kHz for the right. The responses 20 dB lower were from 16 Hz to 23.5 kHz (left) and 25.5 kHz (right). With the record level +10 dB, the-3 dB points were at 21 Hz and about 7 kHz. These results for Scotch 206 are not all that unusual at the high frequency end, but the smoothness over the entire frequency range is excellent indeed.

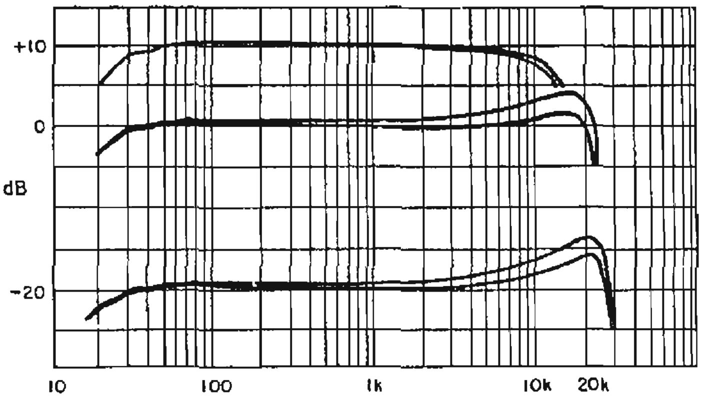


Fig. 3 — Record-playback response using Maxell UD at 7½ ips. (Record 0 reference as above.)

Studer ReVox claims for superior head design are graphically illustrated with the extended low-frequency response with just the slightest indication of head contour effects, superb performance. Similar plots were made using Memorex Quantum at 3 ¾ ips (Fig. 2). At reference level, response was from 20 Hz to 13 kHz. At the -20-dB record level, the upper limit moved out to 23.5 kHz (average), but there was a peak in the response around 17 kHz. With Maxell UD at 7½ ips, the responses extended further than with Scotch 206, but at the expense of excessive peaking at the highest frequencies (Fig.3). This deviation could be reduced greatly with a slight increase in bias. The small discrepancies between the two channels could be eliminated in similar fashion. Studer ReVox, however, takes the position that bias switches or pots should not be readily accessible because of possible abuse by unqualified users. That is a valid point, but I suspect that the more technically oriented owners will drill access holes in the bottom of the cabinet, just as many have done with the A77s.

The playback of a recorded 10-kHz tone had a 30-degree phase discrepancy between channels, evidence of very good alignment between the heads. Phase jitter at 7½ ips was 15 degrees, also very good. The bias residual in the output was down into tape noise, excellent design and adjustment. The playback of a recorded 1-kHz square wave had relatively small tilt (Fig. 4), another demonstration of the superb low-frequency response, and some short-duration ringing at the EQ peaking frequency. Some momentary, partial dropouts appeared in the stored waveform.

Plots were made of HDL_3 vs. record level for Scotch 206 at both speeds and Maxell UD at just 7½ ips (Fig. 5). On a dB-dB basis, the functions were the most linear ever measured, with just a slight upturn near the 3 per cent points. At the low end, HDL_3 continued to reduce to the noise limits of the test equipment, with little evidence of distortion in the electronics. From 30 Hz to 7 kHz, HDL_3 was very low, particularly at 10 dB below reference level. Throughout this entire range, other harmonics were consistently very low, excellent magnetic design. Signal-to-noise ratios with IEC "A" weighting at 7½ ips were 58.7 dBA for Scotch 206 and 59.3 dBA for Memorex Quantum at reference level, and 67.0 and 70.0 dBA for the same tapes for HDL_3 =3 per cent. At 3¾ ips using the Memorex tape, the figures were 57.1 dBA at reference level and 65.2 dBA at the 3 per cent point. These are all excellent figures, and don't forget — without Dolby NR. With CCIR weighting, the relative results were the

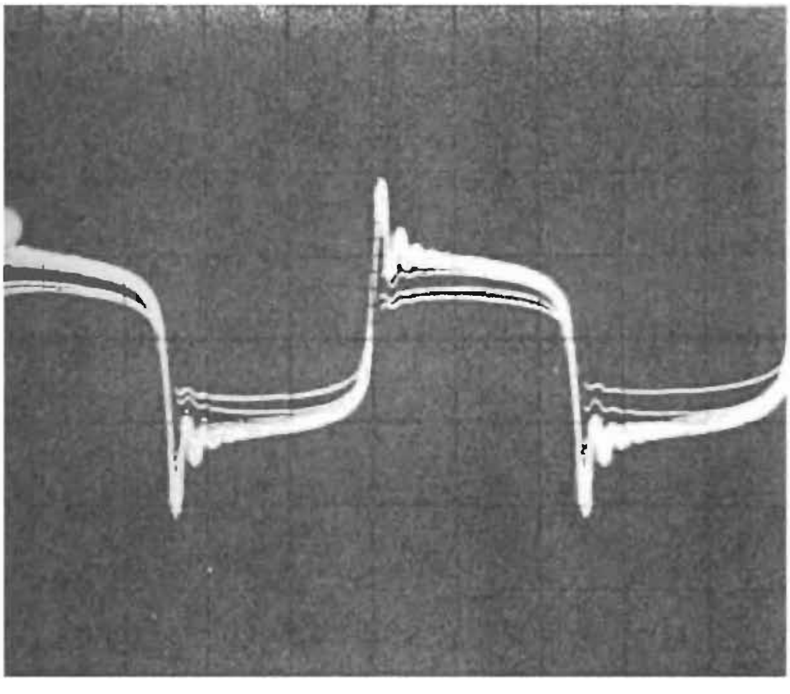
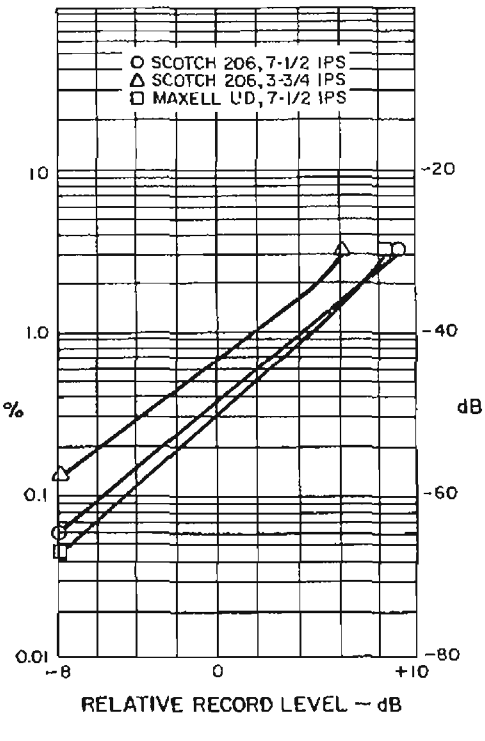
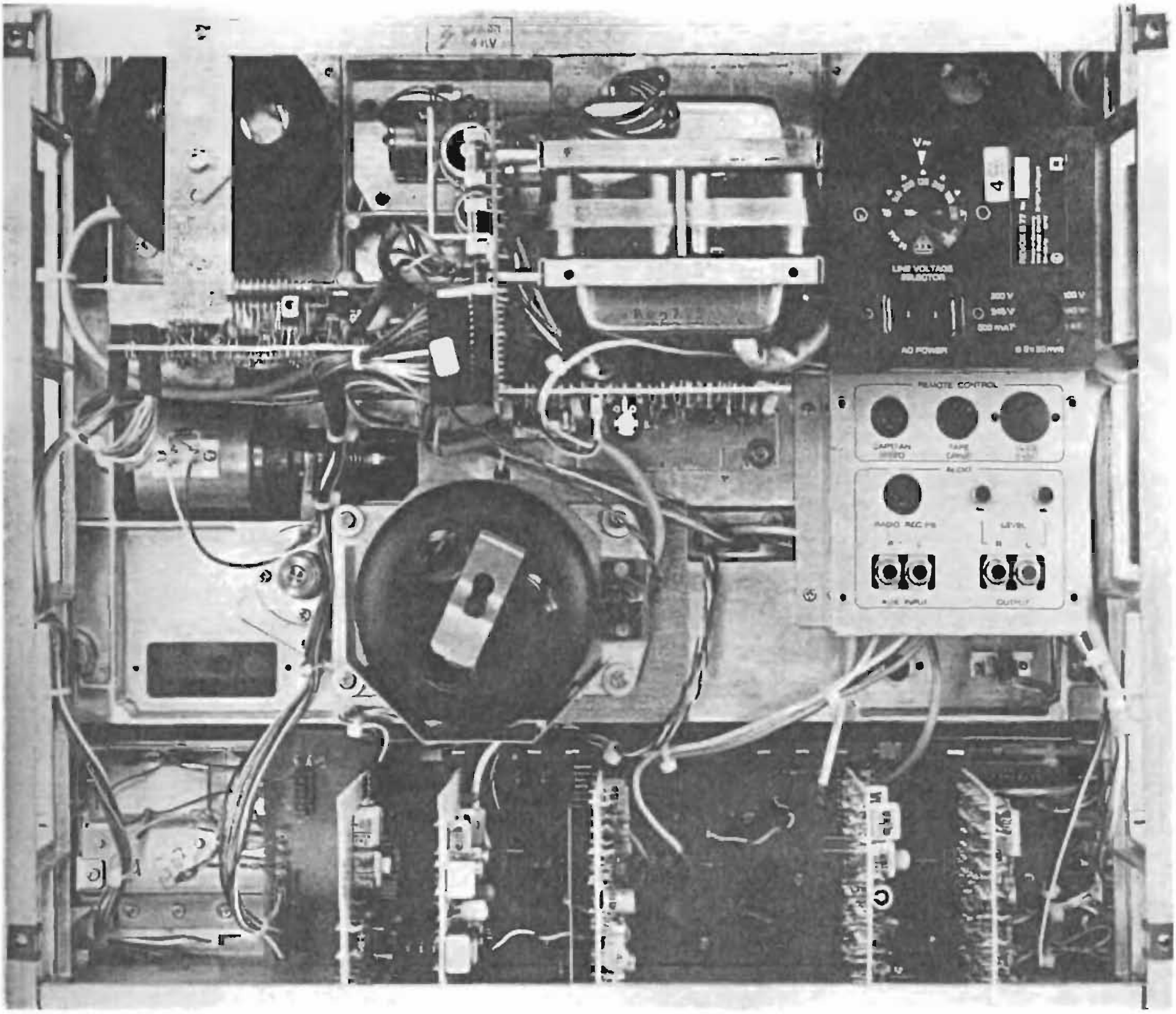


Fig. 4 — Playback of recorded 1-kHz square wave. (Scale is 0.2 mS/div.)

Fig. 5 — Third harmonic distortion vs. record level at 1 kHz. (Record 0 reference level as in Figs. 1-3.)





same, but all of the figures were 8 to 9½ dB less (smaller number). Separation from one channel to the other was a good 51 dB, and crosstalk to the adjacent track of opposite play direction was a good 69 dB. The erasure of a 1 kHz was excellent by any standard, down at least 90 dB.

The input sensitivity for low-impedance mike was all the way down to 0.046 mV, and even with the high-impedance setting only 1.4 mV was required. The line sensitivity was much higher than that normally encountered, just 17 mV for a zero indication. All of the measured sensitivities were six dB or more better than the specifications. Input overload was 48 dB or more above the sensitivity figures listed. Output clipping at the line-out jacks was reached at a level equivalent to a +17 meter indication. The input level pots tracked within a dB for the same pot positions. In accordance with ReVox specifications, the output levels were measured with the meters at +6 The line outputs averaged 1.48 V, just a little less than the spec. The headphone output was 5.5 V open circuit, 2.75 V with a matching 220-ohm load. Although the manufacturer recommends headphones with impedances from 200 to 600 ohms, the circuit will deliver a very adequate 190 mV to 8-ohm loads. (Many so-called 8-ohm headphones are actually 100 ohms or so.) The line-output level-set pots were left at maximum for all tests, although they provide up to 26-dB attenuation for system matching. The sections of the monitor pot tracked within ½ dB over most of its rotation. The total spread in output levels for all positions of the output selector switch was ½ dB, indicative of carefully set internal adjustment pots. The dynamic response of the two meters was in accordance with VU standards, but the extended frequency response to over 200 kHz is subject to question. The peak indicator threshold was very close to +6 VU with a CW input, and was still firing with a toneburst with duration reduced to 15 mS. The meter scaling was very close with maximum errors of 0.2 dB.

All indications from test tapes and a strobe were that the recorder tape speeds were close to exact. There were no indications whatsoever of speed changes from 100 to 130 line voltage. At 3 ¾ ips, the typical weighted peak flutter was 0.06 per cent. At 7½ ips, the flutter was 0.028 per cent on the average, with a maximum value of 0.04 per cent (Fig. 6). These results are well within the specifications and are direct evidence of superb tape motion. There were substantially no shifts in record-to-playback speed, and a slight shift plotted in one case was from drift in the metering. The wind times for a 3600-foot 10½-inch reel was 132 seconds. The time required to go from fast wind to play was less than two seconds at all times, smooth *and* fast.

Listening & Use Tests

Tape loading was easy and direct with little to snag the tape on, and there was good access for regular cleaning and demagnetization. Snapping off the head assembly cover did improve access to the capstan and scrape flutter filter. There was a good feel to all of the pots and switches during all phases of the testing. The logic-controlled, tape-motion switching performed without error, including the usual "torture" test. As stated before, a latch-in pause control did seem a definite lack at times. There was good meter action, and the placement of the overload LEDs seemed ideal in actual use.

There were some troubles with tape motion, but the major

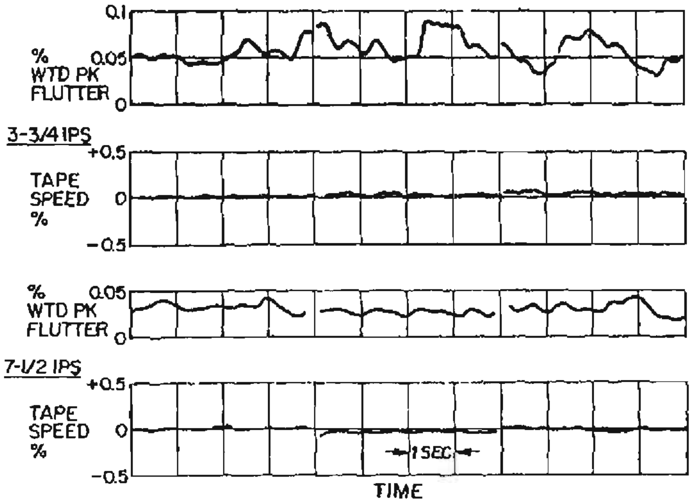


Fig. 6 — Peak-weighted fluttter and tape speed variations at 3¾ and 7½ ips.

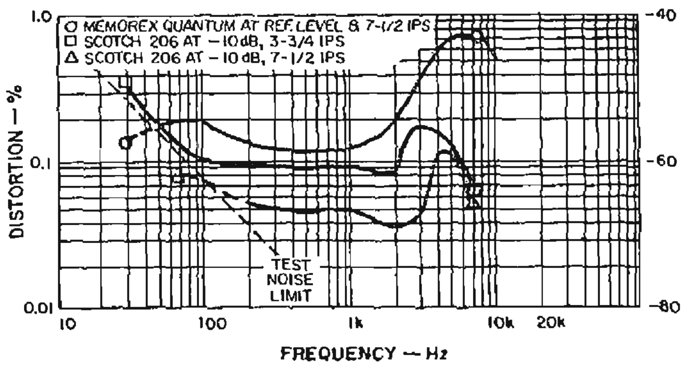
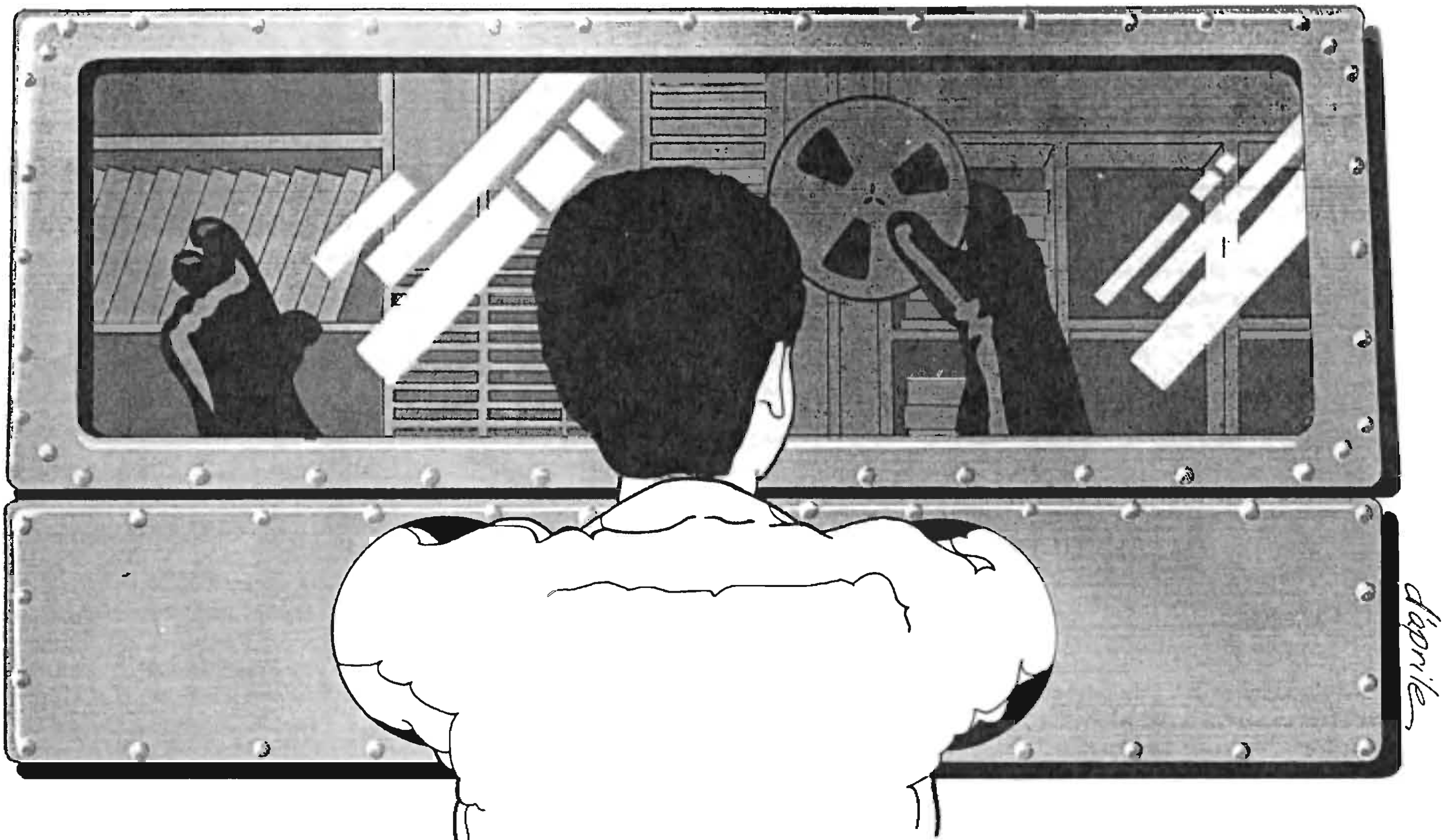


Fig. 7 — Third harmonic distortion vs. frequency.

difficulties disappeared with use. When first started in play, there was a scraping noise from the right reel turntable. Removing the front panel aided in making the tentative conclusion that some brake lining glue was rubbing against the brake band. In any event, the sound was gone by the time the testing was complete. A minor ailment, but annoying, was slippage in the counter. The errors were not great, but they kept creating confusion on where the exact start/stop points were.

The 44-page trilingual instruction book is logical and thorough with good descriptions complemented by clear illustrations. No schematic is included, but there is an excellent signal-flow block, and circuits are given of the remote control units, so you could even make your own. Clicks from pause and stop were 5 dB out of tape noise; the record click itself was covered by tape noise. Different sources were recorded, and there was nothing to fault in the playback. The B77 really shone when it was used to record the revised *Mass* by Leonard Kastle. (The earlier version had been performed on national PBS TV on Christmas 1977). The whole combination of this easy-to-carry high-performance unit with its excellent metering and of the inherent qualities in the music and its performance generated a tape which caused many a comment on its excellence. The $1,295.00 price tag will be more than tough for some, but in a number of the important areas of recorder performance, the B77 is superior to units costing a fair amount more. Studer ReVox may very well have another winner. *Howard A. Roberson*



MAGNETIC SHIELDING

Herman Burstein

A number of readers have indicated their concern that valued magnetic records — audio tape, video tape, magnetic stripe calculator programs, etc. — may be partly or totally erased by magnetic fields. Such fields are almost always about us, particularly 60-Hz fields and harmonics thereof produced by a.c. power lines, a.c. motors, transformers, etc. In addition there are also d.c. fields produced by toy magnets, speaker magnets, powerful lifting magnets, d.c. motors, and the like. Such a.c. and d.c. fields not only present possible hazards to magnetic records but may also interfere with proper operation of a variety of devices such as tape heads, phono pickups, meters, cathode-ray tubes, photomultiplier tubes, reed relays, printed circuit boards, signal leads, etc.

Hence the audio hobbyist may have reason to be interested in magnetic shielding. And this article seeks to inform him about shielding materials and their proper use.

However, this article should not be misinterpreted as a warning to the audiophile that everything in sight had better be shielded to prevent magnetic fields from erasing magnetic records, inducing hum, causing meters to give misleading readings, and the like. Rather, matters are the other way around. Usually there is little or no cause for alarm and the need for shielding is not as great as some may fear. On the other hand — as a number of audiophiles who have run into hum or other magnetically induced problems can attest — the danger from magnetic fields does exist.

Some Preliminary Semantics

To facilitate the following discussion, let us define three terms we shall be using: (1) An "interference field" is a magnetic field that can erase a magnetic record or prevent correct operation of an electronic component. (2) A "generating device" (for example, a transformer) is one which produces an interference field. (3) A "sensitive device" (for example a tape or tape head) is one which is adversely affected by an interference field.

Threat to Magnetic Records

Because there seems to be an appreciable amount of concern about the possibility of magnetic tapes being accidentally erased and a prevalent notion that tapes must therefore be kept well away from speakers, motors, transformers, etc., let's first discuss this problem.

My own experience suggests that as little as three or four inches distance from a generating device minimizes the risk of tape erasure and that six inches provides a very healthy safety factor. Using a very powerful bulk eraser, I have brought it within about three inches from a reel of recorded audio tape without noticeable effect. Only when it nearly made contact with the reel did the bulk eraser produce substantial erasure. The very rapid drop in erasure with a slight increase in separation may be appreciated by noting that the intensity of a magnetic field tends to vary inversely with the *cube* of the distance from the source of the field. For example, the intensity of the field three inches from the source is only about 1/1,000th as great as the intensity one inch from the source.

For a more authoritative view on the threat to tapes and other magnetic records, let's turn to fairly recent studies by the National Bureau of Standards' Institute on Computer Sciences and Technology, NBS for short. (This account of NBS tests is based on an article by Sidney B. Geller, "Erasing Myths About Magnetic Media," which appeared in *Datamation*, March, 1976. Quotations are from that article — H. Burstein.)

The NBS studies dealt with recorded computer magnetic tapes, digital cassettes, and magnetic stripe plastic credit cards. However, the results would also apply to other magnetic media, such as audio tape, because they all have in common the use of ferric oxides or other oxides with similar properties. The studies include the effects not only of magnetic fields, but also of airport metal detectors, nuclear radiation, lasers, radar, and microwaves. Geller summarizes: "The results indicate that magnetic media are really quite safe from most threats, and that they can be easily protected from those things to which they are susceptible."

To investigate the effect of magnetic fields, "a large horseshoe magnet was placed in direct contact with the outer flange of a tape reel and at various distances from the reel. This magnet produced a field . . . strong enough to lift 40 lbs." The conclusion reached was that it is impossible for even a magnet of such strength to cause any loss on tapes stored six inches from the magnet. (On the other hand, much weaker magnets could produce moderate to serious loss if placed very close to magnetic media. One part of the NBS tests revealed that losses could be caused by close proximity to weak generating devices such as telephone receivers, transistor radio loudspeakers, and voltmeters).

In another series of NBS tests, a very large electromagnet of the type used in scrap metal yards "was placed over recorded tapes at various heights. At the closest test distance of 1.3 feet . . . no data loss was incurred." (From this we may gather that only a magnetic field capable of lifting tons of metal could require a separation distance as great as approximately one foot. — H. Burstein.)

The NBS study "found that when devices such as transformers and motors are encased in shielding materials, they cause no signal erasure even when placed in almost direct contact with the media. However, it was also found that some unshielded power supply transformers could cause erasure upon contact with the media when they were carrying high current loads. In all cases, electrical equipment which was enclosed in cabinets that provided at least two to three inches of spacing from the internal electrical components caused no erasure of recorded signals."

Similarly, the NBS studies showed: Airport metal detectors produced "no observed instance of erasure." External leakage fields from microwave ovens "would be unable to cause any data erasure." It is unlikely that radar systems with power as high as 200,000 watts or more "could cause signal erasure unless the media were . . . almost in contact with the antenna." "Extremely high (lethal) X-ray dosages" resulted in no reported data losses. When voltages in excess of 15,000 volts were applied across magnetic stripes so as to produce arcs which struck directly on these stripes, "no signal or data losses were reported." No data loss occurred when magnetic media were exposed to strong nuclear radiation for 1½ hours. "Magnetic media carried within the passenger compartment of numerous automobiles experienced no data losses." Media placed in close proximity to the high voltage circuits or degaussing coils of color TV sets incurred no signal-level losses. Intense light sources ranging from infrared to ultraviolet also produced no data loss. The radiation from a laser beam had no effect if spread over a sufficiently large area of the medium to avoid alteration or destruction of the medium itself.

Altogether, for most generating devices the audiophile is apt to encounter, it appears that a few inches of separation — at most six inches — will eliminate the danger of erasing a magnetic record.

However, the individual who encounters exceptional circumstances or wishes to play it exceptionally safe may want to provide shielding for valued magnetic records. Further, on occasion the hobbyist or tinkerer may find it necessary to protect a sensitive device from an interference field. For example, if a tape playback head or a phone cartridge is picking up radiated hum from a motor or transformer, he may want to cure the problem by use of magnetic shielding. Therefore in the rest of this article we will consider how to achieve such shielding. In order to be specific, the following discussion deals with shielding materials available from Perfection Mica Company, a leading maker of shielding materials effective in the range from d.c. to about 50,000 Hz. (Interested readers may obtain catalogs and price lists from Mr. Charles Neilsen, Magnetic Shield Division, Perfection Mica Co., 740 No. Thomas Dr., Bensenville, Ill. 60106. We are informed that the minimum order is $25.00.)

"Magnetic shielding may be the answer to radiated hum problems"

Techniques of Magnetic Shielding

The basic ways of preventing interference by a magnetic field are: (1) Place shielding material around the generating device, (2) place shielding material around the sensitive device, or (3) do both.

If only one sensitive device is affected, and the interference field is not very strong, a shield around the sensitive device may be sufficient. If a number of sensitive devices are affected by a single generating device, a shield only around the latter may be sufficient. If the interference field is strong, it may be necessary to put shields around both the generating and sensitive devices.

To effectively attenuate an interference field, several layers of shielding material may be required. These layers should be separated by tape or other insulating material to maximize attenuation. However, separation beyond about a quarter inch appears purposeless.

It is not enough to merely interpose a flat piece of shielding material between the sensitive and generating devices. It is best to wrap the material around either device (or both) so as to enclose the component as fully as possible, and to then join the ends of the material. For maximum shielding, one should butt the ends and form a welded seam. However, this technique is probably outside the capability of most audiophiles. The next best procedure is to overlap the ends of the material and join them with solder or tape. However, one should not get solder (or adhesive) between the overlapping ends, thereby preventing them from making good contact as this would seriously reduce the effectiveness of the shield. If a multi-layer shield is employed and if the joints are lapped, these joints should be staggered instead of being placed one atop the other.

The nature of the shielded device or its location may prevent the shielding material from being wrapped completely around it, so that a flat or U-shaped shield must be used instead. Then the ends of the shield should extend well beyond the device. The magnetic flux absorbed by the shield is re-radiated at the edges of the shield, and it is important that re-radiation take place well beyond the device.

Shielding material is available in both sheet form (relatively thick, rigid) and foil (relatively thin, pliable). One should never wrap the foil around an object in spiral fashion (for example, in the way one often wraps tape around signal leads). Perfection Mica warns that a spiral wrap "will produce a magnetic pole on the inside edge of the foil which will re-radiate the interference field."

Foil shielding material is optionally available with two-way, pressure-sensitive adhesive backing. This offers a simple way of attaching the material to the device to be shielded.

Decisions as to choice of sheet or foil, its thickness, and the number of layers depend in large part on the degree of attenuation sought. (Sheet material may also be chosen because of its rigidity, so as to preserve the shape of the shield, enable it to be secured in place, etc.) While formulas are available to assist in selection of the proper shielding material, gauge, and/or number of layers, these formulas yield only approximate results. A trial and error procedure is often necessary. Here an a.c. magnetic field evaluator probe, such as the one made by Perfection Mica, can be helpful to the engineer, technician, or hobbyist. In conjunction with an a.c. voltmeter or oscilloscope, the probe can measure the intensity of a magnetic field before and after shielding, thus indicating the degree of attenuation actually achieved. "With this necessary information," states Perfection Mica, "magnetic shielding material type and gauge can be intelligently selected and shield design optimized."

How Shielding Material Operates

A magnetic shield effective in the audio range performs its functions in two ways, depending on frequency of the interference field: (1) For a stationary field (zero frequency) produced by a permanent magnet, d.c. current, or the earth, the shield offers a path with low reluctance — that is, with low magnetic resistance. Thus the shield acts as a shunt, diverting the field from the sensitive device. At low audio frequencies, the shield behaves in much the same manner. At higher frequencies, however, the interference field produces eddy currents in the shield material. These currents in turn produce an electromagnetic field that opposes the interference field.

The specific shielding materials that we shall next discuss, namely those made by Perfection Mica, consist of an iron-nickel alloy. As previously stated, they are effective from d.c. to about 50,000 Hz; for higher frequencies, other shielding materials, such as copper, are required.

Perfection Mica's two basic types of shielding materials are:

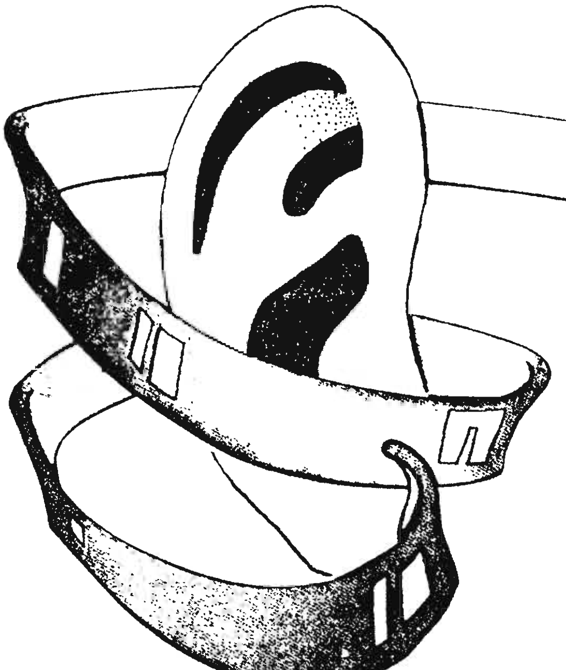
1. *Co-Netic.* This has very high relative permeability, which is a measure of a material's ability to provide a path for magnetic lines of force. The higher the permeability, the greater is the shielding effect. Relative permeability is the permeability of a material relative to that of a vacuum. Air has a relative permeability of about 1. Co-Netic has a maximum relative permeability of about 500,000. For interference fields of weak to moderate strength, Co-Netic can prove an effective barrier when properly used, with a single layer typically providing about 40 dB attenuation of the field. Unfortunately, Co-Netic has a relatively low saturation level. That is, it cannot effectively serve as a barrier to strong magnetic fields. Beyond a point, it can no longer provide a path for magnetic lines of force.

2. *Netic.* This has much lower relative permeability than Co-Netic, but a much higher saturation level. While its permeability is only about 1/100th that of Co-Netic, it can accept more than 100 times as great a magnetizing force before reaching saturation. Hence, when dealing with strong magnetic fields, use of Netic is indicated; however, a single layer will provide only about 1/10th as much attenuation as does Co-Netic.

To achieve a desired degree of attenuation, several layers of shielding material may be needed. For an intense field, the most effective shielding tends to result from a combination of Netic and Co-Netic. The rule to be followed is to place the Netic material closest to the generating device, and the Co-Netic closest to the sensitive device. For example, if combination shielding were to be placed around a transformer, the Netic layer would be on the inside and the Co-Netic layer on the outside; if a combination shield were to be placed around a tape head, the Co-Netic would be on the inside and the Netic on the outside.

Both are available in sheet and foil form, each in a variety of gauges, widths, and lengths. For ease of handling (cutting, shaping, etc.), foil would probably be more advantageous to the audio hobbyist, although sheet has the advantages of rigidity and greater attenuation owing to its greater thickness.

Foil has another important advantage to the hobbyist: It eliminates the possible need for annealing in order to preserve the magnetic properties of the shielding material after it is worked into the desired shape. Annealing consists of heating the material to an appropriate temperature in an appropriate environment and cooling it at an appropriate rate — requiring facilities seldom available to the amateur.



PHASE, TIME, EARS & TAPE

William A. Manly

During nearly 20 years of association with audio and tape recording, I have noticed that technical development tends to be quite faddish. The "in" thing recently has certainly been phase and phase distortion. It really doesn't matter which transducer holds one's interest: Articles and ads can be found in profusion, extolling the virtues of some new technique, or imploring the purchase of some new and expensive piece of equipment. "Transducer" is the key word here — the only electronics which possess any measure of phase shift or distortion are those designed to correct for the non- linearities of some transducer (speaker, phono cartridge, tape head, etc.) in the audio system.

But what is the point? Didn't Herr Doktor Professor Hermann Ludwig Ferdinand Baron von Helmholtz himself say that tone quality did not depend upon the relative phase of its components?[1] Yes, indeed . . . but the statement was very carefully qualified, and Helmholtz tabulated some exceptions. Even those who say that individual system phase distortion is not audible[2,3] will agree that phase differences between stereo channels will displace the stereo image, but the situation is worse than that. I don't intend to take a lot of space here to defend my point of view, but for the purposes of this article I will take the position that certain types of phase distortion are important to audio. The conditions and qualifications will be carefully stated. If you need to be persuaded, I can only recommend Schroeder's marvelous paper,[4] which convinced me beyond a doubt.

Phase

Few readers of this magazine are totally ignorant of what is meant by the terms *phase* and *phase shift,* but there is so much confusion in the literature that it's best to start off with some "basic basics" that we all know. I'll define terms as I go, so that nobody (including me!) gets lost as things get more complicated.

A single frequency can be represented as a constantly rotating vector of constant length, called a *phasor.* To see how this can represent the thing we generally call a *sine wave,* see Fig. 1a. The phasor rotates with one end fixed at "P," and the other end describes a circle. The phasor starts with its moving end at "O." At the end of a certain amount of time, the moving end is at "a," and the phasor has rotated through an angle "A." At the end of twice the first interval of time, the end of the phasor is at point "b," and the rotation has been through an angle "B," which is exactly twice angle "A." This is continued around the circle.

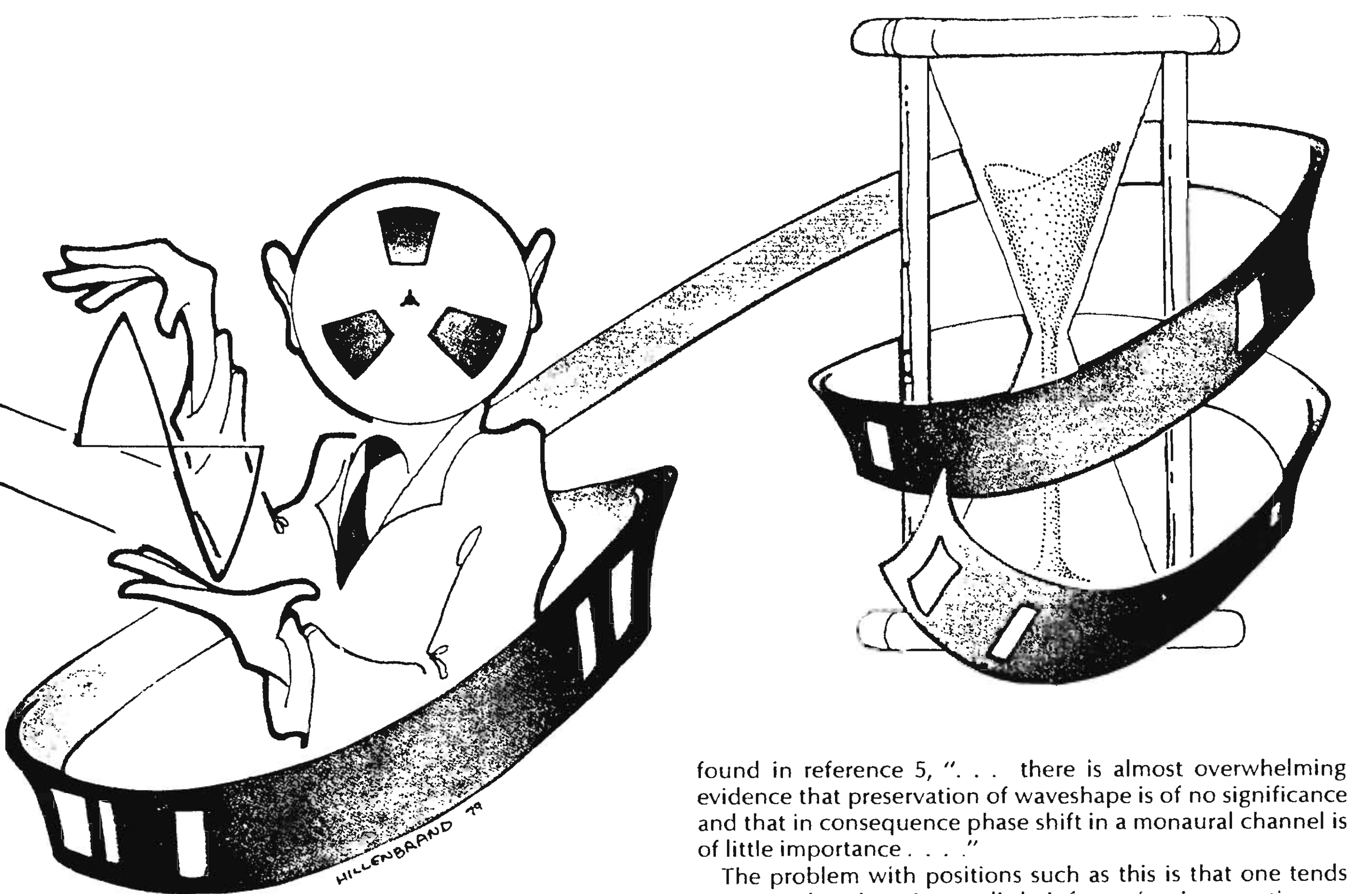
A Cartesian graph is now constructed, with the abscissa extended from the line through the center of the circle and point "O." The abscissa is marked off into equal intervals, and ordinates are erected at each point, labeled to correspond with the points around the circle. Horizontal extensions are made from point "a" to ordinate "a'," from "b," to "b'," etc. A continuous curve (solid line) is drawn through the points where extensions cross their corresponding ordinates. This curve is known as a sine wave. It is a plot of amplitude vs. time of a single frequency. Points along the wave can either be measured by time or by the angle of the phasor. This measurement is known as the *phase* or the *phase angle.* Phase is measured either in degrees (360° for a complete circle) or radians ($2\pi = 6.283\ldots$ for the circle). The points on this particular graph are 20° or $\pi/6$ radians apart. The *frequency* of the signal is just the number of times per second that the phasor makes a complete circle. The *angular frequency* is the number of radians per second through which the phasor rotates or just twice the frequency. *Phase shift* of one sine wave with respect to another can be thought of as if there were a horizontal relative movement of the wave along the abscissa (time axis). In Fig. 1A, the wave shown by the dashed line lags the one shown by the solid line by 90°.

Complex Waveforms

More complicated waveforms can be constructed by adding together sine waves of different frequencies. These frequencies are *harmonically related,* which means that if the *fundamental frequency* is denoted by f_1, the only other frequencies present are the harmonics of f_1, that is f_2 ($= 2f_1$), f_3 ($= 3f_1$), etc. A square wave, for instance, can be constructed with the fundamental (at the same frequency as the square wave) and its odd harmonics, as follows:

Frequency	*Amplitude*	*Phase Shift*
f_1	1	0°
f_3	1/3	0°
f_5	1/5	0°
f_7	1/7	0°
—	—	—
f_i	1_i	0°

with the additions continuing indefinitely. Actually, the frequencies do not go upward without limit in any real system, but the amplitudes of the highest harmonics are so small that they don't contribute very much. If the proper harmonics



with proper amplitudes are added, but the phase relationships are different than shown, the result is not a square wave at all. It may not even be recognizable as a distorted square wave if it is viewed on an oscilloscope screen.[5,6] Nevertheless, what Helmholtz said, is that the ear is sensitive only to the frequencies and the amplitudes, and not in any way to the phase relationships of the various harmonic frequencies. This has been shown many times since by many experimenters, giving rise to such absolute statements as that found in reference 5, ". . . there is almost overwhelming evidence that preservation of waveshape is of no significance and that in consequence phase shift in a monaural channel is of little importance . . . ."

The problem with positions such as this is that one tends to forget that there is very little information in a continuous wave, no matter how exotic its shape. Not many people settle down in their living room chairs for a few hours of easy listening to their favorite square wave! Almost all the information found in speech and music is in the changes found therein, changes in frequency and amplitude, including starts and stops. It also seems obvious that fundamentals and har-

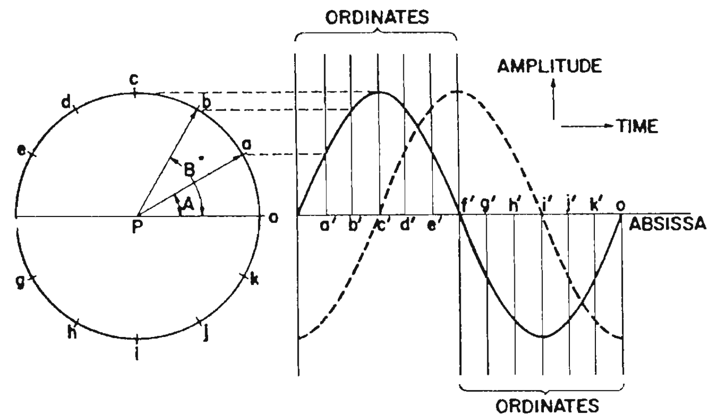


Figure 1A — Construction of a sine wave from a phasor. The phasor is of constant length and has one point fixed at "P." It revolves around "P" with a constant angular velocity. At the end of each unit of angular rotation, the point at the end of the phasor is projected to the proper ordinate representing a unit of time. The height of the ordinate at a' (for instance) is given by h(a') = L sin A where h(a') is the height at a', L is the length of the phasor, and A is the angle of rotation at a. Because of this relationship, the locus of the points is called a "sine wave." The dashed sine wave is of the same amplitude, but it has been phase shifted by 90 degrees.

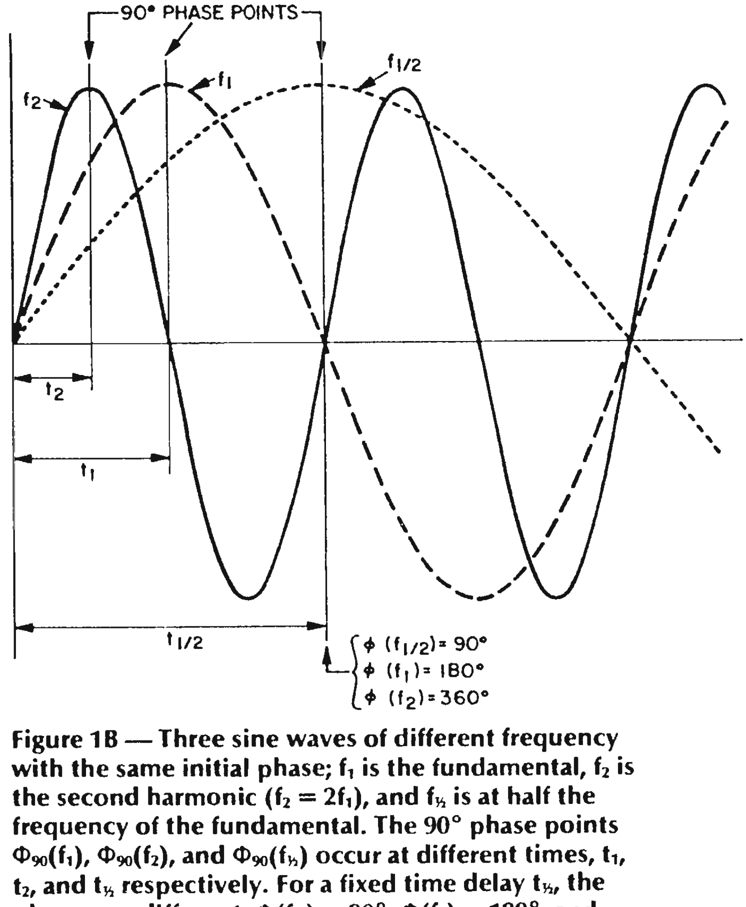


Figure 1B — Three sine waves of different frequency with the same initial phase; f_1 is the fundamental, f_2 is the second harmonic ($f_2 = 2f_1$), and $f_{½}$ is at half the frequency of the fundamental. The 90° phase points $\Phi_{90}(f_1)$, $\Phi_{90}(f_2)$, and $\Phi_{90}(f_{½})$ occur at different times, t_1, t_2, and $t_{½}$ respectively. For a fixed time delay $t_{½}$, the phases are different: $\Phi(f_{½}) = 90°$, $\Phi(f_1) = 180°$, and $\Phi(f_2) = 360°$.

monics must not arrive at audibly different times from each other, so the timing of the changes is important. What all this means, is that the *transient response* (response to changes in the signal) of the system is of great importance — and it is possible for certain types of phase distortions to give rise to audible transient effects.[6]

Time

The phase shift shown between the solid and dashed curves in Fig. 1A can be regarded as a time shift as well. This particular one has a time lag of three units of time. If the frequency was twice that shown, a time lag of three units would give twice as much phase shift, or 180°. For half the frequency, the three units of time shift would give half the phase shift, or 45°. Thus, a constant time delay creates a phase shift proportional to frequency (Fig. 1B). A constant time shift causes no problems, or recordings could not be made now, stored, and replayed later.[5]

Neither is the ear sensitive to a phase shift which is a constant number of degrees for all frequencies. The time delay for a given amount of phase lag is greatest at the low frequencies, and smallest for the high frequencies (Fig. 1B, see how the 90° phase points as a function of frequency affect the time delays in Fig. 1A). This sort of phase change does change the waveshape, as has been previously noted, but the ear ignores it. Of course, if there are two stereo channels with different amounts of phase change, the stereo image will shift its location[7].

Except for minor effects which are caused mostly by nonlinearities in the inner ear, the important effects reduce to two categories, each with two subcategories: 1) Relative time delay, comprised of a) channel-to-channel time delay in a stereo presentation,[8] and b) frequency dependent time delay[8] (this is the same as saying that the phase shift is not a linear function of frequency,[6] and 2) any phase changes in a system when the signal has a positive/negative asymmetry in its waveform, comprised of a) transients,[9] and b) steady state.[4] So far, all of the design effort has been concentrated in dealing with category number 1. Compensating for category no. 2 would require the setting of a phase standard (i.e., if the live music had a positive pressure wave at a certain point, the reproduction should also exhibit a positive pressure wave at that point). Such compensation would also open up a large Pandora's Box; for example, since the phase relations are different at every point in a concert hall, which point is to be chosen for the standard? Nevertheless, category 2 exists, and it can be demonstrated by any well-equipped audio lab, but there has been so little work done on it that we are here forced to concentrate most of our attention on number 1.

Table I — Some time constants of human hearing.

Acoustic Reflex (Gain Control) — middle ear[4]	10 mS
Gain Control — acoustic neurons (Brain)[4]	20 mS
Time for wave to travel the length of the basilar membrane (inner ear)[4]	5 mS
Time for a sound wave coming directly into one ear to diffract to the other ear (calculated)	0.7 mS
Refractory Period (dead time) of an acoustic nerve after firing[4]	1 mS
Electrical pulse (spike) length — acoustic nerve[4]	0.5 mS
Pulse repetition time of acoustic nerve — during onset of strong stimulation[4]	1 mS
Limits of Precedence Effect[10]:	
Lower Limit	1 mS
Upper limit (limit of fusion) — clicks	5 mS
Upper limit (limit of fusion) — complex sounds	40 mS

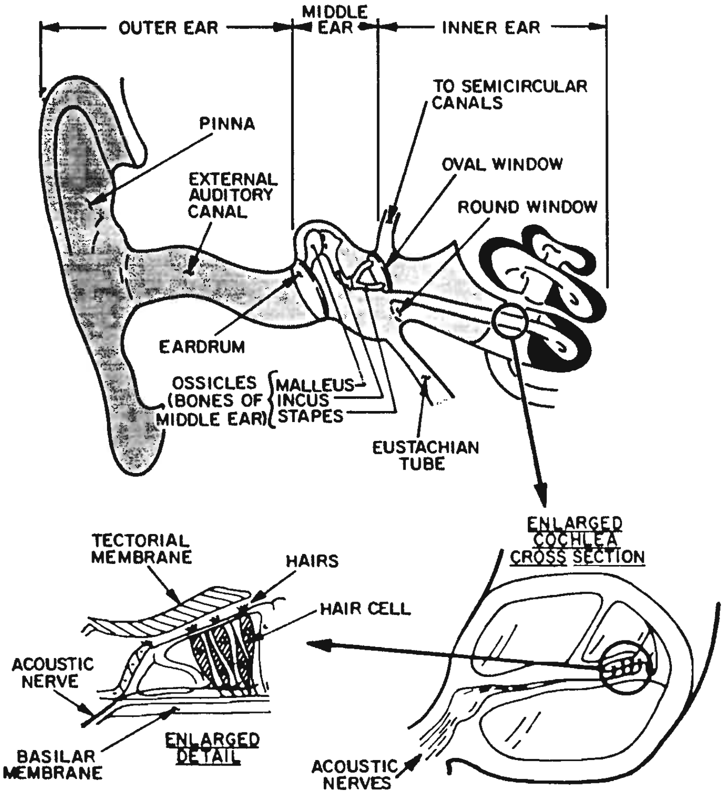


Fig. 2 — The human ear, semi-schematic diagram.[4, 22, 23] The ear consists of three parts, an outer ear, a middle ear, and an inner ear. The outer ear (along with the rotation of the head) assists in the perception of directionality of the sounds. The external auditory canal is a Helmholtz resonator with a resonant frequency of about 4 kHz. Due to the convolutions, the resonance is strongly damped. The ossicles are an impedance matching system to couple the air vibrations to the fluid in the inner ear. The impedance drops by a factor of 22. The resonant frequency of the ossicles is about 1.7 kHz, and it is highly damped. The inner ear is an acoustic transmission line with a low-pass characteristic and a cutoff frequency of about 8 kHz. The cochlea cross-section shows how the acoustic nerves connect to the hair cells. Relative movement between the tectorial membrane and the basilar membrane disturbs the hairs, which sends a signal via the acoustic nerves to the acoustic neurons in the brain.

What we will do now is to place some limits on time delay or phase nonlinearity, then see what restrictions these limits indicate for tape recorder design.

Ears

Table I gives some time constants found in human hearing, and it is well to keep these constants in mind, since one would expect that any time constants of importance would be in the same range as these time constants of hearing. If time effects are described which are not in this range, one should suspect the reality of such effects unless they are satisfactorily explained.

Some description of these effects is in order. The ear consists of three parts (Fig. 2), an outer ear, open to the air and concluded with the membrane called the eardrum; a middle ear, consisting of three small bones (ossicles) suspended in muscle, which transmit the vibrations of the eardrum to another membrane, and the inner ear, which consists of a spiral fluid-filled chamber (cochlea) with a central (basilar) mem-

Table II — Maximum allowable relative time delays for recording.

Between channels (no shift of stereo image)[8]	0.25 mS
Between channels (which are electrically combined to one monophonic channel so that no audible cancellations exist)[8]	0.025 mS
Within a channel — with respect to signals at 1 kHz. These limits are for music and speech.[5]	
at 50 Hz	80 mS
at 100 Hz	20 mS
at 8 kHz	8 mS
Within a channel — between "low frequencies" (700 Hz) and "high frequencies" (7000 Hz) This is a transient limit [11]	2 mS

brane, hair cells, and the acoustic nerve. The *Acoustic Reflex* is a tightening of the muscles around the bones of the middle ear, which dampens the vibrations about 30 dB. The *Gain Control of the Acoustic Neurons*is an electrochemical action that adjusts the firing rate of the acoustic nerurons, which seems to cut the system gain by another 35 to 40 dB, when activated. Both of these are activated only for relatively loud sounds. Standing waves are set up on the basilar membrane, and there is a transient time required for the waves to be set up along the complete length. The acoustic nerves are electrochemical in action and need some time after firing to recover before they can fire again. Each firing puts out an electrical pulse with a characteristic length and with a characteristic pulse repetition rate (which changes to some extent with the amplitude of the stimulation).[4]

Stereo Location

The *Precedence Effect*[10] is best explained by an example. The subject is seated equidistant from two identical speakers in a typical stereo setup. Click-like sounds are caused to come from the two speakers. The click from one speaker is delayed a certain length of time from the other. If the delay is less than the lower limit (1 mS), the subject hears only one click, but is confused as to the origin. If the delay is between upper and lower limits, the subject still hears only one click, but thinks it comes from the speaker where the click occurs first. If the delay is longer than the upper limit, the subject hears two clicks, one from each speaker. Complex sounds, like speech and music, have different upper limits. The precedence effect can be overcome by a difference in amplitude between the two speakers, with the necessary amplitude (to overcome) being determined by the time delay. The precedence effect is sometimes known as the *Haas Effect*, especially by European authors.

There are several effects which can be produced in monophonic and multichannel audio systems as a result of time delay differences. The differences may exist between different portions of the frequency spectrum or between channels. There is some disagreement about the size of these effects or exactly what are the limits, but there is very little disagreement that they exist. The various European standardization organizations for broadcasting, telephone lines, etc., have done the most work to define the acceptable limits of the various differential delays, and they are listed in Table II. Except for the second limit, they all fall in the same general time range as the time constants listed in Table I. The second one differs because it is not a hearing effect at all, but an electrical one. To cancel any sine wave exactly, one needs only to add the same wave shifted by 180° (this is generally known as an "out-of-phase" condition). The time given is exactly half that required to completely cancel a 20-kHz sine wave. The first two limits are for multichannel stereo, while the others apply either to monophonic or stereo signals. The third, which gives limits for audible effects on music and speech,[5] is less restrictive than the fourth, which gives the limits for audible effects on transients.[11]

Table III — Maximum allowable gap scatter to stay within the first two time-delay limits in Table II.

Head/tape speed, ips	Gap scatter limits, microinches: For no stereo image shift	For channel combinations without cancellations
1.875	470	47
3.75	940	94
7.5	1880	188
15	3750	375

Fuzzy Stereo

It seems logical to expect that if a time difference exists between channels which is a function of frequency, that the apparent location of an instrument might wander over the sound field or be spread over a range (rather than be located at a point). A constant amount of phase difference at all frequencies is the same as a time difference which is a linear function of frequency. Bauer [12] says that the effect exists, and that a constant 90° of phase difference will spread the source over the whole sound field between the two speakers. On the other hand, Cooper and Shiga [13] describe a system with a constant 90° of between-channel phase shift deliberately added, and they say that it has no effect on the spreading of sound sources, except when large interchannel phase shifts already exist in the source signals. There's little point in trying to sort this out here, except to point out that the second limit in Table II is so tight that it will not allow a 90° phase difference to exist below 20 kHz. The first limit will not allow 90° of phase difference to exist below 1 kHz.

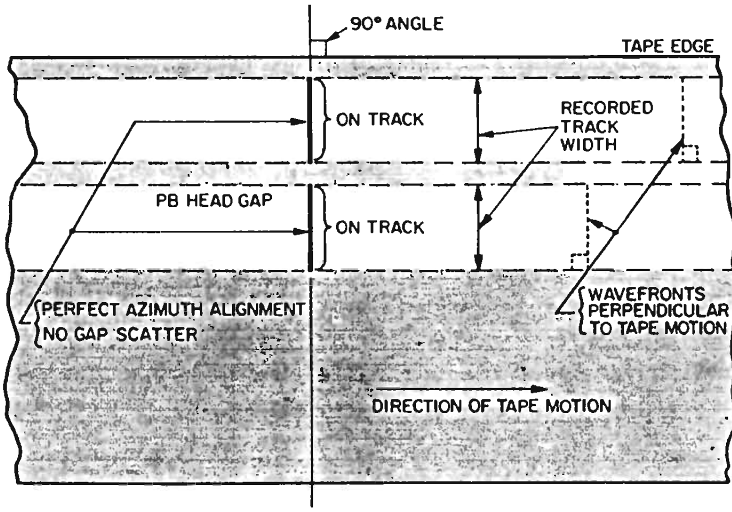


Fig. 3 — Schematic diagram of a magnetic tape with two signal tracks, passing a playback head with two faultless, perfectly aligned gaps.

Tape

Very few of the many facets of magnetic recording can be regarded as linear processes. The transfer function of biased recording, at its best, is an S-shaped curve which is nowhere straight. The high frequency bias, used to "linearize" the transfer function, itself causes large losses at short wavelengths. The high frequencies in the signal look like additional bias to the low frequencies and cause an alarming amount of intermodulation distortion. Mechanical adjust-

Table IV—Sources of amplitude and phase errors in magnetic recording.

Name of Error	Type of Error	Location of Error
Gap loss	A (5)	Playback head
Spacing loss	A (5)	Head/tape interface, on playback
Thickness loss	A (5)	Tape, on playback
Head resonance	A (1,4); P (1,4)	Playback head
Head core loss	A (1,4); P (1,4)	Playback and record heads
Azimuth error	A (1,5); P (2,5)	Between playback and record heads
Gap scatter	P (2,5)	Record or playback heads
Faraday's emf Law	A (4)	Playback head
Recording losses	A (1,3,5)	Tape, on record
Signal demagnetization	A (1,5)	Tape, after record
Gap length/coating thickness geometry	A (1,5); P (1,5)	Record head, in conjunction with the tape
Contour effect	A (1,5); P (1,5)	Playback head contour/tape path geometry
Equalization	A (1,4); P (1,4)	Record and playback electronics

Notes: A=amplitude, P=phase.
(1) Single-channel effect. Any single-channel effect can be a between-channel effect if the channels are different with respect to the effect.
(2) Between-channels effect.
(3) Sensitive to bias setting.
(4) As a function of frequency.
(5) As a function of wavelength, sensitive to head/tape speed.

ments of the transport and heads are very critical, and misadjustments cause several kinds of losses and distortions. Tape magnetization is sensitive to temperature and mechanical stresses on the tape. The list goes on and on, and it's a wonder the process works at all — but work it does and oftentimes very well.

There are a number of things in magnetic recording which cause amplitude losses. Some cause losses of only the amplitude of the signals, while others also cause phase changes. Amplitude losses are very apparent to the listener, so *Equalization* in the playback and record electronics is utilized to compensate for the amplitutde losses, and thus create a system which has a flat amplitude characteristic. The simplest type of equalization unfortunately creates additional phase changes in the system. These additional phase changes in many cases add to the phase changes already in the magnetic recording process, producing systems which are well-known for their inability to reproduce waveforms. [14] A list of the various losses is given in Table IV. [14, 15, 16] We'll take a closer look at these, and see how the phase distortion might be decreased by improved design.

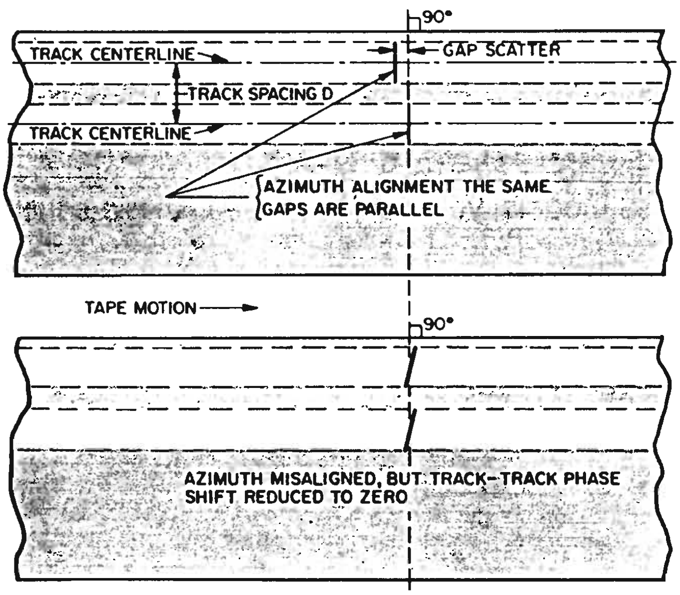


Fig. 4 — Schematic diagram of tapes with two signal tracks, passing playback heads with gap scatter. A) Gaps parallel and aligned, but with between-track phase or timing error. B) Same head, but misaligned so as to remove the between-track phase or timing error.

Playback Losses (Mostly)

The first three sources of loss *(Gap Loss, Spacing Loss,* and *Thickness Loss)* are all amplitude-only loss sources on playback only and do not themselves have any associated phase changes. Gap loss is caused by the finite length of the playback head gap ("length" of a head gap is in the direction of tape travel, though the "width" of the head gap/track may be a much greater dimension). When the recorded wavelength is so small that the gap length is approximately equal to the wavelength, a null occurs in the output (see Fig. 8). There is a sharp drop from about the point where the gap is a half wavelength long to the first null (which occurs where the wavelength is about equal to one gap length). Spacing loss is due to the separation of the head from the tape and is 54.6 dB per wavelength of separation. Thickness loss is similar to a demagnetization loss. The signal on a tape may be viewed as a number of bar magnets laid end to end. When the length of the bar magnets is long compared to their thickness (the tape coating thickness), all the flux lines from the magnets can enter the playback head. When the magnets are short compared to their thickness, some of the flux lines from the parts farthest from the head may not be able to enter the head and thus will be lost to the playback process. These losses may be kept to reasonable values by (respectively): (a) Using short-gap playback heads; (b) keeping the heads and tape surfaces very smooth and the head/tape spacing small, and (c) designing tapes with thin coatings. These three dimensions are measured using the shortest wavelength to be recorded as the measuring unit. Modern systems, especially cassette systems, are designed with these three things in mind.

The next two, *Head Resonance* and *Core Loss,* are losses which are present in all heads, both playback and record. Head resonance is caused by the inductance of the head winding in parallel with the distributed and stray capacitance. Core loss is caused by the fact that the magnetic permeability of the head material is a complex quantity which changes with frequency. With proper design and choice of magnetic material, these can be kept quite small, especially in the record head, which has to work with the high frequency bias signal. Resonance of the playback head has sometimes been used as part of the equalization, but modern practice is to locate the resonance above the audio pass band and utilize all-electronic equalization. This eliminates a large part of the phase shift problem before it ever gets started.

Azimuth Error and *Gap Scatter* are associated with the adjustment and precision of the head's manufacture. When more than one track is in use, it is impossible to adjust the head to eliminate these problems unless the head is perfectly made. Since nothing made by man is perfect, it is always possible to detect errors, even with the best heads available.

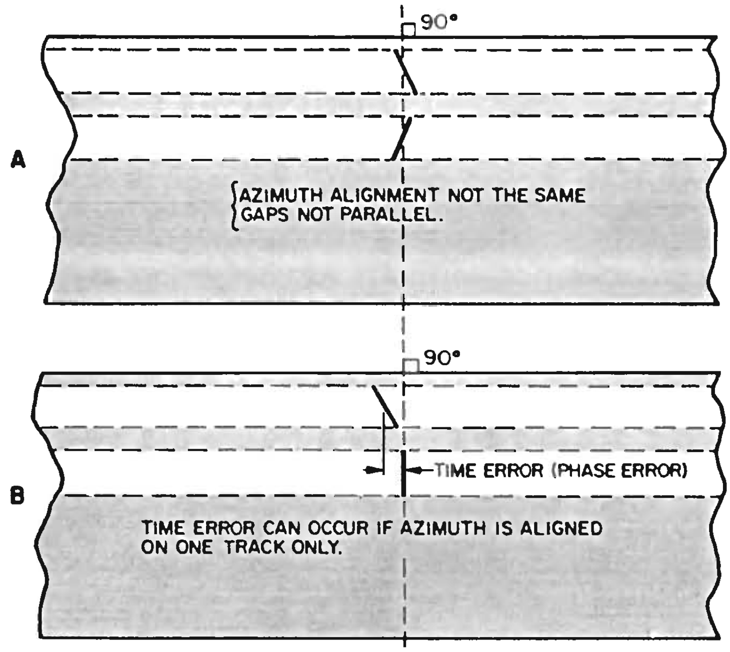


Fig. 5 — Schematic diagrams of tapes with two signal tracks, passing playback heads with gaps which are not parallel. A) Gaps aligned to give zero time error between tracks. B) Azimuth aligned on one gap. Azimuth is misaligned for the other gap, and a time error exists between the two tracks.

Of course, the best heads have acceptably small defects. Let's look at what is meant by these two terms.

Figure 3 shows a schematic of part of a tape passing a perfect two-channel playback head. We assume that the signal tracks have been laid down by perfect record heads, perfectly adjusted. By this, we mean that the record head gaps are perfectly straight and on the same straight line with each other, and that the angle between the gaps and the tape motion (called the *azimuth*) has been adjusted to be exactly 90°. Events which occur at the same time on both channels at record will play back on both channels simultaneously. Also note that the playback head is adjusted so that the gaps are exactly on top of the signal tracks. If the PB (playback) head overhangs the signal track, there will be loss of signal amplitude, but record heads are normally a little wider than PB heads to prevent this, so this particular error is not listed in Table IV.

Figure 4 shows the same situation, except that the PB head has gaps which do not lie on the same straight line, though they are parallel. Figure 4A shows the playback head aligned so that both gaps have their azimuth adjusted properly. The gap scatter is clearly seen. With the tape moving as shown, the gap on top will receive simultaneously recorded signals a little bit before the bottom gap will, thus introducing a time shift between channels. To stay within the first and second limits given in Table II, allowable gap scatter (which depends on head/tape speed) must stay within the limits given in Table III.

The limits in the third column, especially for the 1.875 ips tape speed (used for cassettes), are very demanding of the head-maker's art. The length of the cassette head gap (in the direction of tape movement, remember) is probably about 50 microinches or a bit less. To put this in perspective, green light has a wavelength of about 20 microinches. If you don't want to add to both stereo channels together to make one mono channel, the tolerance for time shift gives dimensions 10 times those shown for stereo channel addition.

Now we are talking about tolerances which any good head should meet, so that stereo image shift due to gap scatter should be relatively rare. Figure 4B shows a condition where the between-tracks time error has been adjusted away, but at the expense of misaligning the azimuth for both gaps (azimuth misadjustment causes an amplitude loss which increases for decreasing wavelength). Methods are in use for making either type of head azimuth adjustment. Both require the use of standard playback alignment tapes, available (at considerable cost) from several sources.

Figure 5 shows a similar situation, but the PB head has gaps which are not parallel. Figure 5A shows alignment such that there is no between-track time error. Figure 5B shows one gap with perfect azimuth alignment, which throws the other gap out of alignment by a serious amount. Proper adjustment for such a head is with the head aligned so that the loss due to misalignment of the azimuth is the same for both channels. There may be some residual between-track time error left after this type of adjustment, but it should not be serious. Most real heads are of this type, but a good one will not require more than about ½ to 1 dB of output drop in each channel at 15 kHz to make the compromise. Figure 5 is greatly exaggerated — the actual angles are very small. Table VA shows just how small — the angles were calculated to give a 1-dB loss in the 15-kHz signal due to azimuth misadjustment. The angles are given in minutes and seconds (for those who have forgotten their geometry, 60 minutes = 1 degree and 60 seconds = 1 minute). If perfect heads have the azimuth deliberately misadjusted to give 1 dB of 15-kHz loss, this will cause an amount of "gap scatter" which is listed in Table VB. Comparing these figures with those given in Table III, we can see that these amounts of "gap scatter" will cause problems with adding stereo channels to form one mono channel, but are too small to cause audible stereo image shift. The track dimensions used in these calculations were taken from a convenient listing published by Nortronics,[17] and the azimuth alignment loss calculations were made using the formula given by Begun: [18]

$$\text{Alignment Loss (dB)} = 20 \log_{10} \left[\frac{\sin\left(\dfrac{\pi W \tan \alpha}{\lambda}\right)}{\left(\dfrac{\pi W \tan \alpha}{\lambda}\right)} \right]$$

Where W is the track width, α is the azimuth angle (α=0 for perfect alignment), and λ is the wavelength of the signal (λ=head/tape speed divided by the signal frequency).

Faraday's emf Law says that the usual inductive PB head has an output proportional to the rate of change of the magnetic flux from the tape flowing in the head. The head output is not proportional to the level itself. The output from

Table V A — Angle for 1-dB loss of signal at 15 kHz.

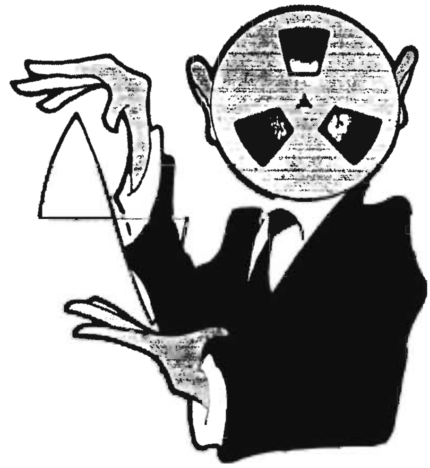
Tape Speed, ips	Track Width, in.	Wavelength, in.	Angle
1.875	0.021 (1)	0.000125	5m, 21.1s
3.75	0.043 (2)	0.00025	5m, 13.6s
3.75	0.021 (3)	0.00025	10m, 42.1s
7.5	0.043 (2)	0.005	10m, 27.2s
15	0.08 (4)	0.001	11m, 14.2s

Table V B — "Gap scatter" caused by 1 dB of azimuth misadjustment at 15 kHz.

Tape Speed, ips	Azimuth Angle	Track/Track Spacing, in.	Implied Gap Scatter, microinches
1.875	5m, 21.1s	0.035 (1)	54.4
3.75	5m, 13.6s	0.136 (2)	206.8
3.75	10m, 42.1s	0.127 (3)	395.3
7.5	10m, 27.2s	0.136 (2)	413.5
15	11m, 14.2s	0.156 (4)	509.9

Notes:
(1) Cassette.
(2) ¼ inch, 2 channel, 4 track.
(3) ¼ inch, 2 channel, 8 track.
(4) ¼ inch, 2 channel, 2 track.

such a head has a constant (+ 90°) phase shift — the same at all frequencies — associated with it. The frequency sensitive amplitude loss (greatest at low frequencies) is normally compensated by an *integrator* in the playback electronics, which has an equal phase shift of the opposite sense (90°). The phase shifts cancel out, except at the ends of the compensation range. We'll come back to this when we cover equalization later.



Recording Losses

The category called "Recording Losses" is actually a collection of several kinds of signal losses which occur during the record process. These are due to [16] *Bias Erasure,* geometry of the record head field, recording field not aligned with the particles, *Switching field* not the same for all the particles, and recording not *anhysteretic* at high frequencies. All of these require some explanation.

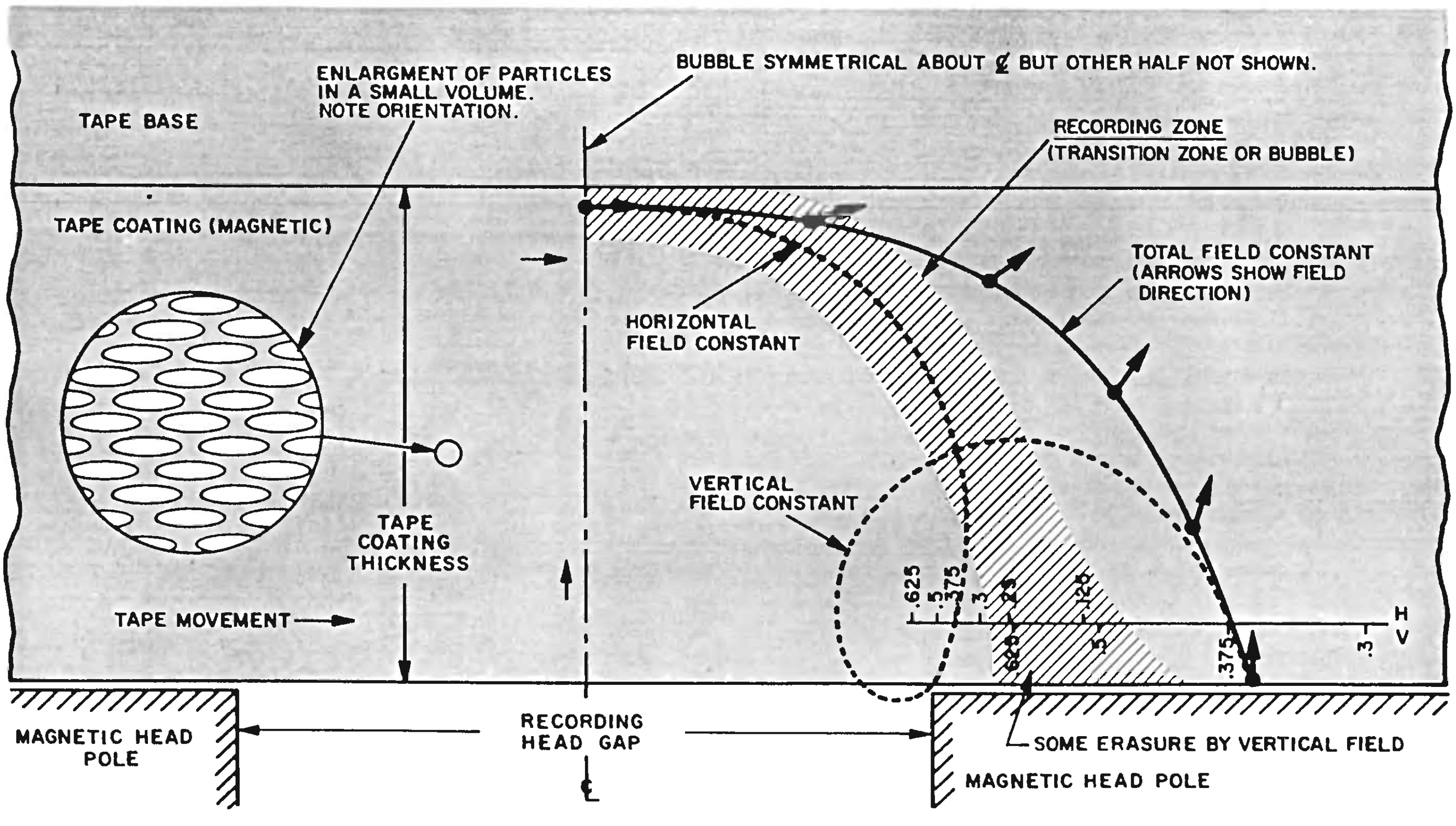
Bias erasure is a description of the appearance of these losses. The bias for maximum sensitivity is not the same for short wavelength signals as for long. The bias is usually adjusted to optimize something such as sensitivity or linearity at long wavelengths, which means that the short wavelengths are overbiased and thus partially "erased." Actually, this "erasure" can be explained in terms of the next four loss sources.

Fig. 6 — Section of the recording head and the tape showing the shape of the recording field "bubble," [16] the shape of the transition zone, and the orientation of the magnetic particles in the tape. In this figure, the record head gap length is of the same order of magnitude as the coating thickness. The scale at the lower right shows the horizontal (above line) and vertical (below line) field values at the points indicated. Note how the vertical field falls off more slowly than the horizontal field as the tape moves away from the record head gap.

The geometry of the record head field is shown in Fig. 6, which is a section through the record head and the tape. Inside of a half-cylindrical "bubble," which lies along the width of the record head gap, the particles in the tape coating are being magnetically switched (having their magnetization change direction) by the high-frequency bias. Outside the bubble, the bias field is too small to switch them. Recording thus takes place at the trailing edge of the bubble, in a transition zone. The reason that this zone is not infinitesimally thin is that all the particles do not have the same switching field, since they are not exactly identical.

Because of the influence of the larger vertical field near the head, the longitudinal extent of the transition zone is greater near the surface of the tape than it is half-way through the coating. The particles in the tape are normally *oriented* in the direction of head/tape movement and are thus easier to magnetize in that direction. Since the field at the surface is only partially longitudinal, it doesn't record on the surface particles as well as it does on the particles in the inner part of the coating. These things don't make much difference at long wavelengths, but at short wavelengths the strongest recorded signal is half way through the coating and not at the surface of the tape, so it suffers from spacing loss at playback. Note that the vertical component of the bias field cannot record upon horizontally oriented particles, but it can erase recording already done by the horizontal component of the field.

One might logically wonder why the *acicular* (needle-like) particles are used and why they are horizontally oriented. Acicular particles have a higher *coercive force,* which means that they are more resistant to demagnetization by external forces and internal fields. Orienting them lowers the distortion and increases the output and sensitivity at long wavelengths. It helps a bit at short wavelengths too, but not as much. Some tapes have a vertically oriented layer at the surface, which allows short wavelengths to be recorded right to the surface of the tape. This gets rid of the problem dis-



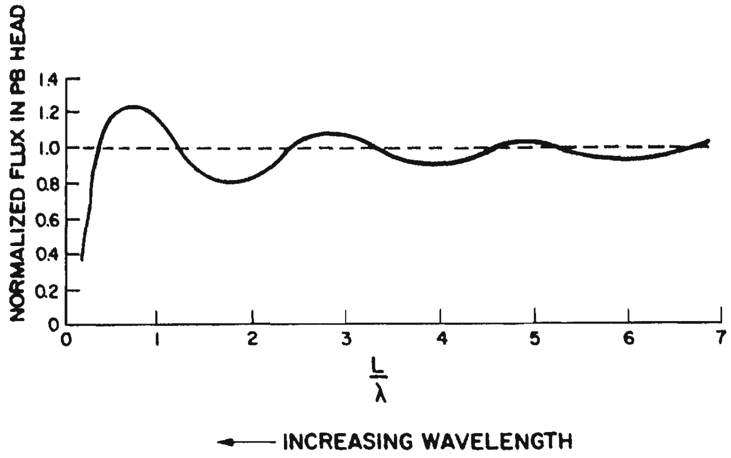
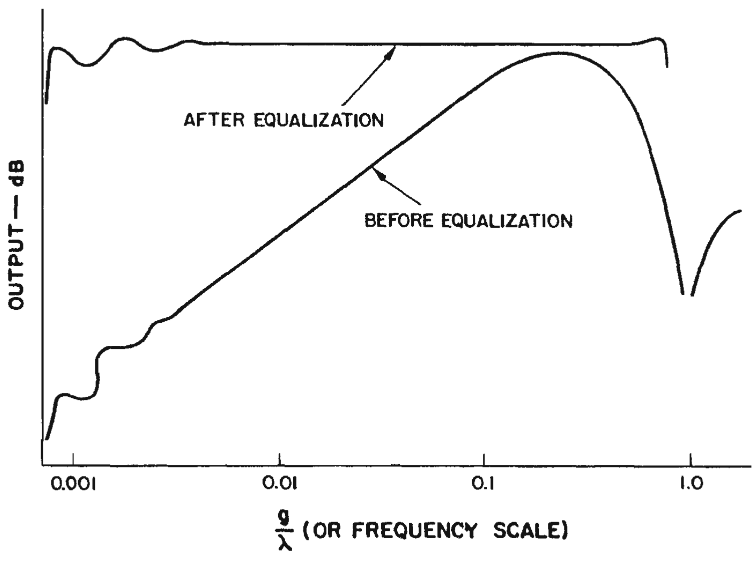


Fig. 7 — Tape flux amplitude circuiting the playback head at very long wavelengths, such that the wavelengths are of the same order of magnitude as the overall length of the head (not the head gap).[19] The interference effect is clearly seen, as is the long wavelength cutoff.

cussed in the previous paragraph, but brings up another: The vertical component of the field does not have a high gradient (drop of the field intensity as the tape leaves the gap) as does the horizontal component. This makes for a wide transition zone, unless a very short gap recording head is used. If the record head gap is too short, there will be difficulty in obtaining good long wavelength recording. The gradient of the horizontal field component is virtually unaffected by change of the record head gap length.

It has already been mentioned that the particles do not all have the same switching field. This means that the particles will "switch" or have their north- and south-seeking poles to exchange places at different applied fields. This causes the transition zone to be wider than it would be if all the switching fields were the same. This wider transition zone causes a smearing of the recorded signal, which blurs the short wavelengths more than it does the long, and results in further short wavelength loss.

Fig. 8 — The total record/playback frequency characteristic before and after the application of equalization to the record and playback circuits. The numbers on the abscissa actually apply to the parameter g/λ, where g is the playback gap length and λ is the wavelength. The frequency is given by f = s/λ, where f is the frequency and s is the head/tape speed, so the numbers on the abscissa differ from the frequency only by a multiplying constant.



To see why the loss of the anhysteretic quality of the recording (at high frequencies) is important, we must first see how anhysteretic recording works. The bias is a strong alternating field which carries the magnetization of almost all the particles through their major (magnetically saturated) hysteresis loop in a cyclic manner, i.e., the particles are magnetized to saturation first in one direction, then in the opposite one. The signal field is much smaller, a tenth of the bias field or less, and it is added to the bias field. When the tape moves out of the head field, the bias field dies out, and the particles stop switching, with some magnetized with their north poles in one direction, and some in the other. The number of left-pointing particles is equal to the number of right-pointing particles with a no-signal condition, but the numbers become unequal as the signal strength rises, almost in perfect proportion to the signal strength. All this requires that the signal change very little as a section of tape moves through the transition zone. This is nearly true for low frequency (or long wavelength) signals, but becomes less true as the signal frequency gets higher (and the wavelength gets shorter). Thus, recording which is nearly anhysteretic at low frequencies becomes less so at high frequencies. Anhysteretic recording is the most efficient type of recording (i.e, it has the highest sensitivity), hence high frequency recording is less efficient than low frequency recording.

After the tape leaves the vicinity of the record gap, it still lies on the high permeability head for a while (permeability can be regarded as a measure of the affinity of a magnetic material for absorbing magnetic field lines). Recorded signals on the tape can be regarded as a series of small permanent magnets, which put out field lines. The head absorbs the field lines from these little magnets almost completely. This absorption of the field actually helps the magnets to retain their magnetization, as there is no field left to demagnetize them (any field from a permanent magnet always has the effect of demagnetizing the magnet to some extent). When the tape leaves the vicinity of the record head, the signals are demagnetized, with the shortest wave-length signals suffering the most. When the tape comes into contact with the playback head (which is also high permeability), some of the demagnetization is reversed, but some of it is not reversible and is permanent. Again, the short wavelength signals suffer in respect to the long wavelengths. Also, the tape with the highest permeability of its own suffers more from irreversible demagnetization than tapes with lower permeability.

Figure 6 shows a record gap which is of the same order of dimension as the coating thickness of the tape. If the record head gap is about twice as long as the coating thickness, this is a balanced situation, with neither the short wavelengths nor the long wavelengths being emphasized. If the record gap length is a lot smaller than the coating thickness, it is difficult for the head field to reach all the way through the coating. Recording tends to be near the surface, and short wavelengths are emphasized with respect to long. If the record head gap is quite long compared to the coating thickness, the field easily reaches all the way through the coating, but the field near the surface tends to be more vertical than horizontal at normal bias levels, the transition zone is correspondingly longer at the surface, and short wavelengths suffer more recording losses than long.

There is only a finite length of the playback head in contact with the tape. As the wavelengths begin to approach this dimension, an *interference effect* occurs. Some wavelengths cause more flux to go into the head than would go into an infinitely long head, while some wavelengths cause less. At wavelengths longer than about twice the length of the head, very little flux gets into the head, and the head is not responsive to longer wavelengths. The flux going into a head of finite length L is shown in Fig. 7 as a function of L/λ where λ is the wavelength (reference 19, page 181). The undulations are known in the trade as "head bumps." They can be minimized by contouring the head (thus, some writers call this "The Contour Effect") so that the tape approaches and leaves the head in a gradual manner. Any head, despite contouring, will have no useful output above some cutoff wavelength. This effect holds even for heads which are flux sensitive and suffer no losses from Faraday's emf Law. A flux sensitive head will read the tape flux at zero head/tape speed, and thus zero frequency, but it will not read tape flux at infinite wavelength.

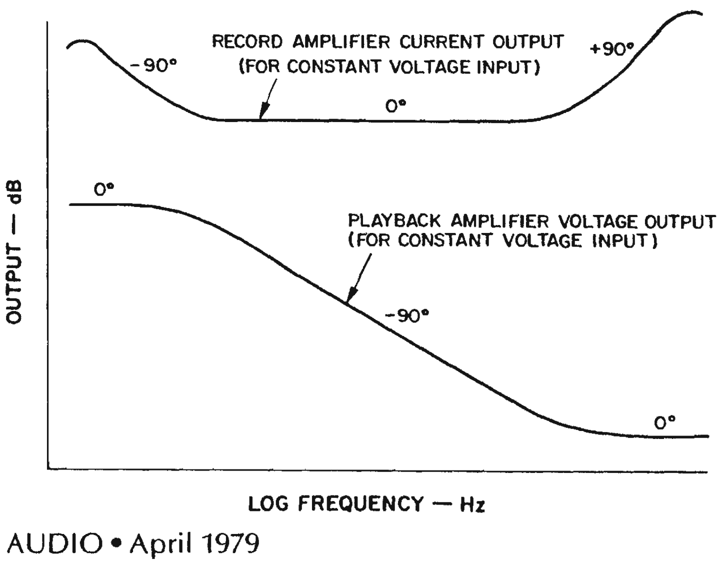
Equalization

The total record/playback frequency characteristic before the application of any equalization is shown in Fig. 8. Also shown is the desired flat frequency characteristic after equalization. Some of the correction is done during recording, and some is done during playback. A considerable amount of literature has been written on just how much to apply in each case. The playback characteristics, being easiest to measure, are usually standardized to be less than the total amount needed for any tape. Only enough record equalization is applied to make the frequency characteristic flat with a particular tape; thus we say that a recorder is "optimized for a certain tape." With any other type of tape (which always has somewhat different characteristics), the overall frequency characteristic will not be flat.

The general type of equalization characteristics applied to the record and playback channels are shown in Fig. 9. A particular machine will sometimes leave out one or both of the shapes at the extreme ends of the record curve. Electronic circuits which have these frequency characteristics generally have an associated phase shift. When the curves change shape (as at the ends), the phase shift is especially pronounced. Because of the very complicated situation shown in Table IV, the phase shifts generated by the equalization never cancel the phase shifts arising from other sources. In fact, they sometimes add to them or create phase shifts where none were before.

In Fig. 9, one can estimate the phase shift caused by the two amplitude characteristics this way. If there is no slope to the amplitude characteristic, the phase shift is zero; if the line slopes to the right by 6 dB per octave, the phase shift is -90°, and if the line slopes to the left by 6 dB per octave, the slope is +90°. These relationships apply to a whole class of electrical networks called *minimum phase networks*. Most ordinary amplifier circuits and simple equalizer circuits fall into this category. With some effort, the designer can construct circuits which have phase shift different from the rules given above. He can then (theoretically) tailor in a phase shift characteristic which exactly compensates for the total of the phase shifts from all other sources, and get a tape recorder with a flat phase shift characteristic. Such a recorder would not suffer from waveform distortion, group delay, or any of the other ills previously outlined. For several reasons, this is not what is presently being done.

Fig. 9 — The general frequency characteristic of the playback and the record equalization for audio machines. The numbers above the curves give the amount of phase shift at those points, if the equalization is produced by minimum phase networks.



Phase Compensation

An excellent example of the current practice is the Ampex ATR-100, a mastering recorder with phase equalization as well as amplitude equalization.[14,20] For all of the errors listed in Table IV except "Faraday's emf Law" and "Equalization," the designers used only a careful and thoughtful engineering design to keep the phase errors suitably low. This design ensured that the major part of the phase shift error would be in the equalization. A standard playback equalization was used to compensate for the Faraday's emf Law. This introduced an extra phase error in the manner which has been described. Then, in the record electronics, an extra phase shift was introduced in the opposite direction which compensated for the phase shift errors in *all* of the equalization. The phase shift adjusts along with the amplitude adjustment, so that compensation is obtained for any tape to which the machine is adjusted. Thus, tapes made on this machine will play back on any other machine with small phase errors, since the compensation is recorded onto the tape itself. A square wave recorded on this machine comes back looking very much like a square wave — something many other machines will not accomplish.

Another example, the Sony 880-2, utilized a different design philosophy. Again, careful engineering design and some carefully made heads keep the introduced phase shift down to a low level. Contrary to the Ampex practice they chose to put their phase equalization in the playback circuit. So long as both recording and playback is done on the same machine, it makes little difference where the phase equalization is, provided the recordings are at a suitably low level. When the recordings are at levels near saturation, phase shift can cause higher than usual *peak factors* (the ratio of the peak value of a waveform to its rms value) to occur. If these peaks are high enough to be clipped by tape saturation or by the electronics, audible distortion will occur. Clearly, it would be much better to correct some of the phase distortion in playback and some in record,[21] to keep the peak factors down to a suitably low level. Sony's engineers probably chose to do it all in playback since any machine sold to the general public will presumably be used to play quite a few tapes made on other machines. The Sony machine will perform phase correction on "prerecorded" tapes.

Now let's take this to its ultimate conclusion. Suppose that all the recorded tape producers use machines with complete phase compensation on all their record circuits. These tapes will play back on all the old machines free of phase distortions. On your brand-new machine with phase compensation in the playback circuits (Sony isn't the only one!), the tape will be overcompensated and will be phase distorted in the opposite sense. Plainly, someone out there ought to be doing some work on standardization of phase correction techniques. Since there still is an argument raging on whether phase compensation is necessary or not, there's not much chance of standardization right now.

None of the foregoing should be construed as a criticism of the engineering decisions which have been made so far. In the absence of standards, all the decisions have been about as good as could be made.

Conclusions

This discussion has looked at a number of different aspects of the audio phase problem. I think that a good case has been made that phase distortions are audible and that phase corrections will result in cleaner sound. We have looked at the sources of phase distortions in magnetic recording, and some of the means now being used to correct for those phase distortions. Lastly, we have seen that there is a great need for standardization in phase correction (phase equalization) techniques right now! Standardization people, what are you doing for us?

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Up in the Air About Metal Tape?

Howard A. Roberson

For the audiophile who has been using cassette recorders, there have been tremendous changes in the performance in this format in relatively few years. It wasn't that long ago (1965) Philips brought out its Cassette-corder, which was strictly for fun, family gatherings, and other low-fidelity purposes. A stereo recorder did appear the following year, but its performance was ho-hum rather than hi-fi. In this same period, of course, there was the start and growth of the software (tape) manufacturers. Ten years ago, development became more rapid, including such new products as TDK's SD (Super Dynamic) tape.

At this time, there was the need for and the practice of greater cooperation between the manufacturers of the recorders and the tape makers. The inherent nature of the magnetic interfaces made this essential: An improved tape needed improved record, play and even erase heads, and improved head designs could benefit from better tapes. It was a great year for the cassette format: In 1970 significant improvements were made in extending frequency response and lowering noise with the introduction of CrO_2 tape and the Dolby B system. Some started using the expression "high fidelity," but "for a cassette" was usually added. The appearance of the Nakamichi 1000 in 1973 impressed many with its one-grand price, but many more were impressed with the sonic results. This was a time of conversion for many skeptics who had concluded that tape speed and width precluded any possibility of acceptable playback.

The next couple of years saw a great expansion in the generation of well-performing cassettes decks and new tape formulations. Some of these were not successful, including early FeCr versions which were not stable and cobalt-doped ferrics which had their own sort of erratic behaviour. TDK brought SA on the market in 1974, and its ion-absorption approach enabled the use of the desirable cobalt. The improvement in straight ferric tapes was continuing all during this period, and continues at this moment. Many other excellent tapes were introduced by Maxell, Scotch, BASF, Fuji, Memorex, and others. The introduction of metal-particle tapes is significant, but it is part of this relatively short history which has included a number of important changes.

Before we get into a discussion of the history of the metal-particle tapes and their magnetic properties, we should review some of the basic characteristics of tape formulations. The reader will note that there are three hysterisis loops in Fig. 1, one for a gamma ferric oxide such as Maxell UD, one for Nakamichi SX (or another similar cobalt-modified ferric), and one for Nakamichi ZX metal-particle tape. Let's make a trip around the loop; follow any one of the three, for the basic story is the same. We also assume that the magnetic material has gone through the loop prior to our discussion, so we are not building up from zero magnetization.

As the coercivity increases in the positive direction, the retentivity also increases, very sharply at first. These steep slopes are desirable and are a goal of the designer, because this shows that the flux in the material is responding well to the magnetizing force. Alas, the material reaches a point where magnetic saturation takes place, and increasing the magnetizing force does not increase the flux in the material. With the lowering of the applied field, when coercivity has been reduced to zero, a certain flux level is retained. This is called retentivity which is the number of flux lines per cm^2 of the tape coating cross-section. This is flux density with the units, gauss, and the area determined by the tape width and the thickness of the coating. This is a fundamental measure of the magnetic performance of the particles, but there is a bit more to this part of the story.

Tape performance is determined by the actual number of lines of flux induced, not just the density. Remanence is the actual signal retention in total lines of flux (in Maxwells), contributed by the tape coating thickness and width. As we recognize that the highest frequencies would not penetrate a thick coating, we can see that the choice of coating thickness facilitates matching low- and high-frequency record sensitivities. Once again, note that retentivity (flux density) goes with the properties of the particles and the remanence (actual lines of flux) goes with the tape product. Squareness ratio is the decimal fraction of flux at zero coercivity to the flux at saturation.

So far in our trip around the loop, we have succeeded in magnetizing the material, but what's required to demagnetize it? If a magnetizing field is applied in the reverse direction, the flux in the particles will be reduced down to zero along the lines in the second quadrant. For the other direction of induced flux, we would travel the other half of the total loop. Let's restrict our attention, however, to the important second quadrant of the figure and make some comparisons among the tapes. Note that there are two dashed lines showing the 5-kHz and 20-kHz demagnetization loss line. These are based upon those appearing in Vogelgesang's article in *Audio* just a year ago.

For the particular tapes shown, the retentivity increases from 1050, to 1550 to about 3500 gauss. There is no doubt about the significant increase in possible flux levels, especially at the low frequencies. Vogelgesang points out, however, that this high flux capacity is needed for good high-frequency performance, along with high coercivity. For this property, the tapes show values of about 300, 550 and 1000 oersteds, showing increasing resistance to erasure. The demagnetization loss lines indicate the much lower likelihood of self-erasure with the metal particle tape. Consider also that with the much higher retentivity, the tape designer has a bit more flexibility in the choice of coating thickness to achieve the best combination of extended response and high record level. The high coercivity is a mixed blessing in that it places much higher demands on the erase and record heads — a definite challenge to the head designer. Later on we'll take a look at some test results with both old and new machines.

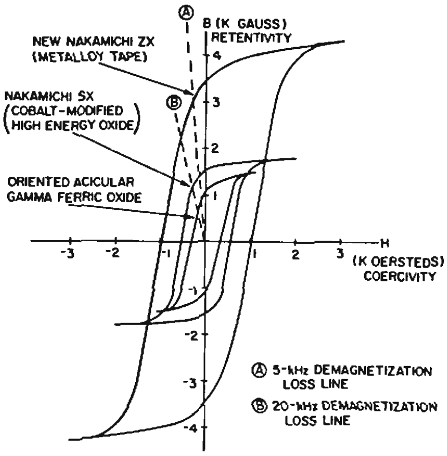


Fig. 1 — Hysteresis loop characteristics of three cassette tape formulations.

Fig. 2 — Particles used in TDK SA tape.

Fig. 3 — Particles used in TDK metal-particle tape.

Metal-Particle Tapes Arrive

The 3M Company has been working on the technology of a tape coated with pure-metal particles since 1965, which is when cassettes themselves started, though the basio work on the particle began two decades earlier. In the last couple of years, the activity in this area accelerated greatly and produced many rumors. Most of the printed discussion of problems of pure-metal particles talked about the need to prevent the actual rusting or oxidizing of the particles. There were some stories about extra fast oxidation, also known as burning, of the material. (Now, that would be a hot tape!) 3M and others, however, were solving the processing problems, and samples began to appear more regularly in early 1978. Scotch will probably gain some points with the use of "metafine" as the name of their metal-particle tape. TDK is using "MA," and Nakamichi has "ZX," but Fuji so far says just "metal." Maxell, BASF, Memorex and Ampex have announced intentions to manufacture metal-particle tapes, but no specific information had been received at the time of this writing. Tests on the first-mentioned tapes, which are reported later, used a Nakamichi 582 deck. Other units compatible with these challenging tapes are being offered by Tandberg, Pioneer, JVC, Aiwa, Lux, TEAC, Sanyo, BIC/Avnet, Technics, Eumig, and others I expect, by the time this appears in print.

The various manufacturers have been gearing up the advertising departments, as well as the production lines, for output of metal-particle tape. Some of the manufacturers have supplied technical data, but some of the "standards" used are not the same. TDK, for example, makes comparison to TDK SA and refers to a TDK standard tape, while Scotch uses DIN references. TDK data shows the possibility of using a bias level 3.5 to 5.0 dB higher than SA bias. Increased headroom at the higher frequencies is listed as 5 to 7 dB greater. Figures 2 and 3 are photomicrographs for the particles used in TDK SA and MA (metal-particle) tapes, respectively. The thinner elements in the MA tape are actually small balls strung together, somewhat like a pearl necklace. Each little ball is about 300 Angstroms in diameter. The SA particles are not in the shape of a chain, but are needlelike, about 0.5 microns in length (5,000 Angstroms). TDK states that the coating thickness of the metal-particle tape is about 4 microns.

Bias for Scotch Metafine is shown as +6.5 dB, but this is referred to DIN ferric bias, so it is not so far removed form the bias for TDK MA as it seems at first. Scotch claims that Metafine has output twice as great as chrome tapes at low frequencies and three times as great at the high frequencies. The manufacturer states that, overall, "this results in 5 to 10 dB greater output over chrome." Additional data on Metafine and TDK MA is given in the tape-tests report in this same issue.

One concept that has received a fair amount of attention lately is the rating of magnetic tape performance by determining its signal capacity. With their extension of high-frequency response, metal-particle tapes have been touted by some as "greatly superior" because of the indicated increase in such signal capabilities. In the next section, consideration is given to this

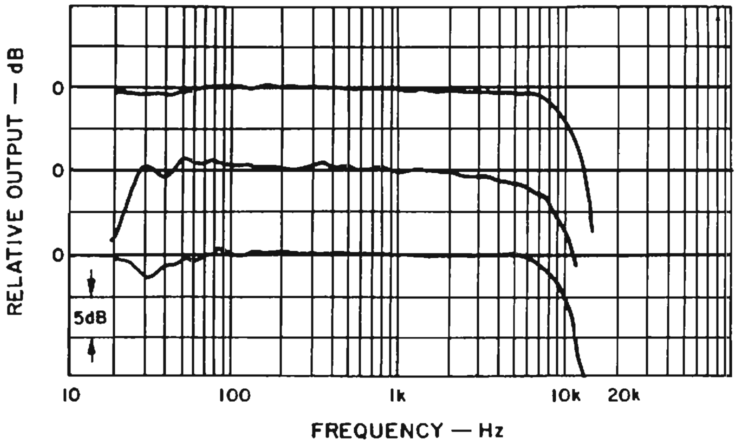


Fig. 4 — Frequency responses at Dolby level obtained using TDK SA ferri-cobalt tape in three recorders: Top, Nakamichi 582; middle, Technics RS-9900US, and bottom, Harman-Kardon HK1000.

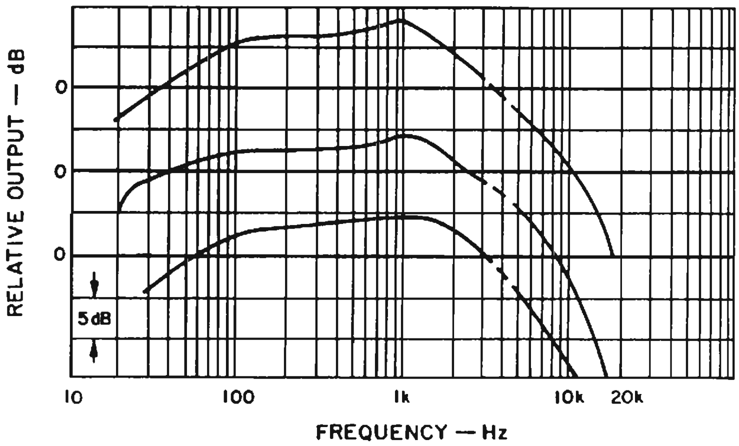


Fig. 5 — Three-percent distortion limit for TDK SA tape in three recorders: Top, Nakamichi 582; middle, Technics RS-9900US, and bottom, Harman-Kardon HK-1000. Zero reference is Dolby level.

and other ways of assessing the improvements with metal-particle tapes.

Performance Improvements: Tape or Machine?

In the process of evaluating the Nakamichi 582, the reviewer was struck by the fact that its performance with FeCo tapes was superior to the majority of machines tested in the past. Further, a review of all the data revealed two interesting things. First, although the results with the metal-particle tapes were superior, there was not as much of a difference as expected. Second, the figures that have been used to indicate the expected improvement were much closer to the difference between the Nakamichi deck with the metal-particle tape and older recorders with chrome-type formulations.

To gain some understanding on the inter-relationships, TDK SA was used for record/playback responses at Dolby level and for maximum-record-level tests. Three recorders were used, the Nakamichi 582, the Technics RS-9900US, and the older Harman-Kardon HK1000. The bias on the first two machines was adjusted to match SA; the HK 1000 had been aligned to SA previously. The maximum record levels were determined with HDL_3=3 percent from 20 Hz to 3 kHz and twin-tone IM distortion = 3 percent from 5 kHz to the upper frequency limit. Figure 4 shows that all three responses are quite good at Dolby leyel; the Harman-Kardon is quite impressive, considering its vintage. When the comparison is made among the machines for the distortion limit (Fig. 5), the Nakamichi has superior headroom across the entire band. The same cassette was used for the three decks, so the differences are machine related, although small shifts in bias could bring some changes.

It might be noted that the 3 percent distortion-limit curves have a slope of about -6 dB per octave and cross zero dB around 2 kHz. With the slope of much music on the order of -3 dB/octave, however, the distortion may reach the stated limit at lower frequencies first. The ability to record a wider, undistorted bandwidth with the Nakamichi comes form greater headroom at *both* ends of the band.

Nakamichi ZX metal-particle tape was exercised in the same way with the same three recorders, but there

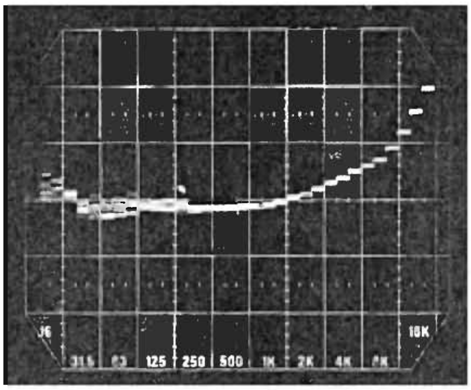
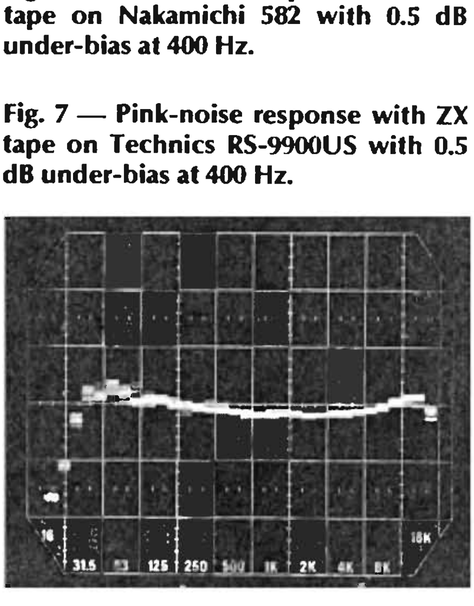


Fig. 6 — Pink-noise response with ZX tape on Nakamichi 582 with 0.5 dB under-bias at 400 Hz.

Fig. 7 — Pink-noise response with ZX tape on Technics RS-9900US with 0.5 dB under-bias at 400 Hz.



were some changes. First of all, no attempt was made to adjust the Harman-Kardon deck bias to match this tape. The results would show what to expect from using such tapes in a deck actually set up for a tape similar to TDK SA. Bias was adjusted on the Nakamichi and Technics decks to match the ZX tape, using pink-noise at -20 dB. The RTA display was as expected with slight under-bias for 400 Hz with the 582 as shown in Fig. 6. Note that the highest frequencies curve upward, as they should under this condition. With slight under-bias with the Technics, however, the rise in the highest frequencies is very mild, indicative of possible self-erasure effects from the high bias (+3.8 dB re: CrO2 zero bias). Figure 8 shows the frequency response plots, with the expected low level and high-frequency peaking on the HK1000 with the severe under-bias condition.

For the great majority of the band (see Fig. 9.), the headroom on the 582 is superior, particularly at the frequency extremes. Note, however, that the Technics has a higher limit from 5 to a little over 10 kHz. To help put a handle on some of the comparisons that can be made, Fig. 10 shows the increase in distortion limit (or headroom) across the band for two cases. The first one examines the improvement by going from the Technics RS-9900US with TDK SA to the Nakamichi 582 with Nakamichi ZX tape. There is an advantage of about 5 dB for most of the band with a rapid increase above 14 kHz. In the second case, just the 582 was used, and the differences between TDK SA and Nakamichi ZX were measured. The average improvement is about 2.5 dB which is worthwhile and nice to have, but certainly less than many of the claims that have appeared. We will get back to look at

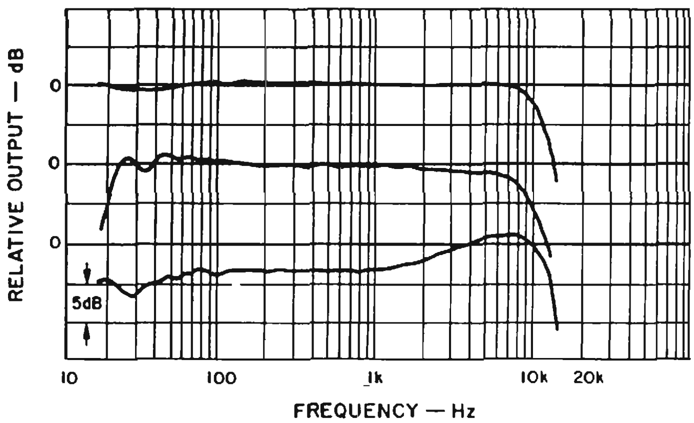


Fig. 8 — Frequency response at Dolby level with ZX tape on three recorders: Top, Nakamichi 582; middle, Technics RS-9900US, and bottom, Harman-Kardon HK-1000.

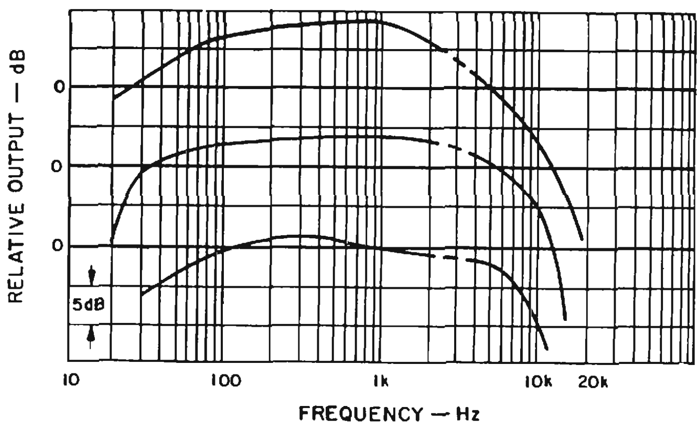


Fig. 9 — Tnree-percent distortion limit using ZX tape on three recorders: Top, Nakamichi 582; middle, Technics RS-9900US, and bottom, Harman-Kardon HK-1000. Zero reference is Dolby level.

these relationships from another perspective after discussing another facet of rating tape performance.

Mention was made earlier of rating tape formulations by their signal capacity. As we have just seen, the performance of a particular tape can vary a great deal from one machine to the other. One has to be very careful, then, about firm conclusions about a tape without being certain of the effects from the machine. If we refer to Fig. 10 again, we could say that there is a great increase in signal capacity if we look at the top plot, or just a useful increase if we use the bottom plot. For analog recording, forecasts of improvements in total performance based upon the increases in signal capacity can be misleading. Some of the formulas being used treat each Hz of bandwidth as equally important. A response or distortion-limit plot on this basis would have linear frequency, such as 2 kHz for each of ten divisions. With noticeable increases in headroom between 10 and 20kHz, there's a great increase in signal capacity. Before you get bowled over by numbers derived in this way, remind yourself of a couple facts. First of all, there is no way that the 10-kHz band from 10 to 20 kHz will ever be as important as the 10-kHz band from zero to 10 kHz for *analog* recording. Second, the levels of the harmonics keep dropping with frequency, except in *rare* cases. The gains with this type of recording should be assessed with frequency on a log basis. If we consider *digital* recording, however, the gains in signal capacity with high-end improvements can be directly helpful, they could be essential for a digital system using the cassette format.

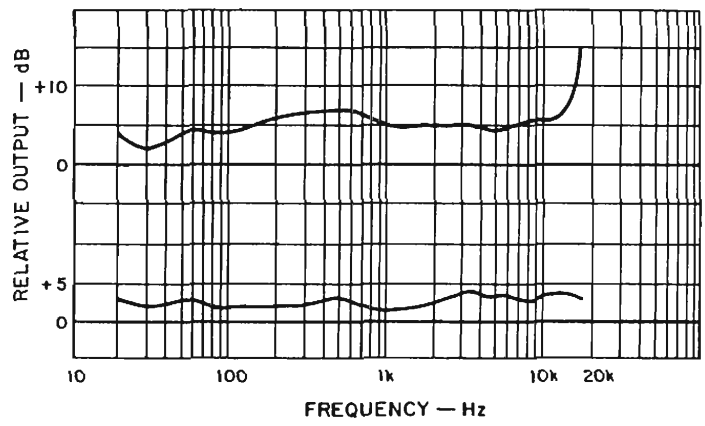
There are some other machine-tape relationships which merit discussion. Scotch states that an erasing field of 3000 oersteds is required, and all manufacturers have commented on the problems of adequate erasure with decks not designed for metal-particle tapes, even if they have the bias capability for record purposes. A few tests confirmed that the problem is real. The Nakamichi 582 was the only one of the three used for the previous tests that was able to erase greater than 60 dB across the audio band. In most places, erasure was greater than 70 dB. On the other hand, the Technics deck had erasure of only 40 dB at lower frequencies with metal-particle tapes. Nakamichi had stated that its deck was able to do a better job than many bulk erasers. I had discounted this claim until I found that I had to use the 582 deck to do what my bulk eraser could not.

The severe challenge to using metal-particle tape in a present deck thus includes many factors. Even if the unit can generate enough bias drive to the record head for the mid-frequencies, limitations in head design could cause a drastic self-demagnetization of the higher frequencies. The user may also have to face the problem of being unable to erase what was put on the tape. Further, he is likely to find out that his bulk eraser can't hack it either. Mine looks impressive, and it says "professional" on it, but it didn't do the job.

Is Metal-Particle Tape Worth It?

The new metal-particle tapes do provide worthwhile improvements in total sonic performance when used with a well-designed deck. It should be clear from the previous discussion that using such tape is not a simple case of throwing another cassette into your present machine. It is quite probable that the new tapes will stick to the 70-μS EQ, which would allow playing pre-recorded tapes with such formulations on existing machines. There are certain to be some hobbyists who will make modifications to their present machines, but the challenges

Fig. 10 — Increase in distortion limit vs. frequency for Nakamichi ZX tape over TDK SA tape. Top, results with ZX tape on Nakamichi 582 deck and SA on Technics RS-9900US deck. Bottom, results with both tapes on Nakamichi 582. See text.



are many, and this approach cannot be given a general recommendation.

There will be an increasing number of new decks that will have the basic capability to utilize metal-particle tapes. As the text above showed, the requirements for record and erase heads are very severe and a great challenge to the designer. Some will obviously be more successful than others. Specific points to check when contemplating purchase include the following: (1) erasure, particularly at low frequencies, (2) headroom across the entire audio band, and, (3) the means of setting and checking bias for best response. The combination of a deck, well designed in these and other respects, and metal-particle tape could very well be a most worthwhile change for many owners of present decks. It is also possible that other new decks will offer improved performance will all formulations, as evidenced with the Nakamichi 582.

The Future

It is to be expected that the metal-particle tapes will continue to improve; the results reported elsewhere in this issue used early-run samples. Perhaps the prices will be quite close to those for FeCo and CrO_2 tapes. And, we should expect to see improvement in performance because of up-dating in the deck designs, particularly the heads. In other words, we will see greater headroom and wider response in the future, with contributions from both types of manufacturers. Areas worthy of particular attention by the engineers are improved consistency, lower modulation noise and lower distortion at the frequency extremes. Perhaps there will be standards established for bias to minimize the possible spread in tape bias requirements without such a guideline.

Refinements should be expected to continue with ferric, FeCo and CrO_2 formulations, upgrading their performance. Du Pont states that CrO_2 still has considerable undeveloped potential for audio cassettes. The FeCr tapes and their future is problematical: There are only three formulations, so deck makers may not want to keep such a switch position with the metal-particle tapes on the scene — and *they are* here to stay.

Tape-to-Deck MATCHING For Best Dolby Tracking

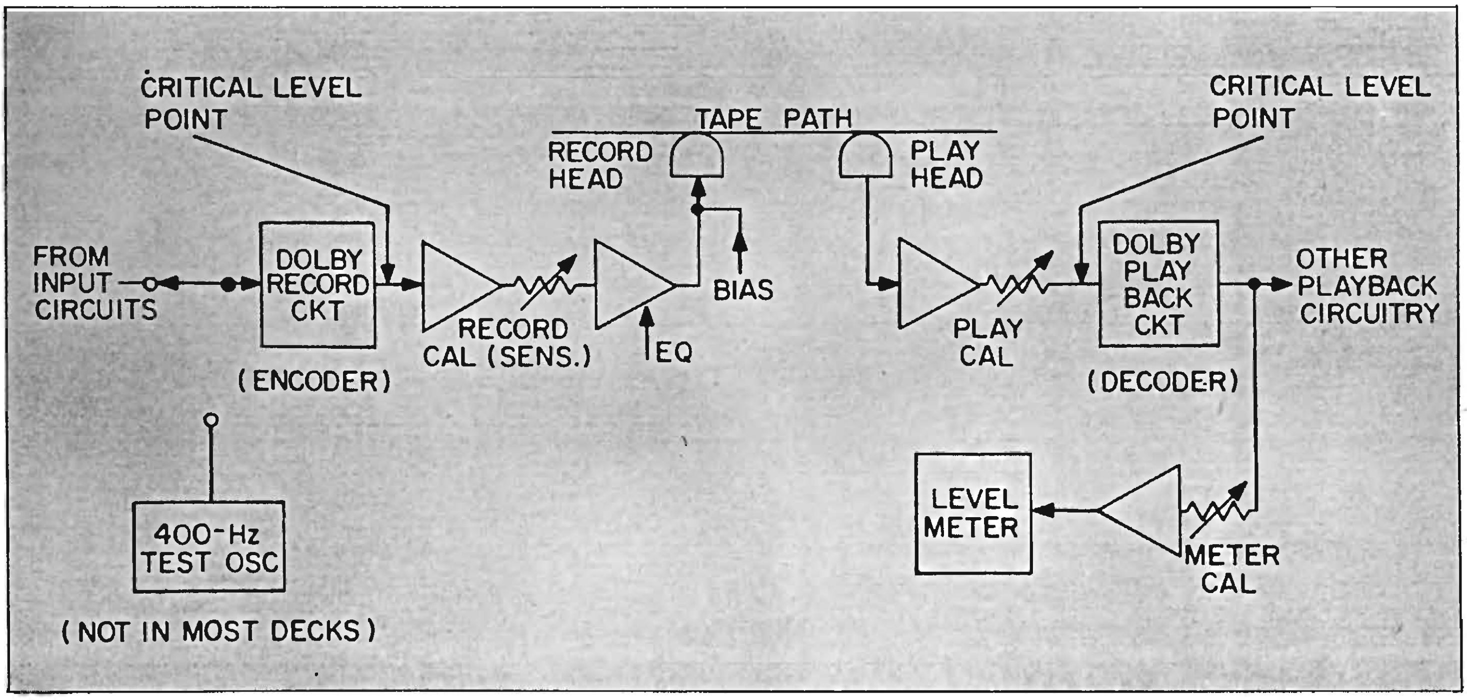
Howard A. Roberson

The inclusion of Dolby noise reduction has certainly been a major factor in the success of the cassette format. Unfortunately, performance in this mode has been unsatisfactory in too many cases because of poor matching between the tape formulation and the Dolby circuit adjustments in a particular deck. A good part of the problem is the fact that most manufacturers do not inform the owners as to what specific tapes were used for set-up.

Let's take a look at the elements essential to Dolby NR calibration. Figure 1 shows the blocks of important parts, as they appear in a few decks. What is shown is for purposes of illustration, such as separate record and play heads and the built-in test oscillator; many other circuit elements, including some switching to the meter, are not shown. The Dolby encoder and decoder perform correctly when fed specific voltages. The output of the decoder should be at a certain voltage at the Dolby level reference, the manufacturer may make an internal adjustment if necessary, and then *Meter Cal* is adjusted for such an indication. Now, when a standard Dolby-level tape (200 nWb/m at 400 Hz) is played on the deck, *Play Cal* is adjusted for the same indication. At this point, playback and metering are calibrated; in other words, metering and decoding are properly referenced to Dolby level.

The critical level points, as indicated in the figure, are at the input of the decoder (which is set with *Play Cal)* and the output of the encoder. To set up the encoder, the manufacturer uses signals specified by Dolby and makes any adjustments necessary, all of which are points inaccessible to the user. If there is a built-in test oscillator, it must be set (usually internally) for the correct drive level, but this is not an adjustment of the encoding function. Between the two points defined as "critical" in the figure, we have some amplifiers, the record and playback heads, the tape path, and the introduction of bias and EQ. In general, we can say, and we do rely on the fact,

Fig. 1 — Essential elements in Dolby N/R loop calibration.



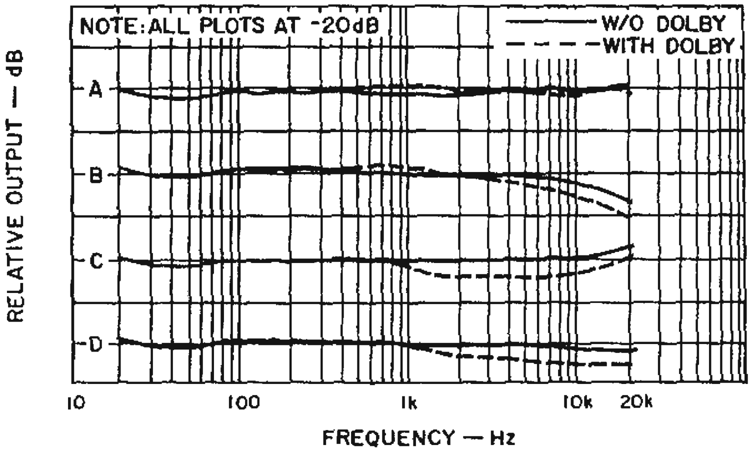


Fig. 2 — How various bias and record sensitivities affect frequency response. A, responses with bias and record sensitivity matched to tape. B, responses with excessive bias. C, responses with correct bias with sensitivity set 3 dB too low. D, responses with bias causing 2-dB drop in Normal and record sensitivity set 2 dB low.

that the amplifiers have stable gain. What does change, however, is tape type with corresponding changes in record sensitivity and EQ and bias requirements.

When the manufacturer sets up the deck, he uses a series of adjustments for each tape for best frequency response and to maintain the level relationship between the two critical points. These same requirements apply to a two-head machine with the single record-play head, where the head and Dolby-circuit functions are switched for playback. If the user of the deck knows what the manufacturer utilized for set-up, he will probably get the best results with the same formulations (or ones very close in characteristics).

If another formulation is used, however, the results may be quite disappointing if bias requirements and/or record sensitivity are different. In Fig. 2, the swept-frequency response plots from 20 Hz to 20 kHz illustrate how this can happen. In the top set of plots, the Dolby response is almost exactly the same as that without noise reduction. For the second set, bias was purposely increased to cause a drop of 3 dB at 15 kHz without Dolby NR. As the dashed plot shows, the response with Dolby is poorer, having about twice as much droop above 3 kHz.

For the third group, bias was returned to the original setting, and record level calibration was set 3 dB too low (at 400 Hz). Here the result is a shelving action with a 2-dB reduction in level, with Dolby, from 2 to 10 kHz. For the bottom plots, bias was set for a 2-dB reduction at 15 kHz, and 400-Hz record sensitivity was set 2-dB low. Here we have a -1½ to -2-dB shelf with Dolby NR with a general fall-off as the frequency is increased. Changes in the sound when switching in Dolby are most evident in the last two cases: There is a definite loss in presence, and most music sounds quite dull.

So, what can you the owner do to minimize such effects? First of all, if you have a deck which includes facilities for checking and adjusting bias, EQ, play calibration or record calibration, you can use what adjustments you have to aid in verifying the performance with Dolby N/R. Proceed cautiously, however, and keep track of any changes made. You may determine that you can get a better match with another tape type. Don't forget that if you want to check play calibration, a Dolby-level test tape is needed. Checking for unwanted shifts in response when switching to Dolby is easiest at 20 to 25 dB below Dolby level. Mistracking should be obvious, and the signal level well above noise. Use music and FM interstation noise, if you do not have test sources.

It's tougher to get to the solution if your deck does not have such adjustments on the front panel. If you're lucky, the manufacturer stated exactly which tapes are best. If there are more than a very few tapes listed, the list is suspect. Unfortunately, most manufacturers hesitate to state exactly which formulations are used for set-up, a reluctance they should overcome. By referring to the two figures and reviewing the discussion above, the reader may correctly conclude that using a tape that plays 400 Hz back at the same indicated level as in record is a matching of record sensitivity. (Check position of output pot, etc.) With a number of tapes that pass this test, record/listen tests with music should pinpoint the best bias/EQ match without Dolby. The tapes surviving these tests should give you good frequency response as well as low noise when in Dolby mode.

Tape guide

Herman Burstein

Which Is the Real Tape Playback Curve?

In *Audio* magazine, tape playback equalization at 7½ ips has been pictured in the manner of Fig. 1, with pronounced bass boost. Elsewhere it has sometimes been presented in the manner of Fig. 2, which shows quite the opposite — bass cut and treble boost. The uninitiated reader may well wonder which is correct.

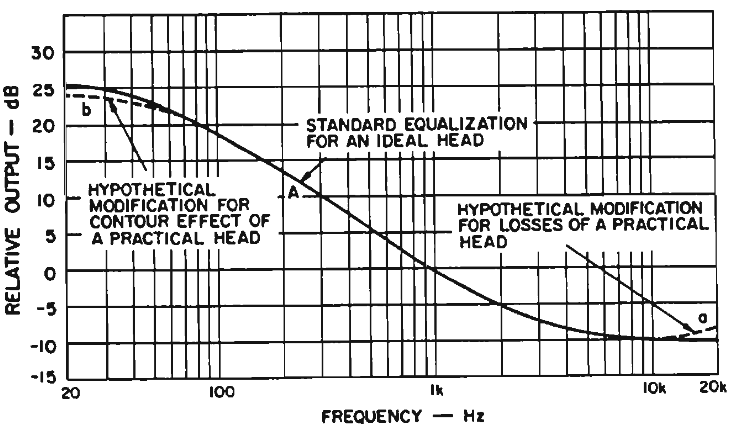
They are both correct but approach the subject from different viewpoints. Before going on, it should be noted that the following explanation applies in principle not only to playback equalization for 7½ ips, but also to equalization for other tape speeds and for various cassette tape formulations.

Conventionally we think of equalization as a change in frequency response performed by an electronic circuit. For example, an FM tuner provides treble cut in order to compensate for the treble boost applied by the broadcast station; this strategy helps reduce noise. A preamplifier provides both bass boost and treble cut when a magnetic pickup is employed to play a phono disc; this compensates for the bass cut and treble boost employed in disc recording to minimize distortion at low frequencies and noise at high frequencies.

Turning to tape, we find that in the complete absence of equalization, record-playback response would take the shape of an inverted U: A combination of severe bass loss and severe treble loss. Bass loss is due to the intrinsic nature of a magnetic playback head, which responds to the rate of change of the signal, so that output varies with frequency. Given a flat signal (constant flux in its core), the head produces an output signal that changes at the rate of six dB per octave as frequency changes. The change in output with frequency may be viewed as either bass loss or treble rise. Here we refer to it as bass loss.

Further, owing to the contour effect—where the playback head as a whole and not only its gap responds to the recorded signal—a practical head may produce somewhat greater output than an ideal head in the low bass region. Thus, Curve A in Fig. 1 might be modified slightly, as shown by Curve b. All in all, however, the playback equalization called for in order to achieve flat response is quite close to

Fig. 1—Standard tape playback equalization at 7½ ips for an ideal magnetic playback head.



that of Curve A in Fig. 1.

In 1965, to emphasize the fact that playback equalization must reflect the irregularities of a practical head, and to get away from the notion that a playback head must inevitably have a six-dB-per-octave characteristic, the NAB (National Association of Broadcasters) decided to present standard playback equalization in a different manner, that of Curve B in Fig. 2. This practice was also adopted by the RIAA (Recording Industry Assoc. of America).

Figure 2 considers the playback head, as well as the playback electronics, part of the playback equalization system. Given a flat signal (constant flux in the core of the head), it indicates that the playback head and the amplifier electronics should *in combination* produce the output of Curve B in Fig. 2—treble boost and bass cut.

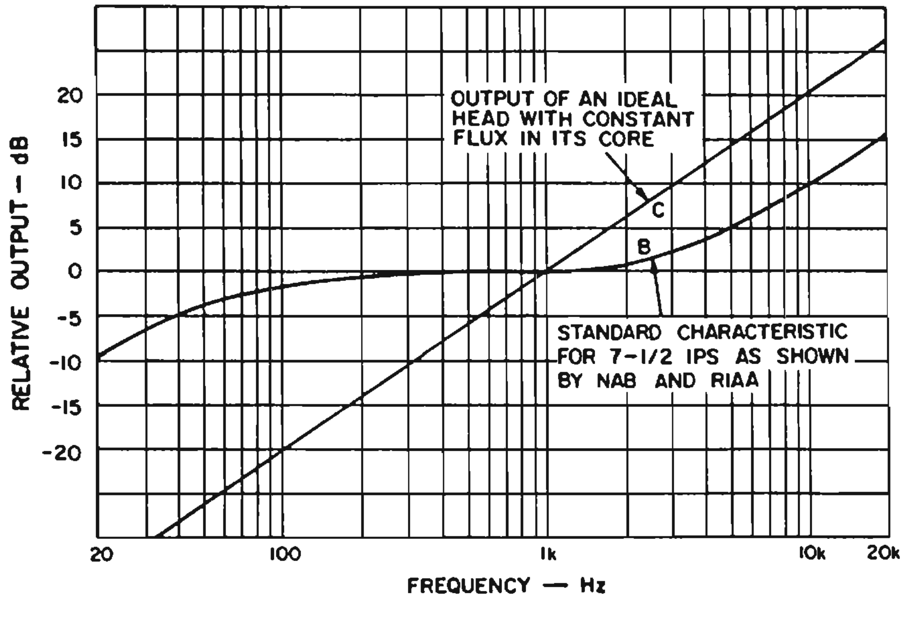
But the output of a magnetic playback head tends to rise six dB per octave with frequency, as shown by Curve C in Fig. 2. Therefore, the difference between Curves B and C must be the equalization supplied by the playback electronics. Specifically, the *difference* between Curves B and C in Fig. 2 is equal to Curve A in Fig. 1.

One way to see this is to plot the difference between Curves B and C on audio graph paper. A second method is to turn Fig. 2 about 34 degrees clockwise until the six-dB-per-octave line (Curve C) is horizontal. In reference to this line, it may easily be seen that equalization supplied by the playback electronics consists of bass boost.

Another way of describing Curve B is that it shows the electronic equalization that would be required if the output of the playback head were flat instead of rising six dB per octave. A Hall-effect head, which has long been in the offing, would have flat output. But up to the time of this writing, playback heads have universally had a six-dB-per-octave characteristic. Therefore, if you measured the playback equalization of a 7½-ips tape deck, you would obtain something quite close to Curve A in Fig. 1.

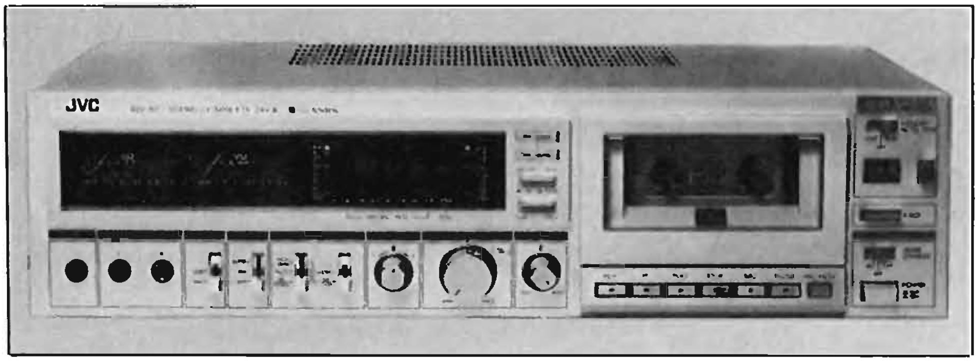
Similarly, at other tape speeds and for the various cassette tape formulations, you would find that playback equalization supplied by the deck's electronics consists primarily of bass boost.

Fig. 2—Standard reproducing characteristic: Reproducing amplifier output for constant flux in the core of an ideal reproducing head.



Ronald G. Ajemian*

THE NEW LEVEL METERS



Introduction

Lord Kelvin, William Thomson, perhaps said it best, in the last century:

"When you can measure what you are speaking about, and express it in numbers, you know something about it; but when you cannot measure it, when you cannot express it in numbers, your knowledge is of a meager and unsatisfactory kind; it may be the beginning of knowledge, but you have scarcely, in your thoughts, advanced to the stage of science."

More succinct is the engineer's axiom: "To measure is to improve."

While it is not trivial to note that we must have some notion of what it is we are attempting to measure before we apply the calipers, since it's difficult to measure volume with a yardstick, we must also fit the scale to the subject — so that yardsticks are not applied to record grooves or to FM reception distances.

Thus, we come to the subject of this article, new techniques in metering circuitry, specifically Light-Emitting Diode (LED) and Vacuum Fluorescent (VF) types. I'll leave it to the author to explain how they work; I want to explain why these techniques are new and why they are important.

First of all, we should consider the older type of meter, the one with the swinging arm or needle. You can still find them in many, many pieces of electronic gear today, and they will probably be with us for at least another generation. When they are well made, they work well and, when proper for their application, should not be discounted in their excellence.

The way this meter works is fairly simple. It's similar to the grade school science teacher's experiment with the magnet and the paper of iron filings; remember how more filings stood up when the magnet was right under the paper than when it was further away. Swinging-arm meters work much the same way; the magnet moving closer to the paper is equivalent to more current being fed into the meter's electromagnetic coil. The trick is to make the meter proper for its application, and here we should distinguish between two basic types of meters — averaging and peak responding.

Perhaps the most common kind of averaging meter is the VU (volume unit) type, which is the basic meter found in recording, both on professional and amateur levels, and in broadcast, the two places where most current types of equipment got their start. While I personally suspect that the originators of the VU meter would have liked to have made their brainchild respond to peaks if they could have, this was probably mechanically impossible at the time. More important, however, was the problem a peak-reading meter would have caused an engineer in a recording or broadcast center — how in the world does one set a level so as to stay up off the noise floor and stay down out of system distortion when the meter needle is jumping around?

The answer is what is embodied in the VU meter, sluggish ballistics, but ones that can be duplicated by any competent manufacturer and that work off voltages and into impedances

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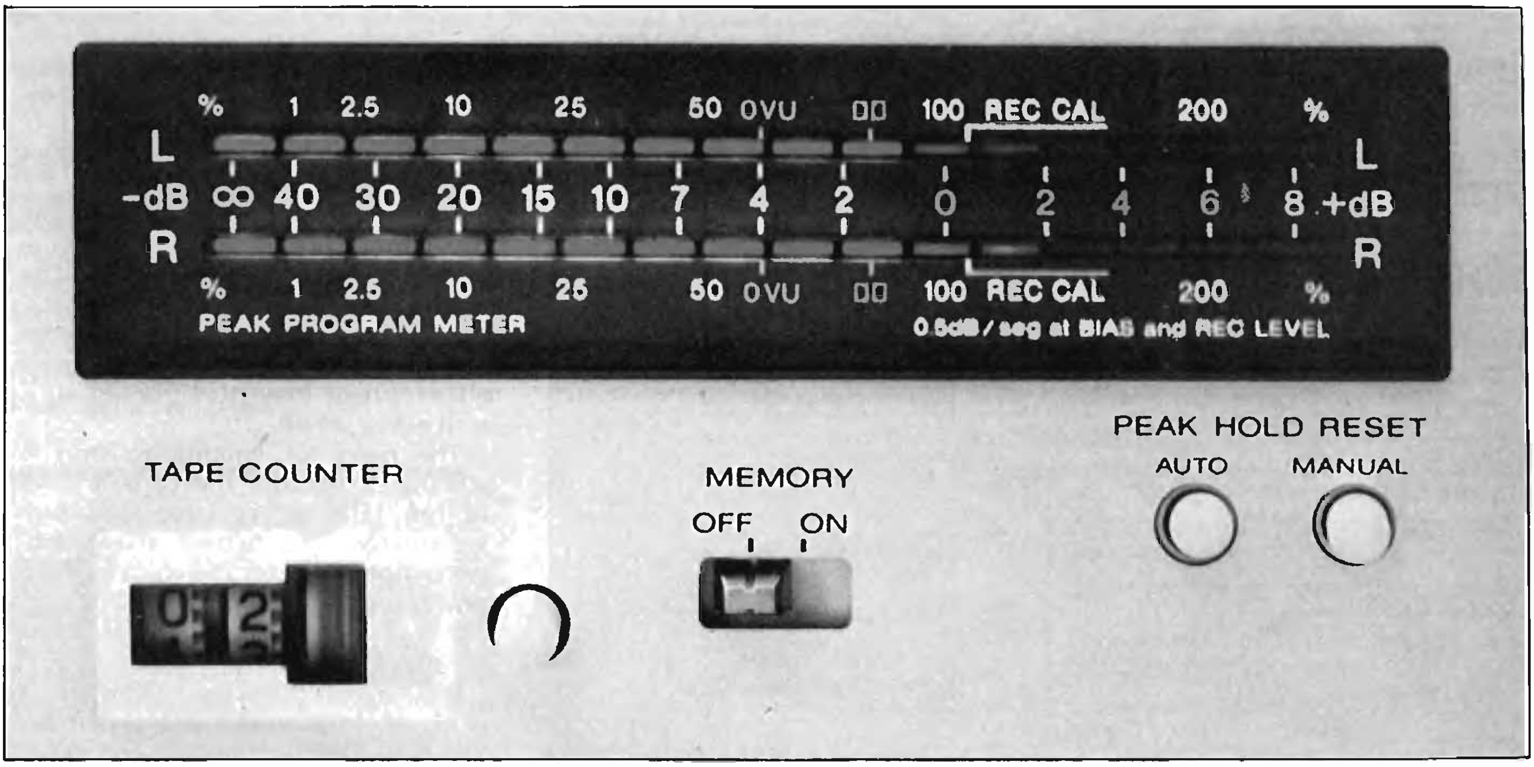


common in the equipment and commonly agreed upon. (The 0 level on a VU meter corresponds to one milliwatt of power in a 600-ohm circuit, i.e. 0.775 volt).

The announced intent of the VU meter was to at least roughly match the ear's response to varying loudness, and it does so quite reasonably well, because the ear-brain combination tends to average out strong and weak sound pressure levels. The position of the VU meter's needle, therefore, seems to mimic one's subjective reaction to different loudness levels.

There are, however, other factors involved. With meter ballistics slowed down, engineers (as well as listeners at home) began to hear bursts of noise and distortion, even though the VU meter was staying relatively still. Usually, this was because the medium was being pushed up into distortion for short periods, and standard VU meter ballistics could not and were not intended to show these peaks.

Thus, engineers were confronted with a need for a meter or system that would show these peak values, which can easily be as much as 12 to 15 dB higher in level than the value shown on a VU meter. While compression and limiting circuits can do some of the work in adjusting signal levels, they too need an initial setup.

How slow is a VU meter? Well, it's pretty quick actually, taking about a third of a second to reach 99 percent of a standardized input. However, the ear is substantially quicker since it takes only about 15 to 30 milliseconds, one-tenth the time, before we begin to recognize distinct notes or echoes. So it's no wonder that we can hear distortion on peaks that are missed by averaging meters.

Admittedly, I am shortchanging the meter makers, since they can make faster meters than the averaging type, of which the VU meter is an example. But the same basic problem of slowness remains, and there are other factors which deserve at least a passing mention. Obviously, the meter should respond to a broad range of frequencies, say 20 Hz to 20 kHz, but should the meter's relative response to different frequencies be tailored to an A-weighting curve following the ear's natural response? Or do we want the meter's response to show us something about system distortion at different frequencies, since that's what we're trying to get rid of (along with trying for high signal levels)? And there are polarity problems too. Some meters do not respond as well to a signal that starts off in the negative direction as they do to a signal that starts off positive.

One solution is in the electronic displays discussed in the other part of this article. Since they are electronic, they can turn on and off virtually with the speed of the signal they are tracking. But, and even better, these displays can be tailored to many, many uses, with infinite variety of refinement, through the use of timing networks and filters. They can have fast attack times to accurately follow peaks and, just as easily, have slow decay times so the eye has no trouble following the bouncing meter. And they can make excellent spectrum analyzers — something not possible with meters.

But let's see how these new displays work. . . . *The Editor*

Currently we are seeing more and more Light Emitting Diode (LED) and Vacuum Fluorescent (VF) based audio power meters being employed in receivers, power amplifiers, tape decks, etc. These electronic displays usually give indications of clipping, which can cause distortion and damage to an amplifier or speakers, and they are also excellent devices for balancing a home stereo system.

Many manufacturers produce the integrated circuits for LED and VF display indicators. Their bar/dot displays are arranged either horizontally or vertically, while some scales are in a semicircle format similar to those found in analog meters. A pointer moves along the analog (moving-coil) meter scales, but it is moving light that does the pointing in LEDs or VF meters.

Operation

National Semiconductor's LM3916, represented in Fig. 1, may be used to illustrate the operation of bar/dot VU-display drivers. This monolithic integrated circuit works on the comparator principle. The LM3916 contains an accurate precision 10-step voltage divider network connected to the positive inputs of the comparators, which scales to provide an electronic version of the conventional moving-coil meter. The circuit will illuminate 10

LEDs consecutively when the input level at pin 5 increases past the threshold points at -20, -10, -7, -5, -3, -1, 0, +1, +2, and +3 dB; it is also equipped with a stable adjustable reference.

The signal input at pin 5 has a low bias-current buffer, which is a negligible load on the signal source. The buffer drives 10 individual comparators referenced to the precision voltage divider network, and typical overall inaccuracy (deviation from ideal) is well below ±1 dB.

The need for limiting resistors is eliminated because the current drive to the LEDs is regulated and programmable. This feature allows efficient operation of the system from power supply voltages of 3 to 25 V.

Displays

It is beyond the scope of this article to list all of the numerous LED displays that are available. However, Figs. 2 and 3 show a few examples of some that are in wide use.

Unlike the LED, the VF display con-

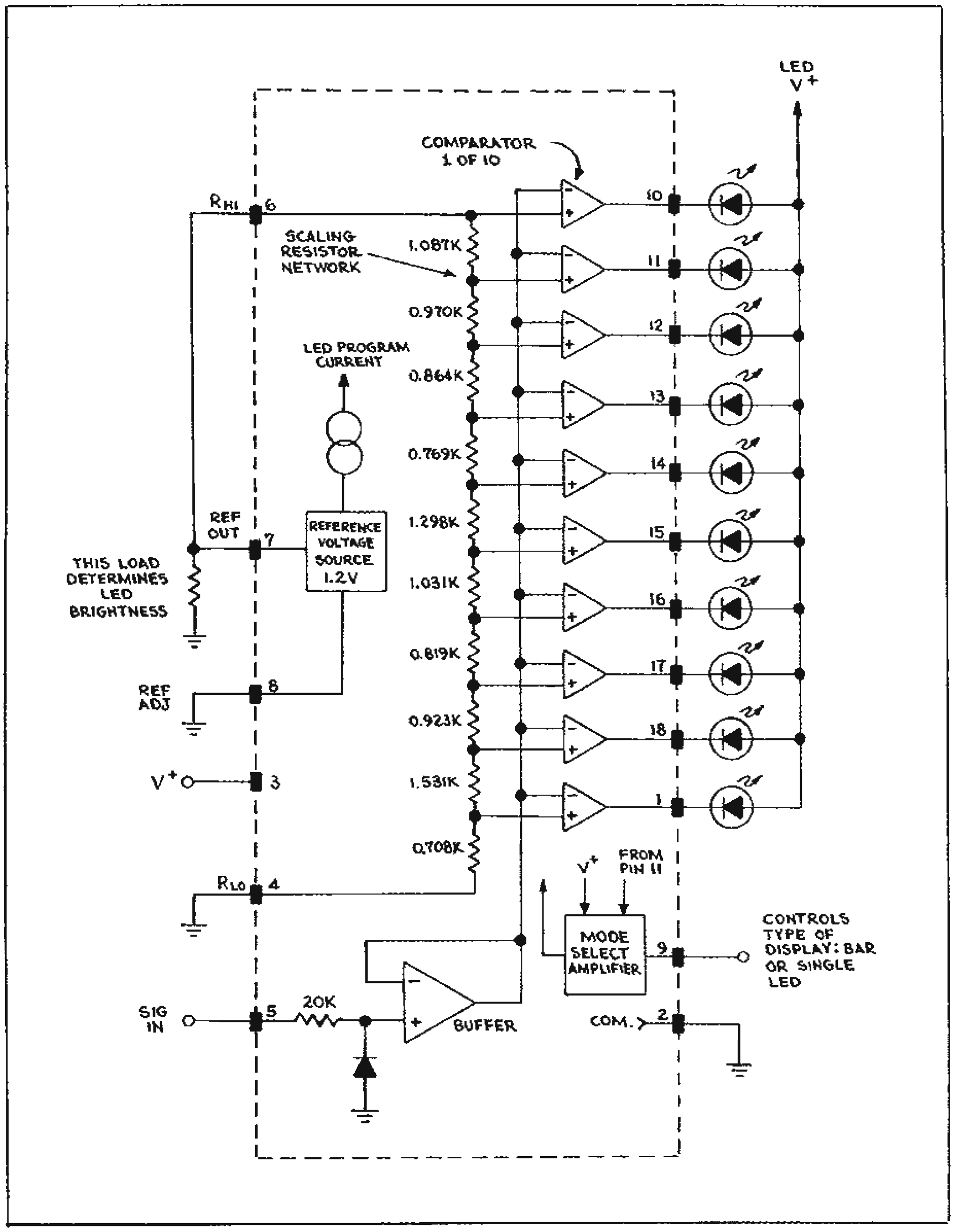
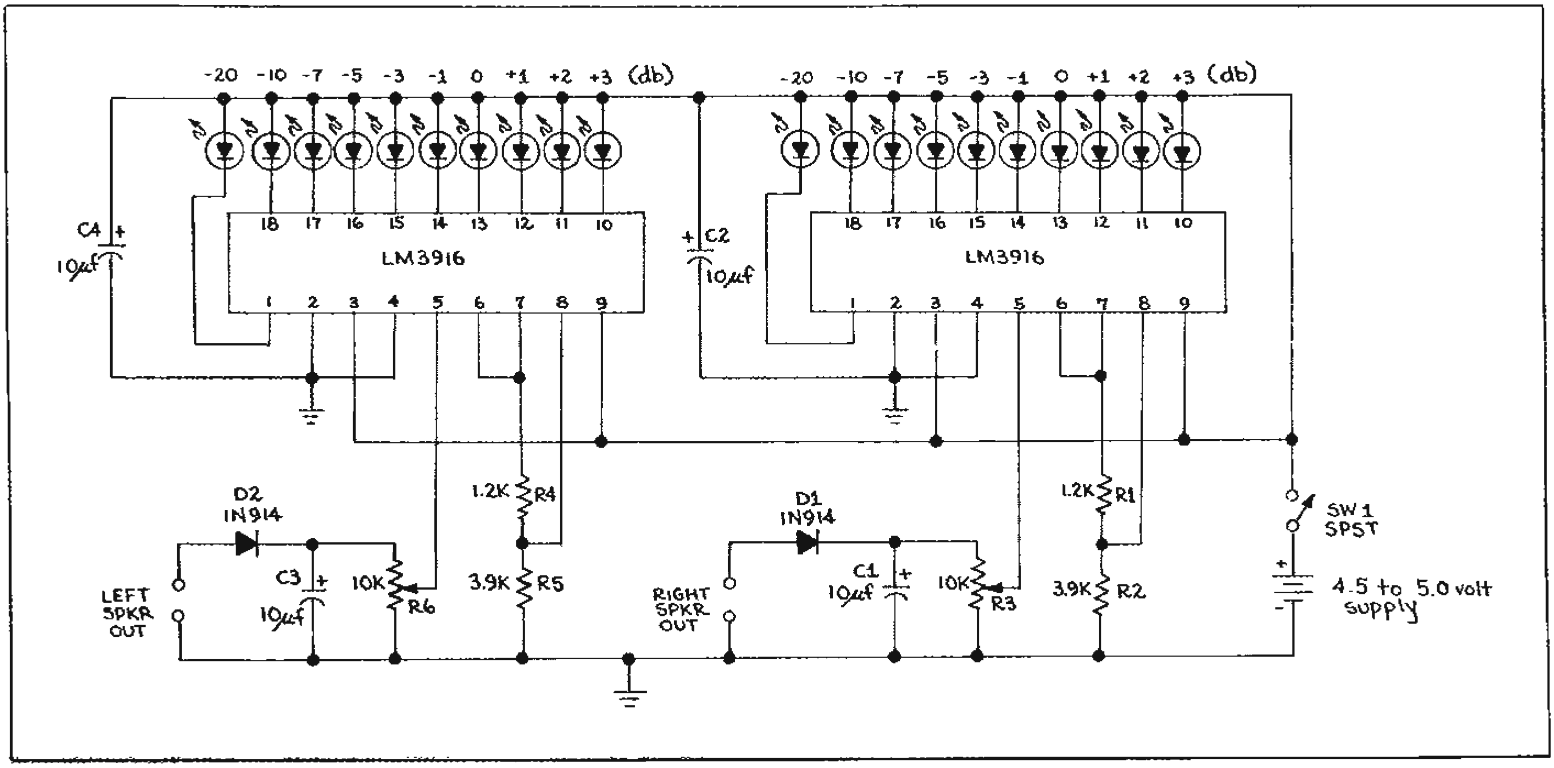


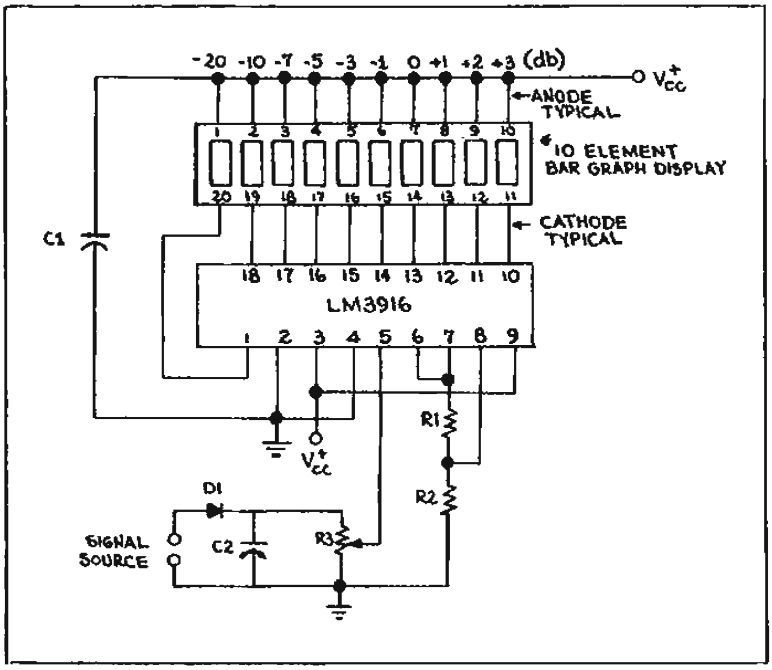
Fig. 1 — Block diagram of National Semiconductor's LM3916 dot-bar display driver, a monolithic integrated circuit.

Fig. 2 — Schematic for an LED stereo power meter based on the LM3916. Notes: C1 to C4 are 10-µ F, 35-volt tantalum capacitors; all resistors are ¼-watt, 5 to 10 percent tolerance; all LEDs are 10 milliamperes.



Schematic Drawings: Ronald G. Ajemian

Fig. 3—A 10-element bar graph display using the LM3916. The display comes as a 20-pin dual-in-line package in General Instrument's MV57164 or Litronix RBG-1000 in red, yellow, or green. Ten individual elements may be used with Hewlett-Packard's HLMP 0300/0301 series (5082-4670) rectangular LED, which is also available in red, yellow, or green.



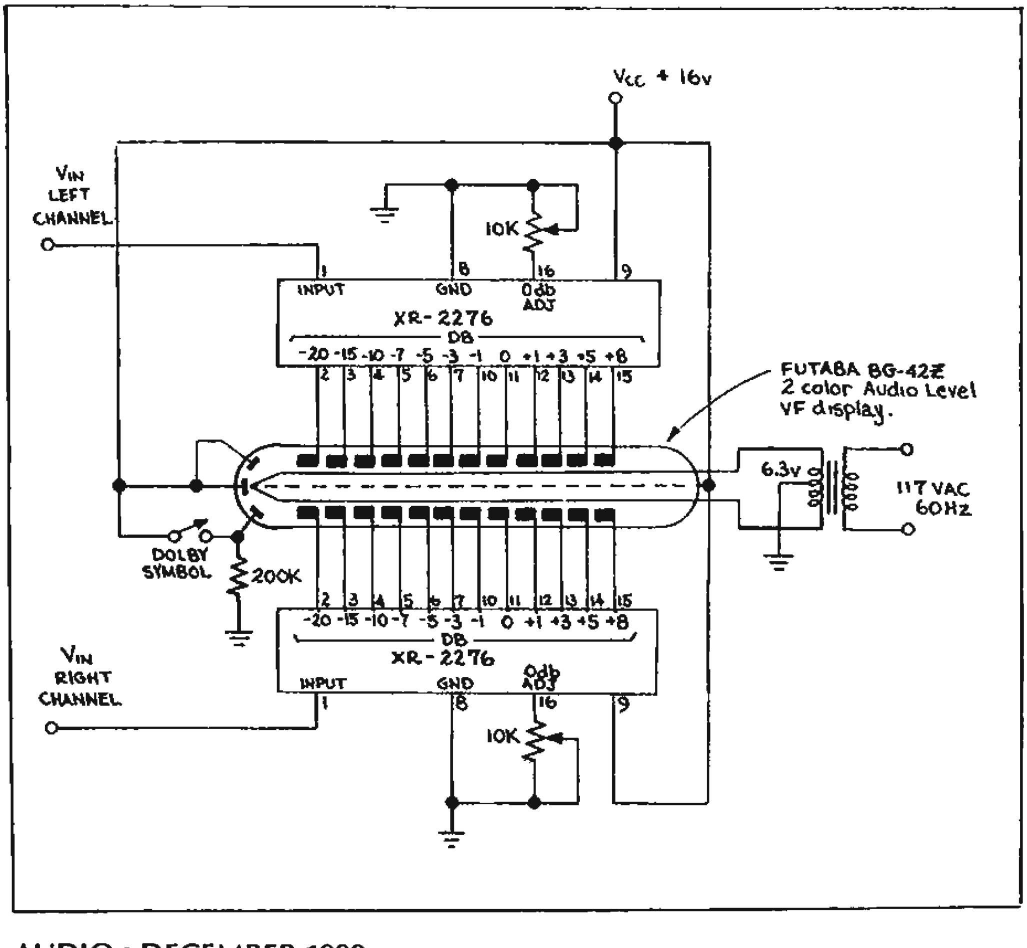


Fig. 4—Vacuum fluorescent level display with drive circuitry.

sists of a grid, plates, and a filament similar to a vacuum tube. VF meters contain fluorescent material, giving designers a choice of colors such as red, blue, green, and yellow. Most new VF displays have operating plate voltages of about 16 to 18 V, and filament voltages of 3 V rms at a current of approximately 100 mA.

A 12-point level detector integrated circuit, the XR-2276 shown in Fig. 4, was developed by EXAR Integrated Systems to drive VF displays directly. This circuit also works on the comparator principle but has 12 comparators, a bias network, and an input buffer amplifier.

The displays discussed here feature 10- and 12-point levels. More levels can be added simply by cascading two LM3916 circuits together to obtain increased resolution over a 28-dB range.

Many types of schemes using integrated circuits in electronic display meters are available from manufacturers. Those readers who wish to build audio power meters with LEDs or VFs may contact the suppliers listed below for data sheets.

Sources

National Semiconductor Corp.
2900 Semiconductor Dr.
Santa Clara, Cal. 95051
(LM3916 Data Sheet)

EXAR Integrated Systems, Inc.
750 Palomar Ave.
Sunnyvale, Cal. 94088
(XR-2276 Data Sheet)

Litronix, Inc.
19,000 Homestead Rd.
Cupertino, Cal. 95014
(RBG-1000 Data Sheet)

General Instrument Corp.
3400 Hillview Ave.
Palo Alto, Cal. 94304
(MV 57164 Data Sheet)

Hewlett-Packard Corp.
1501 Page Mill Rd.
Palo Alto, Cal. 94304
(HLMP 0300/0301 Data Sheet)

Futaba Corp. of America
555 W. Victoria St.
Compton, Cal. 90220
(BG-42 VF Data Sheet)

All That Data:

Tape Deck Frequency Response And Headroom

Howard A. Roberson

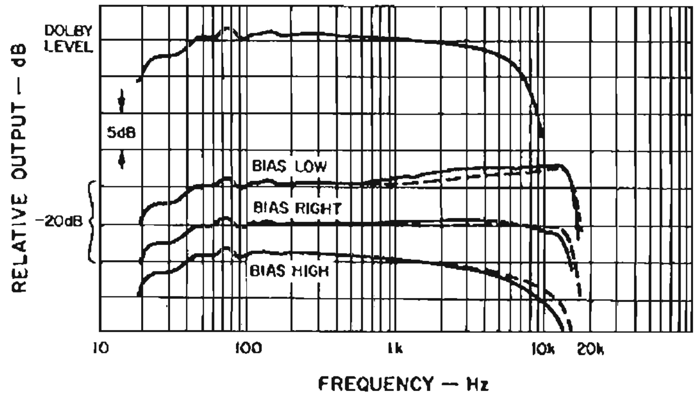
Whenever *Audio* reports on the results for testing a tape deck, we include a number of figures to help explain what the recorder's capabilities are. We usually have three graphs, or plots, that show the frequency response and headroom for three different tape types, usually a Type I (ferric), a Type II (CrO_2 type), and a Type IV (metal). Sometimes, a Type III (FeCr) is used in addition to, or in place of, one of the above.

Strictly speaking, what is commonly called "frequency response" would be more correctly labeled "amplitude response versus frequency." We are concerned with, and what our figures do show, is how the level, or amplitude, of the playback varies across the frequency band. Other high-fidelity components are rated by their frequency responses, of course, but a recorder's response is a long way from the extended, ruler-flat response of an amplifier. One of the goals of this article is to provide some guidelines for examining such graphs with greater perception.

It is best to compare the performance of one deck to another with the same signal or flux-level reference, and to use a reference easily identified by the user or reader. The Dolby standard flux level (200 nWb/m at 400 Hz) is used as the reference level of frequency response, as well as for other tests. Substantially all cassette decks have those little double-D symbols which pinpoint the record level that should gain Dolby flux levels. (It should be noted that because of equalization effects, the 200 nWb/m flux-level figure is correct for 400 Hz only.)

When frequency responses are run at the reference level, there will be a high-end dropping caused by any combination of several things: The tape itself, the design of the record head, the amount of equalization used in record — well, there's saturation and self-erasure mixed in here too. In any event, the plots at Dolby level are important since they show what the *high-level*, high-frequency limits are for the recorder/tape combination.

Fig. 1 — Typical frequency response curves of a cassette deck with and without (- - -) Dolby noise reduction, showing effects of three different bias settings.



Responses are also shown at a level 20 dB lower, where frequency responses are usually specified by makers. This does not mean, of course, that with the record level 20 dB below Dolby level, we will be certain to get that ruler-flat line. There will be both lower and upper frequency limits, with electrical, mechanical and magnetic factors all involved. The results at this level are affected, perhaps greatly, by the closeness of match between the bias current provided by the machine and the bias needs of the tape formulation. The performance in Dolby mode is affected by bias discrepancies, as well as any record sensitivity offsets. Because most users operate their decks in Dolby mode the majority of the time, the responses with NR are considered the normal ones. Responses without Dolby are usually also shown, particularly when there is a noticeable difference.

Examination of the figure with this article reveals that there is a plot at Dolby level and three plots at -20 dB, these last with three different bias conditions. The test signal was a sine wave which was swept from 20 Hz to 20 kHz. By this means, each and every point is a true data point. Discrete-point data are required if it is necessary to go beyond these limits in the usual frequency response curve.

Let's start with a critical look at the

Dolby-level response. Our reference frequency for plotting is 1 kHz, and the trace crosses the reference-level line there. There is a slight rise as we go down to 100 Hz, below which there are some bumps from head-contour effects. The response falls below 40 Hz, but it is not down 3 dB until just above 20 Hz. It would be ideal if there were no head-contour effects, and it is desirable that deviations from 40 Hz up be less than ± 1.5 dB. The significance of the lower limit is twofold: How low in frequency do your music sources go, and are the levels high in this region? Sometimes, but not in the case shown, recording at Dolby level causes a further dropping of the low end, relative to the -20 dB results. This is indicative of low-frequency overload or saturation and shows there is a definite level limit.

It's best if the response continues flat above 1 kHz, but at some point there will be a headroom limitation, and the response will roll off. In the figure, response is down 3 dB at about 6.3 kHz, which would be limiting in any attempt to record at high levels any music with considerable high-frequency content. The distortion also rises rapidly with level in the roll-off region, so it's not just a case of losing a little signal level. We would certainly like to see the response quite flat from 100 Hz to close to 10 kHz, which is possible with many decks. A general tilt, such as is evident from 70 Hz to say 4 kHz, might make the sound a little extra bassy.

Now let's take a look at the group of three sets of responses at -20 dB. The labels show what the deck's relative bias levels would be for a particular tape to get the results plotted. In each case, the trace without Dolby NR is shown dashed. We can see that for *each* bias setting, the plot *with* NR is not quite as flat as *without*, and that when the bias matches the tape, the deviations are at a minimum. (The plot at Dolby level includes traces in both modes; the responses were exactly the same.)

At -20 dB the deck is not operating in a condition of high-frequency overload, so there cannot be a basis for excusing a general roll-off, such as shown in the high-bias case. At the same time, excessive high-end response can be much too bright, with a higher distortion level. So, tape formulations that match the deck are selected for testing. Immediately, we can say that we would like to see a response that is flat within a dB or so, from at least 100 Hz to 10 kHz, and extensions down to 40 Hz or so and up to 13 kHz or more would be nice to have. A very important region is from about 700 Hz to about 6 kHz where Dolby mistracking can cause either excessive presence, because of a boost in this region, or a dull, remote sound because of a saddle-like droop.

In summary, look at the responses at both Dolby level and at -20 dB over the range from about 40 Hz to 10 kHz or so. In both cases, flatness is the goal. Also give attention to head bumps at the low end, headroom limitations at the higher level, and Dolby-mode boost or saddle deviations at the lower level. See how far down in frequency the responses actually extend and whether there is additional roll-off at Dolby level. The extension of flat response above 13 kHz at -20 dB has some value, but this is not as significant as the other times, in most cases.

OPEN-REEL

THE MECHANISM OF MAGNETIC TAPE ERASURE

PETER VOGELGESANG*

Certainly one of the greatest virtues of magnetic tape as a recording medium is the ease with which old information can be erased to make room for new information. Compared to other kinds of recordable media, such as photographic film which is chemically processed and permanently written, magnetic tape can be used thousands of times through the faculty of erasure. Although the process of erasing tape is performed automatically in recording systems and manually with the aid of bulk demagnetizers, the precise mechanism of erasure within a tape is not commonly understood. Perhaps this lack of understanding results from the apparent simplicity of the erasing process — a simplicity which does not challenge the interest of those who use the process. But tape erasure is not as simple as it appears, and perhaps the following description of the erasure mechanism will both interest and enlighten.

To understand tape erasure one must first understand the composition and behavior of the magnetic material used in tape. The most commonly used material is the gamma form of iron oxide, or gamma Fe_2O_3. This material exists as tiny, needle-shaped particles having an average length of 0.4 micron, and is shown magnified 42,000 times in the photomicrograph of Fig. 1. Although other magnetic materials such as chromium dioxide (CrO_2), cobalt modified iron oxide, and metallic materials may have different chemical compositions and physical structures, they respond magnetically in a manner similar to gamma iron oxide.

Each iron oxide particle is a permanent magnet containing only a single magnetic domain. The single-domain structure of the particles dictates that each particle is forever a permanent magnet which cannot be demagnetized. If subjected to an external magnetic field of sufficient intensity, a particle will reverse the polarity of the field it generates, but the reversed field will not be

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Manager, Advanced Recording Technology, Magnetic A/V Products, 3M Company, St. Paul, Minn.

Photo: Susanne Buckler

RECORDERS

FOCUS ON HEAD DEMAGNETIZATION

HERMAN BURSTEIN

Until recently I accepted without question the widely propagated dictum that tape heads must be periodically demagnetized — roughly after about 8 to 16 hours of use. My acceptance became embedded in concrete several years ago, but cracks have developed in this concrete as a result of several letters from "Tape Guide" readers and some inquiries I have made.

At this point perhaps a brief review would be appropriate of the commonly given reasons for demagnetizing tape heads; in other words, The Gospel: The waveforms of most source material are asymmetrical, thus in effect contain a d.c. component, which tends to magnetize the heads. Distortion in the bias waveform has a similar effect. Magnetized heads act as erasing devices, particularly as recorded wavelengths on the tape grow shorter, i.e. as frequency rises at a given tape speed. Furthermore, magnetized heads translate physical and magnetic irregularities in the tape into noise. Altogether, it is claimed, magnetized heads produce treble loss and noise.

Several months ago I received a challenge to this conventional wisdom from Ramon Valdes in Miami Lakes, Florida. He wrote: "Two years ago I did a test to see how important tape recorder head magnetization is. I took a TEAC 7010 GSL tape deck with automatic reverse and optimized it for 3M 206 tape; a test tape was then made with tones from 20 Hz to 15 kHz. I then took a 10½-inch reel of 206 tape and proceeded to record at 7½ ips three hours at 20 Hz, three hours at 1 kHz, and 3 hours at 15 kHz in that sequence for 100 hours. I cleaned the heads every three hours but did not demagnetize them. After 100 hours, I played back the original test tape and also rerecorded the test tones. All parameters (distortion, signal strength) were the same as before the 100 hours of recording. I went another 100 hours before my patience ran out — the results were the same."

My reply to Mr. Valdes was essentially as follows: "If I understand correctly, at 7½ ips you recorded and played tones of 20

Continued on page 30

greater or smaller in magnitude than the original field. Thus, each particle can be thought of as the magnetic equivalent of an electronic flip-flop which has two stable states. Just as a flip-flop can be switched into one of two saturated states, energy applied to a single-domain magnetic particle can cause it to reverse polarization virtually instantaneously but cannot cause it to generate a magnetic field of variable intensity within the particle.

The switching characteristic of a particle is illustrated graphically in Fig. 2. A magnetic field which varies sinusoidally in amplitude is applied to the particle so that the direction of the field is parallel to the long axis of the particle. (A different result is obtained if the applied field is at right angles to the axis.) Particles in most magnetic tapes are aligned along the length of the tape by immersing the tape in a longitudinal field while the magnetic coating is still fluid. Once dried, particles in the coating remain physically aligned in that direction.

As the applied field in Fig. 2 increases from zero, a level is reached (point a) where the coercivity of the particle is equalled, and at this point the field within the particle instantaneously reverses. The particle will not again switch until the applied field has reversed and reached the equal but opposite magnitude. The hysteresis waveform of a single particle will be recognized by electronic engineers as similar to the graphs used to depict the behavior of electronic flip-flops and Schmitt triggers.

Each area of a magnetic tape which is uniquely magnetized during recording contains thousands of magnetic particles throughout the width and depth of the recorded track. The magnetic field emanating from a recorded area is the sum of the fields produced by these individual particles. A magnetic tape is said to be saturated when all the particles are switched with the same magnetic polarity. A discrete area of tape thus saturated will produce the maximum external magnetic field of which the tape is capable. In the erased condition, one-half of the magnetic particles are switched with one polarity while the remaining half retains the opposite polarity, and since the particles composing these halves are closely intermixed, their external fields cancel, producing zero external field from the tape.

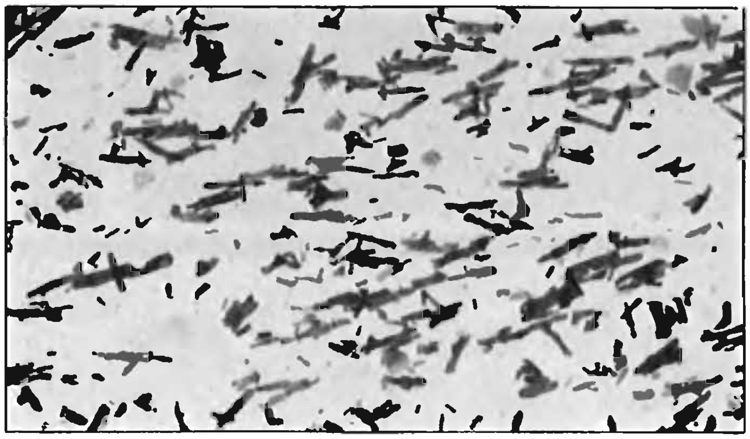

Fig. 1 — Photomicrograph of iron oxide particles used in magnetic tape, 42,000 X magnification. The particles magnetic properties are controlled largely by their shape, and their magnetic characteristics from stimulation fields which are parallel to the particles are different from ones which are perpendicular. The long dimension of the particles is aligned parallel to the recording direction in most tapes.

Varying levels of tape magnetization are produced during the recording process by causing the particles to be switched to the opposite polarity in proportions varying between the saturated (100% switched in one direction) and erased (50%-50%) states. It is seen, then, that the analog recording process in the strictest sense is really not analog at all but is a binary process in which the magnetic outputs of thousands of flip-flop elements are summed in the pole pieces of the reproducing transducer.

It is also apparent from the foregoing that the magnetic transfer characteristic of a tape would not permit the recording of an analog signal if the coercivities of all particles in the tape were identical. A tape made with such magnetic material could exist only in alternately saturated states of two polarities, and would have a hysteresis curve identical to that shown in Fig. 2. (Note: In reality an analog signal could be recorded on a tape having a rectangular hysteresis curve because of the three-dimensional geometry of the magnetic field produced by a recording transducer [1]. In this instance, varying magnetization of a tape would be achieved by modulating the depth of magnetization within the thickness of the magnetic coating. The recording characteristics of such a tape, of course, would be quite different from conventional tapes.) Fortunately, the millions of particles that exist in even a small section of tape have coercivities which cover a broad range. The population of coercivities is similar to the distribution of a normal curve, as shown in Fig. 3. A value of coercivity assigned to a specific tape refers to the average coercivity of the oxide particles, and in most instances, this value will be the peak of the distribution curve.

The task of erasing a magnetic tape can be defined as the necessity of establishing a condition whereby all external fields are zero as a result of a one-to-one ratio of oppositely polarized, closely intermixed magnetic particles [2]. Most explanations of a.c. erasure utilize the expedient of a family of hysteresis curves to show how the mechanism of

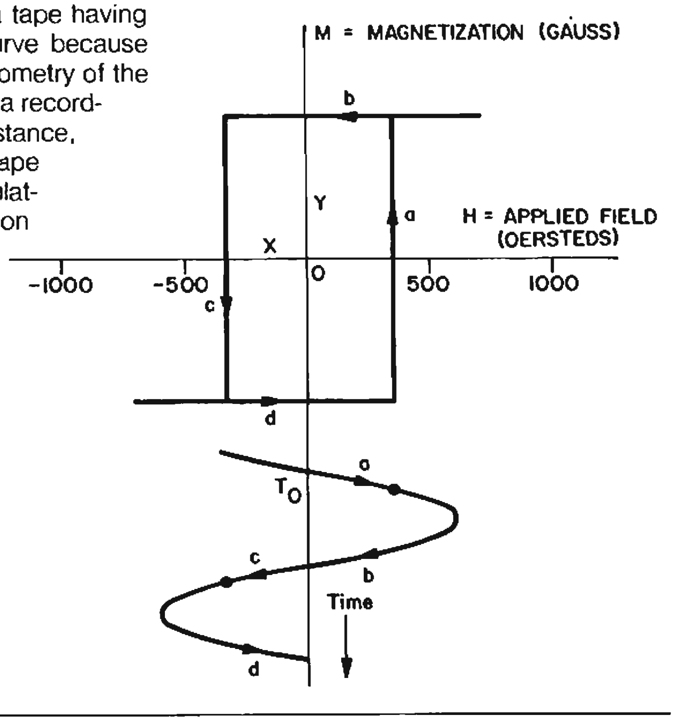

Fig. 2 — Magnetization characteristics of a single-domain particle.

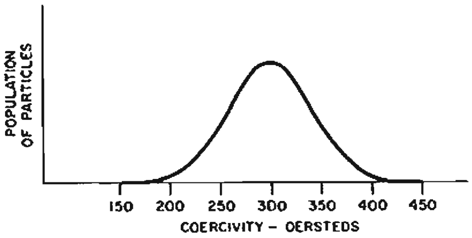

Fig. 3 — Distribution of coercivities in a large group of particles.

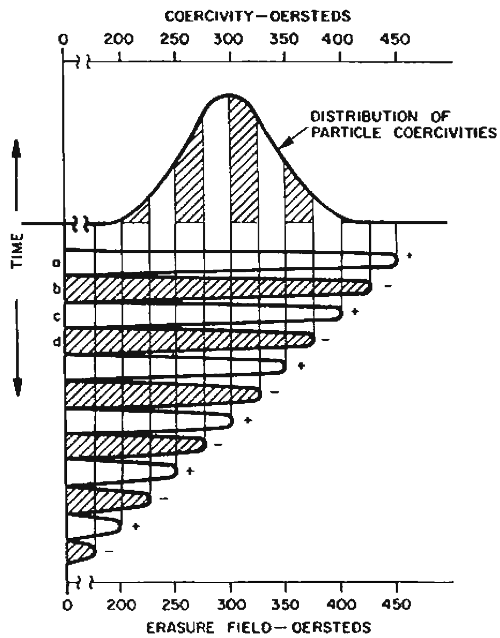

Fig. 4 — Relationship of a magnetic material distribution curve to an erasure field.

erasure works [3]. An alternative explanation which may be more conducive to an understanding of the physical events which occur in the erasure process is presented with the help of Fig. 4. The uppermost curve represents the distribution of coercivities of magnetic particles used in a tape, where the greatest population occurs at 300 oersteds (the specified coercivity of the tape). The waveform below the distribution curve represents an applied a.c. field which varies sinusoidally and which diminishes gradually from a high magnitude to zero. This diminishing field is the kind of field that a tape will experience as a bulk degausser is slowly withdrawn from its vicinity or as a particular area of the tape moves away from the erase head of a recorder. The shaded half-cycles represent a field of negative polarity and for the purpose of illustration are folded over from the negative side of the zero line. The unshaded half-cycle portions represent the positive polarity.

Starting with half-cycle a, the peak applied field is of sufficient magnitude that all particles are switched in the positive direction, and at that point the total magnetic material is saturated positively. Half-cycle b also saturates the tape, but with a negative polarity. Half-cycle c again reverses most of the magnetic particles, but leaves a small percentage in the negative polarity because the decreased magnitude of the field is too low to reach the highest coercivity particles. The largest part of the particles are yet again reversed by half-cycle d, but this time a greater percentage are left in the positive polarity because of the diminishing erasure field. This process continues until the erasure field reaches zero. At this point note that approximately half of the particles remain switched positive and the other half are switched negative; in other words, the sum of the fields produced by all the particles is zero. Note also that the probability of arriving at a one-to-one ratio of polarities is greater as the number of half-cycles is increased (longer diminish time). This analysis shows why a tape must be withdrawn slowly from an erasure field to insure full erasure. It also shows why the field of a bulk degausser cannot be interrupted until the tape being erased is outside the influence of the field. To obtain complete erasure, the applied field must be greater in magnitude than the coercivity of the highest coercivity particles in the tape. An erasure field which is merely equal to the specified coercivity of a tape will not totally erase it; the field will switch only those particles with coercivities in the lower half of the distribution curve, while those particles in the upper half will be unaffected.

Of increasing concern today is the erasure requirement of the so-called "high-energy" tapes which have coercivities considerably higher than gamma iron oxide. As a general rule, the width of the distribution curve of these high-coercivity materials increases in proportion to the increase in specified coercivity. Consequently, a tape which has a specified coercivity twice as great as another will require an erasure field of double the magnitude.

The measured distribution of a 625-oersted magnetic tape is shown in Fig. 5, and it can be seen from this curve that a significant portion of the particles have a coercivity above 1,000 oersteds. The signal which remains after a tape has been erased to a level 60 dB below the maximum output level of a tape may be stored by only one-tenth of one percent of the magnetic material in the tape, and, of course, that small portion of the material will be at the uppermost end of the distribution curve. The tape represented by the curve of Fig. 5 requires an erasure field intensity of 2,000 oersteds to achieve this level of erasure.

The peak field requirement for a bulk degausser is even greater than that of an erase head, because in most bulk degausser configurations the direction of the erasing field is at right angles to the direction of tape orientation. (The excep-

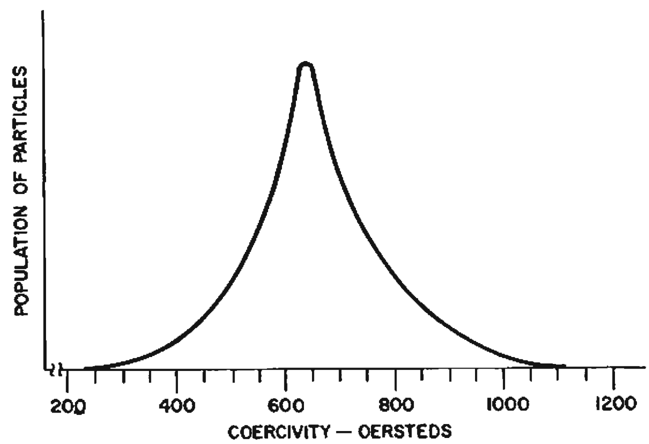

Fig. 5 — Measured distribution of a high-energy tape having a coercivity of 650 oersteds.

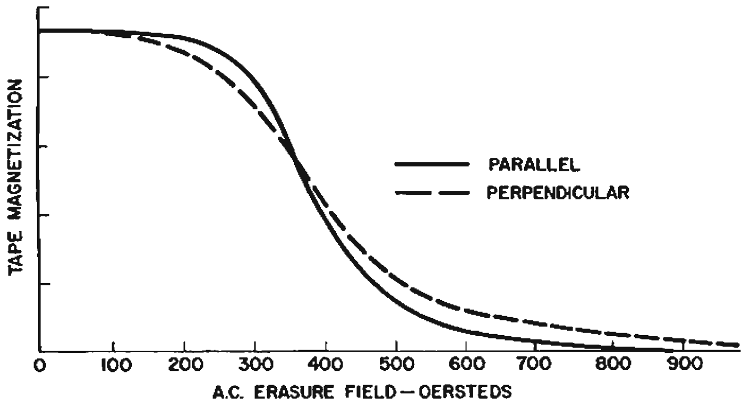

Fig. 6 — Erasure curves for a gamma iron oxide tape using parallel and perpendicular erasing fields. The effective distribution of coercivities is broadened when the applied field is perpendicular to the direction of orientation, substantially increasing the field strength needed for erasure.

tion, of course, is quadruplex videotape, where orientation is across the width of the tape rather than along its length.)

Erasing with a perpendicular field effectively broadens the distribution of the magnetic particles. Figure 6 shows demagnetization curves for a gamma iron oxide tape in which the erasure field is first parallel with the direction of orientation and then perpendicular to it. Note that erasure starts taking place at a lower magnitude of erasure field for the perpendicular case, but that more than 200-oersted greater magnitude is needed to achieve maximum erasure equivalent to the parallel case.

Since bulk degaussers almost invariably use a magnetic field which penetrates the flanges of a reel or the wall of a cassette, the most intense component of the field is perpendicular to the tape. Erase heads, on the other hand, have magnetic gaps which generate intense parallel as well as perpendicular erasure fields and consequently are somewhat more efficient than bulk degaussers. Also, the erase head is in contact with the tape, whereas the bulk degausser must penetrate the thickness of the tape container. While a magnetic field experiences no difficulty in penetrating plastic or cardboard material, the physical separation caused by the container can greatly reduce field intensity. In selecting a degausser to erase a particular tape product, or in the design of degaussers and erase heads, the principal parameters which must be considered are:

1. The specified coercivity and distribution of the magnetic material involved.
2. The direction of the erasure field relative to the orientation of the tape.
3. Separation of the tape from the erase field (in the case of bulk degaussers the field must penetrate the full thickness of the tape pack).
4. The level of erasure which is required.

These parameters will be translated into an erase field intensity.

Most purchasers of bulk degaussers do not have the instruments needed to measure tape distribution or degausser field intensity and so must rely on the claims of manufacturers relative to the suitability of a specific product. Perhaps the best means of determining suitability is to erase a saturated recording with the degausser under consideration and then measure the magnitude of any remaining signal.

The discussion of tape erasure to this point has dealt only with a.c. erasing fields, which are commonly generated by erase heads used in recorders and by bulk degaussers. In terms of ridding a tape of a recorded signal, drawing the tape over the end of a permanent magnet will erase a signal as effectively as any other means. Permanent magnet erasure was used in early recording systems for the sake of simplicity, but it has been abandoned for years in favor of a.c. erasure. Only in a very few applications of magnetic recording is permanent magnet erasure still used.

D.c. erasure was abandoned to minimize the noise which is generated by tape erased in this manner. D.c magnetization of a tape will cause any irregularities in tape surface and thickness to generate fields which appear in the reproducing transducer as noise. Since the field generated by a given area of tape is the sum of the fields of all the particles within that area, it can be seen that localized concentrations or rarefactions of oxide will produce different sums when all particles are polarized in the same direction. The differences in these sums constitute a varying external magnetic field and a noise signal.

Great care is taken in the tape manufacturing process to produce uniformly dispersed oxide in the magnetic coating, but slight density irregularities are unavoidable. These irregularities, combined with slight thickness variations of the coating and slight surface irregularities, produce unacceptable noise when a tape is magnetically saturated in order to erase a signal. (See Fig. 7.)

Strong evidence that d.c. erasure noise is caused by physical irregularities in tape is provided by the simple test of comparing the output noise waveforms from the same area of a tape on two successive passes where polarization of the d.c. erasure field has been reversed. One would expect the noise waveforms to be identical but inverted, as indeed they are in Fig. 8.

A correlation coefficient between two noise waveforms was determined by measuring the waveform amplitudes at 50 different sample points spaced at

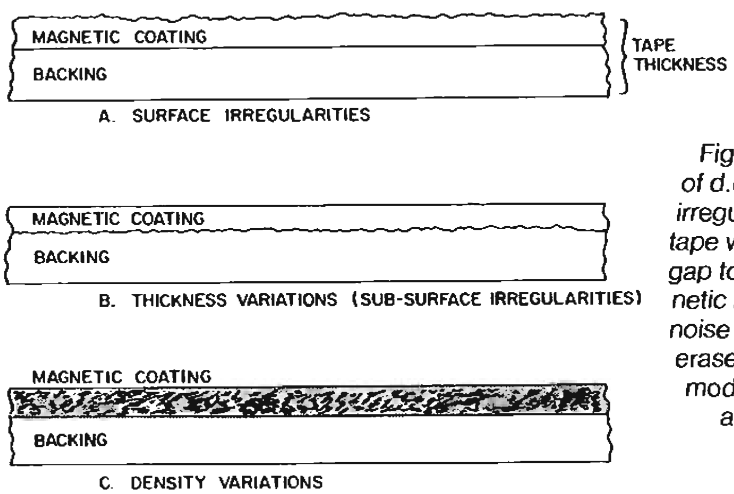

Fig. 7 — Three sources of d.c. erasure noise. Any irregularities in a magnetic tape which cause the head gap to "see" varying magnetic material will generate noise when the tape is d.c. erased and will also cause modulation noise from an a.c. erased tape when it is recorded with an analog signal.

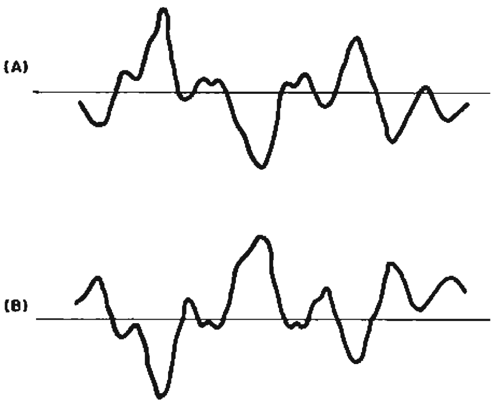

Fig. 8 — Noise waveforms generated as the result of reverse magnetization.

equal time intervals and then mathematically computing their similarity. When the tape was magnetized by passing it over a recording head excited with direct current, correlation of the two waveforms was -0.89, where ideal correlation would be -0.9 (rather than -1.0) because of amplifier and system noise. This high degree of correlation is convincing evidence that noise output of a d.c. saturated tape results from permanent physical features of the tape.

The same density variations that cause d.c. noise also give rise to modulation noise, that is, noise which occurs in the presence of a recorded signal. Imagine, for example, that a tape has been a.c. erased and then recorded with a long-wavelength, high-level audio signal. Insofar as the segment of tape which contains a half-wave of the signal is concerned, magnetization within this segment is no different than magnetization which might have occurred from d.c. erasure. Once again density variations produce external fields, but in this case the noise which is generated has a frequency relationship to the recorded signal and is termed modulation noise. D.c. erasure noise and modulation noise thus have substantially the same origin but are manifested in different ways. While d.c. erasure noise can be virtually eliminated by using a.c. erasure fields, modulation noise is minimized only by judicious design and manufacturing of magnetic tape.

In conclusion, the magnetic recording process is almost always preceded by an erasing process, either by bulk degaussing or by magnetic-head erasure. Erasure is a fundamental step in achieving high-quality recordings. The simplicity with which erasure is performed belies the complexity of the process. It is hoped that the foregoing has created a little higher regard for this seemingly mundane procedure. A

Acknowledgment

The author wishes to acknowledge the work of E.F. Wollack of the Data Recording Products Division of 3M Company, whose research of the recording process contributed significantly to the information presented.

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DEMAGNETIZATION

Continued from page 25

1,000 and 15,000 Hz, each 3 hours at a time, for a total of 200 hours and after such use found no deterioration in the ability of the heads to record and reproduce a 15,000-Hz tone relative to 1,000 Hz.

"However, I have several problems with your findings. The first is that you used a speed of 7½ ips. A number of adverse magnetic tape phenomena, including erasure due to magnetized heads, are wavelength effects. That is, they become more severe as tape speed decreases. I wonder what you would have found if the same test were conducted at a lower speed, say the cassette speed of 1⅞ ips.

"Second, gradual magnetization of heads is attributed to the asymmetric nature of the typical audio signal. For your test, you probably used very pure sine waves devoid of significant asymmetry.

"Further, your letter says nothing about noise. In looking for head magnetization effects, one should look not only for deterioration of high-frequency response but also for increase in noise."

Mr. Valdes shortly replied: "Your first paragraph is correct; that is the way I ran the test.

"Concerning your second paragraph, I only ran a 7½-ips test since that is the (minimum) speed I consider useful for serious recording.

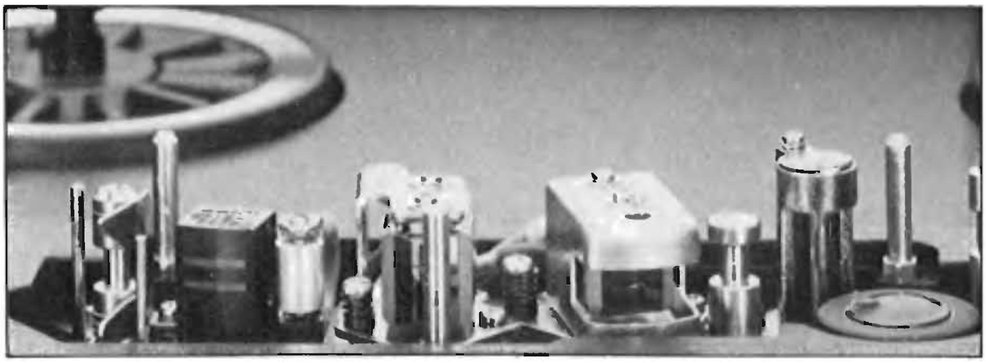
"Concerning the third paragraph, I did use very pure sine waves because that was then the only way I had to measure distortion and frequency response. However, now I have a method of recording precise square waves with different rise and fall times, resulting in asymmetrical waveforms. I can view them with a spectrum analyzer and photograph the screen. Let me know if you think making before and after comparisons, following 200 hours of deck operation, would be a valid test. Using a musical waveform would show whether the heads become magnetized, but only gross distortion would be noticed.

"Concerning your fourth paragraph, the noise increased after 45 hours, but it was due solely to tape erasure by the erase head. When the tape was bulk-erased, the noise disappeared."

In turn I wrote: "It would be interesting to see if asymmetrical waveforms recorded over a long period do raise the level of magnetization of a tape head, and to compare the reproduced waveforms at the end and beginning of the test."

As of the present writing, it is several months since my last letter to Mr. Valdes. This may well be too short a period to permit him to give an account of the results of running his tests with asymmetrical square waves. However, in view of continuing reader interest in the subject of head demagnetization over the years, I thought it best not to delay the present article by waiting further for this account.

About the same time that I received Mr. Valdes' first letter, another challenging letter came from Henry B. Ruh of Owen Valley Broadcasters, Inc., Ellettsville, Indiana: ". . . From my 15 years of broadcast and audio experience, let me state that if your deck needs to be demagnetized, you probably need a new deck! Back in the days when a perma-



nent magnet was used for erasure . . . the continuous stream of one-way magnetized tape particles would over a period of time tend to magnetize items downstream from the erase magnet. Thus, you had to demagnetize the heads, guides, capstans . . . . But when the cost of a high-power a.c. erase system was reduced to a reasonable level and permanent magnet erase heads were eliminated, so was the problem of magnetized heads.

"Although I have checked hundreds of times on many different makes and models of tape machines, I have never found on a modern machine any residual magnetism which could in any way erase any portion of the signal on the tape . . . .

"In conclusion, you can throw your head demagnetizers away and forget the problem. While in the service field, I encountered more magnetized tape heads caused by improper use of demagnetizers than I found blown fuses."

Comments By Several Authorities

After hearing from Messrs. Valdes and Ruh, I solicited views on the matter from several knowledgeable persons in the field of tape recording. While they do not necessarily constitute a representative sample, their positions lend authority and interest to their comments.

Delos A. Eilers, Technical Service Supervisor at 3M Co., St. Paul, Minnesota, wrote: "Head demagnetizing at regular intervals is a good preventive maintenance practice. What that interval should be is difficult to define.

"It is true that improper head erasure can cause more problems than if one didn't demagnetize the heads. If the demagnetizer isn't gradually pulled away from the heads, you won't be showing the head a gradually decaying magnetic field. You will leave the head magnetized. A slowly decaying field is essential for good demagnetization.

"The magnetization that a head can build up over a period of time is not from the earth's field but more from accidental exposure to magnetized particles or build-up of residual magnetization from nonsymmetrical bias or program signals. Thus, we recommend that heads be demagnetized after every 10 to 15 hours of recording."

The chief engineer of a well-known manufacturer of tape decks (who prefers not to be identified) indicated that the need for head demagnetization appears to vary from one deck to another. To illustrate, he cited his ownership of two fine decks, one requiring frequent demagnetization in order to prevent obvious treble loss, and the other apparently never requiring it. To play it safe, he periodically demagnetizes both.

Don Eger, an engineer with Crown International, Inc., gave the following remarks: "Our experience with tape recording heads during the past two decades does not support occasional claims that tape head demagnetization is unnecessary or harmful.

"The effect of using a head that has become magnetized is loss of short wavelength material from the tape, as such material is erased by the magnetism of the head. If the tape is continually exposed to a magnetized head, the recording will lose most of its high-frequency content.

"Occasional dissatisfaction with the demagnetizing process experienced by some recording enthusiasts may be a result of degaussing the head too quickly and/or improperly. If the degaussing coil is removed slowly from the head, it will demagnetize the head properly."

Finally, we quote Henry C. Pollak, an electronics engineer who has spent much of his career designing professional tape equipment and servicing consumer audio equipment, including tape decks. At one time he owned Western Radio Lab of Sunnyvale, California, an audio service shop. (It may be added that he is co-author, with me, of the book *Elements of Tape Recorder Circuits*, which was published in 1957.) . He stated:

"My experience with demagnetizing play heads has been that I have never found any benefit. It's like washing before eating. My mother said to do it, so I wouldn't think of not doing it, although now I'm certain that nothing adverse would happen if I neglected my upbringing. At Western Radio Lab, as a matter of course, we demagnetized all decks that we serviced. The amount of magnetization the play head receives from the tape oxide is extremely small. The narrowness of the hysteresis loop of the play head material is so small that the remanence of the head would be trivial. And, anyway, the next playing of a loud passage would restore the head to its previous condition.

"I would worry more about the harm a head demagnetizer can do (if carelessly used): Bending flimsy tape guides, cracking plastic covers, or scratching head surfaces. Most demagnetizers are too bulky to do a proper job on the cassette heads. At Western Radio Lab, we covered the tips of the demagnetizer with mylar tape to prevent scratching the heads.

"There is an argument for preventing d.c. from flowing through any head — erase, record, or play. And, to minimize noise, it is important to have minimum distortion in the bias and erase waveform."

Summary

From the foregoing, it doesn't appear that a definitive case can be made for or against head demagnetization. However, some conclusions can be drawn:

1. Demagnetization of the heads is advisable as a *precautionary* tactic, for some decks might need it.

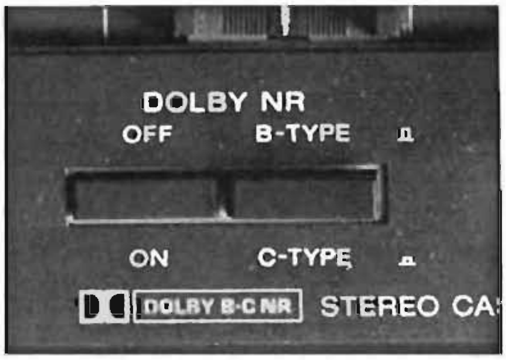
2. If used, a demagnetizer should be operated with proper care so as not to leave the heads in a more magnetized condition than previously. The device should be turned on while several feet from the heads, brought slowly to the heads, moved in slow circular fashion while near the heads, removed gradually, and turned off at a distance of several feet. The tape deck should have its power off during this process.

3. If the demagnetizer is used, care should be taken to avoid physical harm, such as scratching the tape heads with the tips of the demagnetizer, bending any of the tape guides, or damaging a sensitive record-level meter by subjecting it to an excessively strong magnetic field.

The "Tape Guide" will be pleased to hear further from readers who can supply authoritative information on the pros and cons and cautions of tape head demagnetization. Assuming receipt of such items, the subject will be continued.

JOSEPH HULL*

DOLBY C-TYPE NOISE REDUCTION



The Dolby B-type noise-reduction system was introduced to the consumer in 1968; the first cassette decks incorporating it were introduced in 1970. Today, tens of millions of consumer audio products with Dolby noise reduction have been manufactured, and cassette recordings made with the system probably number in the hundreds of millions.

The adoption by the consumer electronics industry of a single noise-reduction system, with characteristics maintained throughout the world by means of a comprehensive licensing program, has ensured compatibility among recordings and recorders. Thus, any recording made with Dolby B-type noise reduction can be played on any machine equipped with Dolby B circuitry, regardless of manufacturer or country of origin. The effectiveness of the system is such that today's high-performance cassette recorders, using good tape formulations, can record the vast majority of available program material without significant degradation.

However, over the past two years or so, there has been a growing desire for a system giving even more noise reduction than is provided by Dolby B. Audiophile disc releases with wide dynamic range, a continuing reduction in the cost of amplifier power, loudspeaker systems with increased accuracy and robustness, the use of equalizers in elaborate systems, and in some markets a serious interest in high-quality live recording on cassettes are all indications of a need for more noise reduction, at least at the higher end of the marketplace. While it remains to be seen just how deeply that need goes, it seemed appropriate in late 1980 for Dolby Laboratories to offer its licensees the option to use a new system, Dolby C-type noise reduction, to supplement the standard B-type system. Dolby C provides 20 dB of noise reduction above about 1 kHz compared with the Dolby B-type system's 10 dB of noise reduction above about 4 kHz. Dolby C-type noise reduction has been designed specifically to deal with the noise spectrum and other parameters particular to slow-speed consumer tape recording formats such as the standard compact cassette and the microcassette.

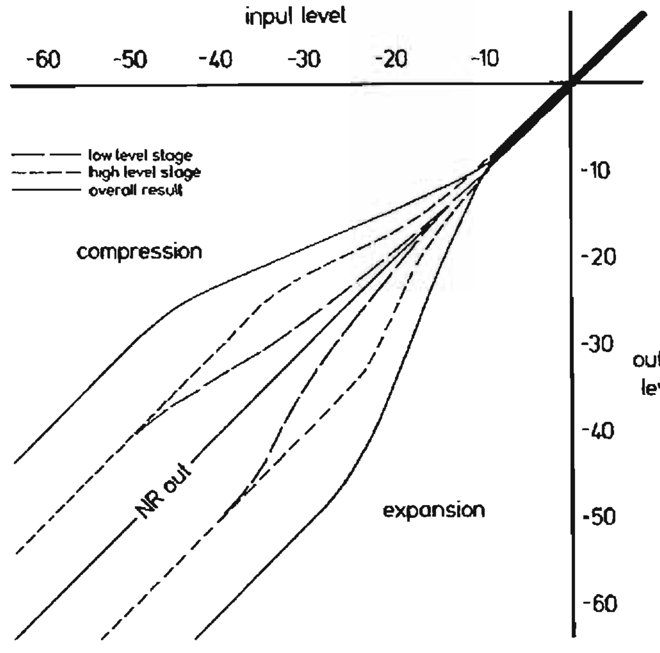
Objectives for Dolby C-Type Noise Reduction

The goal for a new Dolby system was not simply and arbitrarily "more noise reduction." A number of devices, such as wide-band companders, have long been available which provide more noise reduction than Dolby B under some signal conditions, most notably in the absence of signals altogether. However, these devices also introduce such side effects as noise modulation and overshoot distortion under other signal conditions. Thus, an important objective for Dolby noise reduction was minimizing the side effects, and there were a number of other, related goals:

1. *Quantity and quality of noise reduction.* With a companding noise-reduction system, the greater the noise reduction desired, the more the signal must be manipulated by the encode/decode process, and thus the more likely that side effects will be audible. It was decided that Dolby C would not trade more noise reduction for more audible side effects than had Dolby B. This decision is not only reflected in the design of the system, but also in the choice of the amount of noise reduction the new system provides. To minimize the signal processing required, it was necessary to establish the minimum amount of noise reduction required to meet the likely demands of the market. It was concluded that for cassette recording, 20 dB of noise reduction would provide a noise level below that of any current or likely future program source, and would indeed result in tape noise below the ambient noise level of most home listening environments.

2. *Tolerance of normal recorder errors.* For a noise-reduction system to be practical, it must be reasonably tolerant of errors introduced by the recorder at

Fig. 1 — Dolby C transfer characteristics, showing how the effects of the two stages combine to produce 20 dB of compansion.



*Director of Communications, Dolby Laboratories, San Francisco, California.

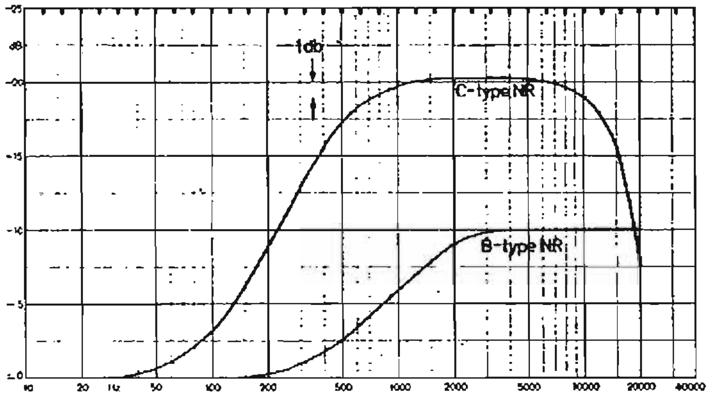


Fig. 2 — Low-level encode characteristic of Dolby B and Dolby C (decoding is precisely reciprocal).

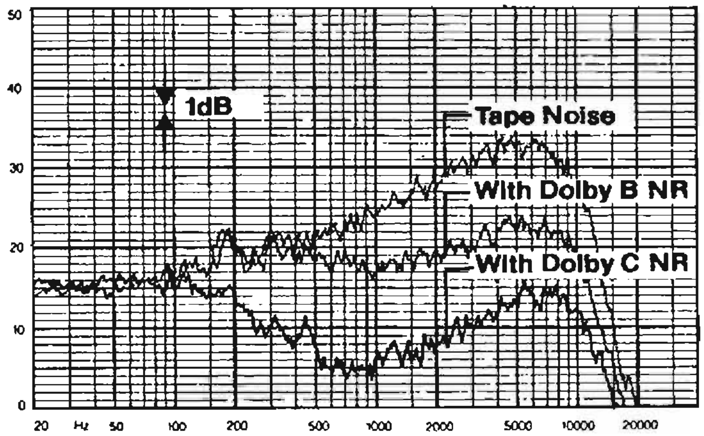


Fig. 3 — Noise from biased cassette tape (70 μS), measured with a constant-bandwidth wave analyzer and weighted (CCIR/ARM) to reflect the ear's sensitivity to noise and noise-reduction effects.

both high and low frequencies between encoding and decoding. Dolby B-type noise reduction has proved sufficiently tolerant of such errors to be practical, so a similar tolerance was an objective for Dolby C. (As it turned out, Dolby C is more tolerant than Dolby B, as the result of developments which will be described later.)

3. *Provision of the Dolby B characteristic.* With so many Dolby B encoded recordings in use, it was thought essential to design the new system so that the Dolby B characteristic could be incorporated easily and economically. Licensees who decide to manufacture products with Dolby C-type noise reduction can provide Dolby B at the same time without the cost and complexity of redundant circuitry.

4. *Cost.* It was obvious that a system providing 20 dB of noise reduction would be more costly than Dolby B-type noise reduction. However, a considerable effort was made to keep its cost appropriate to at least the higher end of the current market. This meant the use of regularly available parts, and also a system which would not put a greater burden on the manufacturer in the way of unusually elaborate production-line testing and adjustments.

5. *The sound of Dolby C recordings played with Dolby B decoding.* A vital factor in the adoption of Dolby B-type noise reduction has been that cassettes encoded with Dolby B sound acceptable on players not equipped with Dolby B circuitry. Thus, assuming there would be interest from duplicators in issuing C-type recordings, consideration was given in the design of the new system to what those recordings would sound like when played on machines equipped only with Dolby B-type or with no noise-reduction circuitry.

The Problem of Noise Modulation

With a companding noise-reduction system, noise reduction occurs at the moment of playback by means of an expander circuit which acts as a gain control constantly adjusting the playback volume. The general idea is to turn the volume down on quiet passages, thus turning down the noise as well, and to turn it back up again on loud passages, which will hopefully mask the noise. If this expanding process is preceded at the time the recording is made by a complementary compression process (i.e., turning the record level up on soft passages and down on loud ones), then the net result on playback should be a restoration of the dynamics of the original music along with noise reduction.

In designing a system which minimizes audible side effects, the trick is to provide a subjectively *consistent* amount of noise reduction, that is, to design a system which does not allow the noise to audibly (and annoyingly) go up and down with level and/or spectral changes in the music. This problem is called noise modulation. As the expander turns up the volume on a loud bass drum rap, for example, it will also turn up the volume of the tape hiss at higher frequencies at the same time so that the bass drum note is accompanied by an audible burst of hiss.

In more than 15 years of research into noise reduction, Dolby Laboratories has discovered only two effective and practical ways to render noise modulation inaudible. The first of these techniques is called band-splitting, and it is used in professional Dolby A-type noise reduction, where the spectrum is divided into four bands, each of which has its own independent compander. Thus, when the hypothetical recording of the bass drum is played back, the expander which acts in the frequency range of the drum lets it through at full volume, while noise in the drum's vicinity is masked by its sound. But in the other bands, where no musical signals mask the noise, the other expanders simultaneously keep the volume turned down, thereby providing full noise reduction in regions such as that of tape hiss.

The band-splitting technique is most appropriate to professional applications because it deals with noise at all audible frequencies; among other things, this characteristic complements the relatively equal distribution of noise in media such as higher-speed professional tape recording. But it is a comparatively costly technique, which fortunately is not required for consumer tape recording where the predominant noise is higher frequency hiss. The second approach, called the sliding-band technique and developed for the Dolby B-type system, uses a single band of compansion which reaches low enough to provide effective noise reduction in the absence of signals, but which also changes its width in response to musical signals. In the case of the solo bass drum note, the companding band will slide up in frequency so that on playback the bass drum is let through at full volume, while the expander lowers the volume, and thus the tape hiss, at the frequencies above the bass drum where there is no musical signal. If the bass drum is accompanied by a violin, the companding band slides further up in frequency to let both the drum and violin through, while still maintaining noise reduction at frequencies above the violin. The result is an effective noise-reduction system with many of the same benefits as the band-splitting technique, but one which is more cost-effective and particularly appropriate to the needs of consumer tape recording. Thus, the sliding band technique has been used for the new C-type noise-reduction system.

Dual-Level Processing

Simply modifying Dolby B to provide 20 dB of compansion at higher frequencies instead of 10 dB is inadequate, as a single processing stage suffers from problems such as overshoots and level uncertainties in manufacturing. Those problems were overcome by the use of two compander circuits in a special way, by which both cover the same frequency band but are sensitive to signals at different levels. One processor is sensitive to signals at essentially the same level as the Dolby B-type processor, while the other is sensitive to signals at a somewhat lower level. Each processor provides 10 dB of compansion; as they are connected in series, their effects add together to provide 20 dB of compansion and so 20 dB of noise reduction.

Figure 1 illustrates by means of transfer characteristic curves how the dual-level processors work together. Note that the total companding action is restricted to middle-level signals. At high signal levels, unlike conventional companders, there is no dynamic action and the system acts as a unity-gain amplifier, while at low signal levels, again unlike many conventional companders, the system acts only as a fixed-gain amplifier. By restricting the system's companding action, such side effects on the signal as overshoot distortion can be minimized; this is characteristic of all Dolby noise-reduction systems.

Noise-Reduction Bandwidth

Yet a further departure from Dolby B was required. The B-type system begins to take effect in the 300-Hz region and increases its action until a maximum of 10 dB of noise reduction is provided in the 4-kHz region and above. The subjective result, due to the ear's sensitivity to the particular spectrum of low-speed tape noise, is an overall reduction of noise, and the remaining noise is subjectively evenly distributed. However, reducing noise in the B-type system's operating range by a further 10 dB, while resulting in very noticeable additional hiss reduction, has the further subjective effect of revealing mid-frequency noise. Thus, a further difference between Dolby B and Dolby C is that the latter reaches two octaves lower, begins to take effect in the 100-Hz region, and provides on the order of 15 dB of noise reduction around 400 Hz and 20 dB in the critical 2 to 10 kHz hiss area. This means that Dolby C, like Dolby B, produces a subjectively even spectrum in what little noise remains (Figs. 2 and 3).

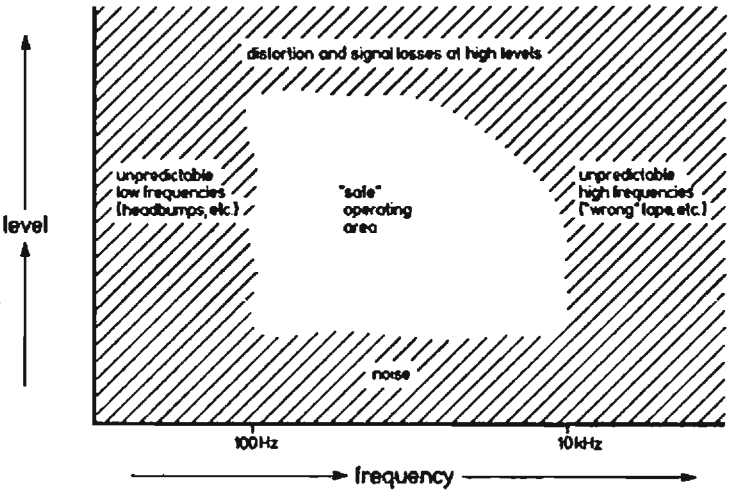


Fig. 4 — To prevent mistracking between the compressor and the expander, a noise-reduction system should operate only on signals which are reproduced reliably by the medium. (The vagaries typical of cassette tape are illustrated.)

As can be seen in Fig. 2, the C-type system's action is sharply and deliberately curtailed below 100 Hz because of the ear's relative insensitivity to low-level, low-frequency noise. The only potential low-frequency noise of any real concern is 50-Hz or 60-Hz hum, which can be kept inaudible (at all but totally impractical listening levels) by proper cassette recorder engineering. This is fortunate, because very low-frequency noise reduction brings up several new problems. For example, it might be tempting for a recorder designer to use the noise-reduction system as a crutch to reduce hum which could (and should) be eliminated by other means. Were there to be considerable hum and were a single-band (now virtually a wide-band) noise-reduction system to reach low enough to affect it, hum modulation would result, whereby higher frequency signals cause the hum level to go up and down audibly. The noise-reduction system could get around this problem if it included a separate low-frequency band, but then the cost and complexity of the system would go up substantially. In addition, a system providing noise reduction at very low frequencies would result in encoded recordings with substantially boosted low frequencies. Such recordings heard without reciprocal decoding would most likely be considered unacceptable due to gross exaggeration of such low-frequency noises in program sources as the room rumble inevitable at most recording sites and turntable rumble components from disc sources.

Dealing with Recorder/Tape Errors

It is important in noise-reduction systems that the expander precisely track the compressor so that the original signal is accurately reconstructed. In tape recording, however, the expander derives the information it needs to reconstruct the original signal *after* the compressed signal has been recorded on the tape. Therefore, if the tape and recorder combination significantly alter the compressed signal, mistracking and potentially audible side effects can occur. To make things even more difficult, the degree to which recorder/tape errors are troublesome is related to the amount of noise reduction desired.

In cassette recording, the likelihood of the recorder and tape introducing upredictable alterations of the signal is very real. For example, different recorders introduce different characteristic head "bumps" at low frequencies so that the signal level coming off any particular recorded tape at those frequencies cannot be predicted accurately. While flat, extended response is theoretically achievable at high frequencies, above 10 kHz it is in reality also unpredictable because the consumer may use a type of tape for which his recorder is not adjusted, and/or the heads may be worn or dirty. Furthermore, he may use a record level too high for the tape, causing distortion and again unpredictable signal losses, particularly at higher frequencies. Ideally a noise-reduction system should operate only on signals in what might be termed a "safe" area, as shown in Fig. 4, where response from the tape can be predicted with some degree of certainty.

Dolby B-type and C-type noise reduction have been designed with these cassette limitations in mind. (For example, as shown earlier, neither system processes signals at unpredictable low frequencies.) But with C-type noise reduction, because more compression and expansion is being squeezed into the same safe operating area, two further developments were necessary to prevent such problems as mistracking, particularly at high frequencies. These are the blocks in Fig. 5 labelled "spectral skewing" and "anti-saturation."

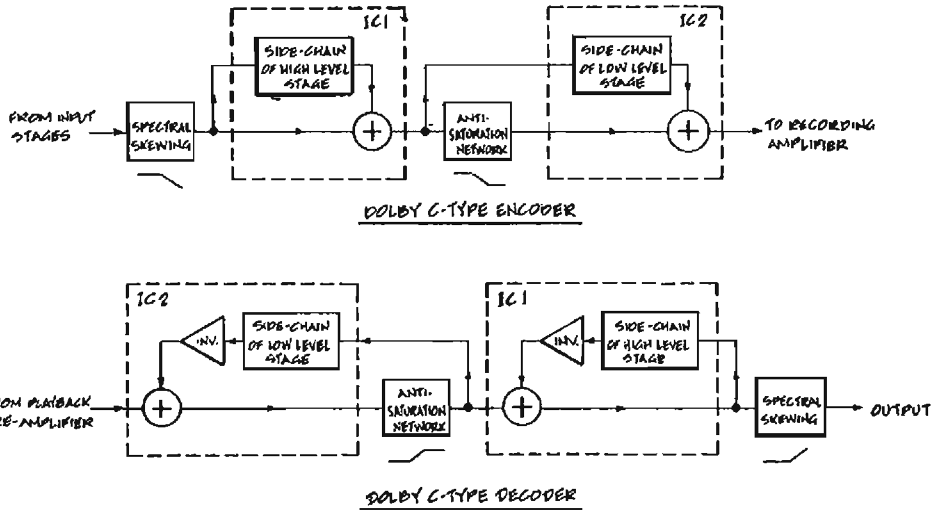


Fig. 5 — Dolby C-type noise reduction block diagram

1. *Spectral skewing.* To prevent expander errors as a result of unpredictable cassette response above 10 kHz (such as those caused by using the wrong tape type), a controlled high-frequency roll-off called spectral skewing is introduced ahead of the compressor when the system is in the encode mode. This roll-off desensitizes the noise-reduction system so that it tends to ignore what's happening above about 10 kHz and so is less likely to mistrack as a result of changes in response introduced by the recording process. A complementary boost of frequencies above 10 kHz is introduced after the expander in the decode mode to maintain flat frequency response. While the noise-reduction effect is thus reduced above 10 kHz, the ear's sensitivity to noise above 10 kHz drops off at an even faster rate than does the noise-reduction effect.

2. *Anti-saturation* helps prevent overloading the tape at higher frequencies, not only to prevent signal losses which might cause expander mistracking, but also to provide higher quality recording overall by reducing the IM distortion resulting from saturation. A shelving network is placed in the circuit in such a way that it affects only the high-level signals which would cause saturation; at low levels, most of the signal passes through the low-level side chain, thus bypassing the network (and thereby maintaining the full noise-reduction effect at low levels). As with spectral skewing, there is a complementary network in the decode (expansion) mode to maintain flat response.

The encode curves shown in Fig. 6 not only illustrate the overall characteristic of Dolby C-type noise reduction, but also show the specific effects of spectral skewing and anti-saturation. The roll-off above 10 kHz at all levels is the result of spectral skewing; the gentler downward slope beginning at about 1.5 kHz on the higher-level curves is the result of anti-saturation.

Implementation of the New System

Ultimately, the most efficient way for the cassette deck manufacturer to provide Dolby C-type noise reduction will be by means of dedicated C-type integrated circuits which are now being developed by several IC manufacturers. As this development will take some time, the first C-type circuits will be based on pairs of standard ICs now being manufactured for Dolby B-type noise reduction in large quantities and at moderate cost by seven different suppliers. Additional circuitry around the IC pairs provides the necessary changes in time constants, the anti-saturation and skewing networks, switching between the B-type and C-type characteristics, and so on. The complexity of the two-IC design is approximately 2.5 times that of Dolby B alone (however, the C-type circuit includes the B-type function as well). That figure should not be taken as a guide to what the premium will be at retail, because greater noise reduction puts greater demands on the recorder performance itself. (Input and output stage noise, for example, must be on the order of 10 dB lower than was necessary before.)

Dolby Laboratories provides its licensees with the C-type system's technology under terms of the current B-type licensing agreement. This means that the royalty paid Dolby Laboratories on a single Dolby noise-reduction processor will remain the same, whether it provides B-type noise reduction only, C-type noise reduction only, or is switchable between both types. (The royalty per processor, which averaged 22¢ at the end of 1980, is not dependent on the number of ICs used.)

So far more than 25 manufacturers and marketers of consumer tape recorder products have plans to introduce models incorporating Dolby C-type noise reduction. Virtually all are likely to incorporate the switchable B-type feature so that existing libraries of Dolby B encoded cassettes will play back properly, and some will incorporate Dolby HX (headroom extension) circuitry in addition. Also, several cassette duplicators have expressed interest in releasing Dolby C encoded recordings, and prototype professional Dolby C encoders have been developed and are being tested for this purpose.

The first products incorporating Dolby C-type noise reduction have just begun to reach the market. It is not yet known how many consumers will elect to pay how much more for a cassette deck with a 10-dB lower noise floor than before. But those who choose these products are likely to find that, with Dolby C-type noise reduction, cassette tape noise, even at the highest playback levels, will no longer be a matter of practical concern.

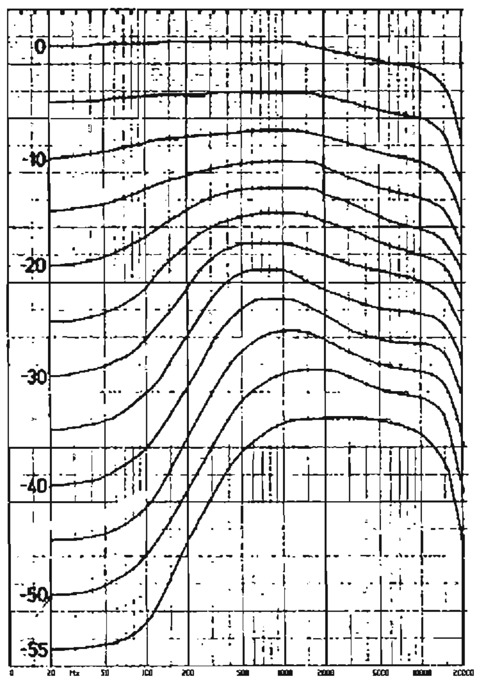


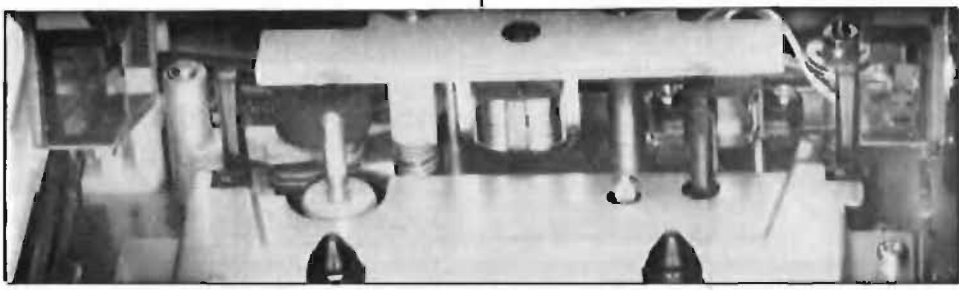
Fig. 6 — Dolby C encode characteristics, showing the effects of spectral skewing and anti-saturation (see text).

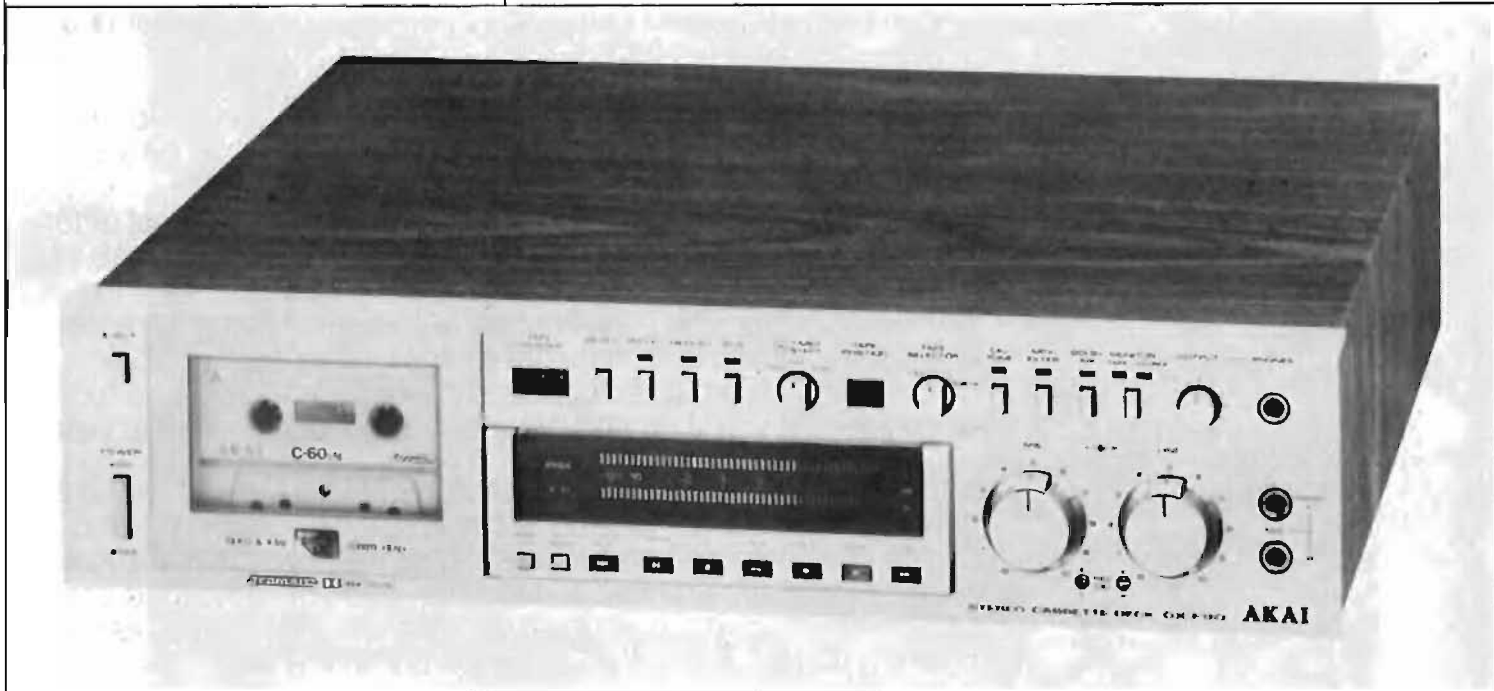
3

AKAI GX-F90 STEREO CASSETTE DECK

Manufacturer's Specifications

Frequency Response: 25 Hz to 17 kHz, 25 Hz to 17.5 kHz with CrO_2 tape, 25 Hz to 21 kHz with metal tape.
Harmonic Distortion: 0.6 percent for 1 kHz at meter zero with metal tape.
Signal/Noise Ratio: 62 dB with metal tape.
Erasure: 70 dB.
Input Sensitivity: Mike, 0.25 mV; line, 70 mV.
Output Level: Line, 410 mV; headphone, 100 mV into 8 ohms.
Flutter: 0.03 percent wtd. rms, 0.08 percent wtd. pk.
Tape Speed: Within ±0.5 percent.
FF & RWD Times: 60 S with C-60.
Dimensions: 17.3 in. (440 mm) W x 4.1 in. (105 mm) H x 14.6 in. (370 mm) D.
Weight: 21.6 lbs. (9.8 kg).
Price: $675.00.





The Akai GX-F90 cassette deck provides excellent performance and has many features of interest, including three long-life heads, fluorescent bar meters, and a logic/solenoid system. The front panel is an attractive brushed aluminum with black designations. There are separate push-button switches for *Memory*, *Repeat*, *IPLS* (Instant Program Location System), *Cal Tone*, *MPX Filter*, *Dolby NR*, and *Monitor*, *Tape* or *Source*. Each of these has a green light which immediately answers any question about deck status. Use of IPLS allows fast winding to the beginning or end of a particular selection, with automatic stop at that point. *Cal Tone* injects a 400-Hz tone for record calibration to ensure accurate Dolby tracking. There are also similar buttons for resetting the counter and ejecting a cassette, and a larger one for power on-off. There are two rotary switches with bar knobs for *Timer Start (Rec-Off-Play)* and *Tape Selector*, which has an adjacent illuminated window to show *LN*, *LH*, CrO_2 or *Metal*.

Eject causes the cassette carrier to move out and tilt down. The door/cover can be removed to facilitate any maintenance tasks. The light-touch tape-motion buttons are logic controlled with substantially any change in mode allowed — including adding record while in play, going into record from fast wind, etc. The *Rec Mute* is right in the same line, as are *Rec Cancel* and the *Peak/VU* meter-mode switches. With the exception of *Stop*, all of the transport control buttons are illuminated: Yellow for the two wind functions, green for *Play*, red for *Rec* and *Rec*

Mute, and orange for *Pause*. This nice feature is made even better with *Play* flashing when in *Pause*, rewinding with *Repeat* or winding with *IPLS*, and *Rec Mute* flashing when it is in operation — each one "'saying'' what must be done next to return to the previous mode. *Rec Cancel* is an unusual feature that will be of definite use to some. If you begin recording what you don't want, a push of this one button puts you back at the start in *Rec* and *Pause*, ready for another try.

The *Peak/VU* switch selects the dynamics of the bar-graph meters, and there are handy status lights to make certain the user doesn't get confused. The bar segments are a light blue up to meter zero and a sort of light brown from there up to the maximum of +8. The Dolby-level reference is at +4 dB. There are two segments for each level step, 24 sets in all, with single-dB steps from -3 to +8 dB, where the best resolution is needed. The peak-level calibration is 7 dB lower than that for VU, which is actually a very good idea because then the maximum meter indications will be about the same for both meter modes. There are dual-concentric level pots for both mike and line, allowing mike-line mixing. The knobs are of a good size and have clear indices, and the friction is just right for making single-channel adjustments.

The screwdriver-adjust, center-detent record-calibration pots are just below the level pots, and the mike phone jacks are just to the right. Use of the left-channel mike jack only will feed that signal to both channels. The knob of the output pot is small, but it is adequate for controlling the level to the headphones and to line out. The line in/out phono jacks are on the rear panel, as is a socket for the optional remote control. Access to the interior was gained with the removal of the steel top and side cover, securely held in place with several machine screws — a touch of quality. The soldering on the p.c. boards was generally very good with little flux residue. The large signal p.c. board, just above the logic board, had all parts identified, including adjustments. Interconnections were made mostly with wirewrap, with some direct soldering and some multi-pin connectors. There were three fuses in clips on the power supply p.c. board. The dual-solenoid, two-motor drive system was judged to be of good construction. The large power transformer was surrounded by a shield, and it and other components were well supported within a rigid chassis frame.

Table I—Record/playback responses (-3 dB limits).

	With Dolby NR				Without Dolby NR			
	Dolby Lvl		-20 dB		Dolby Lvl		-20 dB	
Tape Type	Hz	kHz	Hz	kHz	Hz	kHz	Hz	kHz
Maxell UD-XL I	30	7.2	31	17.0	30	7.3	31	19.3
BASF Studio II	30	7.2	29	18.0	30	7.1	29	20.0
TDK MA-R	30	12.8	29	20.3	30	12.8	29	20.5

Table II—Signal/noise ratios with IEC A and CCIR/ARM weightings.

	IEC A Wtd. (dBA)				CCIR/ARM (dB)			
	W/Dolby NR		Without NR		W/Dolby NR		Without NR	
Tape Type	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%	@ DL	HD=3%
Maxell UD-XL I	62.0	65.0	53.4	56.4	61.5	64.5	51.3	54.3
BASF Studio II	63.0	65.0	54.7	56.7	62.5	64.5	52.5	54.5
TDK MA-R	63.9	66.4	55.3	57.8	63.0	65.5	52.5	55.0

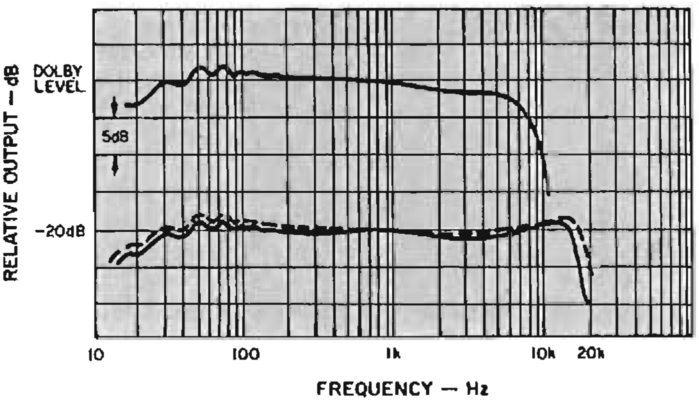


Fig. 1—Frequency responses with and without (- - -) Dolby NR using TDK AD tape.

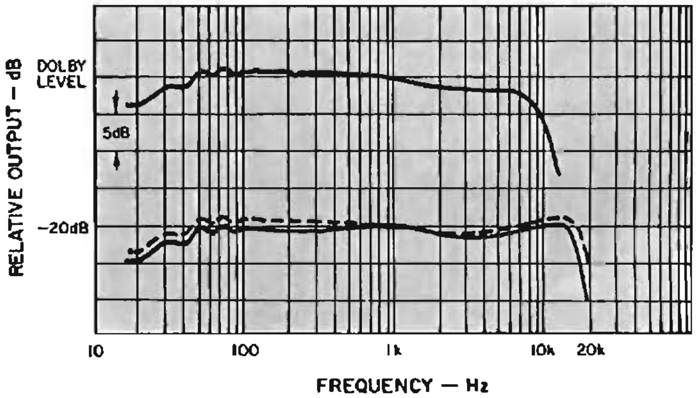


Fig. 2—Frequency responses with and without (- - -) Dolby NR using Maxell UD-XL II tape.

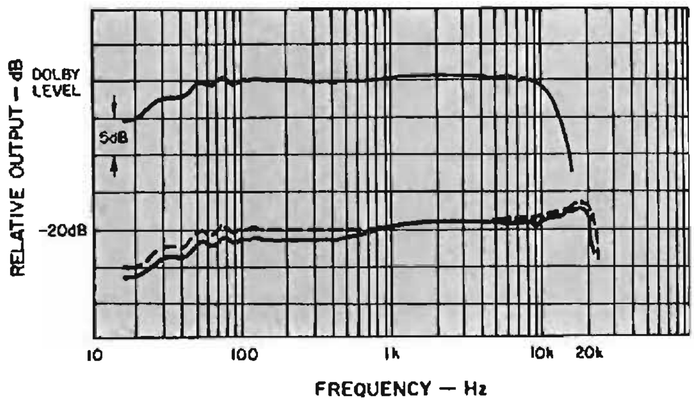


Fig. 3—Frequency responses with and without (- - -) Dolby NR using Ampex MPT tape.

The Akai GX-F90 cassette deck has many useful features and very good to excellent performance in every area.

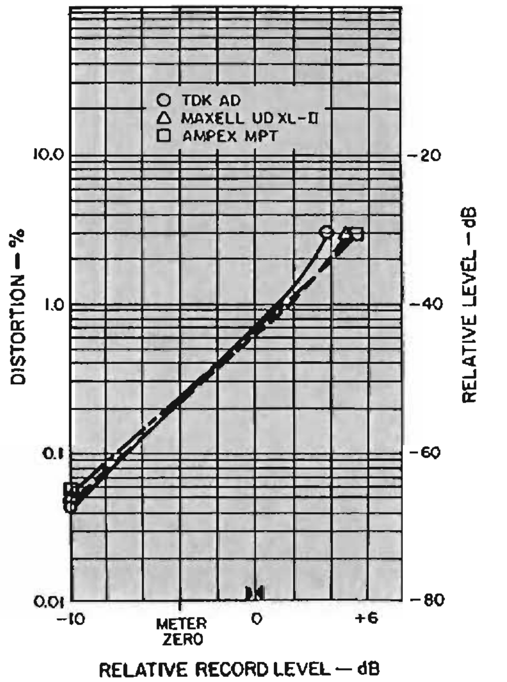


Fig. 4—Third harmonic distortion vs. record level in Dolby mode at 1 kHz using TDK AD, Maxell UD-XL II, and Ampex MPT tapes.

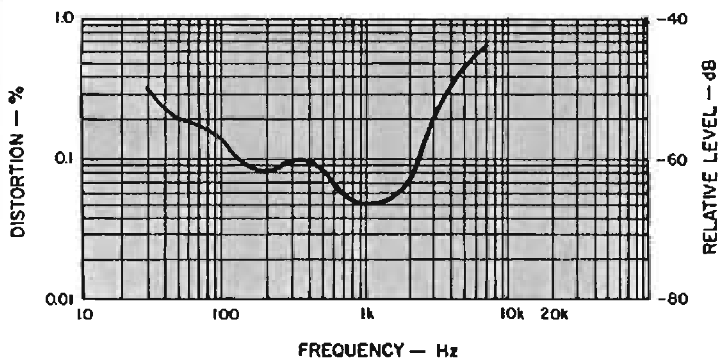


Fig. 5—Third harmonic distortion vs. frequency in Dolby mode at 10 dB below Dolby level using Ampex MPT tape.

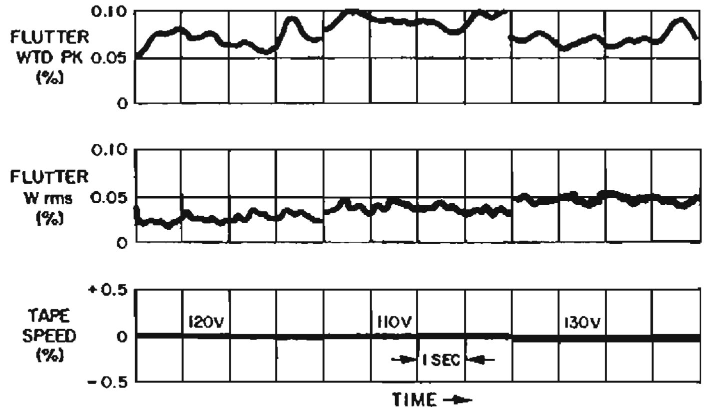


Fig. 6—Tape play speed vs. line voltage and wtd. rms and wtd. pk. flutter, three trials each.

Performance

The first step in the testing was to check the GX-F90 with standard alignment tapes. With both equalizations (120 and 70 µ S), responses were very good at the lower frequencies but were down 3.5 to 4 dB at the highest frequencies. Tape play speed, on the other hand, was very accurate: 0.05 percent slow at most. The level indications on the meters were both within a dB of the standard. Pink noise and a 1/3-octave RTA were used, as usual, for fast surveys of a large number of possible formulations for use in testing. Akai is to be praised for listing both recommended tapes and the standard reference tapes they use for setting up the deck. I found very close agreement with their list although my tapes for test (TDK AD, Maxell UD-XL II and Ampex MPT) were not their reference tapes. Figures 1 to 3 and Table I show the swept-frequency response results with and without Dolby NR, both at Dolby level and at 20 dB lower. With the AD and UD-XL II tapes, there appeared to be a possible influence from the drooping play response mentioned earlier, including the little saddle (droop) around 4 kHz, 1.5 dB at most. The responses with the metal tape are different, being elevated slightly from about 1 kHz out. Nonetheless, all of these results are quite good, and if the deck had bias trim, the responses could have been made smoother, gaining better Dolby tracking.

The 400-Hz (441 Hz actual) calibration tone had about 0.8 percent THD, quite acceptable for the purpose. With Maxell UD-XL II tape, the record calibration pots had a range of adjustment of about ±5 dB. With a 10-kHz test tone, there was a 70-degree phase discrepancy between tracks — typical for the combination head design used. There were 25 degrees of phase jitter, better than most cassette decks. The output polarity was in phase with the input signal. The multiplex filter was 3 dB down at 16.1 kHz, and it was a good 34.8 dB down at 19 kHz. With a 1-kHz test tone, erasure was greater than 80 dB, and separation was 43 dB. Crosstalk was down at least 60 dB, with larger and more desirable figures at times. Erasure of the metal tape at 100 Hz was greater than 70 dB, very good performance. Bias in the output during recording was very low, though there were some beat notes at the very highest frequencies of the swept test tone used for the responses.

HDL_3 (the level of third harmonic distortion) was measured for a 1-kHz tone from 10 dB below Dolby level to the point where the distortion reached 3 percent for each of the three tapes (Fig. 4). The curves are quite linear, with just a slight upward curving at the distortion limit. The 0.05 percent distortion figure at -10 dB shows that there was very low distortion in the electronics, as well as in the magnetic process. The signal-to-noise ratios with both IEC A and CCIR/ARM weightings were measured with and without Dolby NR, and the excellent results are shown in Table II. HDL_3 was also measured over a range from 30 Hz to 7 kHz with Ampex MPT tape at 10 dB below Dolby level (Fig. 5). The results are excellent in the mid-frequencies, and the increases at the frequency extremes are less than with many recorders.

Input sensitivities were 0.23 mV for mike and 65 mV for line, both a little bit better than spec. The input overload points were very high: 56.9 mV for mike and 30 V for line, where waveform rounding first appeared. The output clipped at +18.1 dB relative to meter zero. The two sections of both input level pots tracked within a dB from maximum down about 60 dB, excellent performance. The output pot tracked within a dB down about 45 dB. The 100-mV headphone drive to 8 ohms was fine for most

The GX-F90 has three heads, fluorescent bar meters, and a logic-controlled solenoid transport system.

phones, though the volume was on the low side with an AKG 600-ohm set. The line outputs were 414 mV, and they dropped just slightly (to 410 mV) with the IHF 10-kilohm load.

In VU mode, the bar-graph meters had a response time of 240 mS, slightly fast, and there was no overshoot. In *Peak* mode, the response times were substantially to IEC Standard 268-10, actually being slightly fast: Only 2 dB down with a 3-mS burst, where the standard calls for -4 ±1.0 dB. The fall time was 1.0 second, slightly short for easy reading. *All* of the scale calibrations were exactly accurate, from -20 dB to +8 dB, much better than most meters. The meter response was not polarity sensitive in either mode, and the increased level from a sine wave with d.c. offset (or a single-polarity peak, if you will) was correctly shown. This is additional evidence of excellent design by Akai.

The speed characteristics of the GX-F90 were also impressive. The tape speed at 120 V was one of the most accurate ever measured. Variations with time were very low, and there was just a slight lowering of speed at 130 V. There were the expected variations in flutter with trial and cassette, but typical figures were 0.035 percent wtd. rms and 0.075 percent wtd. pk., certainly better than most decks. Wind times averaged 69 seconds with a C-60, which is over spec, but still fast enough. Run-out to stop time was 2 S, and any change in transport mode was made in 1 second or less.

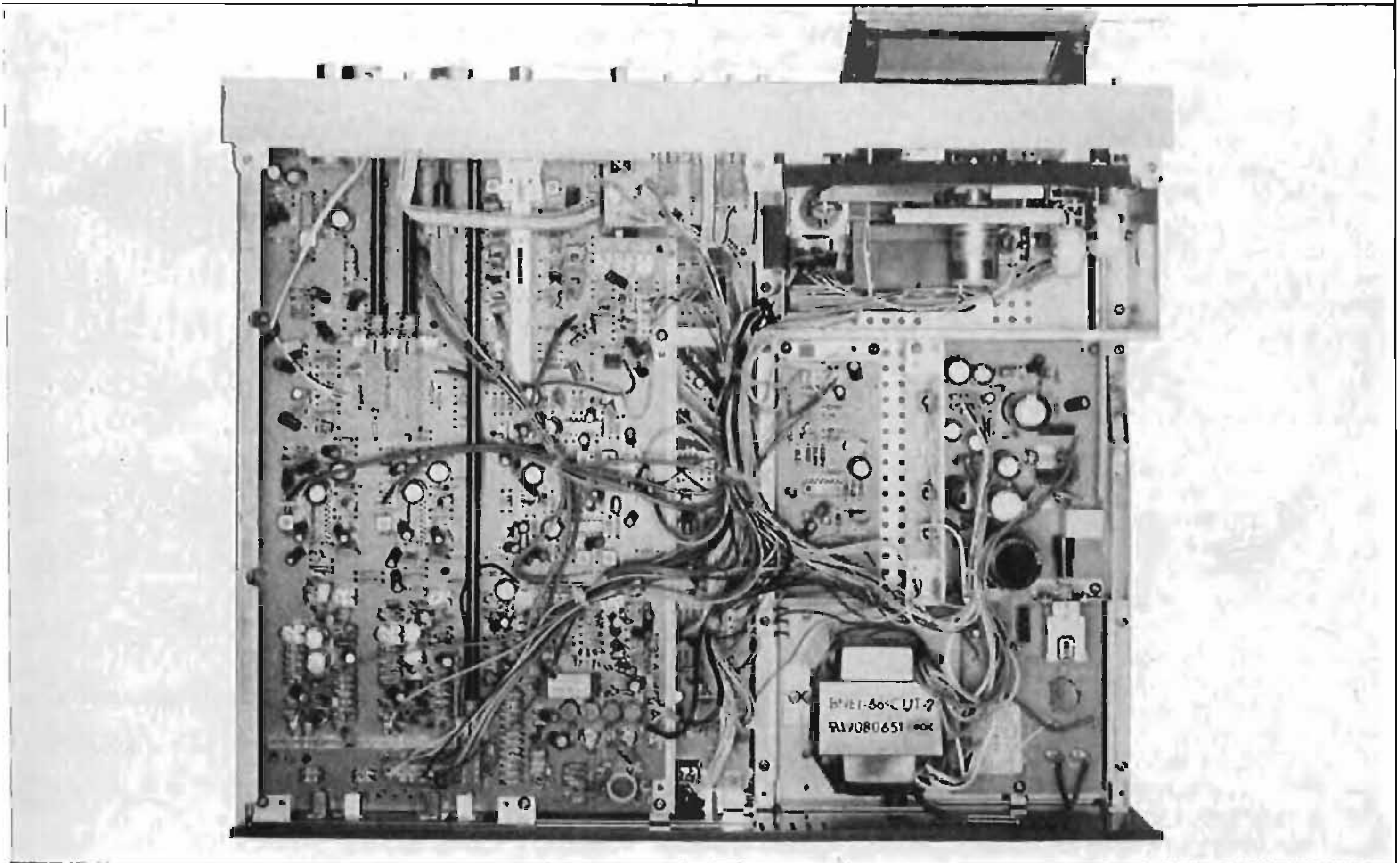
In-Use Tests

Tape loading and unloading was simple and convenient, and there was easy access for maintenance, especially with the door/cover removed. It was a distinct pleasure to use the Akai deck with the many helpful status lights, particularly the flashing *Play* and *Rec Mute* buttons, calling attention to what needed to be done. Setting record levels accurately was quite easy with the good pots and the peak/VU metering. Everything was completely reliable, including IPLS and timer-start functions. The trilingual owner's manual provided sufficient detail in most respects, but the text was confusing on the use of peak metering. There was also a reference to "occasional motor lubrication," which is not as clear as, say, "every XXXX hours of operation."

Record/playback listening was done first with pink noise as a source, and then several records, which included Haydn's *Symphony No. 59* and Bach's *Suite No. 2,* both with Marriner and the Academy of St. Martin in the Field, Diahann Carroll with the Ellington Orchestra (Orinda Records), and the Maxell samplers. The results were really very good, but it was possible to detect some dulling with the noise source and TDK AD and Maxell UD-XL II and some brightening with Ampex MPT. With the music, changes were more subtle, but there were some similar indications, mostly with the vocal by changes in presence. Record, pause, and stop noises were *very* low, just detectable at times.

Overall, the Akai GX-F90 deck has many useful features and very good to excellent performance in every area. It is most worthy of comparison to any of the other decks in its price range. *Howard A. Roberson*

Enter No. 92 on Reader Service Card



HOWARD ROBERSON

FOCUS ON SHELL MECHANICS

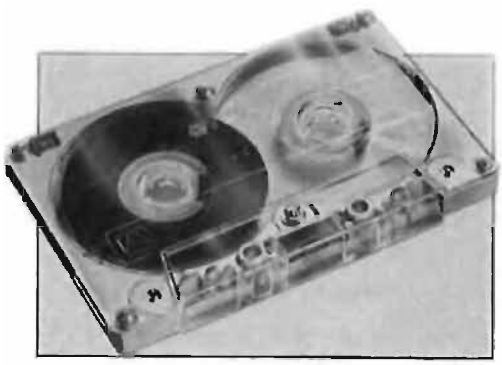


Fig. 1 — TDK's new Metal Alloy audio cassette comes housed in a Reference Standard mechanism/shell. (Photo, courtesy TDK.)

At least once a year, *Audio* tests a number of the latest cassette tape formulations. The effort is concentrated on electrical performance, such things as frequency responses and maximum record levels. It has been a regular practice to check flutter and skew effects, but we have not taken a detailed look at the mechanical inter-relationships that are involved. This article delves into such questions as: Is there such a thing as a low-flutter cassette? Does it make a difference which deck is used? Do all tapes skew? What should I look for in a cassette if I want good mechanical performance? Are some cassettes more reliable than others?

Even with just a cursory examination in the course of buying tapes, you may have noticed that there is a great deal of difference in the way that the various manufacturers package the product. This can be more important than it might at first seem. To minimize the possibility of getting harmful dust on the tape itself, the package should be sealed in some sort of plastic wrap. The little tab seal used by a few manufacturers will assure you perhaps that the tape has not been used by someone else, but the openings in the typical cassette box can allow a lot of dust to enter. After opening the package (a pull tab is a worthwhile convenience), follow good practice in keeping the cassette in the box when not actually being played. Do not store any tapes in dusty environments. If you must do so, try some of your own plastic protection, or stick to cassettes with better quality boxes with tighter closures. The new Memorex boxes provide better sealing from dust than the older standard-design ones do.

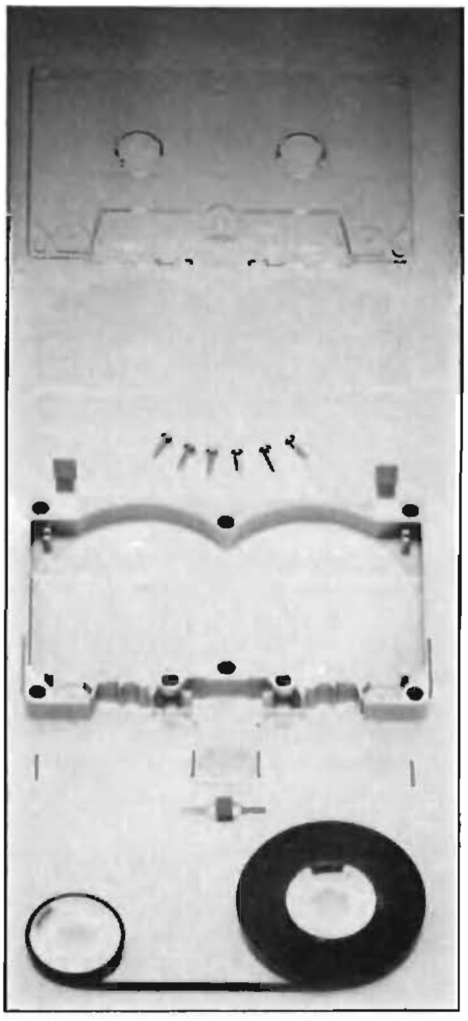


Fig. 2 — Disassembled Reference Standard mechanism. (Photo, courtesy TDK.)

If you have purchased a number of different tape formulations, you will have learned that manufacturers do not agree on what makes the best label. We're not going to dig deeply into this question, but there are a few things to keep in mind. First of all, is there enough space to write in the necessary identification? Some labels are so small, you might need to use some sort of code or abbreviations. What can you write on the label with? Best of all are those which will accept almost anything, but some require a ball-point pen, making changes very difficult. Some cassettes have extra press-on labels, which is a definite help. Take a look at the outside card while you're at it: Some are good, and some have failings similar to those discussed for labels. Now that we've talked about the wrappings and labels, it's time to take a closer look at the cassette shell itself.

Figure 1 shows the assembled TDK MA-R cassette, the most sophisticated design currently available. It is also relatively expensive, of course, so it is not surprising that most other assemblies are less impressive. The great majority of the premium cassettes sold have plastic half shells which are held together with screws. There are just a few that are sonic welded together, and their manufacturers feel that sonic welding can do just as good a job as screws. It is true that screws must be torqued correctly to hold firmly, but not so tight as to introduce unwanted stresses. It is also true that most of the really cheap cassettes are sonic welded, and that the cassette shells which are most nicely finished are all held together with screws. It is possible, of course, for a manufacturer to use more than one quality of shell and mechanism in its product line. TDK actually uses four, the Reference Standard Mechanism for MA-R tapes, the Laboratory Standard Mechanism for MA and SA-X tapes, the Super Precision Mechanism for SA, OD and AD tapes, and the Precision Mechanism for D tapes. We can't say that this is just the way to do it, but there is a great deal of sense in using a higher quality mechanical assembly to go with higher quality formulations.

To aid in the discussion of design requirements for the assembly, let's examine Fig. 2 which shows a disassembled Reference Standard Mechanism. In the center is the die-cast metal frame which must, and does, provide rigidity and stability, accurate outside dimensions and good finish, good support and accurate location for other components to be mounted in the frame, and parallel sides for mounting the cover plates. The cover plates must be flat with some rigidity and accurately dimensioned. The assembly of these two components has to provide the basic inside space for tape storage and guidance. There are two static-free slip sheets for low-friction restraint of tape wander during play or wind. Now, a cassette shell that is made out of plastic should meet the same basic criteria, accurate dimensions, rigidity, stability, etc. Carefully examine the cassettes you are using for surface smoothness, traces of plastic flash, resistance to bending or twisting, etc. Don't try to find their stress limit, but you may be able to weed out some questionable tapes.

When the tape pack is set into the shell, it is threaded over guide pins at each end and then over guide rollers, passing in front of additional guide pins, the tape pressure pad, and the mu-metal magnetic shield. For good mechanical performance, there are criteria that the tape pack itself must meet. The width must be constant, the slitting must not cause any deformation of the edges, the cutting must be a perfectly straight line, with no skew introduced, and the leader must meet the same criteria. As a fast check of the cassettes you have, look at the surface of the tape at the edge of the cassette. It should be perfectly flat and smooth. Any cupping or rippling might cause a range of tape problems, dropouts, changing levels, high flutter, and its getting wrapped around the capstan. It comes down to this: To get the most out of the tape, it must remain in intimate contact with the heads — easy if its own surface is flat (except as shaped to the head), impossible if its own surface looks like a badly misaligned tire.

The tape must be fastened to the hubs, and yet the hubs must not introduce any bumps as the tape winds on. The hubs must also be accurately round and concentric with the hub-drive opening. All the stationary guide posts must be perpendicular to the shell-frame reference plane. Their surfaces must be long-wearing and must not damage the tape in any fashion. The guide-roller surfaces must be smooth (TDK states "seamless"), concentric with the bearing pin (stainless steel preferred) and with flanges that guide and control tape wander without causing any edge damage. The magnetic shield should be accurately positioned behind the pressure pad. There are different approaches to the design of the pad and its support. Do check to see that the pad surface is flat to the tape and not misplaced crosswise. It is impossible to check out the internal construction of a typical plastic-shell cassette, unless you take it apart. Perhaps if all that is not worthwhile, you would want to check what the manufacturer claims he has done to make his cassette a good one mechanically. If the

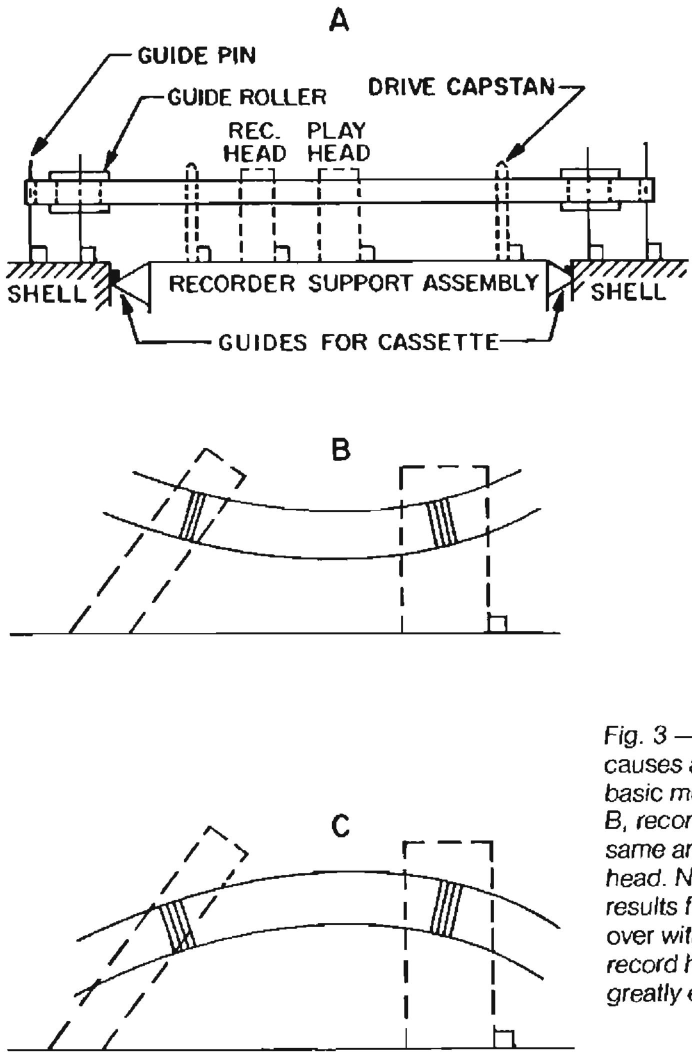


Fig. 3 — How tape skewing causes alignment errors. A, basic mechanical interface. B, record head adjusted for same angle with tape as play head. Note skewing tape. C, results from turning cassette over without readjusting the record head. The effects are greatly exaggerated for clarity.

edge of the cassette mates poorly and is not straight and smooth across, you should be suspicious of any claims for internal excellence. The key words are smooth, flat, perpendicular, parallel, round, concentric, rigid, stable and accurate.

With a few artistic liberties, Fig. 3 shows some of the cassette/recorder alignment relationships and how tape skew affects head alignment. In "A" the guide pins and rollers are shown to be exactly perpendicular to a rigid shell. The molded-in guide pins near the center of the cassette must also meet this criterion. The drive capstans and the record and play heads are shown as perfectly perpendicular to the recorder support assembly, which supports and positions the cassette shell exactly. This is our ideal of course, and if the tape ran perfectly straight, maybe it could really happen. First of all, recognize that if the cassette is not positioned accurately relative to the recorder components, some of the accuracy in the cassette itself is defeated. In other words, when you insert a cassette, make certain that it seats firmly into position. If there seems to be a favored position for best performance, use that all of the time.

Now let's take a look at how tape skewing affects head alignment and record/playback performance. In "B" of Fig. 3, the play head is shown to be perpendicular to the support plate with the tape curving across it. The curvature of the tape and the resultant angles are greatly exaggerated to facilitate demonstrating what happens. The lines on the tape, are radial lines of the curve and perpendicular to the edges. If we adjust the record head to get it in best alignment with the play head, we get the result shown in "B." Next we flip over this cassette ("C") to see how things would work out, *without* readjusting the record head. For the play head, the radial lines are just put at the same angle on the other side of vertical. With the record head, however, the angular error is twice what it was before any adjustment.

These simple figures tell us a number of things about what is desirable and what to expect. Most desirable are cassettes which have *no* skew and which will seat accurately in the recorder. With such tapes, alignment with both heads will remain correct, including when the cassette is turned over. Maybe this

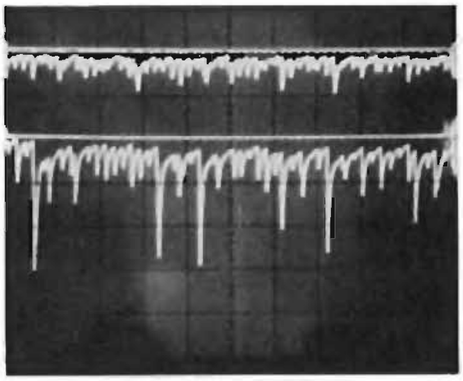


Fig. 4 — Record/playback flutter with Akai GX-F90 deck. Top, TDK MA-R cassette; bottom, BASF Studio I cassette. (Scales: Vert., 0.05%/div.; hor., 1 S/div.)

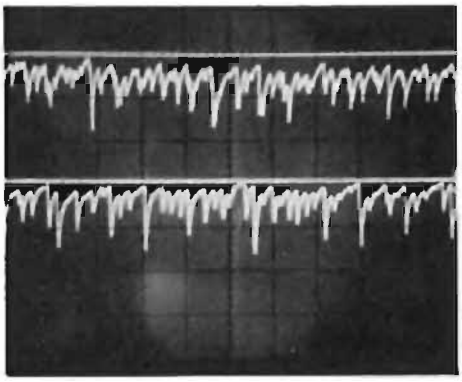


Fig. 5 — Record/playback flutter with Technics RS-9900/US deck. Top, TDK MA-R cassette; bottom, ferric cassette. (Scales: Vert., 0.05%/div.; hor., 1 S/div.)

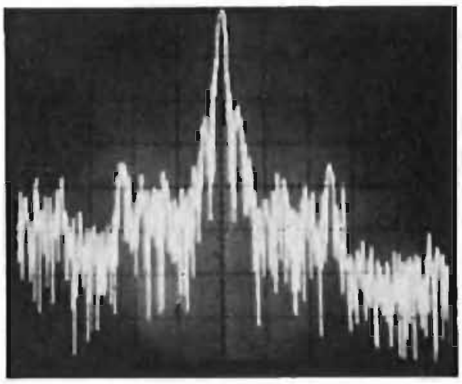


Fig. 6 — Flutter spectrum with Akai GX-F90 deck and TDK MA-R cassette. Scales: Analyzer bandwith, 1Hz; 'scope, vert., 10 Hz/div., hor., 10 dB/div.)

sounds unlikely, but, in fact, we have been able to report on a number of tapes that consistently do not have detrimental skewing from sample to sample and from side to side, including both C-60s and C-90s. Obviously, the problem of poor responses from skewing is more severe with separated record and playback heads. Note, however, that skewing can cause some loss in response even in combination record/playback heads. This is particularly true if you are going to play a tape on another deck than the one used for recording.

We know that smooth tape motion is essential for low flutter and that rough mechanical motion would result in high flutter, but how much does it have to do with the deck? The first part of the investigation utilized an Akai GX-F90 which had shown low flutter, the Nakamichi 582 which had average flutter, and the older-design Technics RS-9900/US. Figure 4 has plots of the record/playback flutter with the Akai deck using a TDK MA-R (top) and a medium-price ferric (bottom). The straight lines are the reference zeros, the vertical scale is 0.05% wtd. pk. per division, and increasing flutter is in the negative direction. The 'scope traces show a number of peaks not indicated on the meter, but there is no doubt about the much lower flutter with the MA-R (0.03% meter) compared to the ferric (0.08% meter). When we tried the same cassettes in the Technics RS-9900/US (Fig. 5), the flutter with the MA-R was higher, the flutter with the ferric was lower, and they were pretty much the same with this deck.

We went back to the Akai deck and plotted (Fig. 6) the MA-R flutter spectrum for 50 Hz each side of our test tone with a 1-Hz analyzer bandwidth. There are sidebands at ±4 Hz, 23 dB down, and at ±24 Hz, 35 dB down. Figure 7 shows the results with the same deck and another ferric cassette, which had shown twice as much flutter on the meter as the MA-R. Note that the sidebands at ±25 Hz are up to -21 dB, and that there is a lot of energy at a higher level at other points. Similar checks with the Nakamichi 582 showed spectra with reduced discrete sideband levels but with considerable "random" energy close to the test-tone carrier. The meter reading of 0.09% wtd. pk. was indicative of the total level of these many flutter components, even though the energy was not concentrated in a couple discrete frequencies.

Subsequent to taking the above data, two Aiwa decks were obtained which had low flutter with many cassettes. A search was made to find cassettes that had mediocre to poor flutter performance. The results from some of these tests are shown in Fig. 8. Please note that the vertical scale is 0.1% per division, as compared to 0.05% per division in the earlier figures. As before, the straight-line traces are the reference zeros, and increasing flutter is downward. The two topmost sets were run with the Nakamichi 582. The so-called mediocre cassette showed no values greater than 0.08% wtd. pk., and an excellent 0.05% was typical. The poor cassette was really that with relatively frequent readings to almost 0.2%, and a few close to 0.3%!

The next two sets were made with the recently introduced Aiwa AD-3600. Preliminary tests had shown very low flutter with many cassettes. The first run with the "mediocre" sample showed most readings below 0.06%, with around 0.04% or less very common. These are certainly excellent figures, but on to the challenge of the cassette that had performed so poorly just before in the other deck. These results (next to the bottom of Fig. 8) were quite unexpected, but the plotted figures were really quite typical — few peaks over 0.06%, with most meter indications less than 0.04% wtd. pk.! This low-flutter result left space for a run with the same cassette in the Aiwa AD-M700 deck, which had also shown well-controlled flutter. There is a noticeable increase, compared to the AD-3600 deck, but the 0.08% maximums are still quite acceptable and much lower than the Nakamichi 582 results. In case there is any question, let it be stated here that exactly the same section of tape was used for each recorder and rechecks were made of all the results.

In general, if you need very low flutter, you must have a good performing deck as well as a good cassette. A cassette cannot force a deck to have low flutter, but many decks are definitely sensitive to the characteristics of the cassette. The Aiwa decks used in the tests reported above were the best seen to date in giving low flutter, regardless of the cassette used. Under a number of condi-

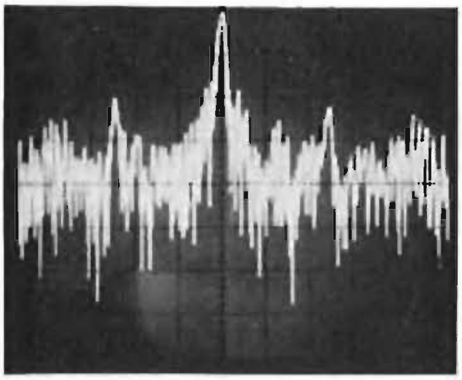


Fig. 7 — Flutter spectrum with Akai GX-F90 deck and a ferric cassette different from the one used to generate the bottom curve in Fig. 5. (Scales: As in Fig. 6.)

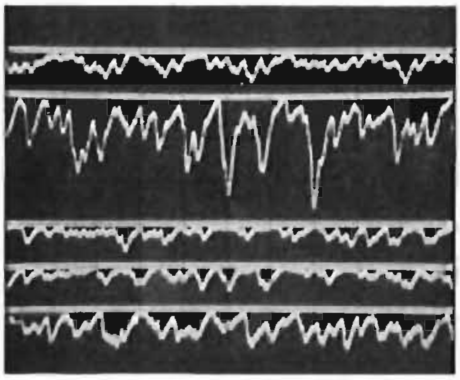


Fig. 8 — Flutter vs. tape and deck. Top, mediocre cassette with Nakamichi 582; 2nd, poor cassette with 582; 3rd, mediocre cassette with Aiwa AD-3600; 4th, poor cassette with AD-3600, and bottom, poor cassette with Aiwa AD-M700. (Scales: vert., 0.1% wtd. pk./div., hor., 1 S/div.)

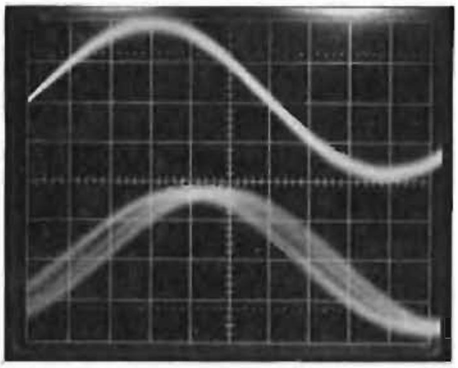


Fig. 9 — Record/playback 10-kHz phase error and jitter between channels. (Scale: Hor., 30 deg./div.)

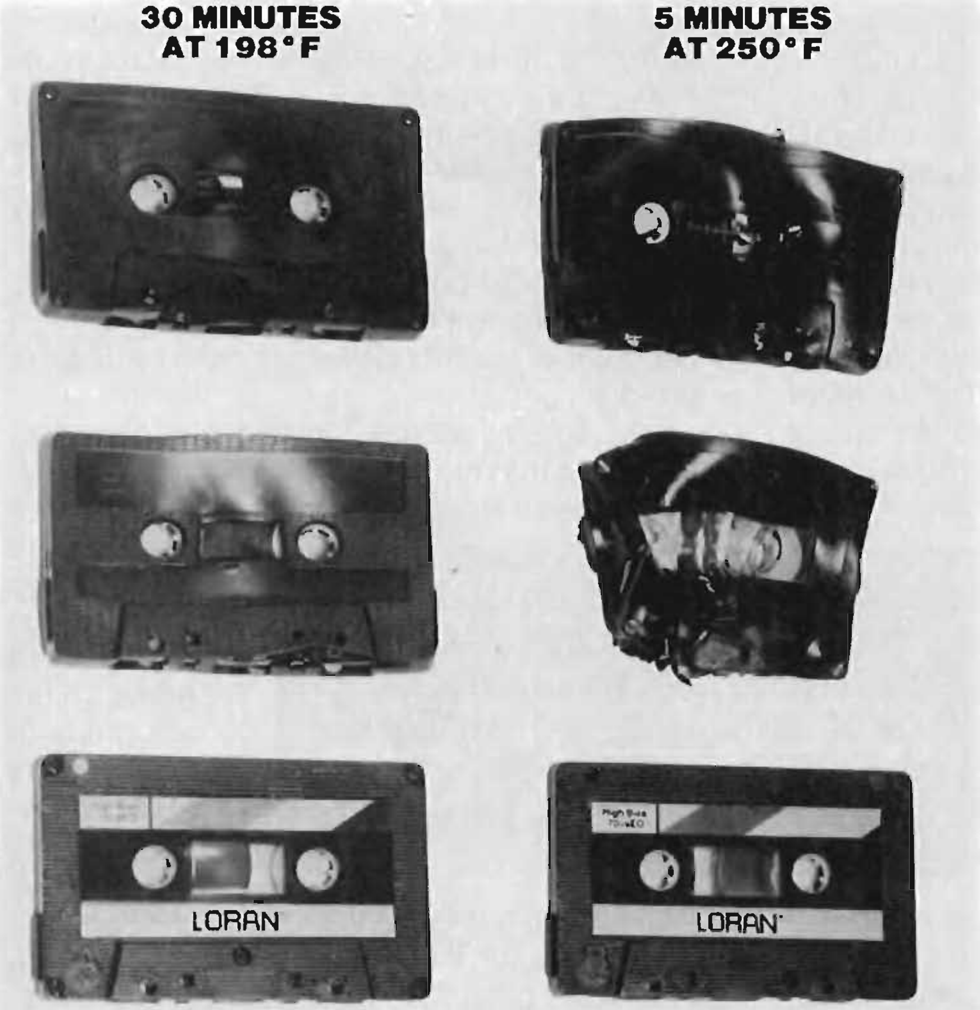


Fig. 10 — The two Loran cassettes shown here were exposed to high temperatures in environmental test chambers along with four leading cassette tapes, labels removed. (Photo, courtesy Loranger.)

tions TDK MA-R appeared to be the lowest flutter cassette. Some of the other good-performing tapes were Ampex EDR, GMI and GMII, BASF Professional II and III, Fuji FX-I, Maxell UD-XL I-S and UD-XL II-S, Memorex MRX-1, Osawa Cr, Realistic Supertape Chrome, Sony SHF, and TDK MA and SA-X. These conclusions must be considered tentative because of the limited, relatively short-time testing and because of the proven influence of the deck.

The last of the tape/recorder interface effects to be discussed is record/playback phase jitter. If the tape motion were perfect across the head, without waving or vibrating, there would be no shifting in time between channel A compared to channel B. Figure 9 shows the output of both channels of a recorder with a 10-kHz test tone, and with the scope locked to "A" (top). The relative phase jitter of "B" causes the trace to move back and forth on the screen, as shown in this timed exposure. The sweep speed was adjusted for 30 degrees per division, and we can see that the total jitter is about 40 degrees, which is fairly good for a cassette deck. A misalignment of about 40 degrees was purposely left in, evidenced by the displacement of the average position of the "B" trace. This angular discrepancy of the 10-kHz tone indicates an 11-µS time difference. Actual jitter and alignment errors can be much greater than that shown. The conclusion drawn after a series of tests with a selection of cassettes and a number of decks was this: Phase jitter is primarily determined by the deck, but the cassette has some influence on the exact results. The deck with the smallest distance between the record and playbcak gaps is most likely to have the least jitter. Recognize that such jitter will exist in any subsequent playback. It is also a fact that jitter and skewing can cause fairly high *level* losses at the higher frequencies when a tape is played back on another recorder, especially when there are head alignment errors.

At the time of this writing, Loranger has just introduced a line of cassettes which have shells made out of Lexan. Among other things, the manufacturer claims that these shells are much more stable with elevated temperatures, such as might be found in car tape players. Figure 10 does show very noticeable damage to the non-Lexan shells, so I subjected C-60 spares to oven temperatures of 120° to 160° for one hour, a temperature period that might well be found on a car dash. I found that the cassettes most sensitive to heat distorted very quickly. Only one-third survived to the end, though some shells were a bit distorted. The Loran Lexan shell showed the least effect, and it should be noted that the Maxell shells showed very little warp. This certainly is a valid area for investigation, and I will try to gather more information for possible publication later on.

Winding

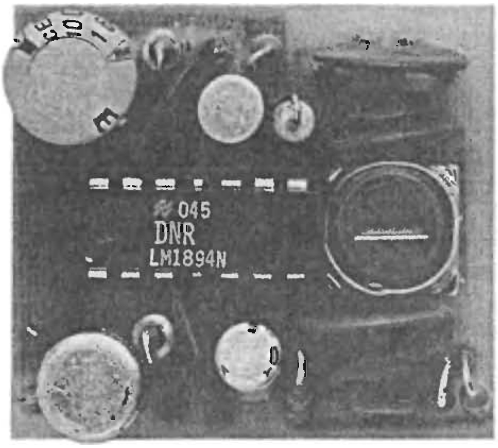
In past years, there were a fair percentage of cassettes that would not survive very many fast winds, and occasional ones that would not even play through one time without jamming. These types of failures are now much less frequent, and there are fewer cases of various types of sounds, squeaking, moaning, chattering, etc. There are still tapes, however, that are very noisy on one deck and most quiet on another. *In general,* it appeared that the cassettes with the smoothest winds (quietest) had the lower flutter and the least likelihood of jamming. There were a number of exceptions, and only very lengthly testing would prove whether there is much of a correlation. Good guidance to the tape pack with slip sheets did reduce over-the-pack failures. Finally — and once again — the total cassette system performance depends upon the mechanical and electrical characteristics of both the tape *and the deck.* ▲

NATIONAL'S NEW NOISE REDUCTION CHIP

RALPH HODGES

National Semiconductor, one of the nation's oldest and certainly one of the leading makers of integrated circuits, is by no means new to noise reduction. Selling Dolby B-type integrated circuits to cassette-deck manufacturers is one of the higher volume activities of its consumer linear division, and you can be sure that other products of its manufacture turn up as gain blocks in many alternative noise-reduction systems. What the company has not conspicuously participated in is the *design* of noise-reduction processors. Indeed, making component parts for everybody else's NR system — and everybody else's electronic anything, for that matter — would seem to be business enough, why should it get further involved? Or, rather, why *has* it, because National Semiconductor's DNR (Dynamic Noise Reduction) amounts to just that sort of involvement?

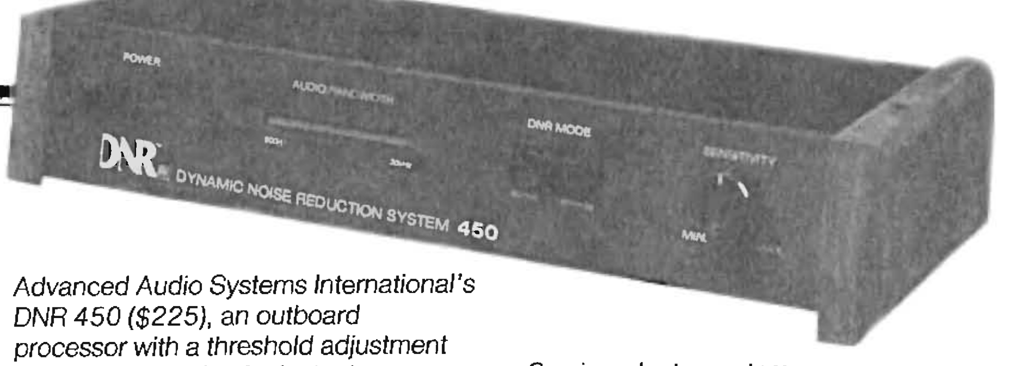
The reason, according to the company, is that noise reduction in its commonly encountered compander form is all well, all good, but all too rare. Efforts by Dolby Labs notwithstanding, FM broadcasts are still largely compander-unencoded. The cassette you play in your portable "tape player cum headphones" may be encoded, but it's unlikely that the player will be able to *de*code it, and the hiss from a diaphragm within an inch of your ear is hard to ignore. You can buy encoded discs from dbx, but per-



The DNR device is essentially a dynamically controlled LPF that is inexpensive, simple and compact in implementation, and reasonably free of audible side effects.

NR

haps not with the performers and performances you'd most prefer. Although tomorrow's videodiscs may be encoded with some form of noise-reduction processing, today's are not. The same goes for the majority of prerecorded videocassettes. The future (AM stereo, stereo TV) remains clouded, but the past is clear, and much of it is made up of recorded material that never had a chance to benefit from practical noise reduction.



Advanced Audio Systems International's DNR 450 ($225), an outboard processor with a threshold adjustment and an LED display for instantaneous bandwidth indication, bears the proprietary logo for the National Semiconductor system. Manufacturer: Advanced Audio Systems, San Jose, Cal.

The DNR scheme seems an appropriate solution to an inherently insoluble problem.

Of course, the situation is not new, but, ironically, new media and program sources are making it more prevalent. In response, National Semiconductor has seized on a solution that is also not strictly new, but which is probably timely. Now, the company believes, is the right moment for noise reduction that can cope with sources which already contain noise. This means a "single-ended" processor — one that steers its way between program and noise, lopping off the latter insofar as it is able to separate such noise out. The DNR device is essentially that of a dynamically controlled low-pass filter, but one that skirts negatives in previous designs of this sort, which as a rule were not (1) inexpensive; (2) simple and compact in implementation, and therefore adaptable to a broad spectrum of products, and (3) free as they could be of audible side effects. DNR is all of these according to National Semiconductor, who expect its appeal to grow rapidly as the word gets around.

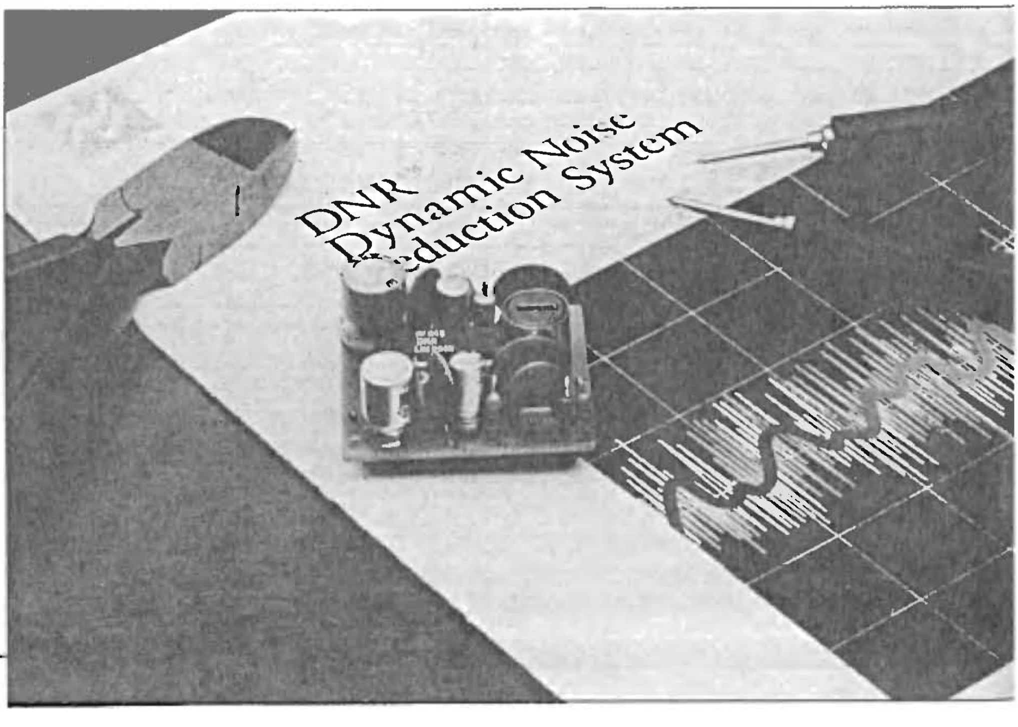
Basics of Operation

In its latest form, DNR consists of a single IC (National Semiconductor LM1894) for two channels, plus a number of external components (see sidebar). As shown in Fig. 1, a single control circuit regulates the filter action of both audio channels, which can vary in bandwidth from 800 Hz to as much as 30 kHz (-3 dB points). Maximum noise reduction (CCIR/ARM weighted) is in the neighborhood of 10 to 14 dB. The filters are single-pole configurations, providing a uniform 6-dB per octave roll-off above whatever corner frequency the voltage from the control circuit dictates (see Fig. 2 for operating parameters).

The control circuit itself derives a control signal from the rectified sum of the two channels. The circuit's response is not uniform with frequency, but increases at a 12-dB per octave rate from about 1 to 6 kHz, flattening out above. A threshold, sometimes fixed but user-variable in the case of one available outboard processor, establishes the noise "floor" of the system, determining what levels of high-frequency energy will be construed as noise (for which the filters will remain closed) and what levels as program (for which the filters will progressively open up). The filters can open (attack time) in as little as 0.5 millisecond, which is consistent with the sharpest transients to be expected in program material. Release time is a more leisurely 50 milliseconds, to avoid the foreshortening of any lingering reverberation.

Considered within the constraints of cost, simplicity and playback-only processing, the DNR scheme seems an appropriate solution of an inherently insoluble problem. The time constants (attack and release) are well chosen in terms of present-day psychoacoustic understanding, and the operational frequency bands are the right ones for maximum suppression of audible steady-state noise (hiss, in other words). Governing the action of the control circuit by higher frequencies alone is a particularly logical idea. It both focuses appropriate attention on the critical area, and avoids control-signal ripples that low-frequency information can impose on a peak-detecting circuit such as DNR employs.

National's LM1894 IC with its external components on a p.c. board.



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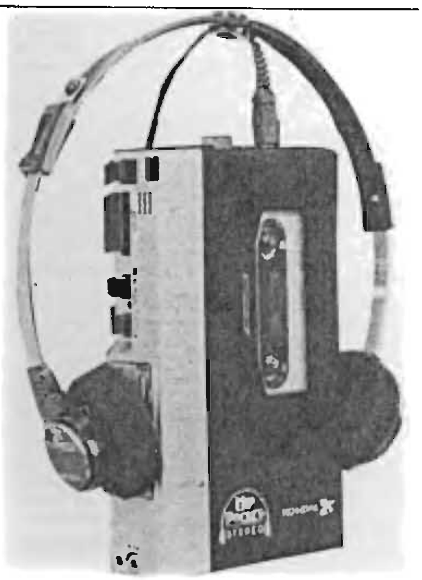
In common with any practical noise-reduction system, DNR depends on auditory masking of noise by program material occurring at or near the same frequency. If this masking does not take place when the filters open to pass high-frequency program, noise will be heard. Worse, noise will be heard going up and down in level with the filter action. Such noise modulation — and arranging for masking to conceal it — is the crux of noise-reduction system design. In less guarded moments, all responsible engineers admit that masking is bound to fail under some circumstances, and can be made to fail predictably if program material is chosen with that end in mind. A proper noise-reduction system considers typical listening fare first and foremost, and trusts the flaws won't loom too large when unusual spectral distributions of program overthrow the designers' expectations.

Program sources that could not previously accommodate noise reduction are obvious candidates for the DNR system.



Technidyne's Hip Pocket Stereo incorporates DNR circuitry.

The claims made for DNR in this regard are certainly not so extravagant as to strain credulity. According to Martin Giles, National's Manager of Consumer Linear Applications, the system will be at its best with material that has signal-to-noise ratios (again CCIR/ARM weighted) exceeding 35 dB for musical ensembles. Certain critical solo instruments may have to start with a S/N of 45 to 50 dB to avoid all masking failures and audible side effects. (These differences have to do with the longer reverberation times of spaces regularly used to record ensembles and the nature of ensemble playing itself.) DNR is not effective with impulse noises such as record clicks and pops; it may alter them in character, but it will certainly not remove them.

Summing the system up, Giles remarks that it will help most of the time, hurt in some rare instances, and not do much of anything audible in those cases where the program material is good enough to stand on its own. But when it is deemed desirable to switch it out, the system is fully out; with compander systems that encode the material, the system can never be fully eliminated once the recording is in existence. For program that is borderline, the threshold control (when provided) will enable the user to set his own compromise between maximum fidelity, minimum noise, and the instrusion of audible side effects.

The Destiny of DNR

DNR has existed for several years now in a two-IC form, and as such has found its way into several portable and home music centers. With the advent of the LM1894, DNR has been adopted by General Motors for use in 1982 car stereo systems, by Technidyne for its Hip Pocket Stereo, by Benjamin in its RAC-10 MK-II DNR cassette changer, and by Advanced Audio Systems in its stand-alone Model DNR-450. Program sources that could not previously afford or physically accommodate noise reduction are obvious candidates, along with new media that have not yet established noise-reduction standards. The company is also hopeful about broader applications and about a supplementary role to existing noise reduction. For example, a tape encoded by a compander system like Dolby B, even though properly decoded during playback, will still not be perfectly quiet if listened to at louder levels. But it will be much quieter if DNR processing is used as a further step in the reproduction chain.

To forestall misunderstanding, it should be emphasized that DNR does not decode Dolby noise reduction or the processing of any other compander system. It cannot return dynamically compressed program material to its original form. It acts only on steady-state noise but does so wherever it is found and whatever its origin. This means universality and compatibility with any source — factors National Semiconductor counts on to carry DNR into the mainstream of audio noise reduction.

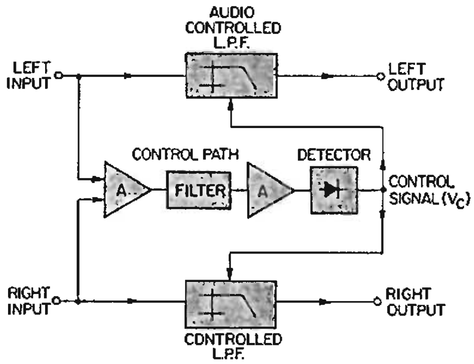


Fig. 1—A breakdown of the essential operators in the DNR system, all of which are contained within a single IC.

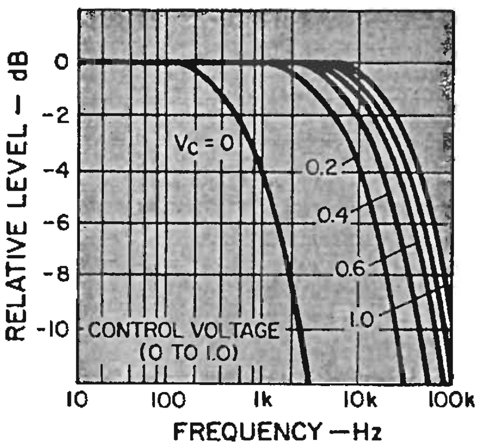


Fig. 2—Increasing control voltages, derived from a network that responds more to higher frequencies in the program, open the DNR passband until it extends well beyond the audio range.

NR

THE CIRCUIT

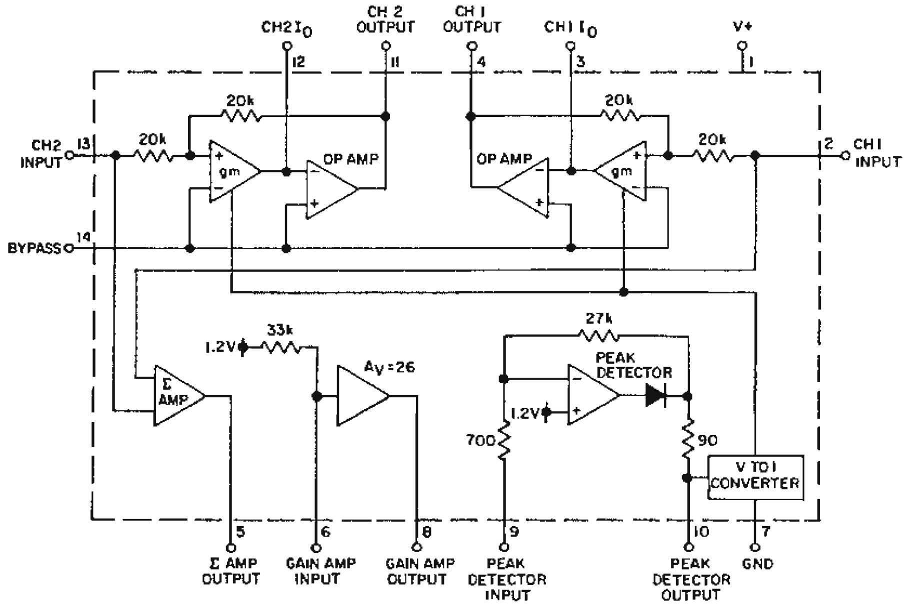


Fig. 3—Block diagram of the LM1894.

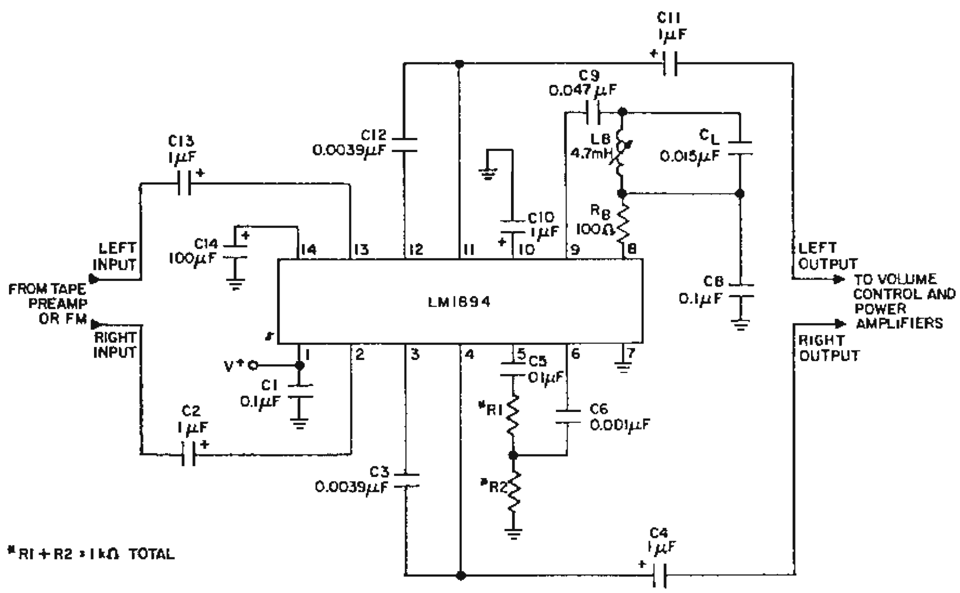


Fig. 4—Schematic of suggested external circuit for LM1894. IC pin designations correspond to those of Fig. 3.

The LM1894 is a 14-pin DIP intended to operate with supply voltages from 4.5 to about 18. Current drawn is 12 milliamperes for a typical supply voltage of 8. Input impedance is approximately 20 kilohms; input overload occurs at 1 volt rms.

Figure 3 is a block diagram of the IC itself; Fig. 4 is a suggested external circuit for the IC. The primary external operators for the audio channels are C3 and C12, which determine the bandwidths passed by the filters. Since bandwidth is inversely proportional to capacitance, the frequency range of the noise-reduction effect can be adjusted by changing the capacitive values. Capacitors C5 and C6 determine the band of program frequencies to which the control circuit responds — in this case roughly 6 kHz and above. The voltage divider formed by R1 and R2 sets the threshold of the control path, which is normally adjusted so that steady-state noise from the program source just begins to open the filters. Resistors R1 and R2 are altered together so that their sum always equals 1 kilohm. Wiring a suitable potentiometer in their place creates a threshold-varying control.

Coil L8 and the components surrounding it comprise a 19-kHz notch filter which prevents the stereo FM pilot signal from affecting the operation of the control circuit. If the DNR module will not be used for FM, or if the tuner has an adequate multiplex filter of its own, these components can be replaced by a simple 0.047-μF capacitor bridging pins 8 and 9.

National Semiconductor foresees and has demonstrated the use of LM1894s in cascaded arrays of two or three, in which case the slopes of Fig. 2 become 12 or 18 dB per octave, and the noise-reduction effect becomes 20 dB or greater.

The LM1894 is available in quantity to manufacturers of licensed products for about $2 apiece. The price is expected to decline as production increases. It is not presently available in small quantities or to unlicensed manufacturers.

HERMAN BURSTEIN

Understanding Equalization and Time Constants

Equalization is used in many, many parts of the audio chain as a standard practice, and indeed there are several equalization standards which *must* be followed closely by manufacturers if their equipment is to interface properly with other pieces of gear and with software. While we think of the "normal" response of this gear as "flat," in many cases we are comparing the unit against a standard which is not at all flat. One example of such a standard is the RIAA response curve established by the Recording Industry Association of America for phono stage response in preamplifiers, integrated amplifiers, and receivers.

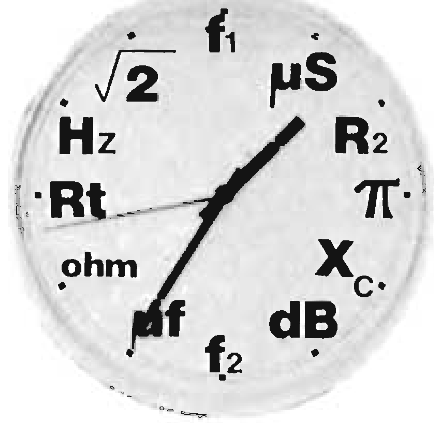
Broadly defined, equalization is the changing of a frequency band or range upwards or downwards in level, that is, changing its voltage level up or down. The precise industry definitions, which insure compatibility between parts of the audio chain, go much further in their accuracy, specifying exactly what the equalization must accomplish and using terms such as time constant and turnover frequency. A filter is the "how" of this equation, the way in which the equalization is accomplished, and filters are composed of resistors and capacitors, Rs and Cs.

For background, let us review the basics of equalization in home audio equipment — tape, phono, and FM. Two essential reasons for equalization are to improve S/N (signal-to-noise ratio), and to compensate for losses in a reproducing medium. Minimization of distortion is a further consideration in deciding how much equalization to apply and at what frequencies.

Industry standards for tape, phono, and FM call for specific playback equalization curves, usually defined in terms of those mysterious µS (microseconds) but also often defined in terms of turnover frequencies (sometimes called transition or corner frequencies).

Without any equalization, record-playback response of a tape would exhibit pronounced bass and treble losses. Factors responsible include the tendency of the output of a magnetic playback head to drop at the rate of 6 dB per octave as frequency declines. Treble losses on the tape are also due to the magnetic phenomenon of self-demagnetization, and due to application of bias in recording; such losses increase with frequency, with reduction in tape speed, with bias; and they vary with tape formulation. Treble losses, usually relatively small, also occur in the record and playback heads.

Therefore, a tape system requires compensating bass boost and treble boost to achieve flat response. Bass boost is supplied largely in playback, and should conform to a standard playback curve stipulated by the industry — which curve depends on tape speed.



Broadly defined, equalization is the changing of a frequency band or range upwards or downwards in level.

Treble boost is provided largely in recording There are no standard treble equalization curves because the required treble boost varies with bias and tape formulation. Hence, the industry standard simply requires that recording equalization, when coupled with standard playback equalization, shall produce flat record-playback response within a specified tolerance.

For a given tape speed and tape formulation (ferric oxide, chrome or chrome equivalent, and metal), the industry's choice of a standard playback curve is largely dictated not only by the loss factors described above, but also by the objective of minimizing both noise and distortion.

In the case of the phono disc, a standard amount of bass cut is applied in recording to avoid excessively wide groove excursions, which would limit the amount of recording time and would result in distorted output of the phono pickup at low frequencies. True, the groove excursions could be limited by recording a lower signal level, but this would reduce S/N. Further, to make possible an improvement in S/N, a standard amount of treble boost is applied in recording; an equal amount of treble cut in playback not only restores flat response but also attenuates noise, which is predominant in the treble region.

The simplest case is that of FM, where stations are required to provide a specified amount of treble boost in order to permit an improvement in S/N — in the same way as for phono discs. The FM tuner should therefore provide standard treble cut, which restores flat response and reduces noise. Standard treble boost (at the station) and cut (at the tuner) are less for a Dolby signal than for a non-Dolby one, with a view to minimizing distortion due to excessive FM modulation at high frequencies.

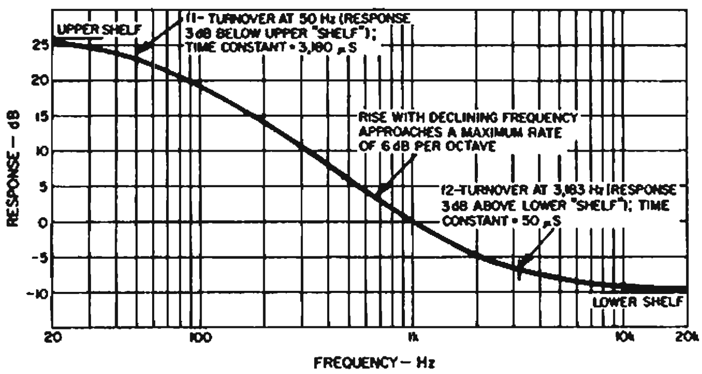


Fig. 1 — Standard playback equalization at 7½ ips, before correction for playback head losses. The 0-dB reference is the level at 1 kHz.

Turnover Frequencies

Equalization for a tape deck operating at 7½ ips serves to illustrate the meaning of turnover frequencies.

Figure 1 shows standard playback equalization, as prescribed by the NAB and the RIAA in the U.S., for a 7½-ips tape system. This curve may be described as either bass boost or treble cut, depending on whether we read it from right to left or from left to right. It is customary to refer to this, and other tape playback curves, as bass boost.

In Fig. 1 bass boost "starts" at 3,183 Hz, where it is 3 dB up. And it "ends" at 50 Hz, where it is 3 dB below maximum. As frequency declines, the curve rises at a rate approaching 6 dB per octave. Total bass boost is 36 dB.

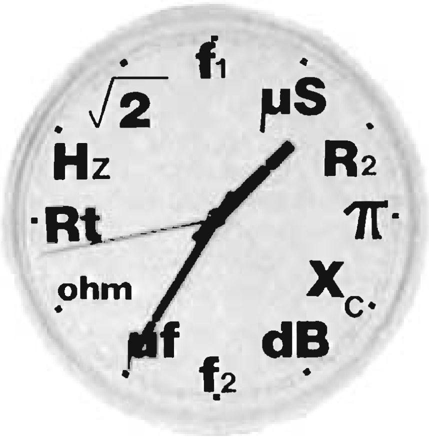
We refer to 3,183 Hz and 50 Hz as the turnover frequencies, respectively f1 and f2, which define the equalization curve. This is coupled with the understanding that the curve rises at a rate approaching 6 dB per octave. For other tape speeds, the turnover frequencies — where response is 3 dB above or 3 dB below a stated level — may differ from those for 7½ ips. At 3¾ ips, bass boost begins at 1,768 Hz (f2) and ends at 50 Hz (f1); at 1⅞ ips, it begins at either 1,326 Hz (ferric oxide tapes) or 2,274 Hz (chrome, chrome equivalent, and metal tapes), and ends at 100 Hz.

Table 1 — Time constants, t, and corresponding turnover frequencies, f

T, μs	f, Hz
25	6366
50	3183
70	2274
75	2122
90	1768
120	1326
318	500
1590	100
3180	50

From Turnovers To Time Constants

At the preference of engineers, standard playback curves are more often defined in terms of time constants than turnover frequencies. However, one definition is easily convertible into the other.



Two essential reasons for equalization are to improve S/N and to compensate for losses in a reproducing medium.

The physical meaning of time constant will be explored later. Here we will just state the simple mathematical relationship between turnover frequencies and time constants, enabling the audiophile to readily convert one into the other:

$$f = 159{,}155/t, \quad (1)$$

$$t = 159{,}155/f, \quad (2)$$

where f is frequency in Hz and t is time constant in μS.

The equalization curve of Fig. 1 is defined as having time constants of t1 = 3,180 μS and t2 = 50 μS. From Equation 1 we obtain:

$$f1 = 159{,}155/3{,}180$$
$$= 50 \text{ Hz},$$
$$f2 = 159{,}155/50$$
$$= 3{,}183 \text{ Hz}.$$

To take an inverse example, the standard FM de-emphasis curve for non-Dolby signals has a turnover frequency of 2,122 Hz (treble response 3 dB down at this frequency, and continuing to decline at a rate approaching 6 dB per octave). Equation 2 converts this into a time constant:

$$t = 159{,}155/2{,}122$$
$$= 75\ \mu S.$$

Table 1 shows the relationship between t and f for the time constants and turnover frequencies generally encountered in home audio systems.

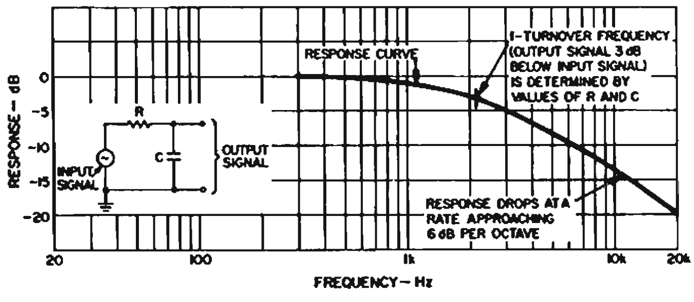


Fig. 2 — Treble loss produced by an RC circuit. (It is assumed that the source impedance of the input signal and load impedance of the output signal have negligible effects.)

Equalization Circuits

Tape, phono, and FM playback equalization customarily employ combinations of resistance (R) and capacitance (C) in what are therefore called RC circuits. To illustrate, Fig. 2 shows a simple but basic circuit containing one resistor and one capacitor. This circuit produces treble cut, such as needed for FM de-emphasis.

Before continuing we should review the terms impedance, resistance, and reactance. Impedance is a general term, denoting opposition to current flow in an electrical circuit. Resistance is the impedance presented by a resistor, while reactance is the impedance presented by a capacitor (or by an inductor, which will not be further discussed here inasmuch as inductors are seldom used for standard playback equalization, although are often used for other audio purposes such as equalizers, speaker crossover networks, etc.). The combined effect of a resistor and capacitor is called impedance; all types of impedance are measured in ohms.

A resistor presents equal resistance at all audio frequencies, while a capacitor presents decreasing reactance as frequency rises. Therefore C in Fig. 2 behaves more and more like a short circuit of the output signal as frequency rises; in other words, treble response declines. By choosing suitable values of R and C, a designer can achieve a desired turnover frequency at which response will be down 3 dB. In the case of a non-Dolby FM signal, the desired turnover is 2,122 Hz (or time constant of 75 μS). For a Dolby signal, the turnover is 6,366 Hz (25 μS).

By adding one resistor to Fig. 2, the designer can produce bass boost, for example that of Fig. 1. The revised circuit appears in Fig. 3. The added resistor, R2, is much smaller than R1; its purpose is to limit the extent to which the output signal can decline owing to the short-circuiting action of C as frequency rises. Instead of an endless decline in output signal with rising frequency (as in Fig. 2), the decline is halted within the desired part of the audio range. To produce the curve of Fig. 1, the R1, R2, and C components would be chosen to produce turnover frequency f1 at 50 Hz, and turnover f2 at 3,183 Hz; the respective time constants are 3,180 and 50 μS. (It may be noted that the circuit of Fig. 3 is only one of several ways of achieving bass boost.)

Time Constants

Let's refer again to the basic equalization circuit of Fig. 2, which produces treble cut. The turnover frequency — at which response is 3 dB down — occurs when the reactance of C equals the resistance of R. Why this is so is explained in the final section.

Capacitive reactance in ohms is $X_c = 1/2\pi fC$, where f is in Hz, C is in farads, and $\pi = 3.1416$. At the turnover frequency, since $X_c = R$, we thus have $R = 1/2\pi fC$. Transposing C, we obtain $RC = 1/2\pi f$ at the turnover frequency.

The time constant, t, is RC. We may simplify by substituting 3.1416 for π, yielding:

$$t = RC = 0.159155/f. \qquad (3)$$

In the circuit of Fig. 2, the input signal charges C through R. It can be shown mathematically or by experiment that RC is the time in seconds required for a constant voltage to charge C to 63.2% of this voltage. For example, if R = 10 ohms, C = 1 farad, and 100 volts d.c. is applied across the series combination of R and C, it would take 10 seconds to develop 63.2 volts across C.

In audio we generally use capacitances far smaller than one farad. A more convenient unit of capacitance is the microfarad (µF) —one millionth of one farad. Since the numerical value of C, and therefore of t, is then increased by a factor of 1,000,000, we must compensate by changing our unit of time from one second to one-millionth second (µS). Thus we obtain:

$$t = RC = 159{,}155/f, \qquad (4)$$

with t in µS and C in µF, but with R still in ohms and f still in Hz.

In audio the time constant is generally expressed as in Equation 4. To illustrate, if the circuit of Fig. 2 consists of a 75,000-ohm resistor and a 0.001-µF capacitor, the RC product is 75, and the time constant is 75 µS. It takes 75 millionths of one second to charge the capacitor to 63.2% of a constant applied voltage.

Of what importance is this to the audiophile? Well. . . the intent of this article is only to explain, not defend, the use of time constants in defining equalization circuits.

As previously pointed out, standard 7½-ips playback equalization, shown in Fig. 1, can be produced by a circuit such as Fig. 3. Turnover frequency f2 — 3,183 Hz, where response is up 3 dB — is determined by R2 and C in Fig. 3. At that frequency the reactance of C must equal the resistance of R2, and the product of R2 × C must be 50, with R2 in ohms and C in µF. If a constant voltage were applied to R2 and C in series, it would require 50 millionths of one second to charge C to 63.2% of the applied voltage.

R1, C, and R2 are all in series with the input signal. We may refer to R1 + R2 as Rt. Turnover frequency f1 — 50 Hz, where response is 3 dB below maximum — is determined by Rt and C. If a constant voltage were applied to Rt and C in series, it would require 3,180 millionths of one second to charge C to 63.2% of the applied voltage.

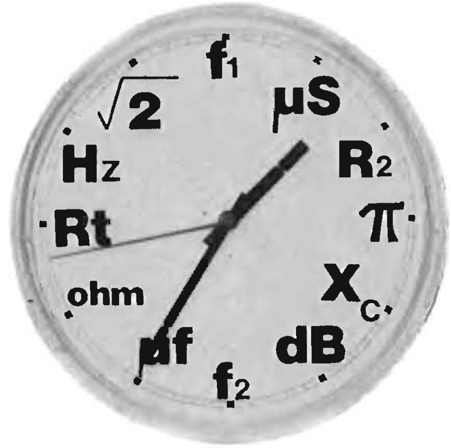
Turnover Response

Referring to Fig. 2, we stated earlier that the turnover frequency f occurs when the impedances of C and R are equal. Accordingly, the signal voltages across C and R are equal. It may seem that half of the input voltage appears at the output, namely across C. If this were true, there would be a 6 dB drop in response at f. Actually, though, the drop in response at f is 3 dB.

The proportion of the input signal appearing across C in Fig. 2 depends on the ratio between the reactance of C and the combined impedance of R and C; in short, on the ratio X_c/Z, where Z is the combined impedance. We cannot simply add R and X_c to obtain Z, because the voltages across R and C are out of phase. Instead we must use vector addition (akin to the manner in which we obtain the hypotenuse from the legs of a right triangle). That is, $Z = \sqrt{R^2 + X_c^2}$. Since $X_c = R$ at turnover, we obtain $Z = \sqrt{X_c^2 + X_c^2} = \sqrt{2X_c^2} = X_c\sqrt{2}$. At turnover, the signal across C is proportional to $X_c/X_c\sqrt{2} = 1/\sqrt{2} = 0.7071$. And 0.7071 of the input signal represents a drop of 3 dB.

Referring to Rt and C in Fig. 3, for the same reason we obtain a 3 dB drop in response at turnover f1, when the reactance of C equals the resistance of Rt. And for similar reasons we obtain a 3 dB rise in response at turnover f2, when the reactance of C equals the resistance of R2.

A review of high school math is not at all what was intended here; rather, we've gone through some of the basic formulas to show how engineers can determine how their circuits will respond with very good accuracy — before they ever put the circuit on the test bench. Whether or not you ever work out one of these problems for yourself, a basic knowledge of what the circuit designer intends, as well as the tools he actually uses in a circuit, goes a long way towards providing a good understanding of many parts of the audio chain.



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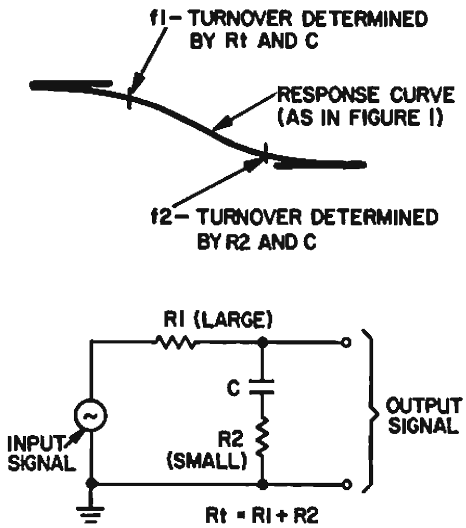
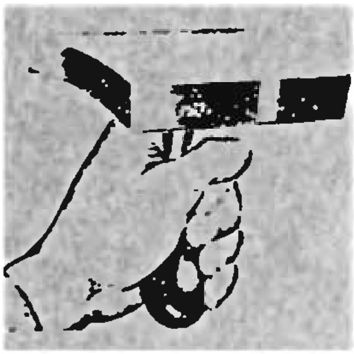


Fig. 3 — Bass boost produced by an RC circuit. (Assumptions about source and load impedances as in Fig. 2.)

TAPE RECORDER MAINTENANCE

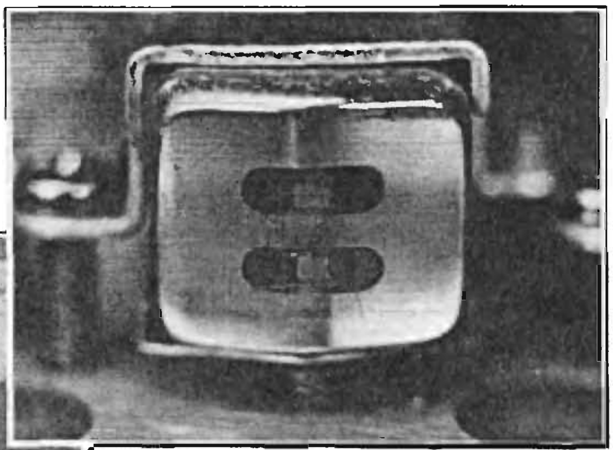
Howard A. Roberson



Just as with the purchase of a home. a car. or a major appliance. your responsibility as an owner of audio equipment does not end with your selection of components. To get the maximum pleasure from your system for the maximum number of years. you must periodically check its condition and perform routine maintenance tasks. The techniques and tools for proper care of open-reel recorder heads will be covered here. and much of this can be directly applied to cassette decks.

Photographer Robert Lewis

Fig. 1—Worn and dirty quarter-track record head.



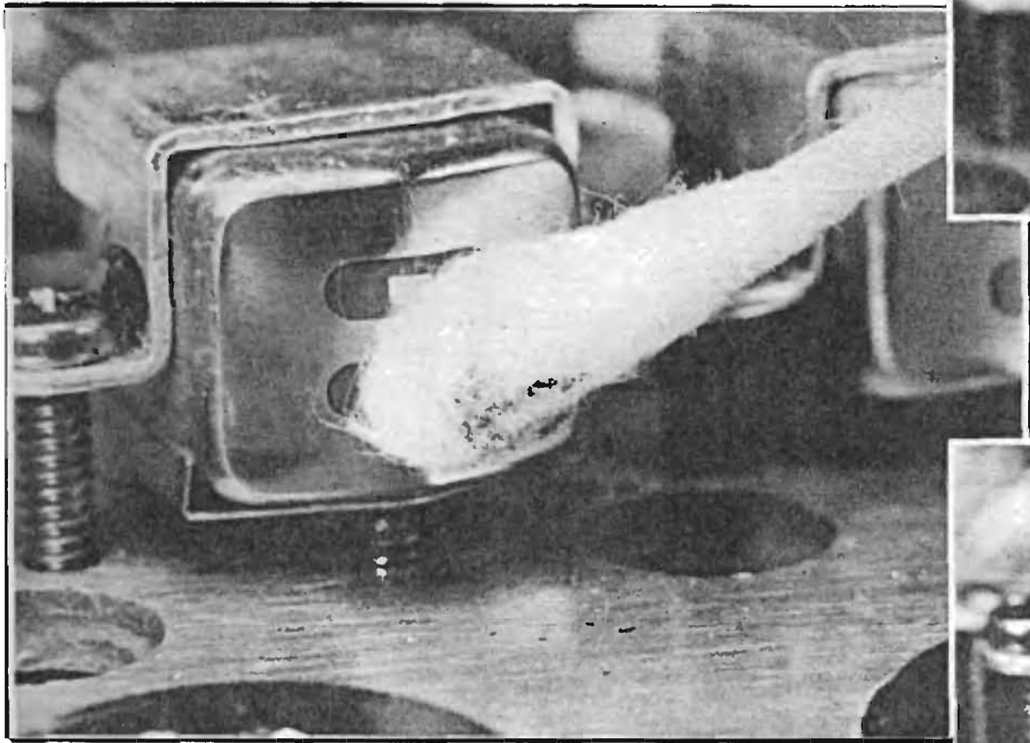


Fig. 2—Partial removal of Aud Vid Com "A" coating.

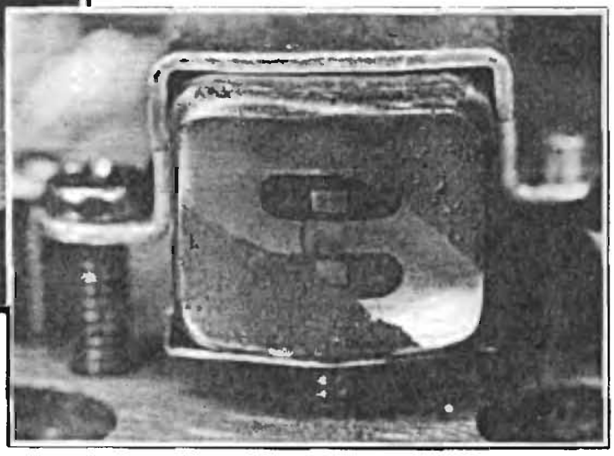
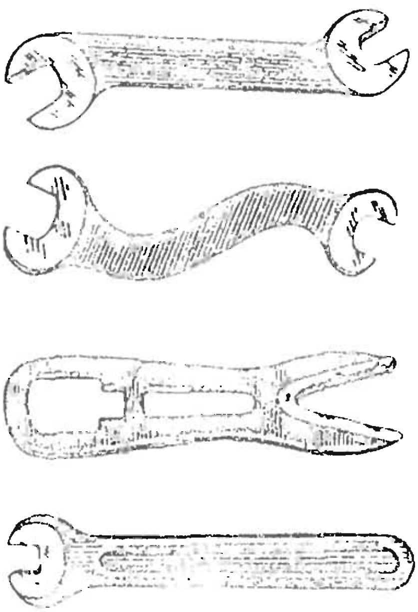


Fig. 3—Dirt removed by rotating a cotton swab.



Keeping Heads Clean

Maintaining clean heads on a recorder demands that the entire tape path be kept clean. If you take a good look at the route the tape follows from the supply reel all the way to the take-up reel, you'll find tape-tension/tape-break arms, guides and posts, the capstan, the pinch roller, perhaps a pressure pad, and a lamp-detector assembly to sense tape run-out. Any of these points is a potential source of scraping the tape surfaces or edges and a potential site for build-up of debris. Figure 1 is a close-up picture of a quarter-track head that is both dirty and worn.

Cleaning Products: Many different types of cleaners have appeared on the market, but most of the present formulations are based upon isopropyl alcohol or TCF (trichlorotrifluoroethane). The advantage of the alcohol is that it is inexpensive and readily available at a drug store. You must be certain, however, that you do not buy rubbing alcohol, which may have such harmful ingredients as lanolin added to the alcohol base. The desired product should be clearly labeled "91% isopropyl alcohol," as is Stericol from Berkeley Drug Co. Alcohol will probably not leave any residue on the heads, but it can be damaging to rubber pinch rollers over the course of time. There is no such disadvantage with TCF, which also will not attack plastics.

I evaluated tape-head cleaners made by Aud Vid Com, Chemtronics, Miller-Stephenson and Nortronics, as well as Stericol. Each product was rated for the ease with which it cleaned the deposits on the head and the extent to which it left the head completely clean. A few unnamed isopropyl-based cleaners did a good job of removing the tape residue, but they left behind a thin layer of something white and streaky. There was no similar deposit from Stericol, available at a fraction of the cost. TCF products, in spray cans from Chemtronics and Miller-Stephenson and in bottle/cans from Miller-Stephenson and Nortronics, all did the task in excellent fashion with no detectable residue.

Aud Vid Com is a two-fluid type of cleaner which the manufacturer claims will polish as well as clean heads. The "A" fluid is applied, allowed to dry, and then (Fig. 2) wiped off. The "B" fluid is used for the final conditioning. The two-step process made for good cleaning, but no better than any of the TCF cleaners, which involve only one step and cost less. Examination of both old and new heads under a microscope showed that Aud Vid Com did fill in some minor scratches in one or two cases, but in most cases there was no observable improvement.

There was little difference among the top cleaners, but my favorite was the Miller-Stephenson, followed very closely by Nortronics, and then by Chemtronics.

Tools and Techniques: I have a number of head-cleaning kits, and they do perform well. Those with angled-handle felts are particularly good for recorders with recessed heads, such as the Revox A77. For most recorders, however, I use cotton swabs, the Chesebrough-Ponds Q-tips, which are well made and won't keep dropping fibers while you clean. Although I occasionally use a spray can, I really prefer to use the liquid TCF in a small, pump-spray (empty Sound Guard) bottle so that I can spray the exact amount I want directly on the Q-tip.

Most manufacturers of cleaners tell the user to "scrub" the head, but *don't* do that: You might scratch the head if hard particles are present. Let the cleaner do its work—loosening the debris for easy removal. Move the Q-tip across all parts of the head while rotating the swab (Fig. 3), as this will lift the dirt off the head and remove it. Many times it is best to use the first swab for rough cleaning of all the surfaces, including the guides and pinch roller. Use as many Q-tips as necessary to make certain that you are not re-depositing dirt already removed.

Demagnetization

Despite improved circuitry in recorders the past few years, it is still possible that there might be some residual magnetism which could be detrimental to an important recording. Demagnetize after every 20 hours of use, before attempting any important recording, and before *ever* playing an alignment tape, if you wish to be absolutely safe. Specific recommendations by the recorder manufacturer should be followed faithfully.

Demagnetizers: Figure 4 shows two Nortronics demagnetizers with bent-tip rods, an old and inexpensive flat-pole-piece unit from Lafayette, and the rugged Annis Handi-Mag. All but the Lafayette have the ends covered with plastic, which is essential to ensure that the heads are not scratched during demagnetization. The Handi-Mag put out the highest flux level, making it my first choice, particularly when there are guides that would benefit from such attention. The large size of its pole pieces, however, prevent its use with recorders with limited head access. In such cases, the Nortronics units can induce more flux into the heads since direct contact can be made.

Demagnetizing Techniques: It is important to realize that demagnetizers will also work on VU meter pole pieces and alignment tapes, so don't be careless. The meters will *not* be damaged by the demagnetizer if used in normal fashion. Make it a practice to turn the unit on and off at least four feet from the recorder. After turn-on, bring the demagnetizer to the head (or guide) slowly, make gentle contact, and then move it slowly away. Do each element in turn, including separate track pole pieces if need be, always moving the demagnetizer smoothly. If you need to turn it off before finishing, move it away first. If you accidentally turn it off at a short distance, do all the demagnetizing over—again referring to the manufacturer's instructions.

Head Alignment

For the serious audiophile or semiprofessional, head alignment can be a prime concern, particularly if the heads need to be replaced. When all of the heads are in perfect alignment, their gaps are perpendicular to the reference mounting plane and the line of tape travel (azimuth), and the head faces are also perpendicular to the plane (zenith). Immediately, a reader may well wonder where that reference plane is on his recorder. Sometimes it is very obvious for there is a flat, metal plate providing support for everything involved in the tape path. With other recorders, there may be just a small plate supporting the head assemblies.

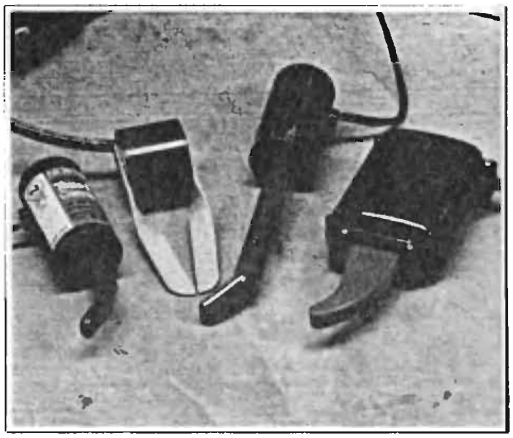


Fig. 4—Head demagnetizers.

Fig. 5—An assortment of tools for head alignment.

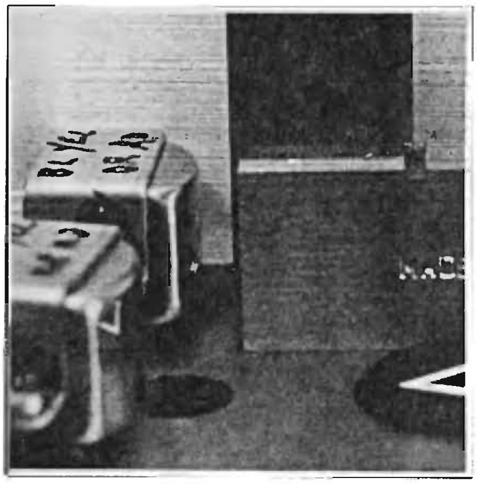


Fig. 6—Using the machinist's square to make the play head face perpendicular to the mounting plate.

What we do know is that the playback head should be in alignment with a standard test tape, the record head should be in alignment with the playback head, and the erase head should match as well. We can all agree on the basic approach, but let's take a look at the tools and techniques.

Tools and Instruments: Figure 5 shows a collection of tools and other aids for head alignment. Going clockwise from the lower left, we see an Ivie IE-20B pink-noise generator, nail polish, a small machinist's square, Nortronics Magview, a flashlight, head cleaner, a Magnetic Reference Laboratory test tape, an Ivie IE-30A ⅓-octave RTA, a pull scale, eraser pencils, and, finally, a steel scale with graduations for each 1/100 inch.

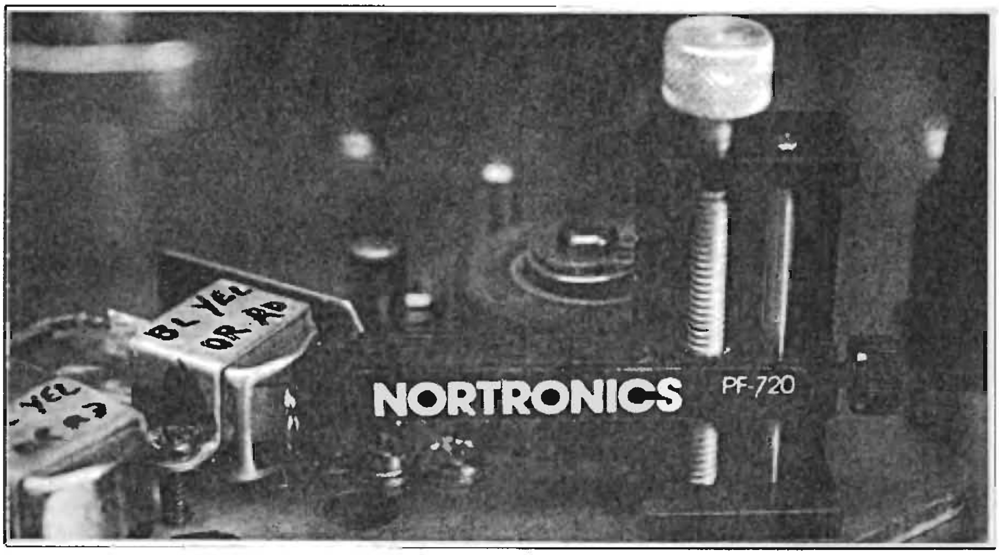


Fig. 7—Using the Nortronics PF-720 height gauge.

There are other things that will be used, of course, including an oscilloscope and audio monitor, jeweler's loupes, to say nothing of a sine-wave source. If the entire head assembly is removable, some work might be most easily accomplished on a Pana-Vise surface plate. Various heights can be checked and transferred with a Nortronics PF-720 height gauge. Additional tools may be helpful at times, but it's time to discuss the procedure for doing the alignment, including what must be done if heads need replacing.

Checking Head Alignment: The first steps are to clean and demagnetize the tape path and to make any adjustments needed on tape tensions. Look for any rough or sharp edges which might call for repair work before putting an alignment tape in your recorder. Clean any pressure pads with light brushing; do *not* use cleaner. Replace any pads that are hardened and/or packed with oxide. The MRL swept-sinusoid test tapes are my favorites for two reasons: The swept response that is shown on the scope can be used for both playback equalization and head azimuth adjustments, and their length of several minutes gives plenty of time for both of these tasks. Use the azimuth adjusting screw to get the maximum output at the highest frequency. Alternatively, make the adjustment for exact (as possible) phase correspondence between tracks, using a two-channel scope. (I am assuming the reader is most interested in two-channel stereo, with quarter-track recording format preferred.)

After playback head alignment, the record head needs to be checked. I prefer to use a pink-noise source at −10 to −20 VU, with the playback fed to the ⅓-octave RTA. The azimuth peaking is done to get the maximum response from the 20-kHz filter, gently centering the adjustment between the initial fall-off points. The RTA display also allows making bias and record EQ adjustments at the same time for the best overall response. Discrete tones can be used, of course, with a low frequency for a rough azimuth setting, shifting to a higher frequency for better resolution. At the same time that all of the alignment checks are being made, observe the stability of the outputs as well as the amount of the high-frequency roll-off. What you see will indicate how soon you might have to replace one or more of the heads.

Replacement of Heads: When there is unacceptable high-frequency roll-off after alignment, and it can't be corrected with bias or EQ adjustments, and/or the levels are bouncing up and down, it is time to think about changing the heads. Take a good look at each head face, aided by plenty of light and perhaps some sort of magnifier. See if there is a definite area that has worn away, perhaps on the order of ⅛ inch wide or more. If so, there will be little shoulders where the top and bottom edges of the tape would normally be. These are points where the tape can be lifted away from the head gap(s) by the unworn shoulder when the tape shifts slightly up or down. Look at the wear areas of all heads very carefully. The edge of the wear pattern should match the upper (or outer) end of the track 1 (left channel) pole pieces and their gaps.

Many times it makes sense to replace all of the heads, even though one may show little wear. The advantage of replacing them all is that the tape path will then be determined by a smooth flow across the new head faces and not by the shoulders in a somewhat worn head. *Before removing old* heads, make notes on all wiring and its color coding. Carefully clean all terminals of the new heads using the pencil erasers. If possible, do not loosen *any* screws that affect the head height. Do note the location and purpose of each screw associated with the head assemblies and write down the number of turns given to any of the screws and determine their pitches (threads per inch).

When a new head is placed in its support carrier (or holder), make certain that it has the same in-out, left-right position as before to ensure getting the correct wrap of the tape on the face. Return all screws to their original positions, adjusting the azimuth screws to make the gaps perpendicular, perhaps aided by a small square. Then, as shown in Fig. 6, use the square to make certain that the face surface is *perpendicular to the plate—placing a* white card behind the head will help you see exact verticality when the square is right up to the face. The Nortronics height gauge, preset to match the older heads, can verify that the height is still correct after any head tilting in zenith to match the square (Fig. 7).

The next step is to find out how the tape actually lies on the heads as far as height is concerned. First, record a high-level (+3 VU), low-frequency (400-Hz) tone on a bulk-erased (or new) tape on all four tracks at one end of the reel. Pull out a section that is recorded and spray it with Magview, oxide side, of course. When the fluid evaporates, there will be a pattern on the tape showing the location of the recorded tracks. If you have been careful, and maybe a little lucky, the pattern will look like Fig. 8, which shows three *equal spaces among the four recorded* tracks. If the spaces are quite different, you will need to move the record head: Up, if the space in the center is larger, and down, if the space in the center is smaller. It is possible to measure the actual spaces directly with magnification and an excellent scale, but it is tricky.

Fortunately, there is a much easier method. Look to see what the ratio is between the outer spaces and the middle space, and then refer to Fig. 9 to determine how much the head should be shifted. *For example, if the top (and* bottom) space appears to be twice as wide as the middle space (TS/MS = 2), the head should be moved down 0.005 inch to put it at the correct height. If the height-positioning screws have 4-40 threads, a full turn would mean a shift of 0.025 inch. We want, therefore, to turn the height-adjusting screws just one-fifth of a revolution. Turn them exactly the same amount to keep the head face zenith setting correct. Recheck with Magview, and do any trimming required to adjust the height exactly.

Record high-level tones on each of the four tracks on bulk-erased tape. With both scope and audio monitors, check for proper erasing, one track at a time. Check the separation between channels, but pay special attention to any crosstalk between tracks 2 and 3 (right channel with opposite play directions). *Disconnect the source while* these tests in playback are being made to prevent any source-to-tape leak-through from appearing as a head-height problem. If erasure is poor or if there is crosstalk, examine the erase and/or playback head heights with the tape running. Shift the head(s) involved to make the tape-edge/end-of-gap alignment more exact, and rerun tests. When completed, perform playback and record alignments as discussed earlier, *after* demagnetizing the heads. Finally, apply a spot of nail polish to lock all azimuth adjusting screws in place.

The instructions and guidelines described here should not involve an inordinate amount of your time. But by following these basic steps, you will prolong the useful life and fidelity of your open-reel recorders and cassette decks, and I hope that the information *I've provided will serve you well.*

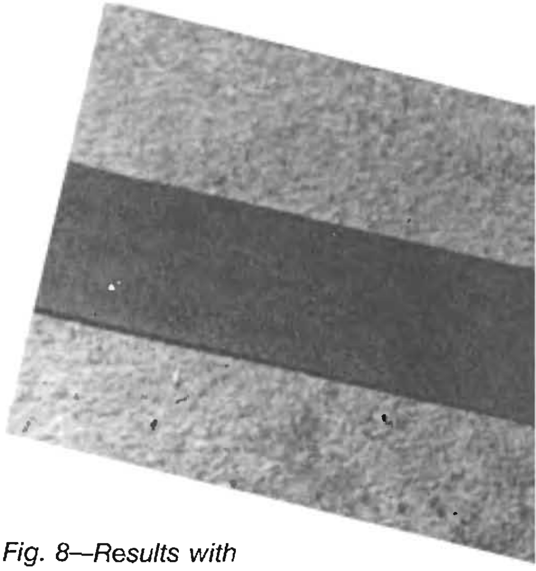


Fig. 8—Results with Magview after record-head height adjustments.

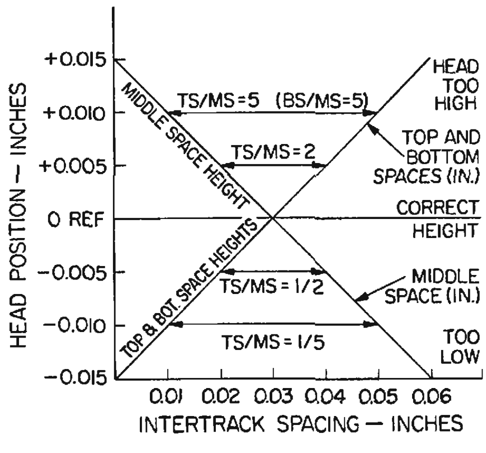
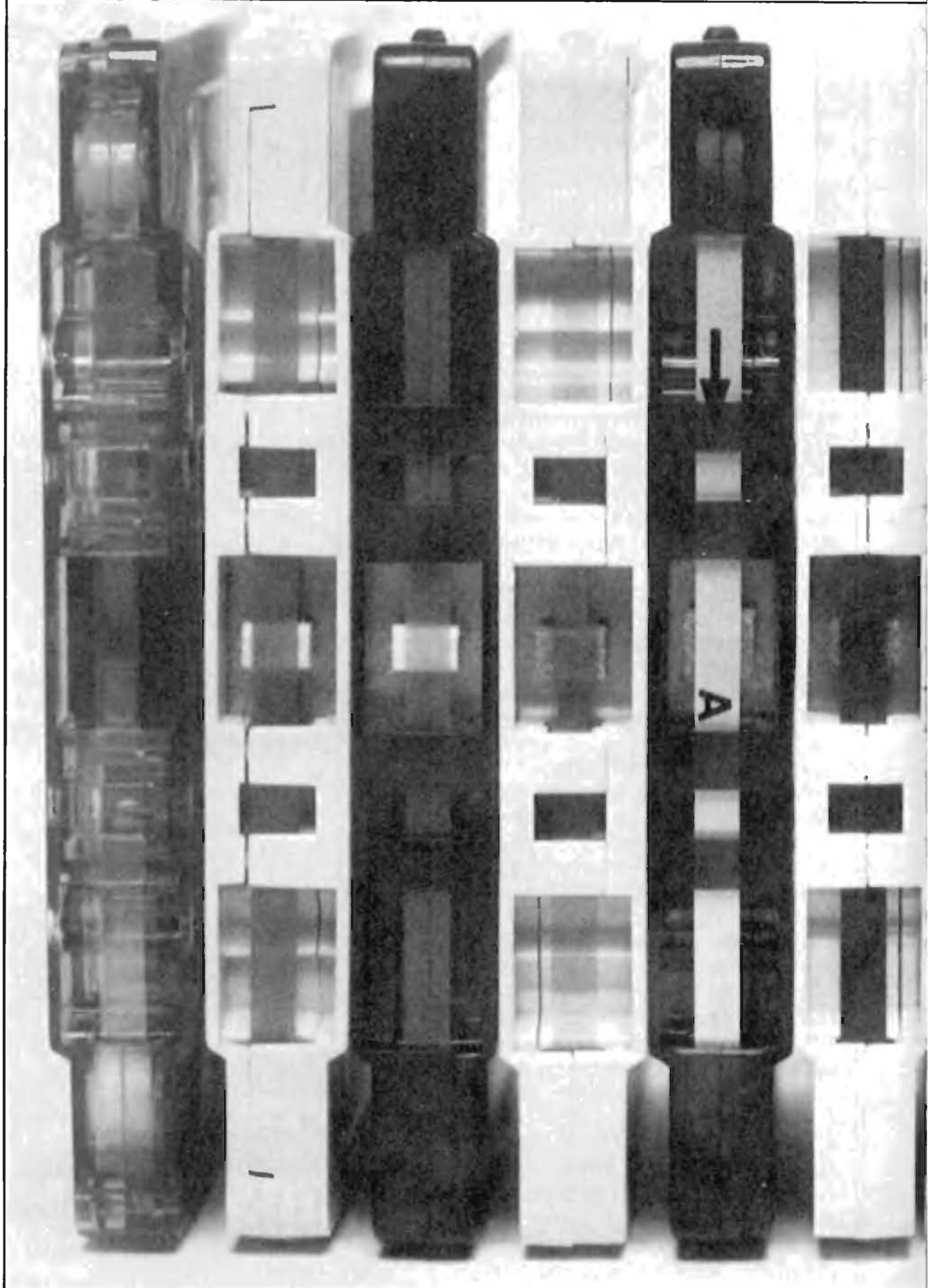


Fig. 9—Intertrack spacing as a function of vertical head position.

HOWARD ROBERSON

BASICS OF TAPE PERF



Tape tests and comparisons are aimed at pointing up performance differences between specific tapes. But the fundamental similarities between tapes are worth examining, too; they are the forest of context for these individual magnetic "trees."

Magnetic Properties

The most fundamental diagram of tape performance is the hysteresis loop, which shows the tape's response to magnetizing and demagnetizing forces. Figure 1 shows loops for three cassette formulations (loops for open-reel tapes have the same basic shape). The plots show flux density as a function of magnetizing force. For each tape there is a saturation point, where the flux density is at a maximum (B_m), and further increases in the applied field will not induce more flux.

If the applied force is reduced from the saturation point to zero, the flux on the tape does not return to zero. This residual flux density (B_r) is called retentivity; without it, there could be no tape recording, as the flux would disappear from the tape when the recording force was removed.

The B_m and B_r points are marked for the metal tape, and the corresponding points for the other two tapes are easily seen. The "squareness" ratio listed on many technical data sheets is the simple ratio B_r/B_m.

If a magnetic field of reverse polarity is applied, the flux will be reduced—all the way to zero, if this negative force is sufficient. The force required to reduce flux to zero is called the coercive force, or coercivity, and is designated as $-H_c$.

We can see the possible advantage in having high saturation flux densities, though that requires higher magnetizing force—a possible challenge to the recorder designer. Coupled with a high squareness ratio, a high B_m provides high retentivity. The high B_m of metal tape (more than twice that of the

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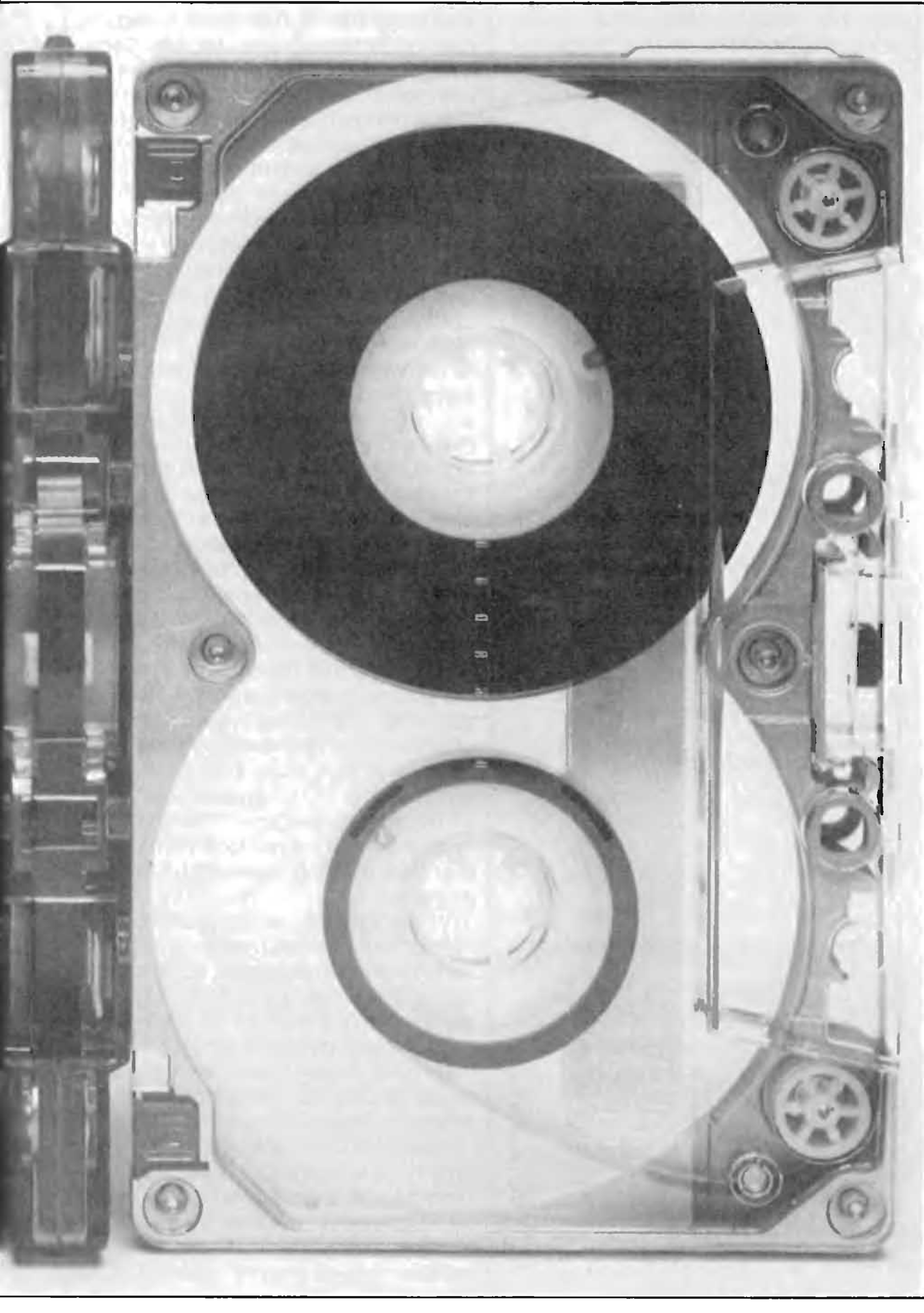

other tape) is quite noticeable in the figure. Quite a few Type I (normal ferric) tapes, however, have retentivities that match those for Type II (high bias) tapes.

Metal (Type IV) tape's advantage extends to its much greater resistance to erasure; Type II tapes are also superior to Type I formulations in this respect. Coercivities are on the order of 1100 oersteds for Type IV (metal), 350 to 650 for Type II, and 280 to 375 oersteds for Type I. These figures are actually developed with d.c. magnetization and demagnetization. Figure 1 also includes two dashed, demagnetization-loss lines, showing the extent to which the saturation loop's B_r might be reduced at higher frequencies by short-wavelength demagnetization effects such as self-erasure. Because of its low coercivity, Type 1 tape is much the most vulnerable to erasure of high frequencies by small magnetic fields such as might come from magnetized heads.

While Fig. 1 shows the tape's performance as it is cycled through saturation, it is, of course, common for magnetization and demagnetization to occur with smaller magnetic fields, resulting in lower flux densities, as in normal recording. A series of nested hysteresis loops, at flux densities from zero to saturation, form the natural magnetization curve. It is most linear in its central region, away from both zero and saturation. A.c. bias effectively linearizes the magnetization curve, and this linear range covers about ± 0.3 B_m. In the 10-dB range from ± 0.3 B_m to $\pm B_m$ (saturation), the curve is nonlinear, and overload distortion increases with increasing level. Examination of the signal-to-remanence curves of several tapes showed researchers that these curves were a nearly exact match, when normalized. From this, an equation was derived have a headroom of 10 dB between the point where the level of third har-

"The fundamental tape performance diagram is the hysteresis loop showing response to magnetizing and demagnetizing forces."

monic distortion (HDL_3) was 1% to saturation. This does not mean that all tapes have the same distortion level vs. recording levels; it does say that changes in recording level will cause the same *relative* changes in distortion.

Effects of Bias

It is common knowledge that high-frequency response and distortion change with bias level. So do several other tape characteristics. Figure 2 shows how the characteristics of one tape, Nakamichi SX, change as bias current is varied. As bias is increased above 1.2 mA, the 3% distortion limit (MML) at 400 Hz increases steadily, but the MOLs (maximum output levels or saturation limits) for 15 and 20 kHz drop a great deal. Distortion at 400 Hz with a flux level of 200 nWb/m decreases very sharply at first, then more gradually, reaching a minimum at about 2.9 mA of bias, then rising as bias is increased beyond that. Bias-noise level changes very little with bias current.

It is not surprising that many audiophiles get somewhat confused when they see such a data sheet, because it's hard to reconcile the idea of flat response with the great discrepancies between the 400-Hz MML and the 15- and 20-kHz MOLs. In actual recording and playback, though, equalization is used to flatten the response. The main point to keep in mind here is that bias is a fundamental factor in frequency response and distortion.

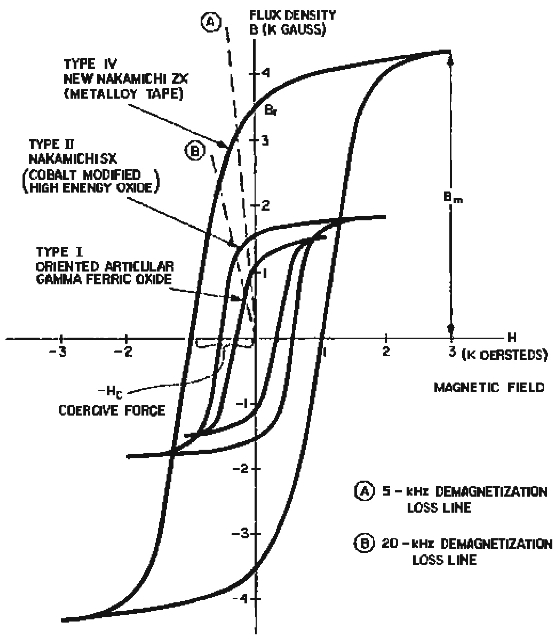

Fig. 1—Hysteresis loop characteristics of three cassette tape formulations.

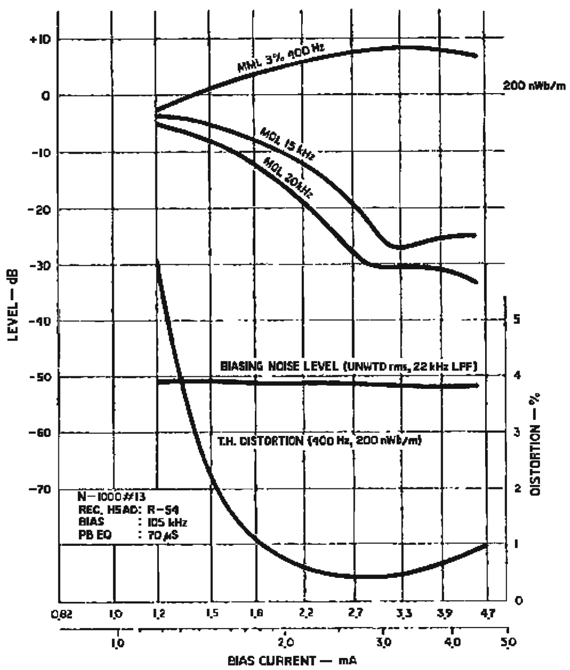

Fig. 2—Effects of bias change on Nakamichi SX tape.

Frequency Response

In general, the perfect amplitude response vs. frequency curve would be exactly flat with no deviations over the entire audio range. (There's little advantage, of course, in responding to frequencies below the lowest musical frequencies—music-search systems excepted—or to noise frequencies above the audio band.)

In the past, tests have shown that by adjusting bias (and equalization), it was possible to make the frequency respose of almost all tapes of any given type (I, II or IV) almost exactly the same for a given deck. It seems quite impossible, therefore, to state that any one tape does or does not have flat response.

While both bias and equalization affect frequency response, recorders which offer front-panel equalization trim adjustments are a rarity, with designers leaving manual EQ adjustment to the manufacturer or service technician. The deck comes with bias and equalization either set up for the manufacturer's choice of tape or under the control of an automatic adjustment circuit. If a user control is included, it will probably be a bias trim adjust; and if the user needs to do a little response trimming, he will probably use bias to do it.

"High-frequency response and distortion as well as other tape characteristics change with bias level."

But bias affects some inter-related problems. First, bias level has a direct relationship with distortion levels, and there is no direct indication of what is happening to distortion when bias is used to trim the response. Another potential problem is that, with many cassette decks, the best noise-reduction tracking may be obtained with the response somewhat un-flat. Recording sensitivity also affects noise-reduction tracking, of course, which is a further complication in selecting tapes and compensating for the differences from the manufacturer's set-up tape. The user is often best advised to try tapes which match the set-up tape's bias needs and sensitivity, to minimize response deviations when using noise reduction.

Recording Media Requirements

Matching the recording medium to the characteristics of the material being taped is a basic challenge. Figure 3 shows two curves of music spectra. One is an average of measurements made by Richard C. Cabot and others of levels and spectra in rock music. The other is based on measurements by Daniel Queen of maximum levels in live music. Placed somewhat arbitrarily on the figure's reference levels, notable similarities can be seen between the curves. Of particular interest is the fact that the fall-off in levels above roughly 1 kHz is close to a −6 dB/octave rate. Tape MRLs show the same slope, as we noted previously.

Work on orchestral instruments by David A. Luce has shown that their upper spectral envelopes have roll-offs of 10 dB/octave and more, with corner frequencies as high as 1300 Hz. An averaging of his curves would show a rough correspondence to Fig. 3, but a steeper roll-off above 1 kHz.

Figure 4 plots maximum octave-band levels of a rock-music FM station and two Mobile Fidelity recordings. The plots (made with an Ivie IE-30A analyzer in its accumulate mode) were deliberately shifted vertically to place their upper-end slopes close to the same reference −6 dB/octave line. The broadcast signal appears rolled-off at both ends. The opening of Respighi's "Feste Romane" contained considerable low-frequency energy, with a general drop in level above that region. The Hall and Oates record had energy distributed across the audio band, with the region around 500 Hz more elevated. It can be seen that if the same high-end criterion is used for all three cases, the mid-band levels will differ greatly for each—and if levels are set for equal mid-band energy, high-frequency levels will differ just as greatly. How, then, can the user set levels correctly to prevent distortion at the high-frequency end?

As a first step, let's consider how recording level meters respond to broadband signals such as music or, for testing purposes, pink noise. A meter that will register a zero-dB level when fed 0.5V at 400 Hz, might register −3 dB for pink noise at the same 0.5V level. The reason is this: The meter will read all the energy in a discrete tone, as long as that tone is within the meter's bandwidth. But if the signal's bandwidth is wider than the meter's, or

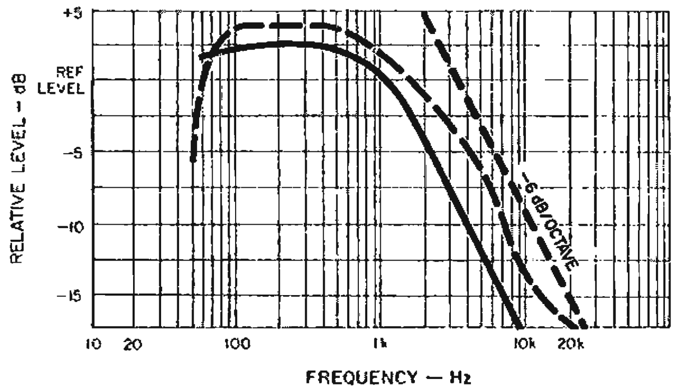

Fig. 3—Music spectra, after Cabot et al. (solid line) and Queen (dash line).

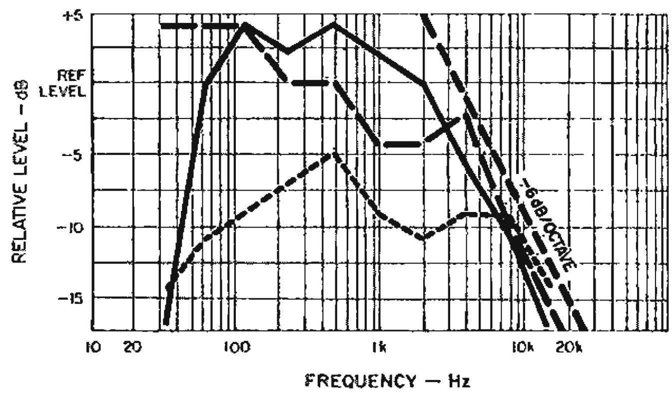

Fig. 4—Maximum octave-band levels from rock-music station (solid line), and Mobile Fidelity versions of Respighi's "Feste Romane" (dash line) and Hall and Oates's *Abandoned Luncheonette* (semi dash line).

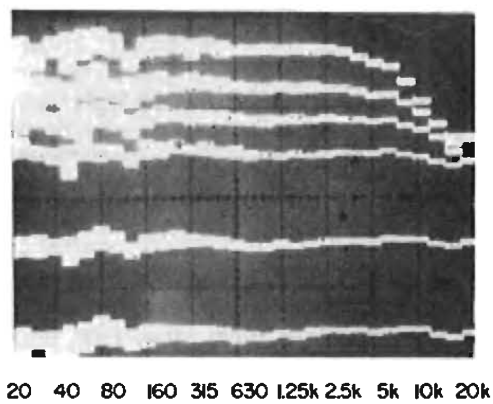

Fig. 5—Spectrum analysis of pink-noise response at +14, +8, +4, 0, −10 and −20 dB recording level.

"Although the meter will indicate the total energy in the music, knowledge of its spectral distribution helps prevent problems."

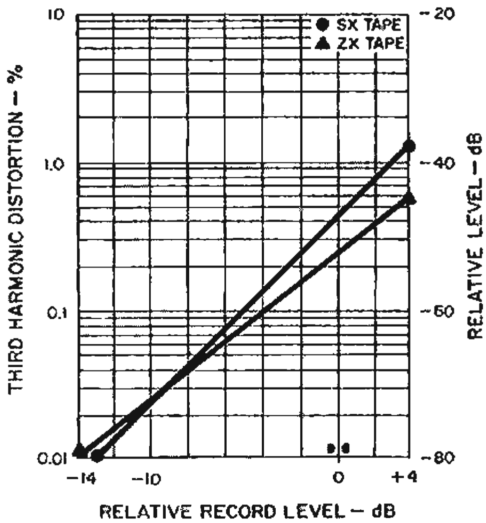

Fig. 6—HDL_3 as a function of recording level at 1 kHz with Nakamichi SX and ZX tapes on 700ZXL recorder.

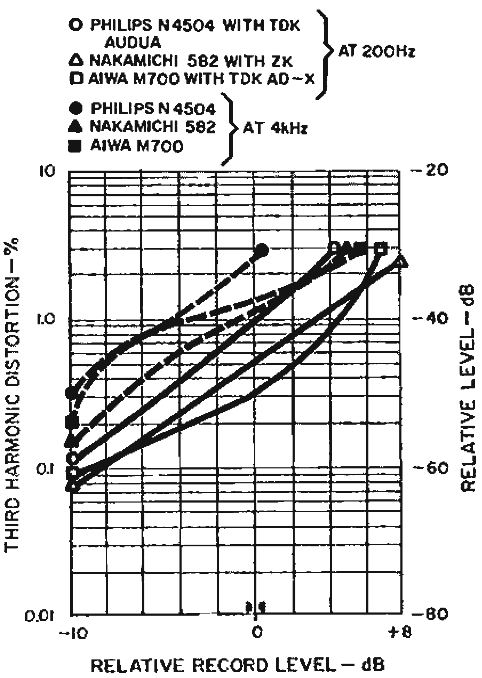

Fig. 7—HDL_3 vs. recording level at 200 Hz and 4 kHz for three recorder-tape combinations.

falls partially outside it, the indicated level will be lower.

The total level for a wide-bandwidth signal is the sum of the power levels of whatever individual bands (octaves, third-octaves, etc.) we are considering. This fundamental fact is one of the reasons that sine-wave testing is of limited use in testing for headroom. The high-frequency test tone exists *without* the presence of energy across the band, as would occur in music.

Figure 5 shows a recorder's record/playback response for pink noise recorded at levels from –20 to +14 on the recorder's meter. Some high-end roll-off is apparent by 0 dB, and the overall compression is obvious by +14 dB. At meter zero, the actual level in each band was close to –15 dB; with 30 bands, the total level is 14.8 dB above each individual band.

Referring back to Fig. 4, with the levels set as indicated by these plots, the rock station produced a +10.5-dB reading, the "Feste Romane" record got an indication of +11.5, and the Hall and Oates recording showed -3 dB. Considering the MRLs of several tapes, we would say that reducing the rock station's input level to normal meter indications should give us plenty of leeway on the high end. A similar reduction would be called for with "Feste Romane," with the possibility that some recorders would still distort its lowest frequencies. The Hall and Oates record is the deceptively challenging one: It has such a nice, smooth sound that the width of its spectrum is not obvious. We can see, however, that increasing its recording level even up to just zero would probably cause some loss at the highest frequencies.

To summarize this section: Although the meter will indicate the total energy in the music being recorded, knowledge of its spectral distribution should help prevent problems at the band limits. Listening carefully to the playback will help in setting record levels high enough for good signal-to-noise ratios.

Distortion

As many readers know, third-harmonic distortion is a fundamental characteristic of magnetic tape. Fifth and higher-ordered odd harmonics also make their appearance at high recording levels. (Even harmonics are usually not characteristic of tape recording. HDL_2, though, does show up regularly in testing, probably from poor bias waveforms, though possibly from leaky capacitors or magnetized heads as well.) The relative third-harmonic level (HDL_3) is a square function; when plotted in dB versus recording level, we can expect a 20-dB change in distortion for every 10-dB change in recording level.

Figure 6 shows measured HDL_3 vs. recording level for SX and ZX tapes on a Nakamichi 700ZXL recorder. Note how close the measured distortion curves (especially that for ZX tape) are to the theoretical 20-dB change (vertical axis) for each 10-dB change in level (horizontal axis). Note also that the curves are still straight, even at –14 dB record level and a distortion of only about 0.01%.

Figure 7 shows a similar plotting for three recorder and tape combinations: The Philips N4504 open-reel deck with TDK Audua, the Nakamichi 582 with ZX tape, and the Aiwa M700 with TDK AD-X. The results are plotted for two frequencies. All recorders showed a considerable increase in distortion with the frequency shift from 200 Hz to 4 kHz, though the increase varied from deck to deck. Even so, there is a general agreement with the 2:1 slope, especially with the 582 deck.

It is interesting that the two cassette decks have lower distortion than the open-reel machine for the same flux level. This is also demonstrated by Fig. 8, which shows the distortion products from recording a 1-kHz tone on the Philips (top) and Nakamichi (bottom) recorders. The open-reel deck had much higher second, third and fifth harmonic distortion, as well as some fourth harmonic not seen in the cassette deck's output. As the vertical scale is 20 dB per division, HDL_3 was about 7% for the open-reel deck and 1.1% for the cassette unit. Let it be said that no concerted effort was made to try other tapes with the open-reel machine or to optimize its bias for the tape used.

Figure 9, made with the same settings on the spectrum analyzer, shows three runs made with the 582 deck with ZX tape. First, a sweep was made with a zero-level 300-Hz tone. Then, a

"The recordist must steer between the Scylla of distortion and the Charybdis of tape noise."

second sweep was overlaid (top traces) with a 1.5 kHz tone. Both tones produced third harmonics, and there's some HDL_2 with the 1.5 kHz.

Finally, both tones were fed in at the same time, with the levels reduced for the same meter reading. The results, in the bottom of the figure, show both expected and unexpected things: Note how the 1.5 kHz appears to have three upper-frequency sidebands, 600 Hz apart. Its own second and third harmonics do not appear, somehow suppressed by the added lower tone. This points out that the generation of distortion products becomes very complex when more than one tone is involved and there are *many* more than that with complex musical wave forms. The recordist must accept the fact that the fundamental approach to controlling the generation of distortion products is to avoid high levels as far as noise considerations permit.

Maximum Record/Output Levels

Most people use the criterion of 3% distortion as the maximum allowable output level, which is, of course, tied to the maximum allowable record level for any particular deck. It has been our practice to use MRLs, rather than MOLs, mostly because it tells the recordist what limits are imposed on the signal being recorded. The MRLs are always a bit higher than the MOLs, perhaps a dB or so, reflecting the compression that appears at higher distortion levels.

At the lower frequencies, the MRLs are based upon $HDL_3 = 3\%$. Above a few hundred Hz, however, more accurate data is obtained using a twin-tone signal and measuring TTIM (twin-tone intermodulation) distortion, with 3% used as the limit.

In the last year or so, certain characteristics of the MRL curves have emerged, mostly because of more careful data taking and the inclusion of other decks in the test process. Figures 10, 11 and 12 are plottings of the range of MRLs for tape Types I, II and IV, respectively, on a Nakamichi 582 deck. Note that in all cases the spread in levels for the 3% distortion points was greatly reduced above 1 kHz or so. A basic conclusion was this: In any one tape type, there was little basis in the *high*-frequency MRLs to select one tape over the other. In the low-frequency MRLs, however, the choices were much clearer, and the data did show that the formulations with the best low-frequency MRLs were also best at the high end, albeit by small amounts.

There are other facets of these plots worthy of attention. The dashed line in Fig. 11 indicates the much poorer performance of two discontinued tapes. This figure and Fig. 12 show a gentle S-curve in the high-end MRLs, believed to be caused by the 70-µS record equalization. Previous tests with Type II tapes had shown that switching to 120-µS EQ eliminated the S-ing, increasing the headroom available; however, the change in equalization also leads to an equal increase in noise, so there is no increase in dynamic range.

On each of the figures is a −6 dB/octave reference line from +10 at 2 kHz to −10 at 20 kHz. We can see that all MRL curves fall off at close to this rate, which our earlier discussions showed to closely match the roll-off in a number of music spectra. Figure 13 shows the MRLs obtained with the

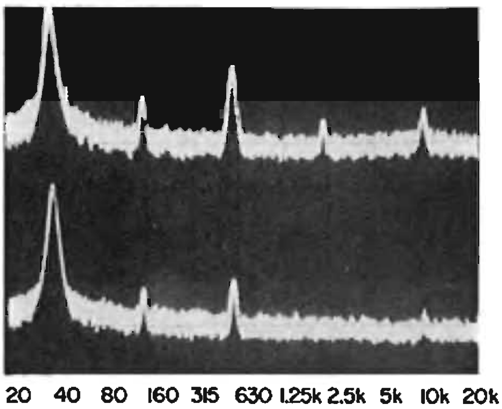

Fig. 8—Distortion products for 1-kHz signal at 200 nWb/m. Top: Philips N4540 open-reel recorder with TDK Audua (see text). Bottom: Nakamichi 582 deck with ZX tape.

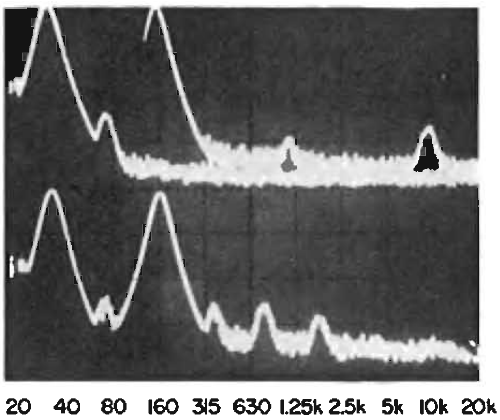

Fig. 9—Distortion with 300-Hz and 1.5-kHz tones. Top: Tones recorded separately, Bottom: Tones recorded simultaneously.

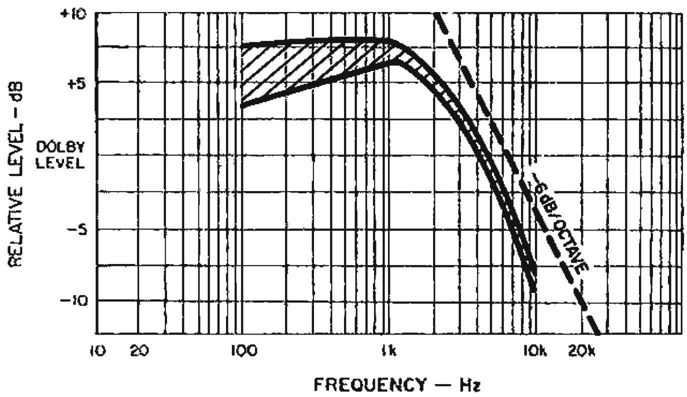

Fig. 10—Range of maximum recording levels for seven Type I tapes with Nakamichi 582 deck.

"Bias should be set high enough for low distortion without unacceptable high frequency losses."

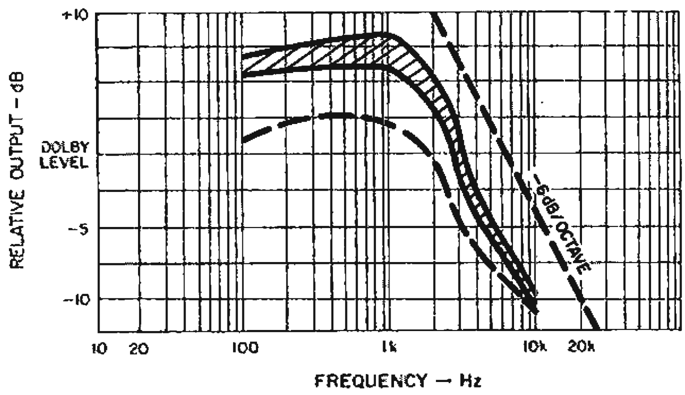

Fig. 11—Range of MRLs for five Type II tapes.

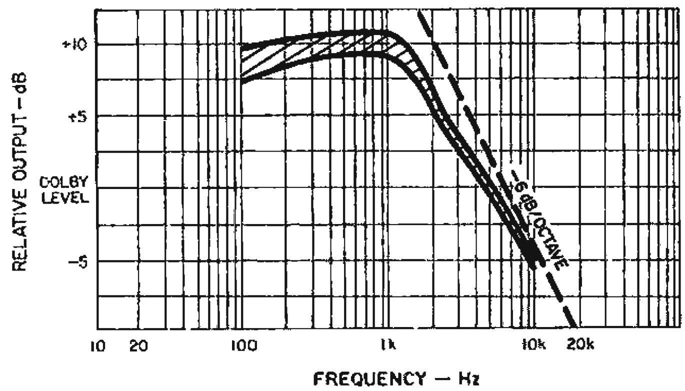

Fig. 12—Range of MRLs for five Type IV tapes.

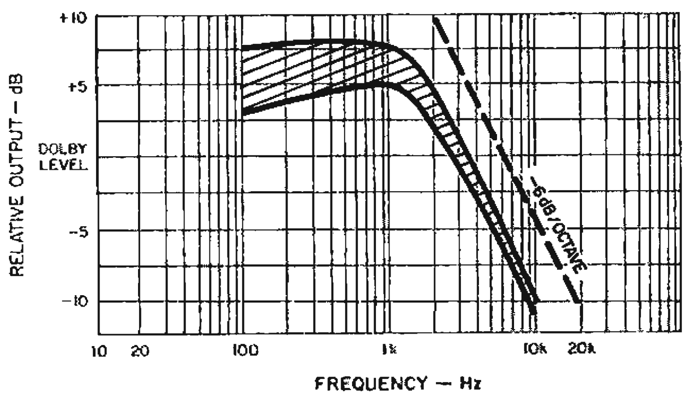

Fig. 13—Range of MRLs with Type I tapes on Aiwa AD3600 deck.

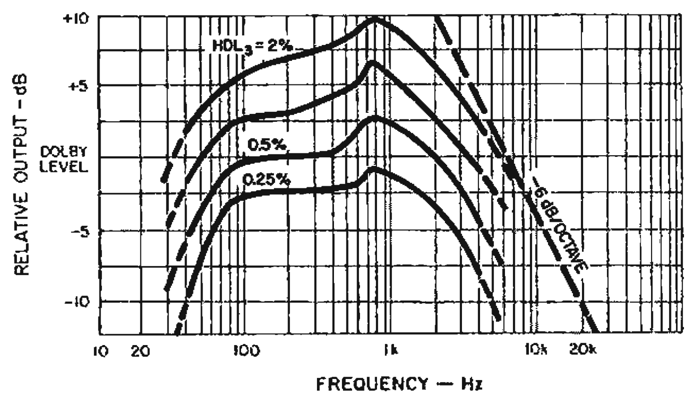

Fig. 14—Output levels for constant distortion percentages, using Nakamichi 582 deck and ZX tape.

same Type I tapes on an Aiwa AD3600 deck. The low-frequency MRLs are nearly the same as for the Nakamichi 582 in Fig. 10, though spread a bit more, and the high-frequency MRLs are a couple of dB lower.

One of the questions on distortion that remained unanswered was: What is the shape of the constant-percentage-distortion curves at less than 3%? Figure 14 presents the results of tests with the 582 and ZX tape from 0.25% to 2% distortion. The curves had quite a consistent shape, but there is some shifting, caused by the onset of compression. It is quite apparent that if a lower percentage of distortion is selected as the allowable limit, record levels must be lower or the spectrum of the music cannot be very wide-band.

Tape Noise

To some extent the recordist may very well feel that he is continually trying to steer his way between the Scylla of distortion and the Charybdis of tape noise. There is no doubt that there is lower noise with 70-µS EQ, and noise reduction has been an essential part of the cassette format in particular for some time. Figure 15 shows tape noise spectra with the 582 and ZX tape for three conditions: 120-µS EQ without NR, 70-µS EQ without NR and then with Dolby B NR. There is no doubt about the successive reduction in noise, and Dolby B shows benefits down to 400 Hz. Figure 16 shows the results of the same tests, but with BASF Professional II tape. The improvement over ZX tape is obvious in all three cases, with some reduction all the way down to 200 Hz. The benefits increase with frequency, up to about 4 dB at the high end. In fact, the BASF tape had about the same spectrum with 120-µS EQ as the ZX tape did with 70-µS EQ. There *are* differences in noise from one tape to another, and the dynamic range at higher frequencies could very well be determined by tape noise. The low-noise BASF tape would be a candidate for recording with 120-µS EQ for higher MRLs, albeit bringing up noise the same amount.

Bias Adjustment

As we stated earlier, there are relatively few recordists who adjust EQ as any sort of regular process. Adjusting

"Distortion becomes very complex when more than one tone is involved, and in music there are many."

bias, however, has become a common practice for quite a few cassette deck owners to get the most of whatever tape is tried. Figure 17 is provided as an aid to understanding the interrelationships of levels and distortion as the bias is changed. The figure is actually a time plot with bias being turned up from minimum to maximum during the sweep. The results shown are to be expected: The 315-Hz and 10-kHz levels increase with bias, but a point is reached where the high-frequency level is dropping rapidly, and the low frequency has reached its highest-level plateau. The 10-kHz level was at −20 dB, so the marked operating point has produced flat response. A short way up the bias-level curve, HDL_3 drops sharply, but then rises to a peak of over 3%. At the operating point, HDL_3 is slightly under 1%. Further increase in the bias does lower the distortion further, but with a severe drop in the 10-kHz level, and the 315-Hz level is starting to drop. The general guideline is that bias should be set high enough to get acceptably low distortion without unacceptable losses at the highest frequencies and without any bringing down of low-frequency levels.

While this article is quite lengthy, some areas have not been gone into in great detail. We hope that what has been presented will provide a framework to understand better the commonalities in tape recording.

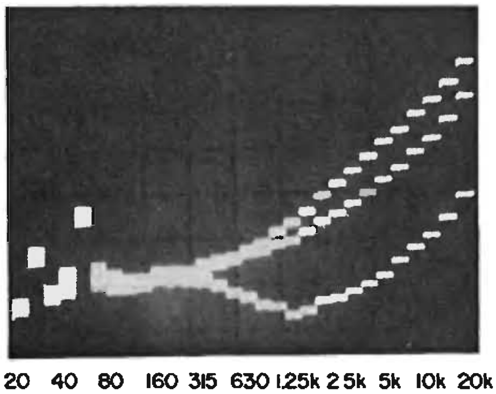

Fig. 15—Tape noise spectra with 582 deck and ZX tape. Top: 120µS EQ without NR. Middle: 70-µS EQ without NR. Bottom: 70-µS EQ with Dolby B.

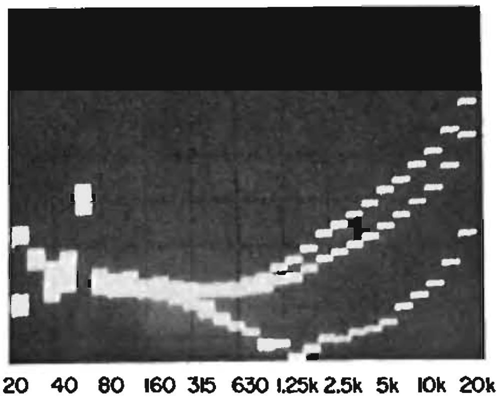

Fig. 16—Same as Fig. 15, but using BASF Professional II tape.

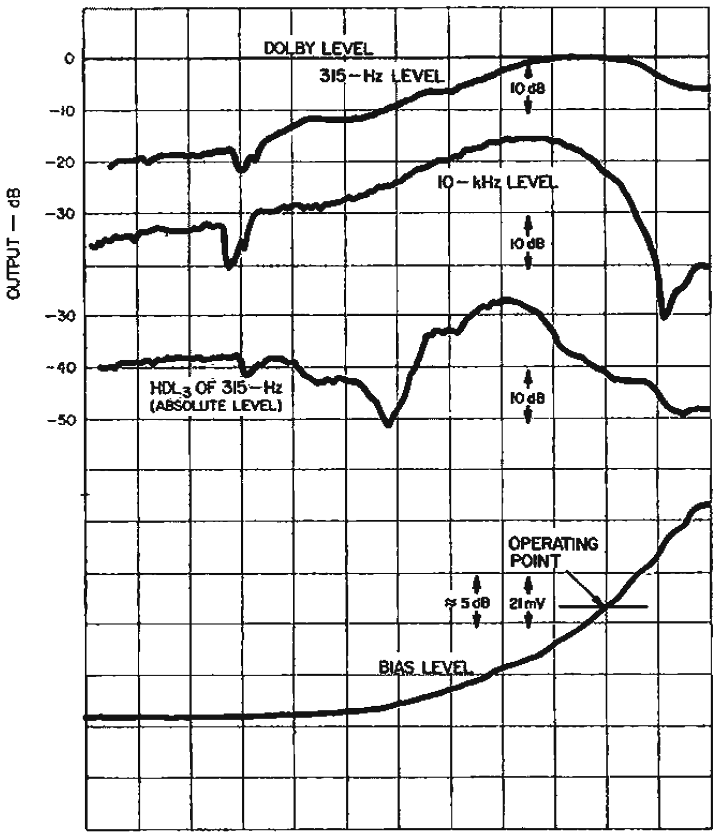

Fig. 17—Variations in 315-Hz and 10-kHz levels and 315-kHz distortion with changing bias level (Maxell XLII-S tape Nakamichi 582 deck).

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REFOCUS REFOCUS ON DEMAGNETIZATION

HERMAN BURSTEIN

Must you really demagnetize your tape deck's heads (and other metal parts contacting the tape) every so often?

The answer which emerged from the various views expressed in my round-up on the subject in the April, 1981 *Audio* is that the need varies with different decks and situations.

We invited reader comment, and many illuminating letters have come in—but still none so illuminating as to reveal one clear, shining, completely unassailable truth. The different viewpoints do increase our understanding of the matter, and here they are:

To begin, Phil Sutterlin, of Ampex, Cupertino, Cal., suggests how the problem occurs: "Heads can be magnetized by an applied magnetic field or a current applied to a head's winding if this field or current is asymmetrical, thus containing a d.c. component, or if it is symmetrical but abruptly changes magnitude; Fourier analysis tells us that there is also a d.c. component in the latter case."

Sutterlin continues: "Magnetic fields can be applied to heads by recorded tapes, head demagnetizers, or other less common sources. Magnetizing currents applied to a head winding can originate in purposely applied signals, namely audio, bias, and erase; d.c. leakage from the amplifier connected to the head, and transients which occur when turning the deck on and off.

"In my experience, heads are most often magnetized suddenly by mistakes such as: Turning the tape machine off while in the record mode (which can magnetize the erase and record heads), or improper use of a head demagnetizer (the extreme case being that of unplugging the demagnetizer while it is in contact with the head). In these cases, a demagnetized head can change into a magnetized one in a fraction of a second, assuming that current is in mid-cycle and therefore flowing through the demagnetizer or the head.

"Slower magnetization does occur from asymmetrical waveforms, whether audio, bias, or erase. For example, I have found that recording a signal with a 10% second-harmonic component—hence an asymmetrical waveform—causes magnetization."

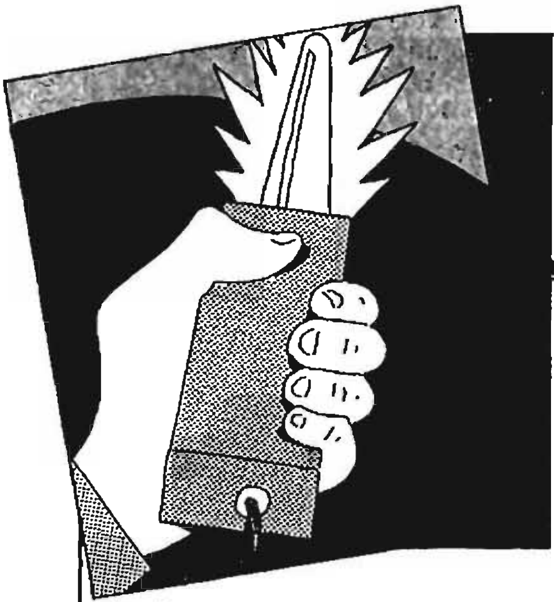
My April, 1981 article stated that a magnetized head produces noise and treble loss. Sutterlin, rightly and importantly, points out that it also causes distortion. "Magnetized heads cause an increase in second-harmonic distortion. Since third-harmonic distortion (caused mainly by the tape) is usually dominant, a spectrum analyzer is required to measure the second harmonics. If a tape deck has very low distortion in its electronics, it is very easy to measure any performance degradation caused by magnetized heads by measuring second-harmonic distortion. Noise and treble losses due to a magnetized head are also measurable with a spectrum analyzer."

I. W. Borner of Revox (Regensdorf, Switzerland), is dubious that the asymmetrical waveform encountered in audio produces significant magnetization, and suggests that routine demagnetization, although widely performed, is based on ritual rather than necessity: "The theory of the asymmetrical waveform, which is supposed to be responsible for head magnetization, is difficult to accept. Would this not mean that the electronics in our recorders must be capable of passing the d.c. component resulting therefrom? Yes, the audio signal may be asymmetrical, but only in that a brief strong positive spike is followed by a longer but weaker negative half; the areas under each are equal, with no d.c. component.

"In querying the younger generation with experience in audio, I find that they have never come across a situation where demagnetization is necessary. Yet, they carry on the practice of regular demagnetization because this is what they have learned from the older generation." (Earlier in his letter, Borner agreed that the tape decks used by the "older generation" did frequently require demagnetization.)

"In essence, I don't believe that magnetism in the heads is something that creeps up with time, like head wear. It is something that may happen as the result of careless handling of magnetized tools, or it may be caused by design shortcomings, for example: A d.c. surge through the record head when it is connected to the record electronics, solenoids which create a strong magnetic field near the heads, or suddenness of the oscillator's attack (and possible decay). Since we usually don't know how a recorder behaves in such respects, we have an uneasy feeling that an important recording may not come out as well as it

Illustration: Kurt A. Vargo



"One professional machine actually showed an a.c. field, caused by a misplaced solenoid, of 34 Gauss."

should. The routine of demagnetizing heads is therefore performed with the same regularity as paying insurance premiums."

Like Borner, John W. Dana (Hamden, Conn.), is disinclined to believe that normal operation of a tape deck produces significant head magnetization; rather, he feels that normal operation tends to demagnetize the heads. "Unless d.c. is accidentally fed to the heads, I don't see how they can become more than minutely magnetized, no matter how much the machine is used. Magnetically recorded tape contains alternating north and south magnetized bits of equal and opposite intensity. If a head or guide were magnetized, it would be demagnetized by the alternating north and south magnetized bits as the tape is played.

"Along this line of reasoning, I believe that two-head decks, which use the same head for record and playback, would have no problem with head magnetization if the machine is used for recording at least once in a while. The alternating signal going through the head would create a strong alternating magnetic field, which would remove any residual magnetism, so long as the recording signal is strong enough to drive the VU meter into the red. One could occasionally record FM interstation noise, turn up the recording level so that the VU meter goes into the red, and *gradually* reduce the record level to zero."

Assuming that a strong a.c. signal through the record-playback head is an effective demagnetizer, Dana's expedient strikes me as unnecessary. Bias current is such a signal; its magnitude is roughly 10 times that of the peak audio signal ordinarily fed to the head.

Dan Dugan of Dan Dugan Sound Design (San Francisco, Cal.), points to a prominent cause of head magnetization and stresses the need for an *efficient* demagnetizer. "In my experience, first as an amateur and then as a professional recordist, the most serious cause of head magnetization is the transient which occurs when the deck is switched into the record mode. I had one machine, which didn't last long on the market, that would sometimes magnetize its head with just one actuation of the record button. But since 1970, many machines have included time-delay circuits in their bias oscillators to soften this effect.

"No discussion of head demagnetization is complete without mentioning the need for a really effective demagnetizer. I've found that most of the demagnetizers sold for consumer use are too weak to do any good, and the persons using them usually have no way of knowing whether demagnetization has been accomplished or not. The R. B. Annis Company of Indianapolis, Ind. sells heavy-duty demagnetizers that really do the job, and it also sells magnetometers to measure head magnetization."

Some types of heads appear to be more susceptible to magnetization than others. Nakamichi, in its "Technical Bulletin 4" published a few years ago, states: "Perhaps one of the most disturbing properties of ferrite heads is the spontaneous build-up of magnetism. . . . There are many stresses imposed on ferrite during the cutting processes because of its great hardness. The high temperature process of melting the crystal glass onto ferrite, furthermore, compounds the stress. Even after cooling, the ferrite head is under constant stress because of the differing coefficients of expansion of ferrite and crystal glass. . . . Whereas . . . it was once believed that physical shock was required to induce magnetism in ferrite heads, it is now known that the constant stresses caused by changing ambient temperature have the same effect as shock."

On the other hand, there is reason to believe that the disadvantage of ferrite heads, with respect to magnetization, may have been overcome to a significant extent, as demonstrated by Nakamichi which does use ferrite heads in its decks, although not for playback.

Still, the point remains: Depending on materials used and type of construction, some heads may be more subject to magnetization than others.

We are reminded that not only heads but also other metallic components may become magnetized and endanger our recordings: The *Standard Tape Manual*, published by Standard Tape Laboratory, Inc. (26120 Eden Landing Rd. #5, Hayward, Cal.), states that magnetization is possible in "the guides, the capstan, and any metallic surface which contacts the tape. Even 'stainless' materials called 'non-magnetic' are capable of retaining some field and are therefore suspect."

The *Manual* goes on to warn of stray a.c. fields which may cause erasure for which the user may blame the heads. "Many top selling machines have been designed without any consideration by the engineering departments as to a.c. magnetic field contamination." As a horrible example, "one professional machine actually showed an a.c. field, caused by a misplaced solenoid, of *34 Gauss* (in nontechnical language, a helluva strong field—H.B.). . . . The field was concentrated at the supply reel side of the transport at the flutter idler guide."

Further, "a poorly filtered d.c. supply may carry enough a.c. ripple to do a lot of damage to a tape passing near a solenoid or relay coil."

We are indebted to Mike Hardwick of Westronix Hi-Fi (Salem, Ore.), for calling our attention to the above information in the *Standard Tape Manual*.

In closing, John J. Swetko (Glen Oaks, N.Y.), writes: "During 18 years of tape deck ownership, which has ranged from tape manglers to top-notch decks, I have experienced only one episode of tape degradation. When I first opened my TEAC I immediately played a tape, and in one pass I immediately destroyed it. After that event, I have never failed to demagnetize after moving a deck. Other than that, my demagnetizing has been haphazard yet without ill effect."

The correspondence indicates that what was once a definite threat is now less of a problem. Nevertheless, head magnetization can occur and caution should be exercised when conditions put recordings at risk.

SIDE BENEFITS AND SIDE EFFECTS

HOWARD A. ROBERSON

Dolby B noise reduction was a vital factor in the acceptance of the cassette format. But as tapes, records and the rest of the audio chain improved, audiophiles felt the need for even greater noise reduction. Dolby C noise reduction, now available on many decks, reduces noise by 20 dB, twice as much as the 10 dB of Dolby B NR. The dbx II NR system, with as much as 30 dB of noise reduction and increased headroom, has been available for several years in outboard processors and has recently been built into decks from several manufacturers.

While everyone knows that these systems reduce noise, their effects on frequency response, distortion and other aspects of system performance are less common knowledge. This article examines these factors, with some suggestions as to how one can best take advantage of each system's characteristics.

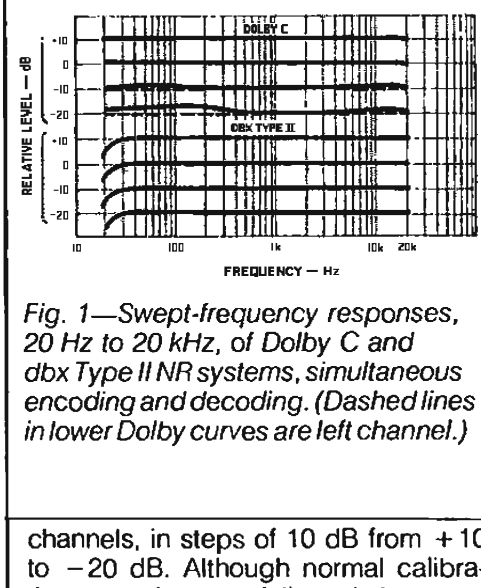


Fig. 1—Swept-frequency responses, 20 Hz to 20 kHz, of Dolby C and dbx Type II NR systems, simultaneous encoding and decoding. (Dashed lines in lower Dolby curves are left channel.)

Responses and Tracking

The investigation utilized the normal collection of test equipment plus a Nakamichi NR-200 for the Dolby C processor and a dbx 224 for the dbx II processing. A Nakamichi 582 cassette deck was used for the record/playback tests. First, the processors themselves were tested to make sure their encoding and decoding were exactly complementary. Figure 1 shows the 20 Hz to 20 kHz swept responses of both channels, in steps of 10 dB from +10 to −20 dB. Although normal calibration procedure was followed, there was a rise of 2 dB or so around 100 Hz at −20 dB with Dolby C NR for one channel. This deviation indicates the need for accurate adjustment with this system. The dbx II responses were perfectly flat at all levels except for the roll-off of 3 dB or so at 20 Hz.

Figure 2 shows a similar set of responses, but they are of the encoder section only. We can see that, with Dolby C NR, there is no compression at the lowest frequencies, but from 100 Hz up there is increasing compression as overall level is reduced. The decoder, of course, has mirror-like responses; by expanding the lower levels, it restores signals to their correct level but reduces noise. Because the curve shapes change with level below zero, level-matching errors will cause at least some frequency-response deviation.

The dbx Type II encoder responses are in steps of 5 dB because of the 2:1 compression. The curve shapes, however, are exactly the same, and the matching (mirror-image) curves of the decoder also do not change with level. Thus, dbx II does not require level matching to maintain normal response. Since dbx also compresses signals above zero, it appears logical that there should be increased headroom. The decoder not only "corrects" the response, but it expands the signal 1:2 relative to zero, back to its correct levels, thereby reducing noise.

What follows immediately is basically a deliberate demonstration of how improper testing methods can make a product look bad. The test signal consisted of the swept sinusoid fed through an MXR ⅓-octave equalizer with the filters purposely set to simulate the response of a poor recorder. The equalizer was inserted in the NR loops, and swept responses were taken. As Fig. 3 shows, Dolby C NR matched quite well at the two levels tried. Dbx II, however, doubled the deviations from flat. It would seem that this demonstrates a failing in the dbx II scheme, but it's really an example of a poor test. Figure 4 shows what happened when I repeated the test but used pink noise,

"Both the Dolby and the dbx II noise-reduction systems lower distortion—but for different reasons."

NOISE-REDUCTION

instead of a swept sine wave, as the source. All three traces look alike, *including* that for dbx II (bottom). The dbx system's gain varies with overall signal level, not with the level in just one frequency band. A swept sinusoid of varying amplitude appears as just another varying-amplitude signal to the dbx II system, which acts accordingly. Since the variations in the test were introduced after the encoder but before the decoder, the deviations were doubled in the expansion. With wide-band signals such as pink noise or music, frequency irregularities have less effect on overall level and so are not expanded.

Figure 5 shows Dolby C NR responses at −20 with the Nakamichi 582 record/playback in the loop. The topmost response was made with a low-pass filter at 25 kHz. The second trace of Fig. 5 was the result of moving the filter out to 50 kHz. The third trace was obtained after switching in the multiplex filter. The highest frequencies were rolled off by that filter, but the smoothing of the overall response is very obvious and certainly desirable. Dolby Laboratories does recommend that the multiplex filter be used as a general practice to prevent mistracking due to energy above 20 kHz—as might come from a synthesizer, for example. The other traces in Fig. 5 show the effects from calibration errors and a rolled-off response from excessive bias.

Figure 6 shows the dbx II responses over a range of the higher levels. The system showed no sensitivity to above-band energy or recorder-response deviations over a very broad range of levels. (Part of the dbx II design is a 10-kHz roll-off in the level-detector loop and a 27-kHz roll-off in the signal loop.) In the top set of four traces, the unity-gain point was set at −5 input/record level, and the input level was increased in steps of 4 dB above that, to a maximum of +7. It can be seen that a roll-off appeared at +3 input level, which was actually −1 on the recorder because of the encode compression. In the bottom four traces, unity gain was set at zero with input levels of −8, −4, 0 and +4. Notice that there is some roll-off at −4, which was actually −2 on the recorder. In this case, with the high unity-gain point, levels were actually pulled up by the encoder compression, leading to the high-frequency loss. The dbx Type II unity-gain point should be set 5 dB or so below the recording level where high-frequency saturation first appears; that will give 10 dB of headroom, with the 2:1 encoder compression.

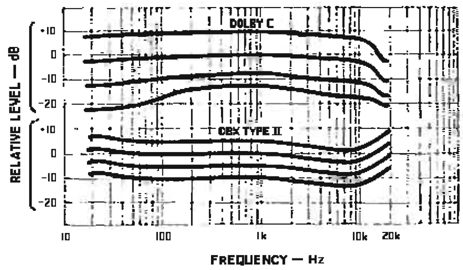


Fig. 2—Swept-frequency responses of Dolby C and dbx II encoders only.

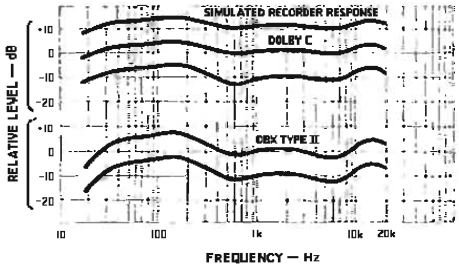


Fig. 3—Response of NR systems with poor "recorder" response, using swept sine-wave signals.

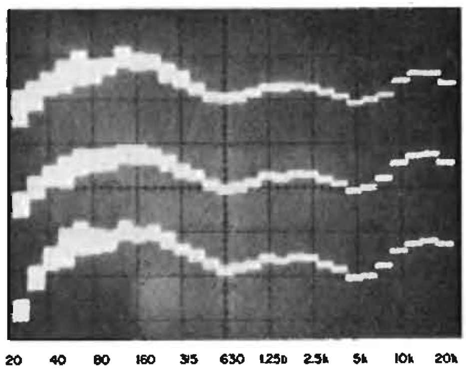


Fig. 4—Response of NR systems to same poor "recorder" response, using pink-noise test signals: "Recorder" response (top trace), with Dolby C NR at −10 dB (middle), and with dbx II NR at −10 dB (bottom). Vertical scale: 5 dB/division.

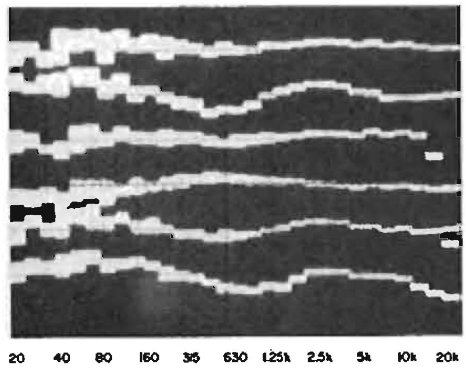


Fig. 5—Effect of high-frequency energy on Dolby C tracking, at −20 dB with tape deck in loop: Pink-noise response with 25-kHz low-pass filter on source (top trace); response with 50-kHz low-pass filter (second); response with multiplex filter added (third); response with Dolby calibration control at +1, multiplex filter out, and 25-kHz filter in (fourth); same, with calibration at −1 (fifth), and response with excessive recorder bias, rolling off at high end, calibration at zero (bottom). Vertical scale: 5 dB/division.

Distortion and Noise Reduction

As a first step, let us discuss some of the basic characteristics of distortion in the recording process. In analog recording with high-frequency bias, the primary distortion components are third-order, even with complex tones. Over the entire range of recorded levels, the absolute level of the third harmonic is a cubic function of the level of the fundamental. The *relative* level of

"The dbx Type II system is appealing when you're making live recordings and can't stay near the recorder's controls."

NOISE-REDUCTION

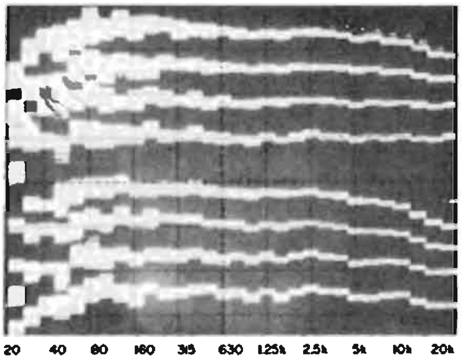


Fig. 6—Effects of dbx II unity-gain setting and input level on frequency response. Upper four curves: Responses at (top to bottom) +7, +3, −1 and −5 dB, with unity gain set at −5 dB; note roll-off in upper two curves. Lower four curves: Responses at (top to bottom) +4, 0, −4, and −8 dB, with unity gain at 0; note the earlier onset of roll-off. Vertical scale: 5 dB/division.

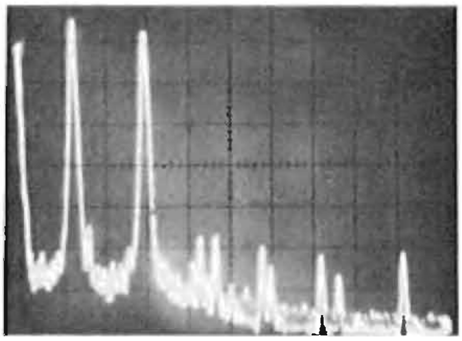


Fig. 7—Record/playback spectrum with test signal of 400, 1,100 and 2,000 Hz, at +5 dB input level, with dbx Type II NR. Scales: Horizontal, 0.5 kHz/division; vertical, 10 dB/ division.

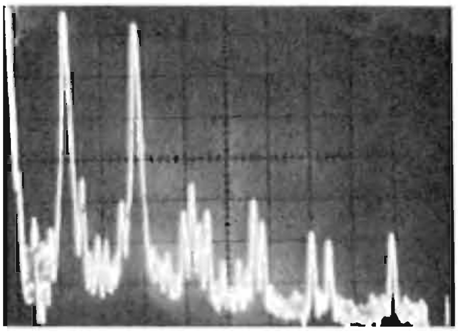


Fig. 8—Same as Fig. 7, with Dolby C NR. Note greater distortion (see text).

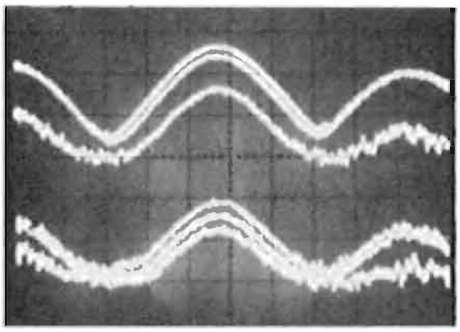


Fig. 9—Distortion components around 2.6 kHz, with same three-tone source Upper set of curves at +5 dB input level: Without NR (top), with Dolby C NR (middle), and with dbx II NR (lower). Lower set of curves same, but at 0 dB input level. Scales: Horizontal, 20 Hz/division; vertical, 10 dB/division.

the distortion, as a percentage of the fundamental, varies as the square of the signal level. This means the distortion will increase 2 dB for each 1 dB increase in record level. Therefore, excessive recording levels can cause very high levels of distortion: At 3 dB above the 3% distortion point, HDL_3 would be about 6%.

Part of the appeal of noise-reduction systems is that by reducing noise, they give some leeway in setting maximum record levels, so there's less chance of distortion. Even when recording levels aren't excessive, both the Dolby and dbx II NR systems lower distortion—but for different reasons.

In the Dolby system, distortion is reduced, like noise, during decoding. That's because distortion, like noise, is added to the signal after encoding and because the Dolby systems' sliding-frequency bands concentrate the decoding on the upper (and, with Dolby C, the middle) frequencies, where undesired harmonics appear. This is why, in tape-recorder "Equipment Profiles," I regularly comment that "distortion without NR was 30% higher."

The dbx Type II system, on the other hand, reduces distortion across the audio band at higher input levels, since its compressing action reduces the actual recording level. (There is no response shaping with any significant effect on distortion.) Below the dbx system's unity-gain point, signal compression increases the recording level, which increases distortion; but the distortion here is low enough to be acceptable, in most cases even with the compression.

Multiple-tone distortion tests (more like music than single-tone signals) confirm this. When the signal was recorded with an input level of +5, tapes made with dbx II (Fig. 7) showed less distortion than tapes made with Dolby C (Fig. 8)—up to 10 dB less, especially for the highest level components. But this is only due to the dbx compression, which reduces the actual recording level to 0 dB. When the input to the dbx system is raised to +15 dB (for a recording level of +5 dB, after compression), record distortion is almost exactly that of a tape recorded at +5 with Dolby C—or of a tape made at +5 without NR, for that matter.

A series of spectrum-analyzer scans indicated that the changes in all distortion-component levels with reducing record level could be measured by examining any selected component. The analyzer was tuned to the 2.6-kHz product ($2f_2 + f_1$), and a narrow scan (20 Hz/div.) was used to get the required detail. Figure 9 shows the results at +5 input level (zero record level for dbx II) and at zero input level (−2.5 recorded level for dbx II). In both cases, the highest distortion level is without noise reduction, the middle level is with Dolby C NR, and the lowest level is with dbx II NR. Note that the dbx II trace appears to be noisier.

Figure 10 plots HDL_3 versus input level at 400 Hz for the two NR systems and for a signal without NR. It can be

"Dolby C sounded more musical in one comparison, perhaps from lower noise at higher levels or lower distortion at low levels."

seen that Dolby C NR reduced distortion for all levels shown. This distortion reduction over a wide level range is a benefit of the sliding-band design of the Dolby C compressor and expander. However, as the level approaches the 3% distortion point, there is less advantage.

It is very evident that the compression in the dbx II encoder around the −5 unity gain point produced a great reduction in distortion at the higher levels. Note, however, that its curve crosses the Dolby C NR curve at −3, and that the distortion below −3 is relatively higher, noticeably so at −10.

If dbx II offers 30 dB of noise reduction to Dolby C's 20 dB, it may seem paradoxical that, in Fig. 9, the high-level signal trace with dbx II was noisier than that with Dolby C. But, as Fig. 11 also shows, output noise at higher levels with Dolby C NR is less than with dbx II. This proves that the Dolby C sliding-band scheme can reduce noise even at higher signal levels. At low recording levels (Fig. 12), noise with dbx II was lower, particularly at the higher frequencies.

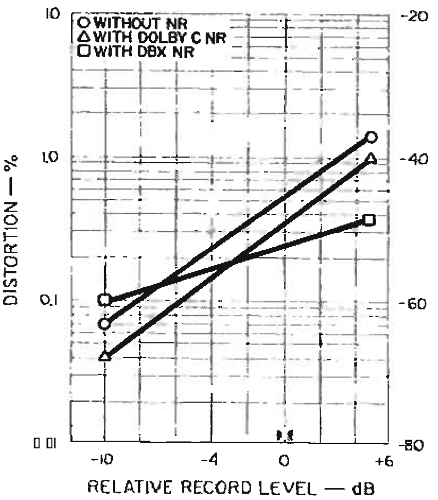


Fig. 10—Third-harmonic distortion vs. record level, at 400 Hz: Without NR (top), with Dolby C NR (middle), and with dbx Type II NR (bottom).

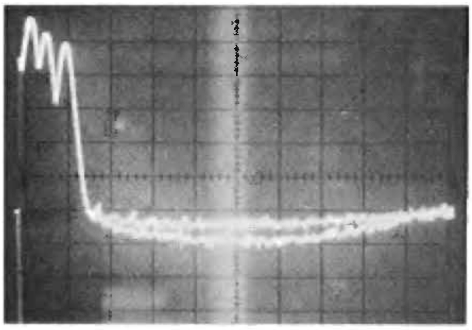


Fig. 11—Three-tone source at −5 dB input level, 20-kHz spectra. Noise levels are higher for dbx II than for Dolby C NR. Scales: Horizontal, 2 kHz/division; vertical, 10 dB/division.

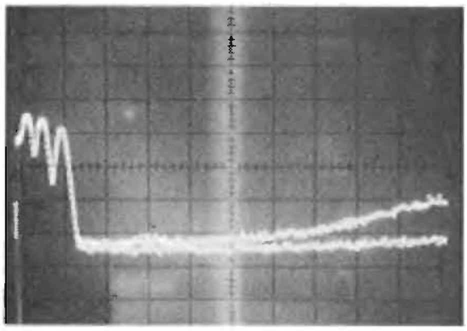


Fig. 12—Same as in Fig. 11, but with −40 dB input level (and analyzer input gain increased). Noise level is now higher with Dolby C than with dbx Type II NR.

Making Choices

Among the advantages of Dolby C NR are its inclusion in many manufacturers' products, very good noise reduction at lower signal levels, some noise reduction even at high levels, and distortion reduction at all normal recording levels. Dolby C also sounded, to my ears, more musical in one comparison where levels could be set as desired. (This may have been from such factors as Dolby C NR's lower noise at higher signal levels and/or from lower distortion at low levels. I did not have the opportunity to correlate the sonic judgment with any test results.) If you're running pink-noise tests or recording synthesizers (which would include energy above 20 kHz), make sure the signal's high end is rolled off *before* the Dolby C encoding, to prevent mistracking. If in doubt, use your multiplex filter.

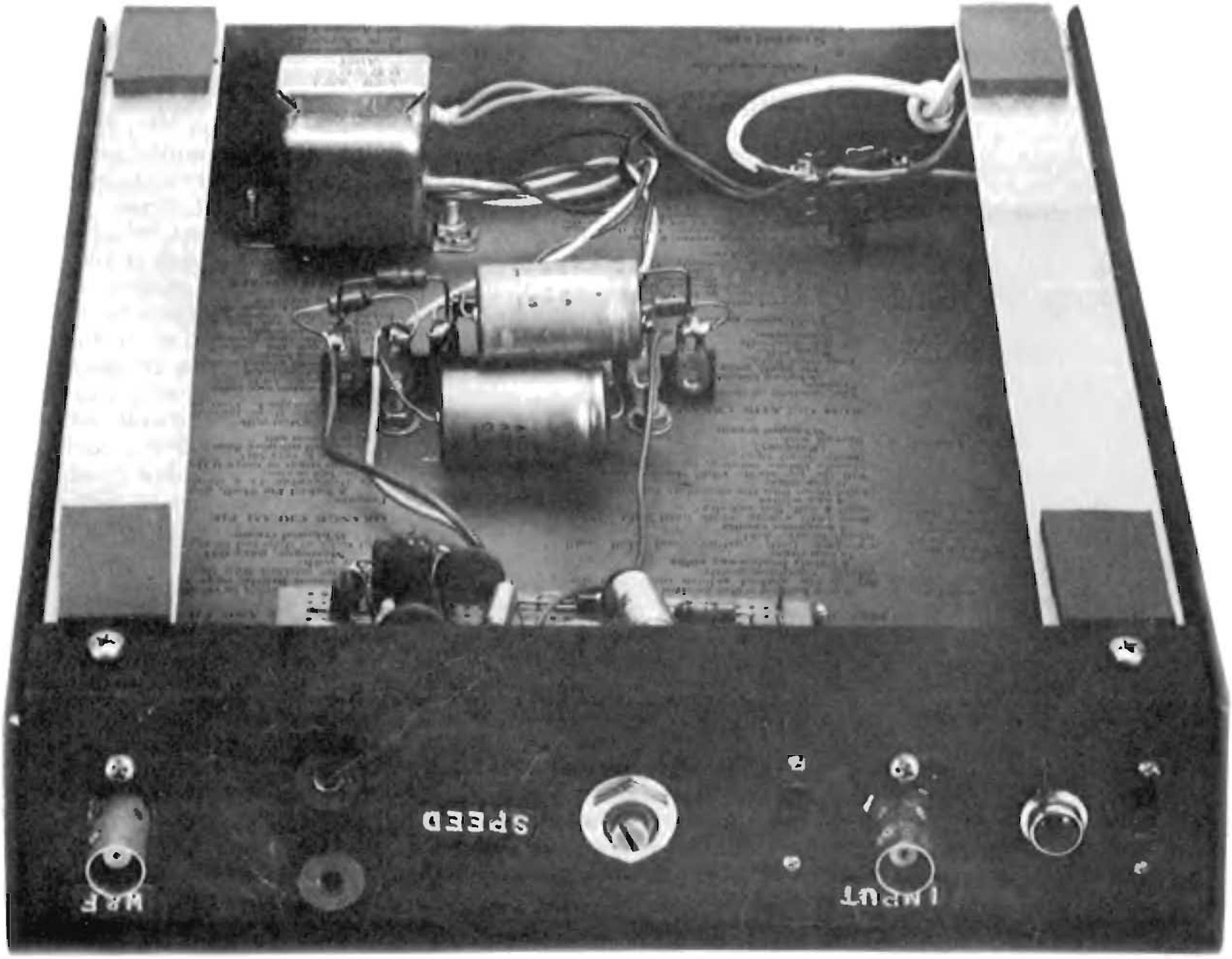
The advantages of dbx Type II NR include increasing availability (even in open-reel recorders), excellent noise reduction at lower signal levels, greatly reduced distortion across the band at higher input levels, no requirement for level matching, and insensitivity to above-band energy. This system is particularly appealing when you're recording live performances and can't stay by the recorder to adjust levels manually. The encoder's 2:1 compression gives you a lot more leeway in setting the input level.

Dolby C's disadvantages are, primarily, its need for accurate level calibration and its sensitivity to above-band energy. In practice, however, accurate calibration is quite simple and direct with good units, and above-band energy can be eliminated by using the multiplex filter, with substantially no loss in music content.

The drawbacks of dbx Type II NR are affected by the choice of the unity-gain point. If too high a level is picked, noise levels for medium-high signal levels will be increased, there will be little improvement in headroom, and distortion will be higher than without NR for anything but high levels. If too low a unity-gain point is selected, noise levels for high signal levels will increase, and there may be some loss of noise reduction at low signal levels. I have occasionally experienced noise modulation with dbx, though only at odd times between movements or when audience applause was dying out, never during the music itself.

I hope the reader accepts that both Dolby C and dbx Type II noise-reduction systems can provide great benefits to their users, and that this article helps you to understand and to apply them wisely.

BUILD A POOR-MAN'S WOW & FLUTTER METER



M. J. SALVATI

This article describes the construction and operation of a low-cost device capable of measuring turntable or recorder speed accuracy and wow and flutter. Cost is kept as low as possible by utilizing the audiophile's own d.c. and a.c. voltmeters to read out speed accuracy and wow and flutter, respectively. However, instructions are given for internal metering, if desired. While this device does not have the same accuracy and features as commercial instruments selling for $500 to $1,000, it certainly will check the specs of medium- and low-cost recorders and will indicate serious deficiencies in top-of-the-line machines. Full specifications for the meter are given in Table I.

Theory of Operation

The output of a turntable or tape machine is fed to a frequency discriminator whose output contains a.c. and d.c. error-signal components representing wow and flutter and speed error, respectively. These error signals are individually processed by the two signal channels, and read out by means of a.c. and d.c. voltmeters connected to the unit. The schematic of the basic device is shown in Fig. 1.

Frequency Discriminator. Op-amp IC1 is a unity-gain buffer that provides a high input impedance for the unit. It drives the input of phase-locked loop IC2. As long as the input signal is above 10 mV and within ±30% of the 3,150- or 3,000-Hz test frequency, the IC's d.c. output level is directly proportional to the frequency of the input signal. As the input frequency shifts (due to recorder speed variations), the d.c. output voltage of IC2 also shifts. Capacitor C5 and resistors R3 and R4 determine the center frequency of the discriminator.

Speed Channel. The error-signal output voltage at pin 7 of IC2 is referenced to its pin 6 voltage, so a differential amplifier (IC3B) with a voltage gain of 75× amplifies the error signal and converts it to a ground-referenced signal. A filter with a very long time constant (resistor R21 and capacitor C13) removes the rapid variations (wow and flutter), so the d.c. voltage appearing at the output of buffer IC5 represents only the (relatively) constant speed error. A d.c. voltmeter connected to jacks J3A and J3B will therefore indicate speed accuracy (or "drift," as it is sometimes called). Because of the drive capability of IC5, even a low-impedance voltmeter can be used for readout.

Wow and Flutter Channel. The error-signal voltage at pin 7 of IC2 is also applied to a 200-Hz low-pass filter. This second filter and amplifier chain removes the test frequency, sets the upper frequency limit of the wow and flutter signal passband, and amplifies the signal to the extent necessary to establish the 1 V per 1% scale factor. Op-amp IC3A uses a small amount of gain (1.7×) to produce a sharp corner despite the use of only two filter sections (R7, R8, C6 and C7). The following op-amp (IC4) provides a voltage gain of 44×, so the total voltage gain for this channel is 75×. Since this gain is the same as that of the speed channel, it means that when the "Speed" output is calibrated (via pot R17 across the output of IC2), the "W & F" output is calibrated simultaneously.

The input coupling to op-amp IC4 is capacitive, to remove the d.c. component from this channel. The output signal of IC4 is applied to the "W & F" jacks (J2A and J2B) via another low-pass filter (R13 and C11), which provides an additional 10 dB reduction of the 3,150/3,000-Hz test frequency.

Internal Metering. Meter M1 (see Fig. 2A) has a zero-center scale, so it is not necessary to switch polarity as tape speed drifts from faster to slower than normal. For wow and flutter measurements, a high-pass filter and precision rectifier are combined to drive meter M2. Op-amp IC8 is configured as a maximally-flat, two-pole filter whose corner frequency is 0.5 Hz. This establishes the low-frequency limit of the 0.5 to 200-Hz standard wow and flutter passband, and also provides some amplification before the wow and flutter signal is applied to the precision rectifier (IC9), which converts it to a unipolar voltage. The "W & F" percentage switch (S4) selects the resistors that establish the scale factors for meter M2.

Table I—Specifications

Operating Frequency: 3,000 or 3,150 Hz.
Input Level: 10 mV to 20 V.
Input Impedance: 100 kilohms.
Speed Readout Ratio: 1 V d.c. per 1% speed error.
Speed Accuracy Ranges: Depends on d.c. voltmeter used; ±5% for internal metering.
Wow and Flutter Frequency Range: 0.5 to 200 Hz.
Wow and Flutter Readout: 1 V a.c. per 1% wow and flutter.
Wow and Flutter Ranges: Depends on a.c. voltmeter used; 0.2%, 1%, 5% for internal metering.

Construction Notes

There is nothing especially critical about layout beyond the common-sense precaution of not routing the power-switch leads near the "Input" connector and its switch.

With the exception of the power transformer, rectifiers and filter capacitors, nearly all of the basic-unit components are mounted on a single piece of Veroboard® or equivalent perfboard. Furthermore, if you obtain compact filter capacitors, there is no reason why all power-supply components other than the transformer cannot be mounted on the same board. However, if you opt for internal wow and flutter metering (Fig. 2B), it might be more convenient to build the additional circuitry on a separate board.

Choice of Parts. There is considerable latitude in the ratings for many of the parts listed in Tables II and III. Where space permits, the alternate choices are given in the parts lists, with the preferred part listed first. For example, the low-power (100 mA, TO-92 case) versions of the three-terminal plastic voltage regulators will do nicely for IC6 and IC7, although any of the equivalent higher-power versions listed work just as well. Similarly, any capacitors having a voltage rating of over 20 V can be used (except for C15 and C16). The most important consideration with capacitors is to locate the ceramic disc bypass capacitors physically close to the associated ICs, and capacitor C16 close to IC7. Where no type of capacitor is specified, you can use paper, polyester, mylar, or ceramic.

Table II—Basic Unit Parts List

- IC1, IC4, IC5—LF351 or LF13741 FET op-amps.
- IC2—NE565 or LM565 PLL.
- IC3—μA747 or LM747 in 14-pin package.
- IC6—78L15, 7815, or LM340T-15 positive regulator.
- IC7—79L15, 7915, or LM320T-15 negative regulator.
- J1—Phono jack.
- J2, J3—Pairs (red/black) banana jacks.
- S1—SPST toggle or slide switch.
- S2—SPDT toggle or slide switch.
- S3—DPDT toggle or slide switch.
- T1—16 to 18 V, 0.25-A power transformer (Mouser 41FJ300 or Digi-Key T102).
- D1, D2—6.0-V, 500-mW zener diodes.
- D3—LED.
- D4, D5—100-PIV, 1-A silicon rectifier diodes.
- C1, C2, C8, C9, C11, C12, C14—0.022-μF, 25-V ceramic disc capacitors.
- C3—0.1-μF, 25-V capacitor.
- C4—0.001-μF, 25-V capacitor.
- C5, C6, C7—0.01-μF, ±10% styrene or silver-mica capacitors.
- C10, C13, C17—0.22-μF, 25-V capacitors.
- C15, C16—220-μF, 35- or 50-V electrolytic capacitors.
- C18—2.2-μF, 25-V tantalum electrolytic capacitor.
- R1—100-kilohm, ¼-W carbon-film resistor.
- R2, R13—4.7-kilohm, ¼-W carbon-film resistors.
- R3—10-kilohm, 15-turn trim pot (Weston 830P or Spectrol 43P).
- R4—5.6-kilohm, ¼-W, 5% carbon-film resistor.
- R5—560-ohm, ½-W resistor.
- R6—13-kilohm, ¼-W, 5% carbon-film resistor.
- R7, R8—120-kilohm, ¼-W, 5% carbon-film resistors.
- R9—430-ohm, ½-W resistor.
- R10—9.1-kilohm, ¼-W, 5% carbon-film resistor.
- R11, R21—4.7-megohm, ¼-W carbon-film resistors.
- R12, R20—10-kilohm, single-turn trim pots.
- R14—43-kilohm, ¼-W, 5% carbon-film resistor.
- R15—1-kilohm, ¼-W, 5% carbon-film resistor.
- R16, R18—10-kilohm, ¼-W, 5% carbon film resistors.
- R17—47-kilohm, single-turn trim pot.
- R19, R22—750-kilohm, ¼-W, 5% carbon-film resistors.
- 7 × 5 × 3-inch aluminum box (Mouser LMB TF-782).

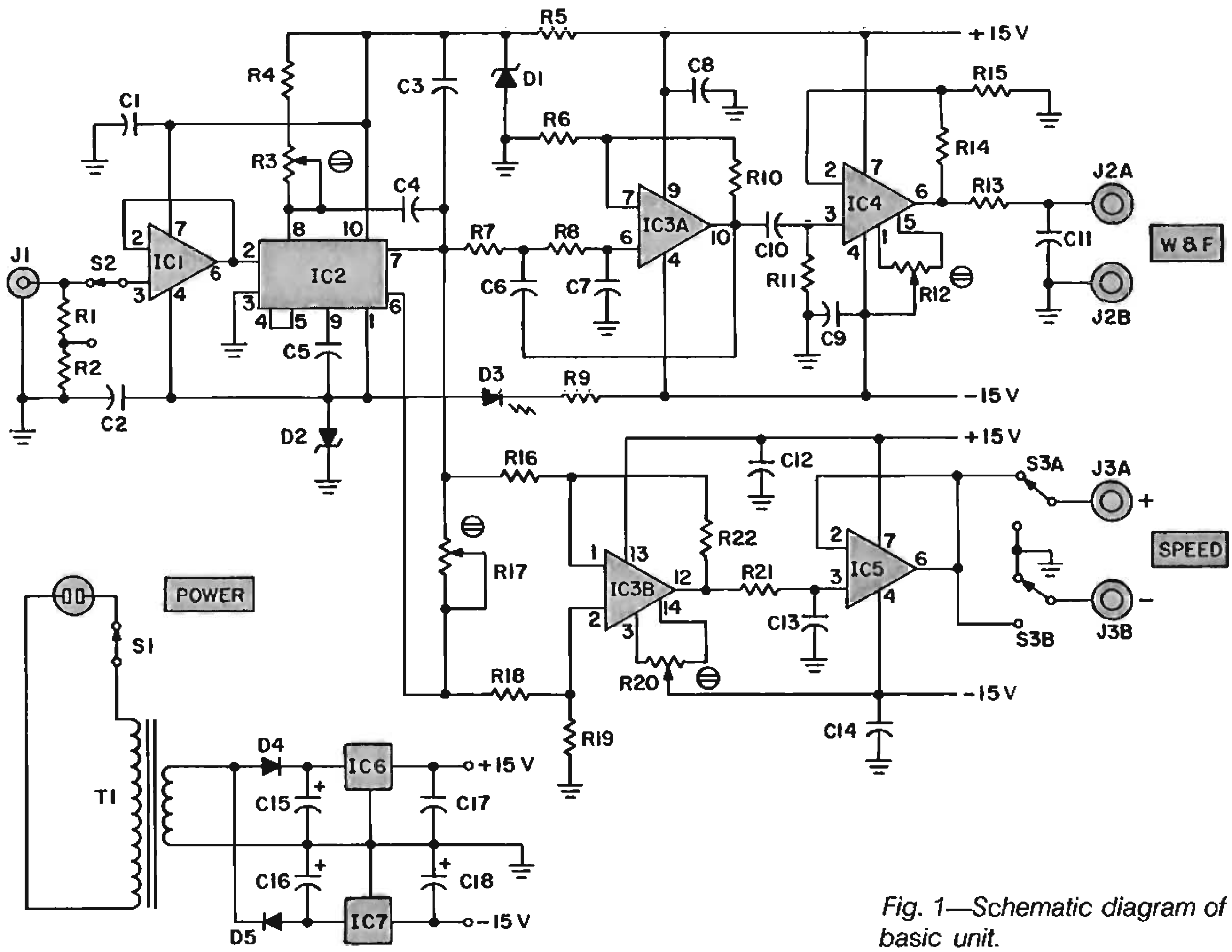


Fig. 1—Schematic diagram of basic unit.

The input and output connectors specified are the most typical for their application. They can (and should) be changed to other types more suited to your own tastes and/or test equipment. (I certainly did this in my prototype!)

Options and Modifications. The circuit shown in Fig. 1 was designed to interface with as wide a variety of test equipment and tape machines as possible. In most cases, you can save a little time, effort and money by eliminating the features not necessary to your setup.

If all your tape machines have at least one output jack apiece whose normal output level is in the range of 10 mV to 1 V rms, you can eliminate S2 and the 4.7-kilohm resistor. Connect the "Input" jack (J1) directly to pin 3 of IC1, and resistor R1 from there to ground.

If you are using an external d.c. voltmeter to read out speed, and that voltmeter has its own polarity ("+/−") switch, you can eliminate S3. Simply connect the red (+) "Speed" jack to pin 6 of IC5, and the black (−) jack to ground.

If you plan to use internal metering for speed readout, eliminate the "Speed" jacks *and* switch S3, and add the parts shown in Fig. 2A. Resistor R23 then connects directly to pin 6 of IC5, and the meter "−" terminal connects to ground.

Using your a.c. voltmeter to read out wow and flutter is both convenient and cost efficient. However, many a.c. voltmeters lack full response down to the 0.5 Hz commonly used as the lower limit measured by commercial wow and flutter meters. Therefore, you may want to build a low-frequency a.c. voltmeter into this device. A suggested circuit, which consists of a 0.5-Hz high-pass filter, precision full-wave rectifier and metering, is shown in Fig. 2B. This circuit provides full-scale wow and flutter ranges of 0.2%, 1%, and 5%. Since multi-scale meters are rarely available as stock items, you will have to use a 0-50 scale microammeter and add the 0-0.2 and 0-1 scales with dry-transfer lettering. The decade zeros of the existing scale graduations can either be erased or painted over with white enamel.

A list of additional and alternate parts needed for internal metering is given in Table III. Note that some of these parts (mostly capacitors) are used *in place of* their equivalents in the Basic Unit Parts List. After you have decided what (if any) options or modifications you want in your unit, carefully go through both parts lists to select only those parts you will actually use.

Packaging. This depends on the degree of complexity selected. The unit I built is housed in a custom-made box. The recommended standard-size aluminum case for the basic unit is 7 × 5 × 3 inches, with a 5 × 3 end used as the front panel. A suggested layout is shown in Fig. 3A. If you use internal metering, a larger case is necessary to accommodate the meters. For both internal speed and wow and flutter metering using the recommended 4-inch meters, a 10 × 6 × 3½-inch case is ideal. The suggested layout shown in Fig. 3B uses one 10 × 6 face for the front panel.

Parts Availability. All of the ICs, resistors, and capacitors are available from consumer-oriented mail-order houses such as Digi-Key and Jameco Electronics. The meters, power transformer, case, and many of the other parts are available from Mouser Electronics. Order their catalogs first to check prices and availability. (Digi-Key, Box 677, Thief River Falls, Minn. 56701; Jameco Electronics, 1355 Shoreway Rd., Belmont, Cal. 94002, $10 minimum order; and Mouser Electronics, 11433 Woodside Ave., Santee, Cal. 92071, $20 minimum.)

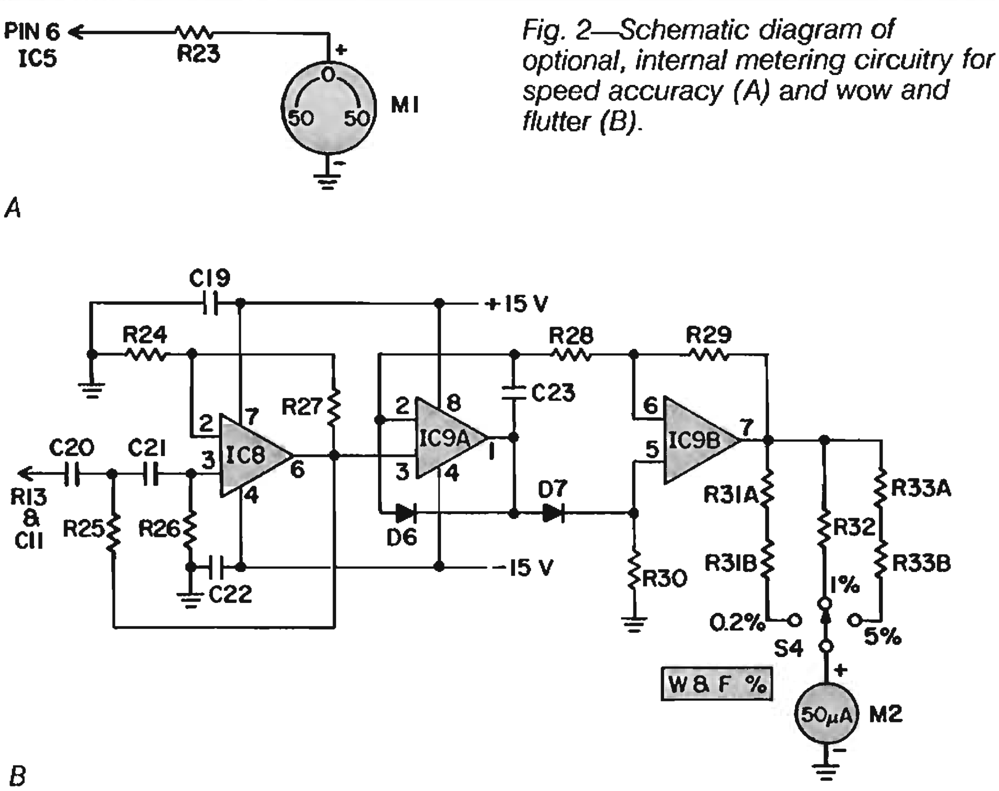


Fig. 2—Schematic diagram of optional, internal metering circuitry for speed accuracy (A) and wow and flutter (B).

Adjustments

To adjust this device, an audio oscillator, a.c. voltmeter, d.c. voltmeter, and frequency counter are needed. Before energizing the circuit, set all pots (R12, R17 and R20) to mid-rotation. Proceed as follows:

1. Plug in the power cord and flip the power switch on. The LED should illuminate.
2. Check the output voltages at IC6 and IC7. They should be +15 and −15 V respectively.
3. Check the voltages at pins 7 and 4 of IC1. They should be +6 and −6 V respectively.
4. Check the voltage at pin 6 of IC2. It should be approximately +4.5 V.

After making these checks, go on to the calibration adjustments, but wait until the wow and flutter meter and test

Table III—Options Parts List

IC8—LF351 or LF13741 FET op-amp.
IC9—LF353, LM1458, RC4558, TL072, or TL082 dual op-amp.
S4—1-pole, 3-position rotary switch.
M1—50-0-50 µA microammeter (Mouser 39LK416).
M2—0-50 µA microammeter (Mouser 39LK414).
D6, D7—1N914 or 1N4148 silicon signal diodes.
C19, C22—0.022-µF, 25-V ceramic disc capacitors.
C20, C21—0.22-µF, ±10% capacitors (Panasonic M1224).
C23—100-pF ceramic disc or mica capacitor.
R23—100-kilohm, ¼-W, 5% carbon-film resistor.
R24—13-kilohm, ¼-W, 5% carbon-film resistor.
R25, R26—1.5-megohm, ¼-W, 5% carbon-film resistors.
R27—9.1-kilohm, ¼-W, 5% carbon-film resistor.
R28, R29—10-kilohm, ¼-W, 5% carbon-film resistors.
R30—4.7-kilohm, ¼-W carbon-film resistor.
R31—5.3-kilohm (5.1 kilohms and 200 ohms), ¼-W, 5% resistor.
R32—33-kilohm, ¼-W, 5% resistor.
R33—168.5-kilohm (160 kilohms and 8.2 kilohms), ¼-W, 5% resistor.
10 × 6 × 3½-inch aluminum case (Mouser LMB TF-784).

equipment have been warmed up for at least *20 minutes*. Do the calibration procedures in the order given, since later adjustments depend on the preceding ones.

D.c. Balance. To adjust the d.c. balance of IC4, connect a d.c. voltmeter across the "W & F" jacks (J2A and J2B). Then, carefully adjust pot R12 for an output voltage of zero, ±0.1 V.

To adjust the d.c. balance of the speed channel, proceed as follows:

1. Connect the d.c. voltmeter across the "Speed" jacks (J3A and J3B).
2. Either connect a jumper from pin 6 to pin 7 of IC2 (best way) or rotate pot R17 so it has zero resistance.
3. Carefully adjust pot R20 for an indication of zero, ±0.1 V.
4. Remove the jumper or restore pot R17 to mid-rotation.

Calibration. The procedure given here will adjust for a scale factor of 1 V d.c. per 1% frequency error at the "Speed" jacks, and a 1-V a.c. output per 1% wow and flutter at the "W & F" jacks, based on the newer standard test frequency of 3,150 Hz. However, the 3,000-Hz figures will be given in parentheses, in case your test tapes use this frequency. After determining which frequency applies, proceed as follows:

1. Connect an oscillator whose frequency can be accurately set (either by dial calibration or by a counter) to "Input" jack J1. Set the oscillator output to around 100 mV rms, and the oscillator frequency to 3,150 (3,000) Hz.
2. Set the range switch of the d.c. voltmeter (connected to the "Speed" jacks) to 5 V and its polarity switch (or S3) to positive. If the prior adjustments were properly made, this voltmeter should indicate 0 V with exactly 3,150-Hz (3,000-Hz) input.
3. Change the oscillator frequency to exactly 3,276 (3,120) Hz, as indicated on the frequency counter or dial. Adjust trim pot R17 for a d.c. voltmeter indication of exactly 4.0 V.

Center-Frequency Adjustment. Pot R3 allows the center frequency to be set to either of the standard test frequencies (3,150 or 3,000 Hz). Again, the procedures given here are for 3,150 Hz, with figures for 3,000 Hz in parentheses. To adjust R3, proceed as follows:

1. Connect a frequency-stable oscillator to "Input" jack J1. Set the oscillator output level to around 100 mV rms.
2. Connect a d.c. voltmeter to "Speed" jacks J3A and J3B.
3. Carefully adjust the oscillator frequency (with a frequency counter) to 3,150 (3,000) Hz, ±2 Hz. Then adjust trimmer R3 for zero d.c. output at the "Speed" jacks.
4. When you have done this, recheck the frequency counter display to make sure that the input frequency is exactly 3,150 (3,000) Hz. Readjust the oscillator frequency if necessary, and readjust trim pot R3 for exactly zero d.c. output. The accuracy of your speed-error measurements depends on how carefully this adjustment is made. *Note:* If you do not have a frequency counter, pot R3 can be set well enough for wow and flutter measurements by using the dial calibrations of the oscillator. However, this alternative is not accurate enough for reliable speed measurements.

Operation

There are two ways to use this device to check tape machines. The simplest method is with your own oscillator as the signal source. This permits an *overall* check of wow and flutter; record and playback performance is measured as a whole. The second method uses a prerecorded test tape to check playback only for wow and

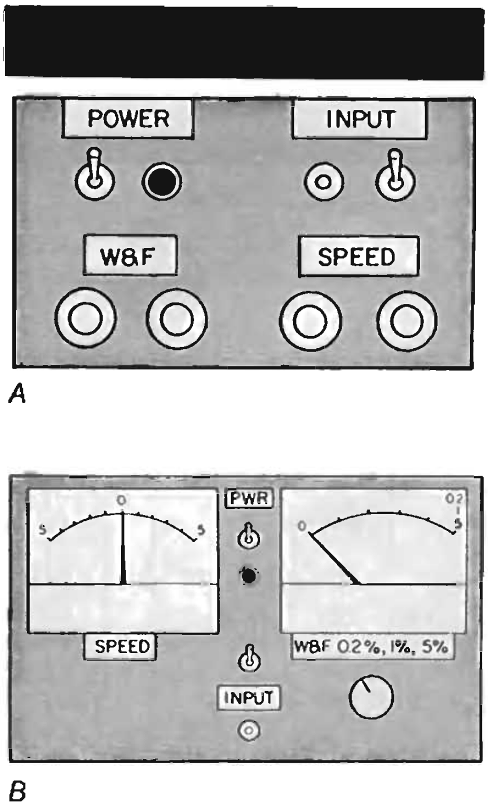


Fig. 3—Suggested panel layouts of basic unit (A) and unit with full internal metering (B).

flutter and speed accuracy. Turntables use this second method with a test disc such as the CBS STR-151. For an overall measurement:

1. Connect an a.c. voltmeter with good low-frequency response to the "W & F" jacks.
2. Connect a stable audio oscillator to the tape machine's input connector. Set the oscillator output to whatever level the tape machine requires for recording. Set the oscillator frequency to the same test frequency (3,150 or 3,000 Hz) as that for which your device was previously calibrated.
3. Allow the equipment to warm up for a few minutes, then record 10 to 15 minutes of test signal on a blank tape. Do this near the middle of the tape reel or cassette for best results.
4. Disconnect the oscillator from the tape machine. Connect the tape machine's output to "Input" jack J1.
5. Play back the recorded test signal, and set "Input" level switch S1 to match the tape machine's output level.

The a.c. voltmeter will indicate total wow and flutter. A fairly steady meter indication means that the output is mainly high-frequency components (flutter). A widely varying indication means that low-frequency components (wow) predominate. However, this is just a generalization, since the damping and ballistics of the a.c. voltmeter affect the amount of pointer movement for any given amount of wow.

Overall wow and flutter measurements will vary considerably from the playback-only wow and flutter specifications listed by most manufacturers. When the tape is recorded and played back on the same machine, the same cyclic speed variations occur in playback and recording, and can either add or subtract, depending on their phase. With test tapes recorded on a different machine of lower wow and flutter, this is less of a problem. In my experience, multiplying the overall measurements by 0.6 or so yields a figure that is *roughly* comparable to playback-only specifications; some testers recommend a figure of 0.7. It also helps, even with commercial test tapes, to make multiple readings and average them.

Wow and flutter test tapes are available from several sources. Standard Tape Laboratory (26120 Eden Landing Rd. #5, Hayward, Cal. 94545) has open-reel and cassette tapes at $40 to $50 for home formats, higher prices for half-inch, 1-inch or higher-speed tapes. LC Engineering Laboratories (9451 North Kostner Ave., Skokie, Ill. 60076) has open-reel, cassette and microcassette wow and flutter tapes, at $19 to $34.

To check speed accuracy and/or playback-only wow and flutter, proceed as follows:

1. Connect an a.c. voltmeter with good low-frequency response to the "W & F" jacks and a d.c. voltmeter to the "Speed" jacks.
2. Turn on the device, the voltmeters, and tape machine. Allow everything to warm up for 20 minutes.
3. Connect the "Input" jack to the tape machine's output jack. Set the "Input" level switch to match the tape machine's output level.
4. Play the test tape in the tape machine. The d.c. voltmeter will indicate speed error at the rate of 1 V (meter indication) per 1% speed error. Positive voltage means the tape machine is fast; negative voltage means the machine is slow. *Note:* if wow is severe, the d.c. readings will also fluctuate slowly.

The a.c. voltmeter will indicate wow and flutter of the playback section alone. Again, a steady meter reading indicates mainly high-frequency speed variation (flutter); if the reading fluctuates, low-frequency components (wow) predominate. The damping and ballistics of the particular a.c. voltmeter used will affect the speed and degree of observed variation for any given amount of wow.

DYNAMIC BIAS CONTROL WITH HX PROFESSIONAL

J. SELMER JENSEN and S. K. PRAMANIK

Not all of a tape recorder's bias comes from the bias oscillator—some comes from the signal being recorded. HX Professional lets that work for, not against, the requirements of good recording.

History has many examples of how a true and deep understanding of a physical phenomenon has been found for the first time through an attempt to solve an apparently unrelated problem. Then, when the problem is solved through this new understanding of its fundamentals, the solution seems so obvious that one wonders why no one thought of it before. And often, when the solution is implemented, it leads to improvements in areas not thought of when the problem appeared, or even when it was solved. All of this applies to the discovery and implementation of the HX Professional tape recording process.

The problem whose solution largely contributed to the invention of HX Professional, while unusual, was not unknown. A quick, accurate, and cost-effective method for fine-tuning cassette recorders was needed for the assembly line. What was thought to be just such a method for adjusting bias and equalization to get flat frequency response was the use of a multi-tone, or comb-frequency, signal.

Such a signal consists of several sine-wave components of different frequency, all at the same level, mixed to form a composite test signal, as shown in Fig. 1A. When this signal is recorded and played back, it should be a simple matter to monitor the output on a spectrum analyzer and make adjustments so that the level of each frequency is equal, as in the input signal. When the adjustment is complete, the recorder would presumably be set up for flat frequency response.

The only problem with this method is that it does not work. No matter how carefully a recorder is adjusted using a comb-frequency signal, conventional frequency-response measurements on the same machine yield a curve something like the one in Fig. 1B. Tape saturation (the condition when tape is magnetized to its uppermost limit so that no further increase is physically possible) leads to a similar result. But, since the same thing happens at low recording levels, any suggestion of tape saturation as the cause has to be discarded. This effect is not confined to any particular recorder or tape formulation, but may be easily reproduced on any standard machine, whatever its price, quality or specifications.

Obviously, the reverse is also true. A recorder set up to give a flat frequency response using a standard sine-wave signal gives a response which looks like Fig. 1C, if tested with a comb-frequency signal. Since speech and music signals are almost never a single frequency, but consist of combinations of large numbers of tones, more like a comb-frequency signal, the frequency response recorded on any normal recorder with real-life signals also looks more like Fig 3. The amount of deviation from flat frequency response depends on the high-frequency content of the audio signal and is constantly changing. The error that is produced may be called dynamic frequency-response error. This error is also not confined to any particular kind of recorder, but may be reproduced on any standard machine.

To understand what happens, it is necessary to go back to fundamentals

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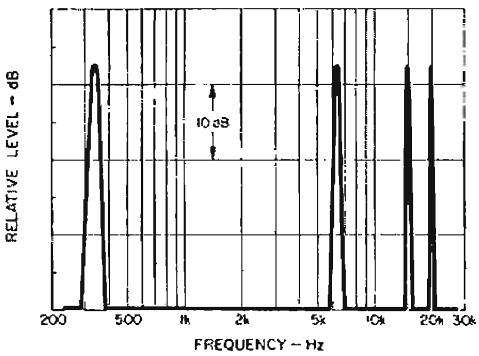
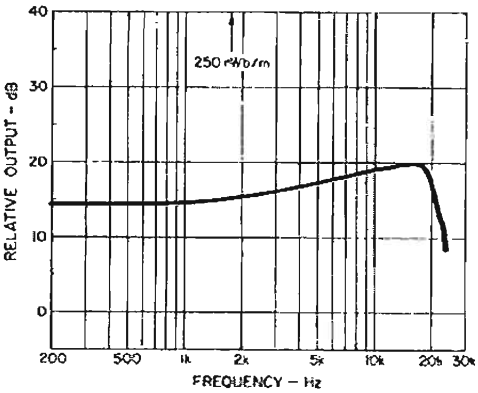
Illustration: Gene Greif

Fig. 1—Theoretically, adjusting a tape deck for flat record/playback response with a comb-frequency signal (A) should result in flat response. In practice, it yielded a peaked curve (B) due to dynamic biasing. On a deck set up for flat response, comb-frequency response declines at higher frequencies (C).



A



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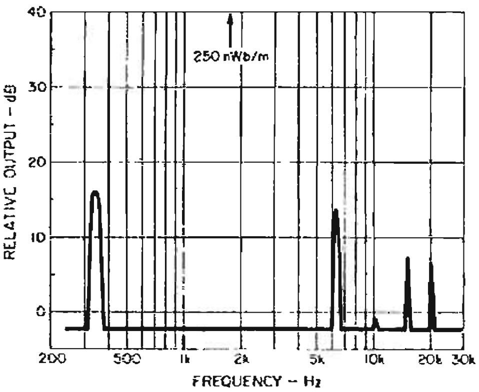


C

and examine the physical properties of tape and the recording process itself.

The Physics of Recording

Magnets are known by most people as pieces of metal that attract anything made of iron and are attracted to or repelled by other magnets. But magnets are also useful pieces of hardware, forming the basis for (among other things) motors and electrical generators—and for magnetic, or tape, recording.

If a current is passed through a coil, a magnetic field (or flux) is created around the coil, similar to the magnetic field created by a permanent magnet. The strength of the field depends on the amount of current, the number of turns of wire in the coil, and other factors. Obviously, the strength of the field that is generated can be regulated by controlling the electric current in the coil. If a suitable magnetic material is held in the flux, it will become magnetized, to a degree that depends on the strength of the flux passing through the magnetic material as well as on the properties of the material itself.

Conversely, if a magnet is moved near a coil of wire, a generated current can be taken from the coil. The amount of current depends on the strength of the magnet, its distance from the coil, the number of turns of wire in the coil, and so on.

These two fundamental facts are the basis of all magnetic recording and reproduction and, in the consumer field, of recording on and playback from cassette tape.

Recording is the process of creating a tape magnetized along its length to a degree which varies exactly as the sound signal received by the microphone. The microphone converts the sound it receives into an identical electrical signal, which is amplified and fed to the recording head in the recorder. In the recording head, the current passes through a coil, while the tape, of a suitable magnetic material, moves past the coil. The flux generated by the coil magnetizes the tape to a level proportional to the signal so that the audio signal gets recorded.

Similarly, during playback, a tape with the recorded audio signal passes by the playback head, generating in that head's coil an electrical signal similar to the pattern of magnetization on the tape. With luck, the voltage from a playback amplifier connected to the head will be an exact replica of the audio signal that was originally recorded. When amplified and fed to a loudspeaker, the reproduced sound will then be exactly like the original sound received at the microphone.

Although these are the basic physical laws of recording, in practice things are a little more complex. A modern cassette recorder is, in fact, quite an intricate piece of machinery, where each part interacts with all the others in a very precise manner. The degree of precision is a measure of the performance of the recorder, which, in a high-quality machine, can reproduce a very convincing copy of the original audio signal.

Recording on Tape

Recording tape is made of a thin layer of magnetic material deposited on a ribbon of plastic film. The magnetic material is not homogeneous, but is formed from millions of tiny particles, each of which is a magnet. The properties of these magnets depend on the material of which they are made (such as ferric oxide, chrome dioxide or iron powder) and are the reason for the familiar tape-type designations. Their physical shape and size, together with their density in the magnetic layer, determine the properties of the recording tape. It is important to remember that we are dealing with these tiny magnets, and that the physical laws that apply to magnets in general also apply to these magnets.

For the purposes of this article, we will assume that the playback chain can be made as perfect as we wish, and concentrate only on the process of recording the tape.

The audio signal to be recorded is constantly changing, and if these changes are to be accurately recorded as different levels of magnetization, then the flux must be concentrated at a very small portion of the tape. The coil in the recording head is therefore

Theoretically, adjusting bias with comb frequencies should yield flat response. The only problem is, it doesn't work.

formed around poles, which focus the recording field at a very thin-line air gap. The tape transport is made so that the tape slides past the air gap at a constant speed. It would thus appear that, provided we can feed an accurate copy of the audio signal to the recording head, we should get the required recording. But again, this is too simple. To understand why, we must look a little closer into the physics of the tiny magnets on the recording tape.

The strength to which a material is magnetized is, of course, related to the strength of the flux to which it is subjected. In turn, the flux generated is related to the current in the coils formed around the pole pieces in the recording head. But in both cases, the relationship is not linear. In other words, the strength of the resulting magnet is not always proportional to the flux, and the flux is not always proportional to the current in the coil.

The relationship between the residual strength of the magnet and the flux is shown in Fig. 2. The curve, derived from a hysteresis loop, shows that, for an increasing flux, the strength of the magnet increases at first at an increasingly rapid rate. Once beyond a certain point, however, the rate of increase slows, and finally magnetization reaches a maximum value above which no further increase occurs. The magnet is then said to be saturated. The important fact is that there is only a small part in the middle of the curve that is a relatively straight line, and it is only in this region that magnetization is proportional to the flux.

If the different magnetization levels required for a recording can be limited to the linear part of the hysteresis curve, an undistorted recording should result. Audio signals are both positive and negative, so recording will normally be made in the nonlinear portion of the curve near the axis. If a separately generated signal, called a bias current, is added to the audio signal, the average level of the audio signal may be raised so that only the straight portion of the curve is used for recording. In the early days of magnetic recording, a constant d.c. signal was used to bias the audio, but for the last 50 years, a bias current consisting of a high-frequency a.c. current has been used. This is called high-frequency (or H.F.) bias, irrespective of its actual frequency.

If a very-high-frequency signal is mixed with the audio signal, the sum looks like Fig. 3. The audio looks as if it is riding on the peaks of the high frequency. If the sum is fed to the coils, then the audio signal will be recorded at a higher part of the hysteresis curve, as shown in Fig. 4. If the high-frequency bias is at the correct level, a low-distortion recording is possible.

To obtain the best results, different formulations of tape require different levels of bias, because the shapes of the hysteresis curve for the different tape formulations are not the same. This is why many recorders permit the user to adjust bias for the type of tape being used, or, alternatively, the recorder is factory-adjusted for certain recommended types of tape. Once set, the bias, supplied by an oscillator, remains at a fixed level, which may be called the fixed bias of the recorder.

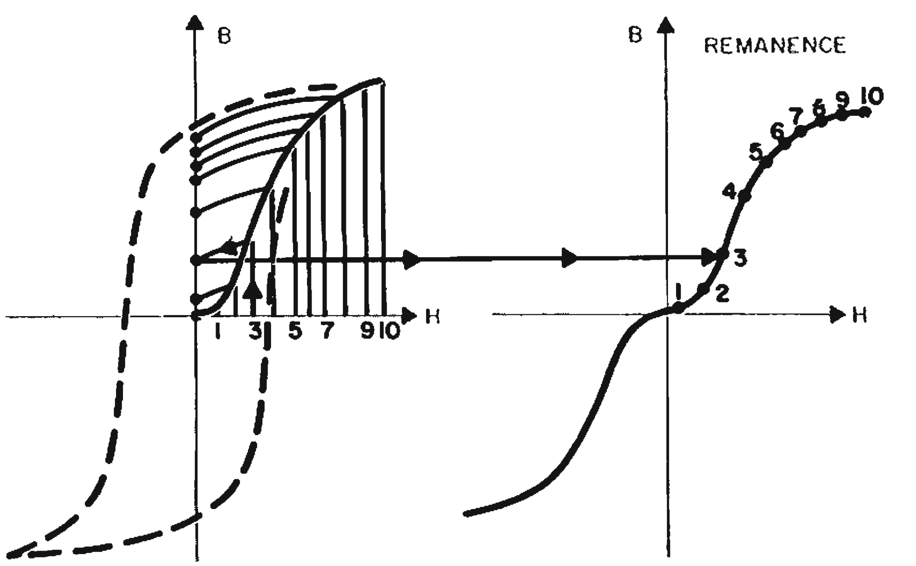
Bias for the lowest possible distortion is also different for different frequencies. In addition, bias affects other parameters, such as sensitivity and maximum output level (MOL).

Sensitivity is the ratio of the output signal to input signal, with all other variables kept constant. If the input is kept constant and the bias is changed, it will be seen that the output will vary, as shown in Fig. 5. Not only that, but the shape of the curve is different for different frequencies, as shown.

The same is true of MOL, the maximum output level that can be reproduced from tape. Above this level, the magnets on the recording tape are saturated, and distortion increases dramatically if a recording at a higher level is attempted. Again, MOL is dependent not only on bias but also on frequency, as shown in Fig. 6.

These factors are now getting complex, especially if we note that minimum distortion and maximum output do not occur at the same bias level. Of course, we always wish to record at the highest level consistent with acceptable distortion to keep tape noise or hiss as far below the recorded signal as possible. Thus, a correct setting of the fixed bias, while very necessary for good results, is always a compromise between low distortion and high recording level, for both high and low frequencies. For optimum results, setting fixed bias is best done by an experienced technician using accurate

Fig. 2—This transfer function shows the relationship between record-head flux ("H") and remanent magnetic field ("B") when recording without bias. Note how the curve's shape alters the relationship between signals of different amplitudes.



instrumentation, and, in fact, the spectral content of the type of music to be recorded should also be taken into account.

The Comb-Frequency Paradox

Now that we know "all" that there is to know about the physics of recording, we are now in a position to solve the paradox of the comb and single sine-wave frequency responses. We saw that bias, as used for recording, is a very high frequency compared to the audio signal. Thus, if we wish to record a bandwidth of, say, 20 kHz, it would be normal to use a bias frequency of

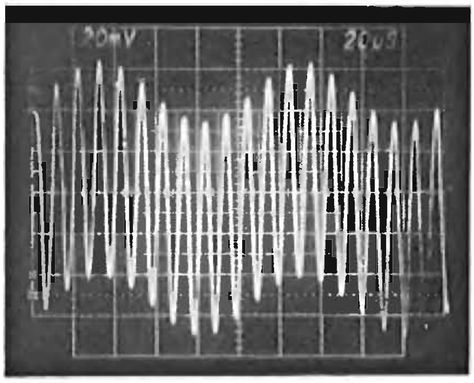


Fig. 3—A low-frequency audio signal riding on a high-frequency bias wave.

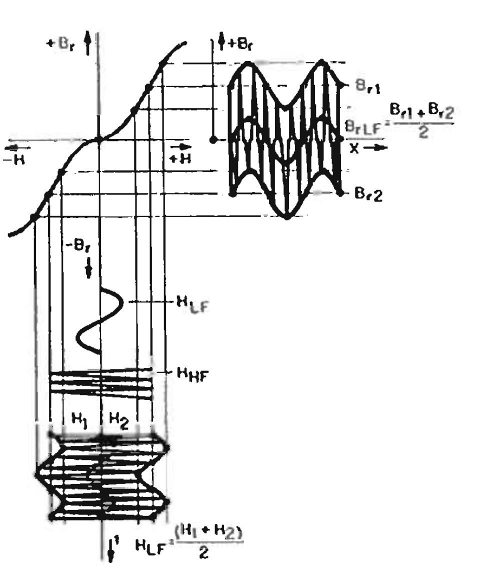


Fig. 4—How bias shifts the audio signal to the linear portions of the hysteresis curve, lowering distortion.

five times that frequency, or 100 kHz. The criterion is that the bias frequency must be sufficiently high as to allow the low frequency to ride on the peaks of the high-frequency signal.

If this is so, then bias at a frequency of 10 kHz will be fully adequate to bias an audio bandwidth of 2 kHz, and 1 kHz for 200 Hz, etc. But when the audio signal is a mixture of low and high frequencies, the low frequencies do not know whether high frequencies were put there by the recorder's designer to act as bias, or whether they are just a part of the audio signal itself. When a high frequency, whatever its source, is superimposed on a lower frequency, the high frequency will tend to raise the point on the magnetization curve where the low frequency is recorded, just as bias does. Thus, in an audio signal composed of many frequencies, each frequency acts either partially or fully as bias for all lower frequencies.

When the high-frequency part of the audio signal acts as bias, the bias seen by the lower frequencies will be the sum of the original fixed bias to which the recorder was adjusted plus the biasing effect of the higher frequencies. Therefore, bias will constantly change with the high-frequency content of the audio signal, and, for the low frequency, all parameters related to bias (such as sensitivity, distortion and MOL) will change with the high-frequency content in the same audio signal.

That this is true is seen from the case of the comb and sine-wave frequency responses. If the bias in a recorder is adjusted to a flat response with a swept sine wave, then all frequencies see constant bias while the test is conducted, that is, the fixed bias to which the recorder is adjusted. When multiple frequencies are present simultaneously, each frequency below the highest will be subjected to altered bias, which is the fixed bias plus the biasing effect of all the higher frequencies. As bias changes, the sensitivity for any frequency will not remain the same as before. The output at the different frequencies will therefore change, and the frequency response, as seen on an analyzer, will no longer be flat. In other words, we have dynamic frequency-response error.

It may not be out of place to repeat that this is what happens with all recorders that use fixed a.c. bias. This includes not only cassette tape recorders, but also reel-to-reel machines, studio machines and high-speed duplicators, although the amount of dynamic frequency-response error will vary from one type of machine to another and for different tape formulations. The amount of error is related to the ratio between the audio and bias currents fed to the recording head. The smaller this ratio, i.e., the closer the magnitudes of the audio and bias currents are to each other, the larger the error will be.

Active Bias

To sum up the problem, we have seen that the loss of high frequencies when recording at high levels is due less to tape quality than to the fact that there is an increase in the effective level of the bias current when high frequencies are a part of the audio signal. If active bias is too high, then sensitivity at high frequencies decreases, and the output at these frequencies drops. The larger the content of high frequencies, the greater the drop. Once this fundamental principle governing high-frequency performance has been established, the answer to our problem becomes almost obvious. This has been implemented in the recording circuit, HX Professional.

The total bias seen by the low-frequency component of any audio signal, as described earlier, is the sum of the fixed bias and the biasing effect of the high-frequency part of the same audio signal. It will be remembered that it is the immediate value of this sum which determines the recording conditions for the audio signal. It will be useful to use another term for the sum of the bias and the biasing effect of the audio current, and we will call this sum active bias.

Experiments using a high-quality sound source with high treble content and a cassette recorder using ferric tape show that active bias can vary by

When the audio signal contains both high and low frequencies, the highs act as added bias for low frequencies.

as much as 6 dB at different parts of the recording. This is equivalent to saying that bias is incorrectly adjusted by that amount, at least for some part of the recording. The consequences are obvious for anyone familiar with the procedure of adjusting bias. The disappointing result, a loss of treble performance—particularly with the less expensive tape formulations at high recording levels—is most often attributed to poor tape quality, tape saturation or other causes, rather than to a constantly changing bias.

Low frequencies see a uniform level of high-frequency bias, provided active bias is kept constant. This is accomplished when bias from the oscillator is reduced until active bias returns to the level of the "fixed" bias whenever high frequencies are present in the audio signal. The term fixed bias, of course, now loses its original meaning, as bias from the oscillator no longer remains fixed. So we define another term, no-signal bias, which is the bias level to which the machine is set with zero signal at the input, and it is equivalent to fixed bias in a conventional machine. We shall return to this, and see how no-signal bias can be optimized more effectively with the HX Professional circuit.

Keeping active bias constant implies two operations: First, monitoring the active bias continuously, and second, changing oscillator bias as necessary to keep active bias constant.

The black portion of Fig. 7 shows a typical recording circuit in a conventional recorder. The signal output from the recording amplifier is mixed with the high-frequency bias current, typically five times the frequency of the highest audio frequency to be recorded. Bias current or, more correctly, no-signal bias current, is set to the optimum level for the tape in use. The mixed audio and bias current is then fed to the recording head.

What is added to implement HX Professional is a high-impedance monitor for active bias. The ideal place to measure active bias is, of course, at the point where it acts, at the recording head. The sum of the audio and bias currents is first passed through a filter, and the voltage is monitored to determine the flux being generated by the head. At this point some wise men will shake their heads and say we should be measuring the current, as it is well known that the flux generated is proportional to the current, and not to the voltage.

While this is theoretically true, in practice such a measurement fails to take into account losses in the magnetic and electrical circuits of the tape head. For various reasons, these losses are not proportional to the current, and magnetization of the tape is therefore also not proportional to the current. This is a failing of certain systems designed to improve recordings, which base the improvement on a measurement of recording current.

So, in the HX Professional system, the voltage across the head is monitored and integrated to give a calculated value of the useful flux generated in the tape head. Flux is actually proportional to this calculated current, as the influence of head losses on the voltage across the head is very small. The calculated value of flux is therefore a very close approximation of the effective magnetizing field. Thus, if the characteristics of the filter are suitable for the purpose (that is, it is designed to reflect the level of active bias accurately and constantly), it can be incorporated into a feedback loop that will keep active bias constant.

The Filter

The characteristics of the filter may be found by experiment. Since we wish to minimize any changes in frequency response caused by the high-frequency content of the audio signal, we may be tempted to proceed as follows:

Using a low-frequency input signal of, say, 200 Hz, bias is adjusted to give the lowest possible distortion of that signal. Then, the low-frequency signal is recorded on tape at a fairly high level, say, 20 dB below the level at which the tape is saturated (that is, 20 dB below MOL). Its output level is measured as a voltage on the output terminals of the recorder. The same frequency is then recorded at the same level, but this time with a high-frequency signal superimposed, let us say 15 kHz, at a level sufficiently low to ensure that saturation does not occur.

When this is done, it will be seen that the output level of the low frequency falls, because sensitivity has changed due to the change in active bias. The amount by which it falls will be seen to depend on the relation between the two frequencies, as well as their relative levels. For this particular case, we can now reduce the bias current until the output of the low-frequency signal returns to the same level as the one it had before any high frequencies were superimposed.

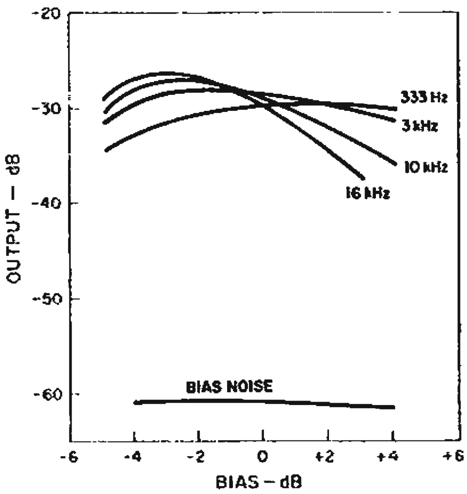


Fig. 5—The relationship between tape sensitivity and bias level varies with frequency.

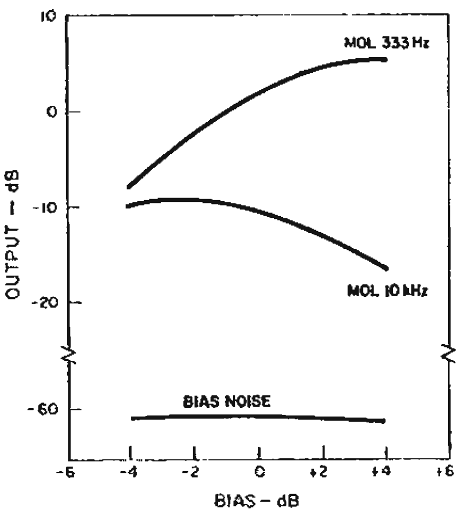


Fig. 6—MOL also differs with both bias level and signal frequency.

We have now achieved the condition, albeit manually, where no change in frequency response occurs due to the presence of the high frequency. The amount by which the oscillator bias needs to be reduced is then the biasing effect of the higher frequency on the lower. This procedure might be repeated for a number of different frequencies, at various input levels, when it should then be possible to work out a mathematical relationship between the ratio of the frequencies and their relative levels.

However, using a filter designed to this relationship does not lead to a flat frequency response. A slightly altered procedure is required, which will give the primary condition of a conventional flat frequency response and considerably reduced dynamic frequency-response error. The procedure to find the true value of active bias is similar in principle but uses extrapolated points on bias-related parameters as reference, rather than a simple frequency-response parameter. However, if the relationship derived from this procedure is fairly constant at various frequency ratios and relative levels, then a filter for this purpose may be economically constructed.

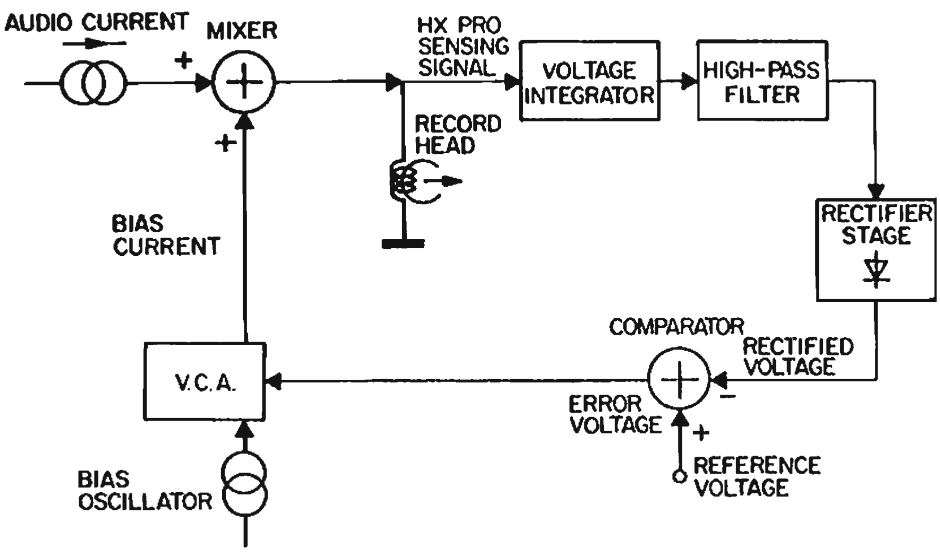
It is found experimentally that the biasing effect of a sine-wave signal on a signal very close to it in frequency is

virtually zero. As one of the frequencies increases, its biasing effect on the lower frequency increases at 6 dB/octave, provided they are both at the same level.

It turns out that the filter we require is one of the simplest among electronic circuits. In fact, it is so close to being a simple filter at 6 dB/octave that any attempt to get a more accurate form, at least for cassette tapes, is superfluous. In other words, if a low-pass filter, passing all frequencies below the highest the machine is designed to record, is put into the control loop, the active bias can be correctly set for all audio frequencies. And this setting adjusts itself, by its very nature, to all static bias levels and for all tape formulations.

Once the form of the filter has been determined, our problem is solved. It only remains to construct a suitable circuit that will implement the filter as part of a feedback loop to control the bias level. For a competent circuit designer, this really should present no particular problem.

Fig. 7—In conventional recording circuits (shown in black), low-frequency response can be diminished by the biasing effects of high audio frequencies. With the addition of the HX Professional circuits (shown in color), bias level is controlled by the total high-frequency signal (audio plus bias) at the recording head.



The Circuitry

It will come as no surprise at this point to see the final circuit, which is shown in Fig. 7. As stated earlier, the audio line feeding the recording head is conventional except that the signal is sensed at the head, to derive the signal required for HX Professional. An advantage of measuring at this point is that it allows the audio signal path to be kept totally free of any manipulation by the HX Professional circuit and therefore free from any possibility of degradation. Further, any necessary manipulation of the signal, such as pre-emphasis or noise-reduction circuits, is automatically taken into account when the signal is monitored at this point.

The signal monitored at the head, after passing through an integrator and the filter, is rectified to give a d.c. voltage proportional to the active bias. This is then compared to a previously set reference voltage. The presence of any high frequencies in the signal will alter the level of the rectified voltage by an amount dependent on the amplitude and frequency composition of the signal and the characteristics of the filter. When the rectified voltage changes, it will differ from the reference voltage, and a control signal proportional to the difference is generated. The control signal is used to alter the gain of the voltage-controlled amplifier, which in turn alters the bias current from the oscillator.

The reference is a stable, adjustable d.c. voltage set to a value that gives the required no-signal bias current. The difference between the rectified signal and the reference voltage is used as a control signal to adjust the bias level from the oscillator. It can be seen that in the absence of any audio signal, no-signal bias current at the tape head can be accurately adjusted for any tape formulation simply by changing the value of the d.c. reference voltage. Once this adjustment has been made, the circuit requires no further adjustment or correction.

The bias circuit for HX Professional differs from the conventional in that a voltage-controlled amplifier is placed between a conventional oscillator, which is also used in the erase circuit, and the point at which the audio signal is added to the bias in the recording circuit. This amplifier, as its name suggests, changes its gain under the control of a d.c. voltage to control the amount of bias current that is fed to the mixer. The d.c. voltage used to control the amplifier is, of course, the signal that is derived from the feedback circuit, which senses the signal at the tape head.

What now happens is that, for an audio signal composed of high and low frequencies, the active bias increases, and oscillator bias is reduced by a corresponding amount. The low frequencies see a total bias equal to the sum of the reduced oscillator bias and the biasing effect of the high frequencies. By definition, and courtesy of the HX Professional circuit, this value remains constant. The high frequencies see the reduced bias from the oscillator, but are far less sensitive to the biasing effect of the high frequencies themselves. In other words, they see a reduced level of active bias.

As we have seen earlier, high frequencies require a lower level of bias for optimum recording conditions than low frequencies do. Thus, the lower oscillator bias is advantageous for the high frequencies in the audio signal, and results in better MOL, besides lowering the dynamic frequency-response error. Together, these two factors lead to a dramatic improvement in the quality of recorded high frequencies, particularly when high recording levels are used. The improvement is more marked on less expensive tapes, but also substantial with the most expensive formulations.

HX Professional also has other advantages as byproducts. As mentioned, conventional recorders use a "fixed" bias adjusted to a value which is a compromise between levels that give the lowest distortion at low frequencies and the highest MOL at high frequencies. A compromise is necessary, as the optimum values for these characteristics occur at different levels of bias. The compromise is, of course, chosen to be somewhere between these two optimum values.

After implementing HX Professional in a prototype recording circuit, it was realized that the conventional compromise may be considerably reduced. Since the high frequencies take care of themselves, i.e., oscillator bias falls to accommodate high frequencies, no-signal bias may be adjusted close to the optimum value for the best low-frequency distortion. This lowers overall distortion in the recording at the same time that high-frequency recording is improved.

HX Professional and Dolby HX use similar principles but differ in their aims as well as in their implementations.

Finally, it was found that the tape overload characteristic for high frequencies, at levels above those where tape saturation begins to occur, is more gentle with the HX Professional circuit. Distortion rises at a slower rate as the normal maximum recording level is exceeded, permitting higher recording levels and a better signal-to-noise ratio without audible distortion at unexpected peaks in the audio signal.

HX Professional vs. Dolby HX

A description of HX Professional will be incomplete without mentioning the circuit from which it gets its name, Dolby HX. Although the names are similar, as are some of the principles on which the two circuits are based, they are in fact quite different in their aims as well as in implementation.

K. Gundry of Dolby Laboratories, the inventor of HX, realized that a wide-band audio signal has a self-biasing effect and that this is part of the cause of high-frequency losses in recording. The problem led him to develop a circuit where oscillator bias is changed when high frequencies are present in the audio signal, as is done in HX Professional. But his aim was to permit the maximum level of high frequencies to be recorded—in other words, to get the highest possible level of MOL. Hence the name HX, Headroom eXpander system.

However, in order to maximize MOL, oscillator bias must be reduced by more than is necessary to keep active bias constant. Also, a frequency-response error (in a sense opposite that of a conventional recorder, and at an unacceptable level) is introduced due to overcompensation. This was also recognized, and an ingenious correction was introduced.

Since Dolby noise-reduction circuits already measure the high-frequency content of the audio signal, in Dolby HX the same circuit was used to derive a signal to control oscillator bias. In addition, this control signal was also used to alter pre-emphasis to continuously correct the dynamic frequency-response error. The resulting circuit required fairly complex design and adjustment for both the oscillator and pre-emphasis to track correctly, and the problem became even greater when using different tape formulations.

Although dynamic frequency response, even with correction, was substantially less accurate than with HX Professional, HX did in fact function marginally better than HX Professional in improving MOL. The reason it did not become more popular was probably because tape recorder manufacturers were not willing to cope with the complexity of the adjustments required for correct operation.

HX Professional was developed at the same time as the Dolby circuit, although independently, but it was released later. And it is so fundamental to recording technology that Dolby Laboratories not only took part in the final stages of development, but agreed to license the circuitry, on behalf of the inventors, to manufacturers not only of cassette recorders but also of other recording devices (such as high-speed duplicators).

This explanation of the function of HX Professional is necessarily simplified in the interests of an explanation of the broad principles at the cost of rigorous detail. The authors hope that readers will take this into account. More exact formulations will be found in the references below.

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History of Magnetic Recording

PART I

BY ROBERT ANGUS

PHOTOGRAPHY BY ROBERT LEWIS

Thirty years ago, magnetic tape pioneer John Herbert Orr stood before a committee of the Alabama state legislature, trying to explain exactly what he did to produce a brand-new recording medium. "Son," drawled the chairman, "Y'all mean to tell us that what you do is take a strip of plastic and coat it with *barn* paint?" By the beginning of the 1980s, strips of plastic covered with iron (not barn) paint had become far more than a scientific curiosity, or the toy of audiophiles. The computer age, the universality of audio cassette recorders, and the popularity of home video recording had put those plastic strips covered with red paint well on the way to becoming, as past TDK President Sho Okiyama

observed recently, "the writing paper of the 21st century."

Even today, a strip of recording tape may not look like much. But behind it lies 90 years of intrigue and imagination, adventure, discovery and just plain fun, a story whose chapters are too unlikely to pass for fiction yet hard to accept as fact. It's also a timely story, because several noteworthy anniversaries take place this year, ranging from the development of recording tape in Germany and the founding of one of the world's largest manufacturers of recording tape in Japan exactly 50 years ago, the U.S. introduction of the compact audio cassette and the first home video recorder 20 years ago, the introduction of U-Matic VCRs 15 years ago, to the invention of Betamax 10 years ago.

Apparently, the first person to think of magnetic recording was Oberlin Smith, who in 1888 wrote an article containing the basic ideas for *The Electrical World*, an American magazine. After describing the use of woollen thread impregnated with iron powder, Smith confessed that the lack of a laboratory prevented him from testing his thesis.

The practical side of the story begins in 1893, in Copenhagen. Valdemar Poulsen, son of a judge, was about to graduate from the University of Copenhagen as an electrical engineer, with the promise of a job with the Copenhagen Telephone Co. before him. As a class project, Poulsen proposed a device which flew in the face of established scientific theory at the time—a device for recording sound on piano wire by means of magnetization. Of course, physicists of the day knew that it was possible to magnetize wire, but they also knew—or thought they did—that the impulses would flow along the wire and quickly disappear or evaporate into the air. Nevertheless, the Poulsen idea showed imagination and effort, and it earned him a top grade.

Poulsen had envisioned his Telegraphone, as he called it, as a means for improving the efficiency of already overloaded telephone lines, and it was this idea which attracted the interest of the telephone company. There is evidence that Poulsen and a fellow student, Peder Oluf Pedersen, produced a working model as early as 1894, but it wasn't until December 1898 that he received a patent for it. In order to pursue the Telegraphone, young Poulsen, drawing on his family's resources, quit his job with the phone company.

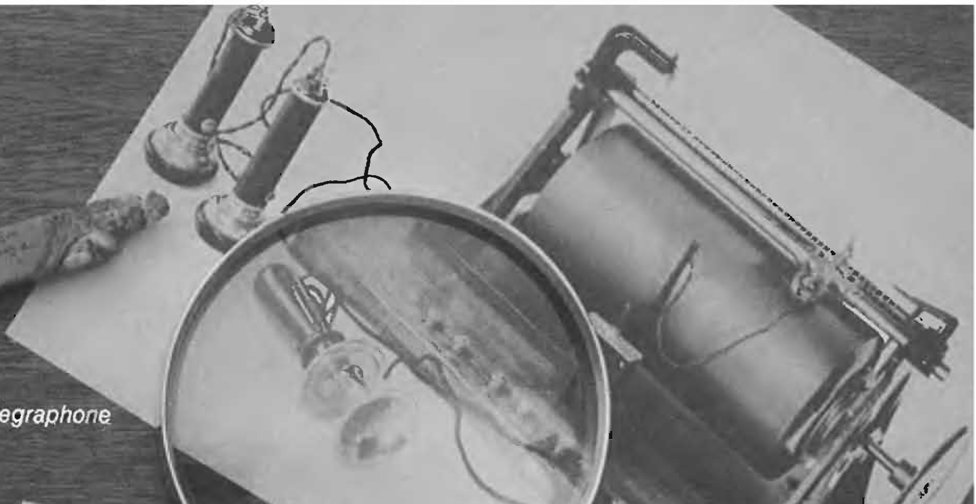
By the following summer, Poulsen had been manufacturing wire recorders for a year, at which time he realized that his recording device had at least as big a potential in the business world as Thomas Edison's Dictaphone. The problem was to publicize it, to raise money for manufacture and promotion.

Poulsen's first step was to pack up one of his machines and head for France, where it created a sensation at the Exposition Universelle in Paris. One visitor who was intrigued by the Telegraphone's possibilities was Emperor Franz Josef, the septuagenarian head of the Austro-Hungarian Empire. He insisted on making a recording—a testimonial to Poulsen's inventiveness which survives today.

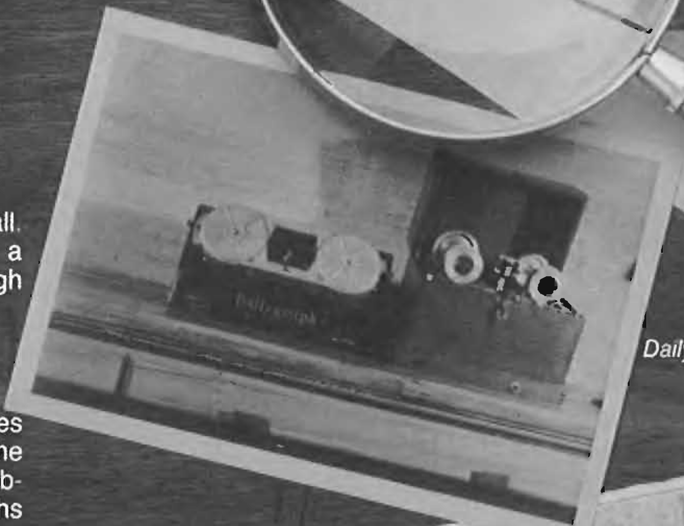
Shortly thereafter, Poulsen set sail for America, where he hoped to establish a company which could do a proper job of manufacturing and selling the Telegraphone throughout the world. For a variety of reasons, the American Telegraphone Company got off to a rocky start, running through its initial capitalization of $5 million (an enormous sum in those days) in eight years without, apparently, producing a single recorder. Then, something happened: A New England industrialist, Charles Dexter Rood, acquired control of the company and moved its factory to his home town of Springfield, Mass. Five years later, the factory was turning out several different variations of Poulsen's original design and selling them to such customers as Rensselaer Polytechnic Institute, E.I. du Pont de Nemours & Co., and the Imperial German Navy.

Another of American Telegraphone's customers was the Atlantic Communication Company, then engaged (1912) in erecting a giant shortwave transmitter and antenna complex at Sayville, Long Island. Ostensibly an American firm, Atlantic's station was linked to POZ, a station near Nauen, Germany, and there was suspicion that the company was trying to conceal ownership by the German company, Telefunken. The Sayville station, designed to carry commercial messages between North America and Europe, utilized the most advanced radio equipment and techniques, including automatic transmission systems which used paper tape and piano wire (the latter system, in fact, was an American Telegraphone).

With the outbreak of World War I in August 1914, radio amateurs along the East Coast began hearing strange transmissions from Sayville. There were commercial messages which made no sense and long series of numbers. And there were transmissions so fast that no operator could take down the Morse code signals, not to mention sounds which didn't seem



Telegraphone



Dailygraph/Telegraphone

like Morse code transmissions at all. "A musical note like the buzzing of a titanic bumblebee which sped through space," one writer described it.

There were rumors that Sayville was transmitting instructions to the U-boats then prowling the North Atlantic and that the coded messages were, in fact, intelligence reports. The rumors grew stronger when the submarine *Deutschland,* in the months shortly before U.S. entry into the war, paid goodwill visits to several American ports and members of the press who visited her saw two Telegraphones aboard.

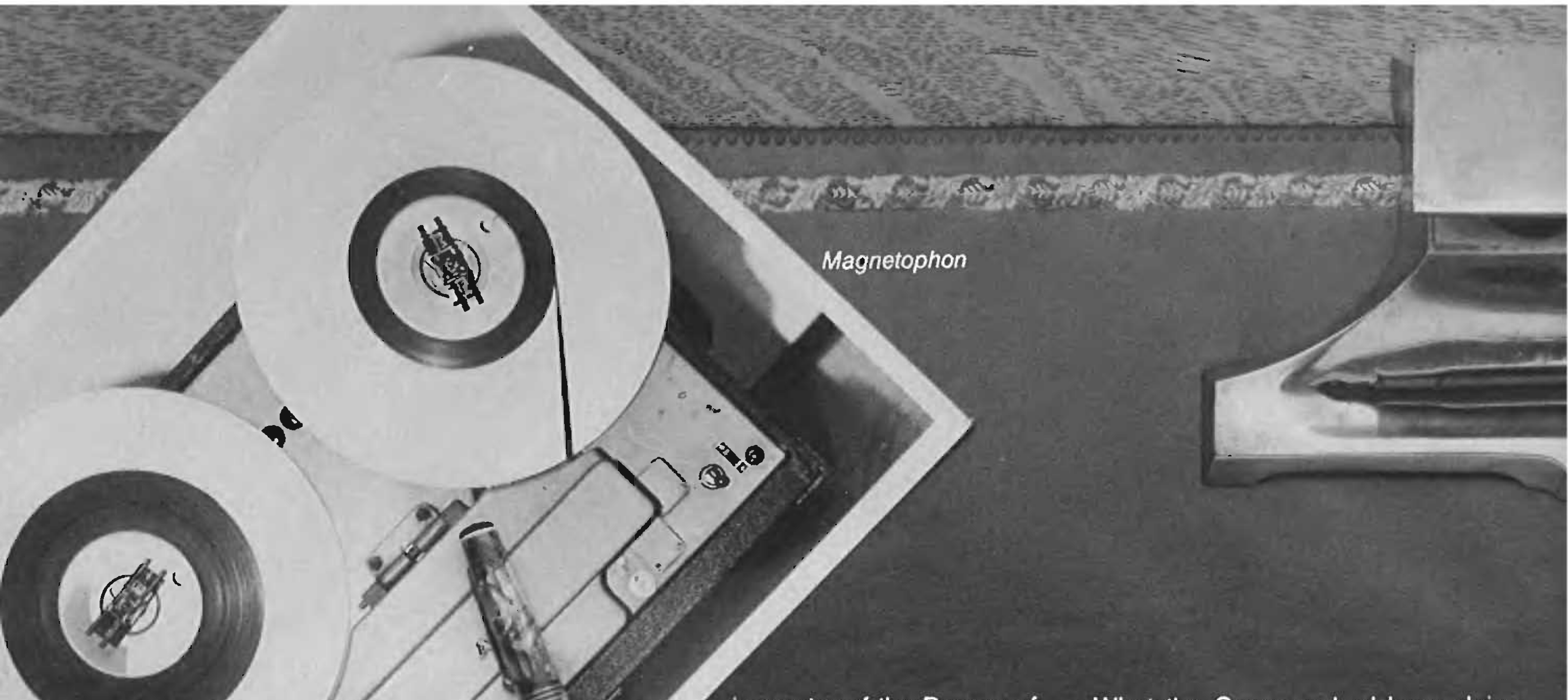
Just why the Telegraphones were there became obvious one morning late in June 1915. Charles Apgar, a New Jersey radio experimenter, puzzled by the high-pitched whistles emanating from Sayville, had managed to record some on a hand-cranked Edison cylinder phonograph. Forgetting to rewind the machine, he played one back and was astounded to discover, as the cylinder slowed down, that the whistle resolved into the normal dots and dashes of Morse code.

For two weeks, Apgar recorded Sayville transmissions, and then he took his find to L. R. Krumm, the chief radio

Sayville, N.Y.

Rood

Poulsen



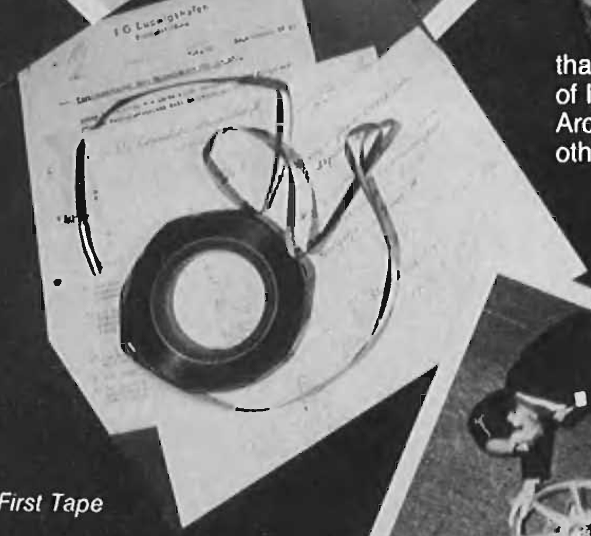
Magnetophon

inspector of the Bureau of Navigation in New York. What Krumm heard caused him to summon W. J. Flynn, the Secret Service's top agent in New York. Ten days later, the U.S. Navy seized the Sayville station and clamped a lid of secrecy on it which was to last more than half a century—until the Freedom of Information Act enabled the National Archives to make the cylinders and other data available.

What the Germans had been doing with their Telegraphones was exactly what Poulsen had proposed as a means of improving telephone efficiency. He had suggested that messages be recorded at normal speed, played over a telephone line at a much faster speed, recorded on a second Telegraphone, and then replayed at normal speed. The idea was that messages could be transmitted in a quarter to a tenth of their original time, thus cramming more onto existing phone lines. The trouble with the idea was that it made no provision for two-way conversation—but that hadn't been necessary for the German Navy.

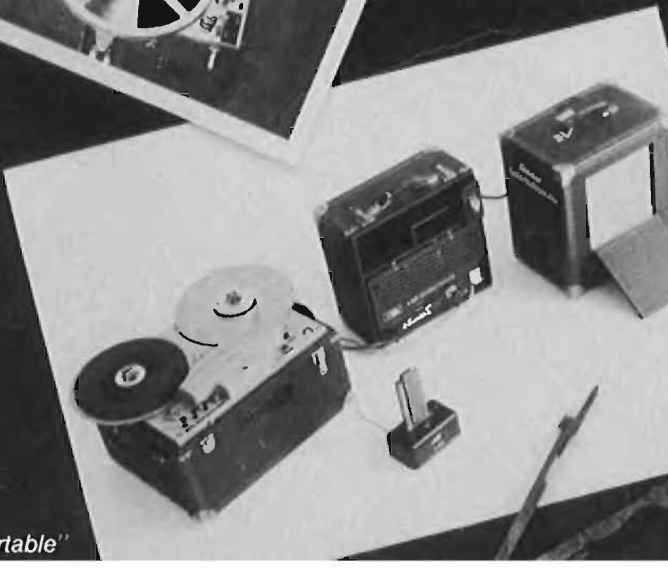
Actually, the Germans had been American Telegraphone's biggest customers. During the war, the Army Signal Corps had attempted to buy machines for its own use, but the company had delivered only a few defective ones. In 1918, the Poulsen patents expired, and American Telegraphone slid into a long period which produced lawsuits and charges against its president, Rood, rather than the Telegraphone recorders.

It would be another decade and a half before Poulsen's wire would give way to magnetic tape as we know it. In the meantime, a succession of German inventors and companies experimented with wire and metal strips. Of these efforts, perhaps the most spectacular was the Blattnerphone, a 6-foot-high monstrosity that looked something like two ancient Irish spinning wheels joined together and standing on end. It used a steel band, 6 mm wide, which one edited with metal shears and joined with solder. Purchased by the BBC in 1931, it was still in use in the 1940s when, as former BBC engineer H. Burrell



First Tape

Lorentz-Stille



"Portable"

Hadden recalls, "We lived in terror of those soldered connections coming apart, which they did every now and again. When that happened, strips of steel were lashing about and you ducked for cover to avoid being decapitated." It took two strong men to change reels on the Blattnerphone. In addition, this machine's speed accuracy was notoriously unreliable, varying according to room temperature and relative humidity, the day of the week and how much steel ribbon was on which spool.

In 1932, Badische Anilin und Soda Fabrik undertook a joint venture with AEG to develop a magnetic recording system along the lines of the Blattnerphone, but less expensive, more reliable and with better performance characteristics. BASF, then a division of the I.G. Farbenindustrie chemical combine, would develop a recording medium; AEG would make the playback equipment.

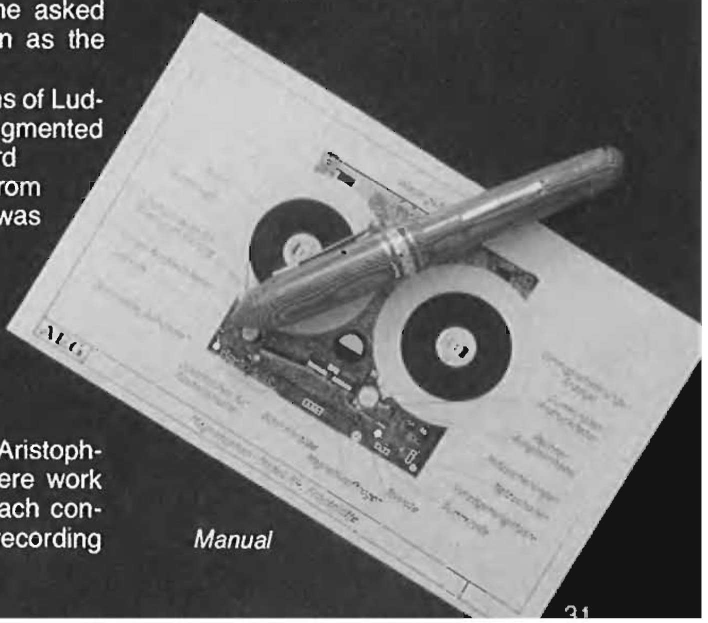
BASF had recently signed a contract with Fritz Pfleumer, an independent inventor who, in 1928, had proposed coating a strip of film or paper with a magnetizable powder. So, using Pfleumer's technology, the company managed to produce some 165,000 feet of tape, in time for AEG to demonstrate its recorder at the 1934 Berlin Radio Fair. For some reason, the formal demonstration was cancelled at the last minute, and AEG and BASF took their respective products back to the laboratory for more work. The following year, however, both were ready, and the new recording system was one of the highlights of the 1935 Radio Fair.

The next year, 1936, Sir Thomas Beecham and the London Philharmonic Orchestra were touring Germany. On the evening of November 19, they were scheduled to appear at the Feierabendhaus in Ludwigshafen to perform for BASF's workers and their families, and BASF's engineers thought this would be a wonderful opportunity to try out their new recording system. The musicians arrived two hours early, for rehearsal, and Sir Thomas was in a testy mood. It didn't get any better when, just as the orchestra was getting ready to rehearse the Mozart symphony scheduled for that evening, the mayor of Ludwigshafen stepped out on stage and proceeded to make a speech. It finally came to an end, and the orchestra began to play. Only then, when the conductor was able to evaluate the really fine acoustics of the hall, did his mood begin to improve.

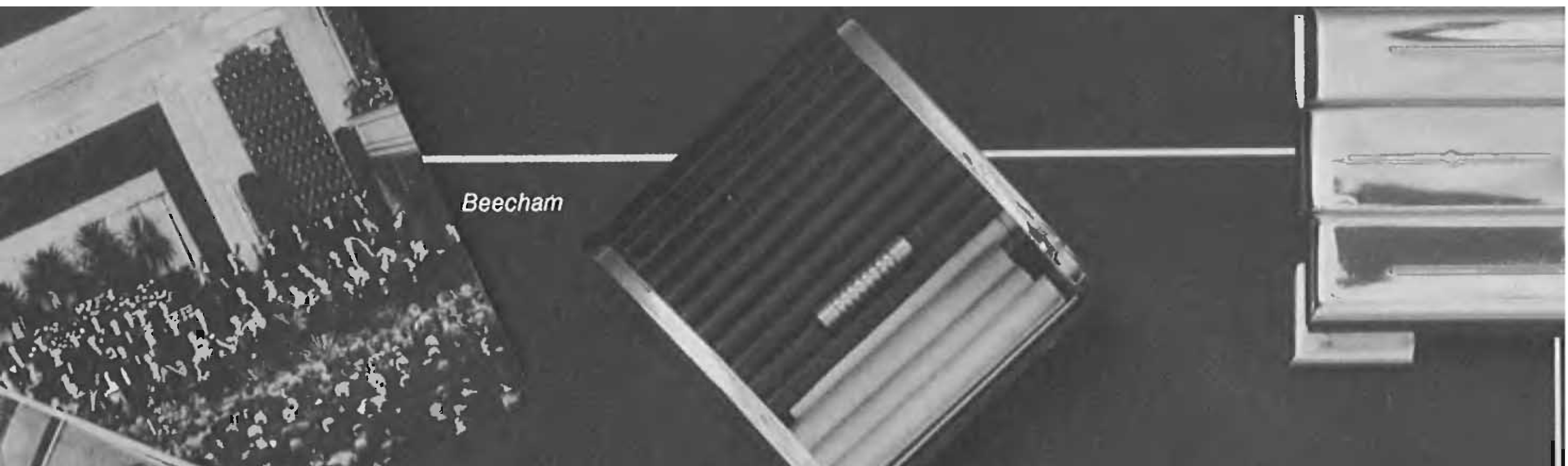
At that point, somebody told him that the concert would be recorded using a brand-new film technique instead of discs. Sir Thomas always took a keen interest in the technical side of recording and music-making, and he asked to hear the playback as soon as the concert was over.

At 8 o'clock, the good citizens of Ludwigshafen took their seats, augmented by music-lovers who had heard about the concert seemingly from all over Germany. (Beecham was a noted interpreter of Mozart, and besides, this tour included two works by Frederick Delius receiving their first performances in Germany.) The lights went down, and Sir Thomas started with Vaughan Williams' "Wasps of Aristophanes" Overture, also a premiere work for the tour. The tape reels each contained less than 20 minutes recording time, so the engineers waited until the second item on the program, the Mozart Symphony No. 39. They had set their levels, and as the orchestra played, the reels turned. The tape ran out before the music ended, and Sir Thomas was well into Delius' "Summer Night on the River" before they were able to rethread the reels. They then proceeded to capture the next two items on the program, "On Hearing the First Cuckoo in Spring" and a spirited performance of Rimsky-Korsakov's "Coq d'Or" Suite.

When it was all over, there was tremendous applause. The BASF directors had scheduled a gala banquet to honor the conductor, and a number of important people had been invited to attend. But Beecham wasn't anywhere in sight. He'd stopped backstage to listen to the playback, then demanded to



Manual



Beecham

Magnetic Recording: An Aural History

Crosby

Apgar

The seven cuts on this collector's-item Eva-Tone Soundsheet record represent important milestones in the history of magnetic recording. They have been assembled from archives around the world and include some sounds believed lost forever.

1. The first voice you hear is that of the Emperor Franz Josef of Austria, speaking at the Paris Exposition of 1900. It is the oldest magnetic recording known to exist and was made by the inventor of the Telegraphone, Valdemar Poulsen. In it, the Emperor acknowledges Poulsen's scientific achievement and expresses his thanks for the opportunity to make a recording.

2. Charles Apgar—the man who exposed a radio station in Sayville, Long Island, as transmitting intelligence information to the Germans—told a nationwide radio audience, in December 1934, how he managed to crack the "code" the Germans were using. (It was actually wire-recorded telegraph code, sent at high speed.) The occasion was the opening of a radio museum at Rockefeller Center in New York City.

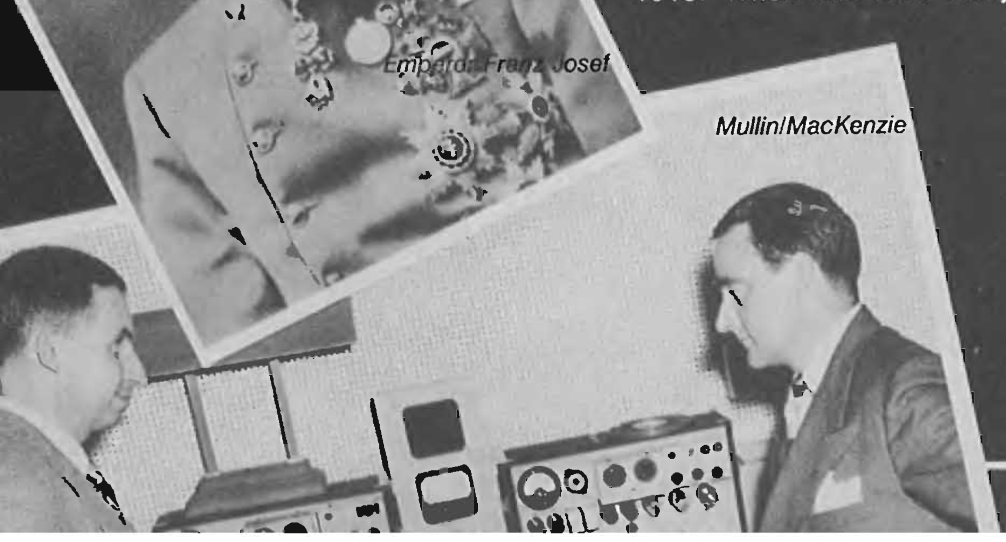
3. This message, in Morse code, was recorded by Apgar on an Edison cylinder machine in June 1915. When the U.S. Navy seized the radio station at Sayville, it also clamped a lid of secrecy over the Apgar recordings which was not lifted officially until the 1970s, under the Freedom of Information Act. This rare sample was provided by the Antique Wireless Association of Holcomb, N.Y.

4. On the night of November 19, 1936, the London Philharmonic Orchestra and its conductor, Sir Thomas Beecham, were scheduled to perform at a concert hall in Ludwigshafen, Germany. Engineers at BASF, whose chemical plant was located there, thought it would be a wonderful opportunity to test their new magnetic recording tape. What resulted was a remarkable sonic document: A candid portrait of one of the world's great conductors in his prime before a live audience—a first on magnetic tape—and a demonstration of just what the new process of tape recording could do. The musical excerpt is from the second item on the program that night, Mozart's Symphony No. 39.

5. In 1931, the British Broadcasting Corporation acquired a magnetic recorder which used a ribbon of steel as the recording medium. Used to record King George V's Christmas Message that year, the ungainly recorder quickly became a mainstay of the BBC's new Empire Service. The man responsible for its operation was Lynton Fletcher, who later became head of BBC Sound Recordings. In a recording actually made on a Blattnerphone, Fletcher recalls what it was like.

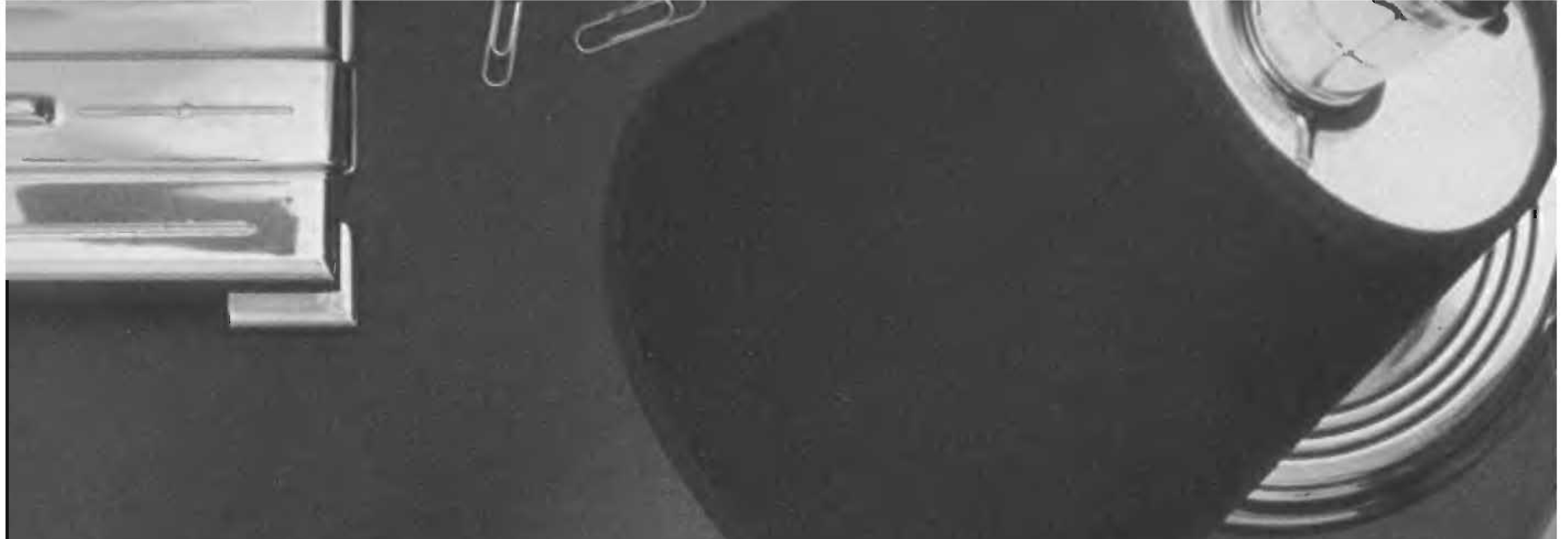
6. For the first time, in his own words, Jack Mullin recounts the steps which led up to his discovery of the Magnetophon, a high-fidelity tape recorder, at the end of World War II. Mullin points out that the decision he took that day changed his life forever. It also changed the broadcasting and recording industries in America and around the world.

7. Toward the end of August 1947, as an experiment, Jack Mullin recorded this, the first Bing Crosby radio show of the 1947-1948 season, using one of the Magnetophons and captured German recording tape which he'd brought back from Europe. Mullin edited the tapes into a finished program which then was transferred to disc for broadcast on the night of October 1, 1947. This recording is from a copy Mullin kept as a souvenir of the event. (Used by permission of HLC Properties Ltd.) *R.A.*



Emperor Franz Josef

Mullin/MacKenzie



hear parts of it again and again. One director anxiously reached his secretary and said, "Six hundred people are waiting for Sir Thomas." By the time he had his fill of listening to the miraculous new medium and discussing it with the technicians, Sir Thomas wasn't interested in banquets. He got into his car and headed straight back to his hotel.

This was the first-ever tape recording of a live concert, and BASF recently re-created that concert in the rebuilt Feierabendhaus (the original was destroyed during World War II). As a result, a digital recording of the complete program from that historic night exists.

By the late 1930s, radio stations throughout Germany were busy building libraries of recorded music—mainly classical, and ranging from short piano pieces to complete operas—all on tape. In Britain and the United States, music, particularly classical, meant either live performances on the networks or late-night concerts featuring noisy 78-rpm discs, with breaks every three or four minutes. Although it seems incomprehensible today, nobody outside Germany paid any attention to the taping activity there.

In the United States, one day in 1939, a junior at the Illinois Institute of Technology was faced with a problem: His cousin was studying singing and needed a low-cost device which would enable him to hear himself as he practiced. There were disc recorders, of course, but the recording time was woefully short, the cost of discs high and the machines bulky and difficult to operate. So Marvin Camras began experimenting with a device which could record sounds on piano wire, something like Poulsen's. By the time he graduated the following year, Camras had developed a machine which would record a wide variety of sounds for electronic playback. That wire recorder, which he took to class to show his professors, landed Camras a job with the Armour Research Foundation. And during World War II, that foundation would land a contract to supply wire recorders to the Navy (altogether, some 10,000 recorders between 1942 and 1945). Among Camras' achievements was his independent discovery of a.c. bias. [See the article, "Dynamic Bias Control with HX Professional," in this issue.] Unbeknownst to him, he had been preceded by others (including Dr. Hans-Joachim von Braunmuhl, a German radio engineer, in 1938 and an American, W. L. Carlson, in 1927).

Before we leave the 1930s, however, there are three other developments which attracted little attention at the time but are worth mentioning today. In 1934, just 50 years ago, the Fuji Photo Film company was formed in Japan, and its primary business has been the manufacture of photographic and motion picture film. The firm is also celebrating one 20th and two 25th anniversaries this year. It has been 25 years since Fuji produced its first videotape (it still supplies tape to Japan's NHK television network) and opened a U.S. office. And 20 years ago, Fuji started making audio tape (8-track cartridges, with open-reel tape the next year, 1965, and compact cassettes in 1968).

In 1935, the Tokyo Electro-Chemical Company (TDK) was formed in Japan to exploit a patent for ferrite cores. At the beginning, the company and its products had nothing to do with magnetic recording; the ferrite products were used in radios and industrial and communications equipment. It would not be until 1952 that TDK would turn its attention to the manufacture of recording tape and magnetic heads for tape recorders. But this firm would make up for lost time during the next three decades, with a string of innovations

which, by the early 1980s, would make it one of the largest manufacturers of recording tape in the world.

In 1936, Luigi Marzocchi (hardly a household name, then or now), a Milanese inventor, came up with a brand-new idea in recording which was to make reasonably priced home video recording possible 40 years later. Apparently unaware of the developments in Germany, Marzocchi developed the idea of mounting two or more electromagnetic heads in a single assembly which rotated at an angle to the tape's travel. This laid down a series of helical tracks across the width of the tape rather than a single, longitudinal track. Through this process, Marzocchi hoped to cram more signals onto a given length of tape. Late in the 1960s, an American inventor, Chester Newell, would resurrect the idea.

Just about this time, AEG attempted to interest General Electric in a license for the Magnetophon, as the tape recorder was then called. GE's experts, however, decided the product had no future.

During World War II, AEG and other German manufacturers developed a family of audio recorders designed for

Continued on page 96

If it weren't for the work of two Army Signal Corpsmen, Mullin and Orr, we might still be recording on steel wire.

MAGNETIC RECORDING
Continued from page 33

just about everything, from field use by military commanders and journalists to high-fidelity broadcast use. At the same time, the U.S. and Allied commands made do with wire recorders.

We might still be recording on wire if it weren't for the work of two Army Signal Corpsmen, John T. Mullin and John Herbert Orr. The fall of 1945 found Mullin in occupied Germany, part of a team looking for anything of electronic interest left behind by the defeated German army. One day, while he was checking out an antenna site north of Frankfurt, he met a British officer on a similar mission.

"Have you heard the tape recorder they have down at Radio Frankfurt?" the officer asked. Mullin had heard German tape recorders before—lots of them, abandoned in the field by retreating troops and correspondents—and he had been unimpressed by their sound quality.

Still, it was a glorious day for a drive, the afternoon was still young and Mullin had nothing else scheduled, so he set off for Bad Nauheim, where Radio Frankfurt actually was located. The station was being operated by the American Armed Forces Radio Service, but the technicians were predominantly German. When the officer in charge learned what he wanted, Mullin recalls, "He summoned an assistant, who clicked his heels and ran back into a room and came out with a roll of tape and put it on the machine. That's when I really flipped, because I had never heard anything like that. To my knowledge, there had never been anything like that in recording before. You couldn't tell whether it was live or playback; there was no background noise. I was thrilled."

So thrilled was Mullin that, after he had rounded up the two working Magnetophons required by the Signal Corps, he managed to find two more for himself. These he disassembled into parts small enough to be shipped home to his mother. "My biggest problem was getting the case into a mailbag. You couldn't send anything that wouldn't fit in a mailbag, and, of course, the case had to enclose everything else." Nonetheless, he did it, mailing back 35 packages in all, including some recording tape. After his discharge and return home, Mullin began reassembling his machines.

Meanwhile, another Signal Corpsman, Major John Herbert Orr, had been assigned the task of getting BASF's tape manufacturing plant at Wald Mittelbach back in operation as quickly as possible. There, he worked with Dr. Karl Pflaumer, a chemist who was responsible for the oxides BASF had been using to coat tapes all during the war, and the two developed a close personal relationship.

Toward the end of his tour of duty, Orr was involved in an automobile accident which put him in the base hospital. While recuperating there, as he told it later, he was attended by an Irish nurse who helped him pass the time. In appreciation, he was later to name the company he established to manufacture tape in the U.S. after her—Irish Tape. When Orr finally was discharged from the hospital and headed homeward, Dr. Pflaumer gave him a going-away present, a brown paper bag. Inside the bag was the iron-oxide formulation BASF had been using to make its tape. Back home, Orr bought himself an abandoned prisoner-of-war camp on the outskirts of his home town of Opelika, Ala., and turned it into a tape factory.

The regular meeting of the San Francisco chapter of the Institute of Radio Engineers was scheduled for May 16, 1946, at the NBC studios. The featured speaker was John T. Mullin, who would talk about and demonstrate the two tape recorders he'd been using to do studio sound recording for several months. Among those jammed into the studio audience was Harold Lindsay, who would later join a small company named Ampex (which had been making precision motors for the Navy). Lindsay told Ampex founder Alexander M. Poniatoff what he had seen. Sometime later, Murdo MacKenzie, Technical Director of Bing Crosby Enterprises, also heard about the Magnetophon and asked Mullin to repeat the demonstration for him.

MacKenzie realized at once that Mullin's tape recorders were the solution to a serious problem: "The Bing Crosby Show" was the only major network radio program not broadcast live.

Mullin's tape recorders were the solutions to a serious problem, editing the Crosby Radio Show down to the proper time.

Crosby preferred to record his show in advance, using 16-inch transcription discs for the purpose. To edit out fluffs and to cut programs (which frequently ran overtime) down to size, Crosby's engineers dubbed from one disc to another, with a resulting loss in sound quality with each transfer. Since some transfers were made three or four times, the final version didn't always sound very good.

At the end of the demonstration, MacKenzie asked Mullin to bring his machines back two months later to record and edit the first show of the season, a kind of dry run. It went so well that the network (ABC) dubbed Mullin's edited tape onto transcriptions for broadcast, and he undertook the job of taping shows for the rest of the season. Meanwhile, Ampex's Lindsay had taken careful measurements of Mullin's Magnetophons, and Ampex engineers were busy designing their own recorder, based on them. Mullin's machines stayed in use until Program 27, when the first Ampex recorders showed up and 3M's Scotch recording tape arrived to replace Mullin's original German-made tape.

"All I had was the original 50 rolls of tape I'd brought with me from Germany," Mullin recalls. "I didn't dare throw anything away when I edited, because I didn't know where I could get any more. So after every show, I'd go through and take the tape apart, splicing all the little bits together so that I could use them over again." By the time the first American-made tape arrived, he continues, there were more splices than tape in the original rolls.

BASF's early tapes had used magnetite, a black powder of ferric-ferrous oxide. By the end of 1936, magnetite had given way to gamma ferric oxide, a brownish powder, which formed the basis for virtually all American recording tapes from the introduction of Scotch 111, by 3M, in the late 1940s until the introduction of chromium dioxide in 1969. Unlike the 1980s, when hardly a week seems to go by without the announcement of some dramatic improvement in tape and particle technology, the 1950s and 1960s were stable times, an era when Scotch 111 could remain the standard to which all other tapes were compared for a full 20 years.

Photo Captions and Credits

Page 27, Telegraphone by American Telegraphone, Springfield, Mass., from the author's collection.

Pages 28 & 29, clockwise from top, Poulsen Telegraphone with steel wire on reels (photo by C. G. Nijsen, Philips International, in "—And the Music Went Round and Round . . . , Part 2," *JAES*, April 1984); Dailygraph c. 1920 and Telegraphone from the collection of Prof. Harold Layer, San Francisco State Univ.; transmitting equipment at the Sayville, N.Y. station; Valdemar Poulsen, inventor of the Telegraphone, and Charles D. Rood with neighborhood children, c. 1906. Props: Lamp, Guinevere Antiques, N.Y.C.; desk set, Niccolini Antiques, N.Y.C.

Pages 30 & 31, counterclockwise from top, AEG Magnetophon in J. Herbert Orr's collection (photo by Donna Foster Roizen); a sample of BASF's first trial of tape production in 1932 (photo from BASF); Lorentz-Stille steel tape recorder, c. 1933, now on display in Swiss Broadcasting's Basel studio, holds 3,000 meters of steel tape for 30 minutes playing time at 1½ meters/second, viewer is C. G. Nijsen (photo from Nijsen/*JAES*); AEG "portable" Magnetophon which weighed over 100 pounds and included recording module, amp, speaker, and mike (photo from BASF), and page from the manual for the K4 Magnetophon, c. 1940 (photo from BASF).

Page 32, from top, the London Symphony Orchestra, led by Sir Thomas Beecham, on Nov. 19, 1936, played the first concert ever to be recorded on tape (photo from BASF); Bing Crosby first used tape in 1947 as the basis for syndication of his radio program; Charles E. Apgar unravelled the high-speed radio code transmissions from Sayville, N.Y.; the Emperor Franz Josef was very interested in Poulsen's Telegraphone at the Paris Exposition of 1900, and John Mullin with Murdo MacKenzie, who was Bing Crosby's technical producer. Props: Mood Indigo, N.Y.C.

Part II of "History of Magnetic Recording" will appear in the September issue.

Part II

HISTORY OF MAGNETIC RECORDING

ROBERT ANGUS

In 1947, while John Mullin was proving the worth of the Magnetophon tape recorder he had brought back from Germany, a research team at Bell Laboratories (consisting of physicists John Bardeen, Walter Brattain and William Shockley) was developing a device which would revolutionize tape recording along with just about every other aspect of electronics—the transistor. Manufacturers of magnetic recording equipment didn't pay much attention—they were too busy developing wire and tape. Magnecord, which had been organized in 1946 to produce wire recorders (first for broadcast use and almost immediately for public consumption) abandoned that format in favor of tape in the fall of 1948. Magnecord's PT-6 quickly became the workhorse of recording studios and radio stations, as well as finding favor among hi-fi buffs.

One day the following year, David Apps of General Motors Laboratories approached Magnecord with an unusual request: Could the company build a tape recorder which could capture sound stereophonically? GM had been using a PT-6 to analyze automobile noise, but the results were unsatisfactory because they lacked spatial perspective. Perhaps a stereo unit might solve the problem. Magnecord quickly modified a PT-6, installing two record/play heads 1-5/16 inch apart along the tape path. The upper one recorded left-channel sound, the lower one the right. Having made its first stereo recorder, Magnecord made two more for display at the 1949 Audio Fair in New York.

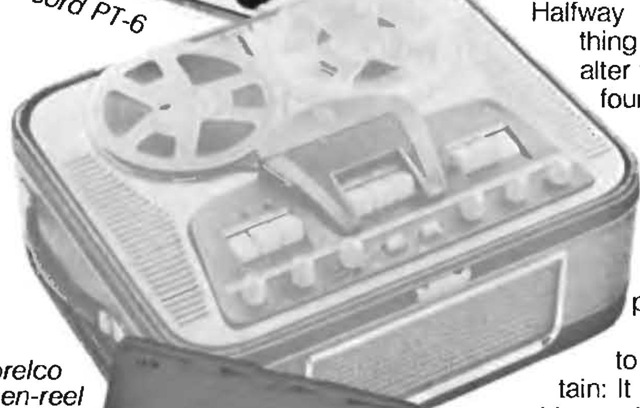
The musical applications of the stereo recorder were obvious to Magnecord's engineering-oriented management team, so in 1950 they hired a musical coordinator, a young recording engineer and audiophile named Bert Whyte. Whyte's first chore was to use the stereo machine to record as many different types of music as possible. A highly personable and persuasive man, the Chicago-based Whyte proceeded to talk just about every musical agglomeration that blew through the Windy City into letting him show them what stereo sound could do for their music. Benny Goodman, Lionel Hampton, Jimmy McPartland, the U.S. Navy Band, Woody Herman, and the Chicago Symphony all passed before Whyte's Magnecord for a series of recordings, many of which still exist. None, however, has ever been released commercially because of union problems and the fact that they were recorded as experiments.

Talking Woody Herman and Goodman into letting him record was child's play compared to Whyte's next challenge, explaining stereo to James Caesar Petrillo, then president of the American Federation of Musicians. Six years earlier, Petrillo had brought all commercial recording in America to a screeching halt while he held out for a recording trust fund for his members. The major record companies, who had been bloodied in that battle, weren't about to tangle with Mr. Petrillo again.

Magnetic recording was known before the turn of the century and recording on tape has its 50th anniversary this year. But it was only after WW II that tape recording truly came of age—thanks to important developments like polyester backing for tape, stereo recording, the Philips cassette, chrome particles, video recording, noise reduction, etc.

Norelco 2500 cassette player

Magnecord PT-6



Norelco open-reel

Scotch 111 "Living Letter"

Whyte and Magnecord, on the other hand, had nothing to lose. So he made an appointment to see the union leader, bringing with him a tape he'd made of a Leopold Stokowski concert in Urbana, Ill. "He sat there and listened," Whyte remembers, "and as he listened, I could see his face clouding up. When the tape ended, he scowled and said to me, 'Of course, you realize that we'll have to work out a pay scale for stereophonic recordings. Since there are two channels on the tape, I'd consider that two recording sessions. Each man in the group would have to be paid twice.' "

Whyte realized that stereo would never go anywhere under that arrangement. The record companies simply wouldn't sit for it. "Yes, Mr. Petrillo," Whyte replied, "but don't you see? If stereo catches on, everything will have to be rerecorded since there are no stereo recordings now." Petrillo thought that over, considering the increased work for his members—and the jump in record company contributions to his new trust fund. The scowl gradually faded into a benign smile. "Yeah," he answered, and the meeting was over.

Halfway around the world, something was happening that would alter the tape scene just as profoundly as had any of the previous developments. In the fall of 1945, a handful of Japanese engineers had leased some space in a burned-out Tokyo department store to start a company. They weren't sure exactly what they were going to do, but one thing was certain: It would be different from the things other companies were doing. Their first efforts were handmade pieces of broadcasting equipment to be used by the government radio network, NHK.

One day in 1948, the company president, Masaru Ibuka, was delivering a piece of equipment to the NHK studios when he happened to see a device on the desk of one of the U.S. occupation officers who oversaw NHK operations. It was a tape recorder. Suddenly Ibuka knew what the company was going to make.

It took the company which was to become Sony two years to do so, partly because nobody in Japan knew how to make recording tape. Ibuka and his associates experimented first with cellophane as a base, but found it stretched under tension and expanded in humid weather. Next, they tried paper. The results were better, but the paper tore every time the tape caught in the transport. "It was a blessing in disguise," Sony's Board Chairman, Akio Morita, whose cousin supplied the paper, recalled recently. "We had to take painstaking care in designing and manufacturing all of the parts along the tape path, to be sure the tape wouldn't snag." Old-timers say that at first the paper was laid out on the floor of Sony's "new" factory, a warehouse in the Shinagawa district of Tokyo. Workers then ran back and forth along its edges, spraying on the magnetic powder and binder with an airbrush.

Morita today admits that those first tapes weren't very good, particularly when compared to 3M's product. But they improved, and by late 1949 Sony had a prototype tape recorder. Early in 1950, the first Sony recorders went on sale in Japan. They cost $400, a lordly sum in postwar Japan, weighed over 100 pounds and were about the size of a small steamer trunk.

Back in the United States, some visionaries were talking about the day when tape might replace the ubiquitous long-playing record. Recording Associates, as early as 1950, published a catalogue of recorded music on tape, and other entrepreneurs talked confidently of putting the major record companies out of business. The record companies responded initially by ignoring the whole thing.

As a result, companies who wanted to offer music on tape either had to record their own or acquire recordings from European sources who hadn't been able to make deals with the independent U.S. labels. The result generally was music of less than compelling interest performed by artists nobody'd ever heard of. One of the first to record specifically for tape—and to observe professional quality standards—was

Chet Smiley with several Ampex open-reel machines.

Bert Whyte

James C. Petrillo

Chet Smiley of Livingston Audio Products. Smiley packed up a stereo Magnecord in 1951 and flew to Florence, Italy, to record a number of classical performances at the May Festival. To these he added organist Hack Swain in a number of programs of old favorites and finally the *piece de resistance*—Lenny Herman and the Mightiest Little Band in the Land, then performing at the Hotel New Yorker.

Then, in 1954, the dam broke, with RCA Victor the first major label to produce its own prerecorded open-reel tapes. In quick succession came Vox Productions, Mercury Records (for whom Whyte had been consulting) and Westminster. The tapes' cost was high—$12.95 for a half-hour stereo program, at a time when discs cost $4 and frequently contained 50 percent more music.

Everybody loved tape's high-fidelity sound—and the fact that, in the early 1950s at least, it was the only way of hearing true stereo. But cost was an important consideration. So was convenience. Together, those two factors spawned a number of unlikely products, beginning with a device which fit on top of conventional professional turntables, converting them into tape decks. The three-wheeled unit, made by Presto, included takeup and feed tape reels, a playback head assembly and a capstan wheel driven by the turntable spindle. Then there was the Garrard cartridge, two full-sized reels of tape sealed inside a plastic shell. The British record changer manufacturer had visions of producing a tape transport which would hold and drop these cartridges in sequence for automatic play. Midway through the 1960s, Sony would offer a machine which changed conventional tape reels automatically. If you wanted one with your name engraved on a silver plaque on the chassis, it would have cost you $995.

Primitive as the Garrard cartridge was, it was the first step toward what would be the biggest revolution of all in audio recording, Philips's development of the Compact Cassette. But first came the RCA cartridge, an oversized precursor of the Philips cassette, which used standard quarter-inch-wide tape at a speed of 3¾ inches per second. Then came a short-lived square cartridge developed by Columbia Records and 3M (the takeup reel was inside the player). And then, in 1963 and 1964, the Compact Cassette. All were attempts to get over the public's fear of tape handling and threading, and were designed to make tape players more compact, efforts which would pay off some 15 years later in the video era.

Video, in fact, was only being hinted at in tape recording circles during the 1950s. In fact, Bing Crosby Enterprises had shown a black-and-white open-reel video recorder as early as 1951 and on January 4, 1927, a Russian immigrant to Britain, Boris Rtcheouloff, had applied for a patent based on Poulsen's invention, describing a system for recording pictures and sound on wire or "a travelling strip of magnetic material." The Rtcheouloff application even goes so far as to describe a rudimentary video camera.

It wasn't until 1956 that video recording became a reality at the professional level, with the Ampex VRX-1000. It was first used by CBS for "Douglas Edwards and the News," on November 30. Shortly thereafter, NBC became the first network to broadcast an entire program recorded on videotape, "The Jonathan Winters Show." And just after the New Year, 3M would begin marketing reels of videotape at $307 each.

The first hint of video recording for the home also came in 1956, when RCA announced that it was working on what it called a See-Hear video tape player, which would enable homeowners to play prerecorded programs through their television sets. Like other RCA pronouncements on video, this one wasn't followed up with actual products.

Beginning in 1954, with the first use of Bell Telephone Laboratories' transistor in a consumer product—this one a radio, manufactured in Texas for Regency—the move toward complete elimination of vacuum tubes in recording devices began. That same year, and not coincidentally, Sony produced the first Japanese-made transistors, under license from Bell. It would take 10 years before transistors completely displaced tubes in tape equipment, but the handwriting clearly was on the wall, and it said "Solid State."

By the early 1960s, the sale of open-reel tape recorders was booming, along with other audio products. Consumers, unaware of the changes taking place beneath the surface, were

Wollensak Model 1515s, c. 1974, left, and c. 1954.

Du Pont chemist Dr. Paul Arthur, Jr., inventor of chromium dioxide, six grams of which are needed for a C90 cassette.

paying as much as $399 for so-called professional models from Concertone or Bell Sound Systems. You could buy a do-it-yourself kit from Eico for $300 or pay $200 for the Wollensak 1515, a recorder which was fast becoming the Coke bottle of the tape field. Not a thing of beauty, the mono version went on sale in 1954. With very minor changes in its cosmetics and profile (but significant improvements in its interior, including a stereo version in 1963), it lasted until 1978, when the last ones were made for schools.

But the days of the home open-reel boom were numbered. In Holland, Philips had developed a compact dictating system which it would introduce in Europe during 1963 and in the United States the following year. Also in 1963, the first video recorder intended for the home made a fleeting appearance in the basement of the Cinerama Theatre on Broadway, in New York.

Of these, the one producing the greatest impact initially was Philips' Carry-Corder. Over the next 20 years, it would spell the end of open-reel tape recording in the home, open the whole new field of portability and spur the development of new types of tape better suited to the Compact Cassette's peculiar requirements.

For the first year or two, purchasers of Norelco Carry-Corders got a single blank tape (later three) with their machines. The manufacturer felt that since the units would be used for dictation, there really was no need to supply blank cassettes in quantity.

The Dutch hadn't counted on the cussedness of people. Nobody ever accused those first cassette recorders ($199 at first, later $149, and about the size—but not the weight—of two bricks side by side) of being high-fidelity instruments. But almost from the first, people began recording music on them to play back where their home stereo systems couldn't go. Accordingly, the need for blank tape arose almost at once—and the leading tape brands of the day (Ampex, Reeves Soundcraft, Audio Devices, BASF and Scotch) attempted to meet it. By 1966, Maxell was producing Japan's first cassette tapes.

The cassette might have remained in that rough-and-ready state if it weren't for a development taking place in, of all places, India. There, a young engineering graduate of Stanford University and Cambridge was finishing out a tour of duty with the Peace Corps. Ray Dolby (formerly a member of Ampex's first video-recorder design team) was a music lover who was intrigued by the problem of noise in tape recording, which seemed to be inherent. The nights in India were long, and Dolby had lots of time to think about the persistent hiss he heard in the quiet passages of his favorite tapes. Then he hit on the idea of breaking up the frequency spectrum during recording, with certain frequencies and certain volume levels of the signals recorded at higher levels than other combinations. Then, during playback, these signals would be suppressed by exactly the same amount, along with the accompanying background noise or hiss. Dolby's invention found immediate application in professional recording, where Decca Records in England bought every unit he could produce during his first five months in business.

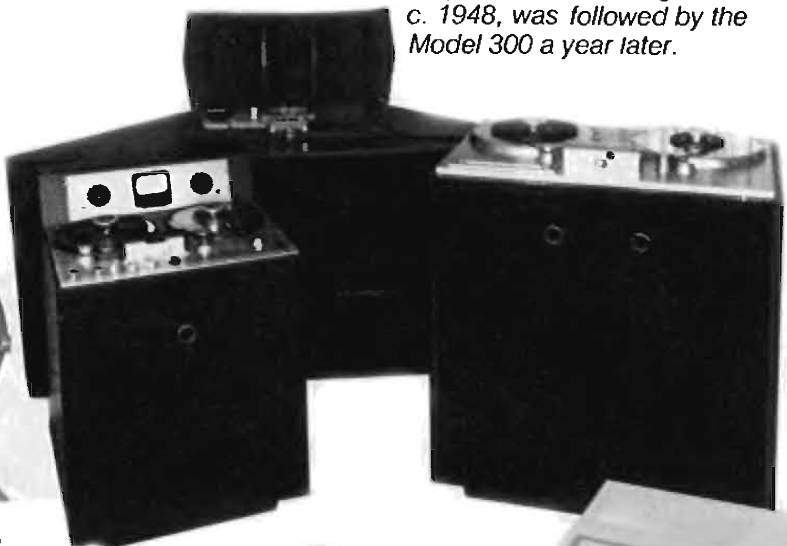
What turned the cassette into a high-fidelity recording medium, however, was the development of a low-cost version of the system Decca was using, one which could be built into home recording equipment. The idea came from Henry Kloss, then president of KLH. The recorder in which he planned to use it was an open-reel model, but when Kloss moved on to establish Advent, he took the idea of a low-cost Dolby noise-reduction circuit with him and applied it to that company's first cassette deck.

During this period, Dolby appeared before the Audio Engineering Society to describe his system, and mentioned casually that there was no reason why the circuit KLH was using couldn't be reduced to an integrated-circuit chip and incorporated in every cassette recorder. In 1973, Signetics produced such a chip. Harman-Kardon then beat Advent to the punch by using the new chip to upgrade a cassette deck it had just introduced.

The coming of cassette helped bring Scotch 111 and the other first-generation red-oxide tapes to the end of their

An original Philips "Carry-corder" (Photo; Courtesy C.G. Nijsen, Philips and J.A.E.S.)

Ray Dolby



Ampex Model 200, right, c. 1948, was followed by the Model 300 a year later.

long careers. At 3M and at Philips, work had begun on a new tape particle, pure metal, rather than the oxides which had been used for 30 years. The chemists and engineers were trying to do the same thing that Dolby did, improving the signal-to-noise ratio of tape recordings by making backgrounds quieter. "We all knew that it was the ferrite bits in the oxides which did all the work," 3M technical manager Del Eilers says now, "that the oxidized bits caused all the noise because no signal was recorded there. So we figured that if we could coat a tape with particles of pure iron, we could dramatically reduce residual noise levels, while doing something dramatic about frequency response and other characteristics." The catch was that pure iron filings or powder proved highly unstable. Exposed to oxygen in the air, they would immediately oxidize, sometimes with explosive results.

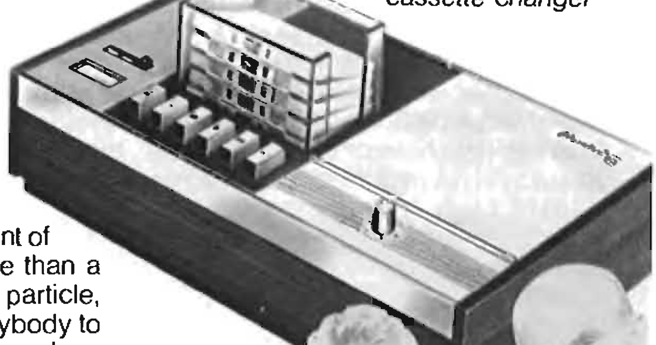
There is no evidence that anyone was ever killed or injured in these explosions, but both Philips and 3M experienced them, and there was apparently some damage to equipment. So the idea of metal tape was put on the back burner.

Meanwhile, E.I. du Pont de Nemours & Co. had patented a process for producing chromium dioxide, a product not found in nature. The trouble was that, at first, nobody seemed to know what to do with it. The scientists at du Pont thought that because of its high coercivity, it might have an application in magnetic recording. Accordingly, in 1969, the company sent a marketing/engineering team to New York to explain to the tape industry how chromium dioxide could be used to make superior audio recordings, particularly in cassette recorders designed specifically to accept it, as well as in video applications. Interest was high, and du Pont signed licensing agreements with BASF, making it the exclusive manufacturer in Europe, and with Sony, which had all rights to the product in Japan. That meant that any Japanese tape manufacturer wishing to produce chrome tape would have to talk to Sony first. If they produced the tape commercially, they would have to pay Sony royalties as well.

The predictable result was a race to develop a chrome substitute. The first step in that direction came in the early 1970s when Pfizer Inc. developed 2228, a high-quality gamma ferric-oxide particle which could be mass-produced. Previously, some tape makers had formulated their own particles from raw materials supplied by chemical companies like Pfizer, while in other cases, small firms, many engaged in producing paint pigments (as was Pfizer), purchased finished particles.

In any case, the development of 2228, which was nothing more than a very good quality ferric-oxide particle, made it possible for almost anybody to go into the tape business and produce a high-quality coating. It was compatible with the ferric-oxide tapes which had been produced during the 1950s and 1960s in terms of signal biasing and equalization, but it offered much better uniformity and particle size ratio. It could also be combined with cobalt particles to achieve even better frequency response and signal-to-noise ratio—but only when biased and equalized at a level somewhat higher than previously used and specified by Philips in its patent licensing agree-

8-track and cassette tapes

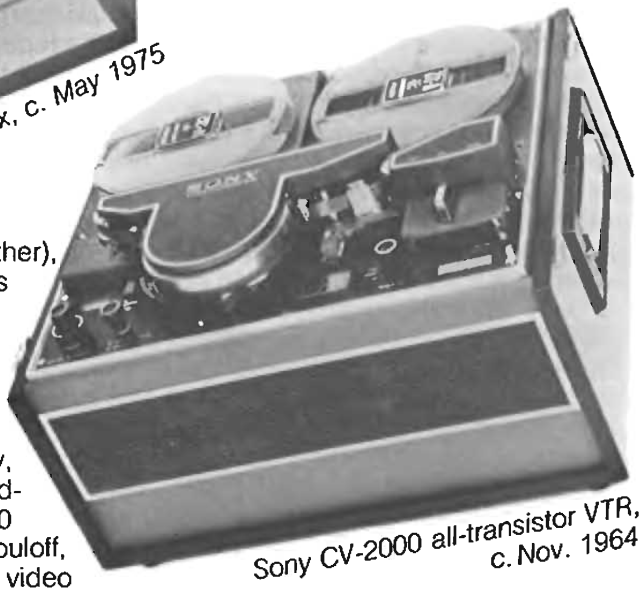


Norelco 2502 cassette changer

Lindsay and Poniatoff of Ampex.

Sony VP-1100 U-Matic, c. Oct. 1971

Sony SL-6300 Betamax, c. May 1975



Sony CV-2000 all-transistor VTR, c. Nov. 1964

ments. Since its introduction, virtually every improvement in ferric oxide tape has involved the use of 2228 or an equivalent as a basic building block.

Indeed, du Pont had run afoul of the Philips patents when it first tried to use chrome for cassette recording. So after protracted negotiations, the Dutch company amended its standards to permit biasing and equalization for chrome. As it happened, these specifications worked equally well for cobalt-enhanced 2228 particles, and soon every Japanese tape manufacturer other than Sony was offering its own version of chrome-biased ferric-oxide tape. By 1976, it had become necessary for companies like Maxell to clearly identify each tape's bias and equalization characteristics on the cassette or package.

By far the most bizarre event of that decade of change was the announcement, late in 1963, by a previously unheard-of British electronics firm, Telcan Ltd., that it had developed a home video recorder which could sell for the incrediby low price of $94. It used half-inch tape at a voracious rate—an 11½-inch reel provided only 11 minutes of playing time. The day of Telcan's demonstration was a cold and wintry one; the pictures reaching the black-and-white monitor in the basement of the Cinerama theatre from local TV stations were smeary, unclear and unacceptable. So were the recordings the Telcan unit made of them and played back to the assembled press corps. Sniffed the highly respected *Television Digest* in its report, "Telcan't."

Telcan didn't go away immediately, but neither did recorders appear in stores. Instead, the company reorganized and talked about a kit version for $164 (it never appeared, either), while other kitchen inventors offered working models of their own so-called brute-force recorders.

The whole idea might have died there. Instead, Ampex, Panasonic and Sony, using the helical scan recording idea advanced nearly 40 years earlier by Boris Rtcheouloff, introduced black-and-white video recorders designed for home use and priced between $1,000 and $1,200, or about three times the cost of the best audio cassette deck at the time. These recorders provided much better results than Telcan's, but they had their own problems. One was that tapes recorded on one machine might not play back on another unit, even if both were the same brand. Another was that the Ampex, at least, was a service technician's nightmare. Once something went wrong with it, it usually went into the closet and was forgotten.

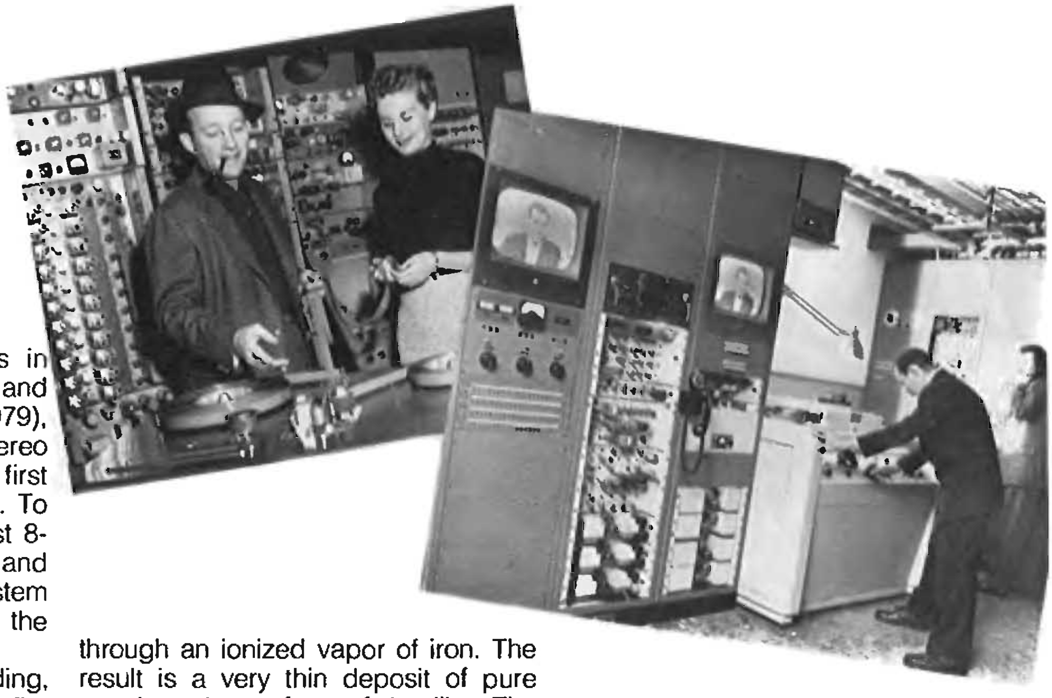
Akio Morita believed that it was Sony's task not only to develop and produce products which were revolutionary, but also to teach people how to use them. Accordingly, Sony bought television time in New York to explain to viewers who didn't know that they needed home VCRs what the recorders were for and how easy they were to use. The commercials zeroed in on the time-shifting feature, which Morita has always felt was the most important. They didn't sell many recorders—all three models disappeared within a year or two—but they did raise consciousness when it came to video recording and to Sony.

Sony's first real video recorder hit was the introduction of U-Matic recorders in 1970, in conjunction with a number of other Japanese manufacturers who had agreed in advance on a single format. This time, recordings were fully interchangeable from one machine to another and from one brand to another. The recordings were in color this time, and the tape was protected by a cassette-type shell. However, the manufacturers made no real effort to sell U-Matic as a home medium; instead, the recorders were sold to educational and industrial users, to cable television systems and even to local broadcasters.

Two years later, home viewers got their chance, with a console called Cartrivision. It contained a 25-inch color TV set in addition to the cartridge video recorder, at a cost of $2,400. Very few people had that kind of money, and many of those who did had no use for another 25-inch receiver. Besides, the sets didn't always work, and very few programs were available for the system.

Home video finally became a reality in 1975, when Sony introduced the Betamax. Panasonic and JVC followed up quickly with VHS machines early in

The Crosby Enterprises VTR used high tape speed past fixed heads, but was abandoned when Ampex showed its unit in 1956. First use of Ampex's VRX-1000 was for "Douglas Edwards and the News" in November 1956.



1976. Since then, improvements in these systems have come thick and fast—the first portable VCRs (1979), the first VCRs with high fidelity stereo sound (1982) and, in 1983, the first one-piece Beta camera-recorders. To them were added in 1984 the first 8-mm camera-recorders from Kodak and the first really compact VHS-system camera-recorders from JVC, using the ultra-small VHS-C videocassette.

Like the story of video recording, that of recording tape remains unfinished as these lines are written. As home video developed, recorder manufacturers found that the high-bias audio chrome and cobalt-absorbed ferric-oxide tapes, which began appearing around 1970, were ideal for recording video as well.

In 1979, 3M's technicians finally found a way of producing metal-particle tape without explosions, and without the tape oxidizing the first time the user took it out of the box. The secret: Allow the thin surface skin to oxidize just enough to protect the rest of the metal coating underneath and combine traces of other metals with iron to inhibit oxidization. 3M contacted a number of recorder manufacturers before going public with its tape. Of the lot, Tandberg was the first to develop a recorder designed to take full advantage of the ultra-high-bias tape.

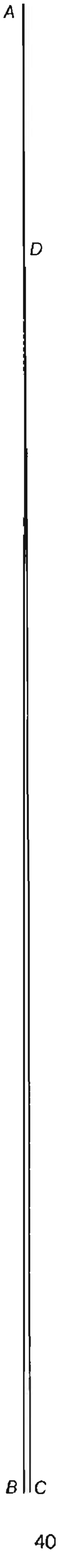
3M thought it had stolen a march on other tape manufacturers. It was aware, of course, of Philips research on metal particles, but it believed that the Dutch had not yet solved the stabilization problem. The company was astounded, therefore, when a Japanese maker, Kanto Denko, not previously involved in tape manufacture at all, announced that it had a metal particle, and proceeded to sell it to any Japanese manufacturer who cared to buy.

Metal particle tape has yet to find a niche in video recording, though the recent introduction of an 8-mm recorder-camera designed to use it suggests that it is about to do so. In the meantime, there is another form of metal tape which has been used for several years in the manufacture of microcassettes. "We originally called it metal-plated," a 3M technician told me, "when we began research on it back in 1963." He explained that the process involves passing a charged plastic film through an ionized vapor of iron. The result is a very thin deposit of pure metal on the surface of the film. The coating is much thinner than one of metal particles, but it's also more even, with virtually no gaps or dropouts. Panasonic patented a process which works in the early 1980s and labelled the result Angrom. Indications are that we'll be hearing more about both types in the years to come.

And what about the Compact Cassette, which for practical purposes buried the open-reel home recorder in the early 1970s? Well, you can mark 1984 down as the year in which it finally surpassed the long-playing record in terms of prerecorded sales, due in no small part to another idea of Sony Chairman Akio Morita. Back in 1980, Morita reasoned that what the world needed was a very compact, very lightweight high-fidelity cassette tape player that could be enjoyed through high-quality, comfortable headphones. Virtually everybody at Sony tried to talk him out of it. For one thing, they pointed out, nobody had ever heard of a portable player without a loudspeaker. And if you put in a speaker that sounded like anything, the unit would be heavy. No record function? Likewise unheard of. Morita stuck to his guns (after all, he was the chairman of the board). He even had a name for the new player: Walkman. Some of his associates cringed. It would never play in America, they said, and persuaded him to let the players be introduced under another name there. Within 18 months, everybody had forgotten the first name (Soundabout); the Sony Walkman was the hit of three continents and the Sony headquarters building in Shinagawa had a museum in the basement containing blatant copies by several dozen competitors.

The magnetic recording industry entered a new era—or at least the audio portion of it did—in 1978 when Sony introduced the first PCM digital recorder. Since then, PCM units have come down in price, size and complexity to the point where virtually any audiophile can afford one, alongside his hi-fi video recorder and professional-quality Compact Cassette deck. In the 1980s, it's virtually an embarrassment of riches for the tape-oriented audiophile.

The Ampex VRX-1000 included three items basic to all current VTRs: Spinning heads, FM recording of the video signal, and time-base error correction. The design team included Fred Pfost, Shelby Henderson, Ray Dolby, Alex Maxey, team leader Charles Ginsberg, and Charles Anderson.



An error of 20', or one third of a degree, (see text).

How Important Is Tape Azimuth?

HERMAN BURSTEIN

Not least among the factors that enable a cassette system to maintain essentially flat response out to 15 kHz or more is accurate azimuth alignment of its record and playback head or heads. A considerable number of queries received by my "Tape Guide" column grow out of the problem of achieving and maintaining such accuracy.

Azimuth refers to the orientation of the head gap with respect to the direction of tape motion. To simplify discussion, we'll temporarily assume that a tape head has but one gap (rather than the two needed for stereo). In the *absolute* sense, correct azimuth alignment denotes that the gap of a tape head is perfectly perpendicular to the direction in which the tape moves, as in Fig. 1. If not, we can say there is absolute azimuth error, depicted by the angle in Fig. 2; for clarity, the angle is greatly exaggerated.

Absolute azimuth error may be caused by the head being mounted with its gap slantwise to the tape path or by the tape skewing with respect to the gap when in motion, as in Fig. 3. Such error tends to produce increasing loss of signal as frequency rises (Fig. 4); we can call this azimuth loss. The loss occurs not on the tape but in playback.

Whether azimuth loss does occur depends on *relative* azimuth error—on the azimuth of the playback head relative to the azimuth of the record head. For compatibility among tape decks, it is desirable that both heads be correctly aligned in the absolute sense, as in Fig. 1; therefore, both will then have the same azimuth, so that relative azimuth error is nonexistent.

However, if absolute azimuth error does exist, but is the same in both the record and playback heads, the errors cancel; there will be no relative azimuth error and no azimuth loss.

Thus, it is advantageous to have a two-head deck, so that the same head is used for record and playback. Even though the head may not be in absolute azimuth alignment, the errors in record and playback will tend to cancel, and there tends to be little or no azimuth loss when playing a tape recorded with that head. (The possibility of some azimuth loss, usually slight, still exists because, depending on the quality of the cassette and of the deck, the tape might skew differently in recording than in playback, resulting in relative azimuth error.)

Significant azimuth loss can more easily happen with a three-head deck, one which employs separate heads for record and playback. Such loss occurs still more easily when different decks are used to record and playback, as with borrowed, traded, and commercially prerecorded cassettes and those tapes recorded on a formerly owned deck.

In sum, it is relative azimuth error which concerns us (although this does not diminish the importance of absolute azimuth alignment to insure compatibility among decks). Hereafter I will simply say azimuth error to denote relative error. Also for simplicity, I will assume that azimuth is absolutely correct in recording, with error occurring only in playback; after all, it isn't until playback that the consequence of azimuth error was manifest.

Azimuth Loss and Tape Format

We have already noted that azimuth loss increases as frequency rises. The loss also increases as tape speed is reduced. (Both causes of increased loss are really different sides of the

Illustration: Lou Brooks

same coin: The loss increases as the wavelength decreases—and wavelength, which is speed divided by frequency, grows shorter as frequency rises or as speed drops.) Because of this, the azimuth problem is more acute at the cassette speed of 1⅞ ips than at the speeds of 3¾ and 7½ ips commonly used in home open-reel decks.

On the other hand, azimuth loss decreases as track width is narrowed. So the cassette's narrower track (0.024 inch, versus 0.043 inch on four-track open-reel tape) partially offsets the effect of its lower speed as compared to open-reel tape (Table I). But the azimuth problem still remains greater for cassette systems.

Azimuth error angles of less than 1° can produce disastrous treble loss. Therefore, since the angles we deal with are very small, they are usually not stated in degrees but in minutes (60ths of a degree)—for example, 12′ instead of 0.2°.

From Table I, we see that an azimuth error of only 12′ (0.2°) produces a possibly acceptable loss of 3.07 dB at 10 kHz and a truly unacceptable loss of 7.77 dB at 15 kHz. Greater azimuth errors produce losses far out of keeping with the concept of high fidelity. (It should be recognized that the losses in Table I are additional to treble losses caused by other factors in a tape system, such as tape saturation, electrical characteristics of tape heads, insufficiently narrow gap in the playback head, improper equalization in recording or playback, tape properties, etc.)

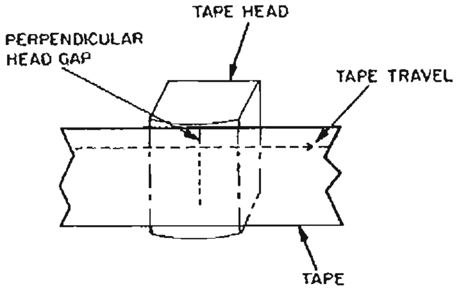


Fig. 1—Correct absolute azimuth.

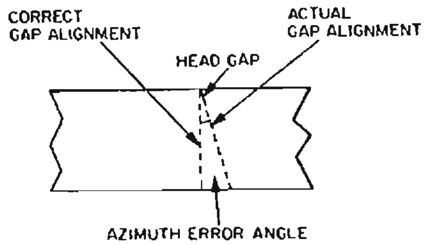


Fig. 2—Absolute azimuth error (exaggerated, for clarity) due to head misalignment.

For a cassette system to maintain fairly good response out to 15 kHz, Table I indicates that the azimuth error angle, α, must be kept under 9′; for good response out to 20 kHz, $\alpha = 6'$ is at the edge of acceptability. Preferably, α should be kept to no more than about 3′, which produces a loss of less than 1 dB at 20 kHz.

It may be of interest to visualize how small the values of α under discussion are, and thus appreciate the care that must be exercised by deck and cassette makers and by others in dealing with the azimuth problem. To do so, the following takes only a few moments. Assume a gross, totally unacceptable error of 20′ (or one-third degree). On an 8½ × 11-inch sheet of blank paper, draw a light, thin vertical line down the middle from top to bottom, with Points A and B respectively designating top and bottom of the line. One-sixteenth inch to the right of Point B, mark Point C. Connect Points A and C with a light, thin line. Lines AB and AC will now form an angle of 20′. Darken line AC for 1-13/16 inch, starting from the top, and designate the bottom of the darkened line as Point D. If AB corresponds to tape width (0.150 inch), AD corresponds to the upper gap of the head of a cassette deck (0.024-inch long). Although we *know* that AD deviates from vertical, our eye cannot *detect* the deviation. Clearly, azimuth alignment by eye is out of the question. (On the other hand, some people can do a fairly good job of aligning the playback head by *ear*, using a tape containing program material with substantial high frequency content or using an azimuth alignment tape containing a high frequency.)

It is clear from Table I that open-reel tape, with its higher speeds, can tolerate greater azimuth error than cassette, despite the cassette's narrower track. This is especially true of 7½-ips open-reel format. For example, Table I shows that an azimuth error angle of 6′ is at the edge of acceptability (down 3 dB at 20 kHz) for cassette, while with 7½-ips open-reel tape, a 12′ error is within the edge. Looking at Table I from another point of view, we see that, for a given azimuth error, 7½-ips open-reel tape suffers much less treble loss than the 1⅞-ips cassette.

At 3¾ ips, however, the azimuth problem is only slightly less for open-reel than for cassette. The advantage of faster speed is nearly cancelled by the disadvantage of greater track width. Hence 6′ azimuth error is approximately at the edge of acceptability for both formats; Table I shows that such error produces a loss of 3.07 dB at 20 kHz for cassette, and a loss of 2.42 dB at 20 kHz for 3.75-ips open-reel tape.

In the past, a few cassette decks have incorporated 3¾ ips as an extra

Table I—Playback loss (in dB) due to azimuth error in common home tape systems.

Format	Frequency (kHz)	Azimuth Error, α (Minutes of Arc) 3	6	9	12	15
Cassette	10	0.18	0.73	1.67	3.07	5.03
	15	0.40	1.67	3.97	7.77	14.64
	20	0.73	3.07	7.77	18.65	∞
Open-reel, 3¾ ips	10	0.14	0.58	1.33	2.42	3.93
	15	0.32	1.33	3.12	5.95	10.47
	20	0.58	2.42	5.95	12.05	∞
Open-reel, 7½ ips	10	0.04	0.14	0.32	0.58	0.91
	15	0.08	0.32	0.74	1.33	2.11
	20	0.14	0.58	1.33	2.42	3.93

speed for improved fidelity. One of the improvements is greatly reduced azimuth loss. For example, an azimuth error of 9′ produces only 1.67 dB of loss at 20 kHz with a tape speed of 3¾ ips, compared with 7.77 dB of loss at 1⅞ ips. For cassette, the edge of acceptability goes from about 6′ at 1⅞ ips to about 12′ at 3¾ ips.

The azimuth problem becomes all the more pronounced in two slow-speed tape systems, not yet discussed here, that have become part of the audio scene—1⅞-ips open-reel and 15/16-ips cassette. To illustrate, assume an azimuth error of 6′, which is at or within the edge of acceptability for other systems discussed thus far. This would produce losses of 2.42 and 5.95 dB respectively at 10 and 15 kHz in a 1⅞-ips open-reel system, and of 3.07 and 7.77 dB in a 15/16-ips cassette system. Such losses are unacceptable for high fidelity.

General Principles

If we stay within the area of "moderate" losses—say, not exceeding about 10 or 12 dB—we may infer from Table I and other data that square-law principles apply (in approximate fashion) to azimuth loss. Azimuth loss increases roughly as the square of any increase in angle of azimuth error, frequency or track width, and as the square of any *decrease* in tape speed. In other words, if you double the azimuth error angle, the recorded frequency or the track width, the loss in output will quadruple; if you double the tape speed, the azimuth loss will diminish to one-fourth of its original value.

In the area above "moderate" azimuth losses (say above 10 or 12 dB), the square-law principles give way to losses that move precipitously toward infinite as azimuth error increases, frequency rises, tape speed declines, and track width increases.

The table and other data also show that maximum acceptable azimuth error tends to decrease in proportion to any decrease in tape speed or increase in track width. To illustrate, maximum acceptable azimuth error tends to be halved if tape speed is halved or if track width is doubled.

The Colinearity Problem

For simplicity, Figs. 1, 2, and 3 show only one head gap. But cassette tape heads ordinarily have two, one for each stereo channel. Each gap is about 0.024 inch long, and the two are spaced about 0.011 inch apart. Ideally, they should be colinear, that is, both in the same straight line, so that if one is correctly aligned for azimuth the other is too.

However, depending on the quality of the head, the gaps may depart from colinearity, as shown in Fig. 5 (greatly exaggerated, there, for clarity). In such circumstances, the optimum alignment is that yielding equal azimuth loss in each channel.

Calculating Azimuth Loss

When azimuth error is stated (in an equipment review or elsewhere), it is given either as an angle (α) or as phase shift (P) at a specified frequency. If one has a pocket calculator with trigonometric and logarithmic functions, the following equations enable one to readily translate α into azimuth loss or P into α.

$$L = 20 \log \frac{\sin(180T)}{\pi T} \qquad (1)$$

where
- L = Relative level in dB,
- T = (Tan $\alpha \times F \times W$) ÷ S,
- α = Azimuth error angle, in degrees,
- F = Frequency in Hz,
- S = Tape speed in ips, and
- W = Track width in inches.

(Common track widths are 0.043 inch for quarter-track open-reel, 0.024 inch for stereo cassette. The 180 and π factors convert degrees into radians, the "20 log" term converts the answer to dB.)

To illustrate, assume that azimuth error α = 0.1° (or 6′), and we wish to know the azimuth loss at 15,000 Hz for cassette. In this case, F = 15,000, W = 0.024, tan α = 0.0017453, and S = 1.875. Therefore, T = (0.017453 × 15,000 × 0.024) ÷ 1.875 = 0.3351032, so that

$$L = 20 \log \frac{\sin(180 \times 0.3351032)}{3.1415927 \times 0.3351032}$$

$$= 20 \log \frac{0.8687921}{1.0527576}$$

$$= 20 \log 0.8252536$$

$$= -1.67 \text{ dB.}$$

(Note that the answer appears as a negative, signifying a signal loss.)

To convert phase error into azimuth error requires a different formula:

$$\alpha = \text{arc sin} \frac{P \times S}{360 \times D \times F} \qquad (2)$$

where
- α = Azimuth error angle in degrees,
- P = Phase error in degrees,
- S = Tape speed in inches per second,
- D = Distance between track centerlines, and
- F = Frequency in Hz.

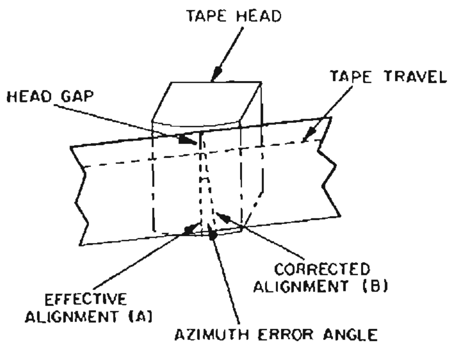


Fig. 3—Tape skew can effectively misalign a head positioned properly with relation to the tape deck. The effective alignment that results (A) is at an angle to the tape; repositioning the head produces corrected alignment (B).

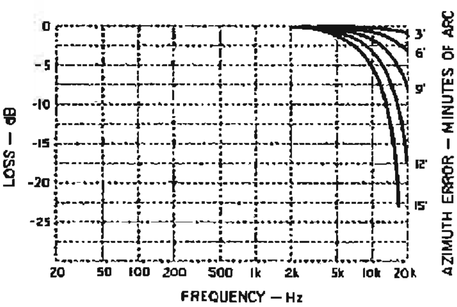


Fig. 4—Loss vs. frequency, as a function of azimuth error, for cassette tape.

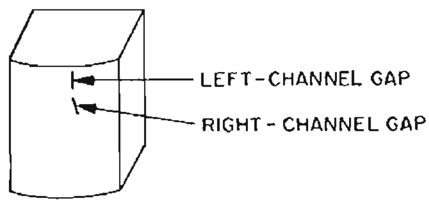


Fig. 5—Noncolinear head gaps (greatly exaggerated).

Compatibility in Tape Decks

HOWARD A. ROBERSON

It would seem that if the same alignment tape were used to adjust the playback heads on several three-head tape decks, and then each record head was adjusted for an exact match to its respective playback head, the recorded flux would have the same orientation to the tape on each deck. We might further conclude that a tape made with flat response on one deck could be replayed with flat response on a second deck.

Unfortunately, the situation is more complex than indicated. To get data on what really happens, I used a BASF Type II Alignment Tape to adjust the playback heads of three decks, all with separate record and playback head gaps. Two of these, the Nakamichi 582 and Tandberg TCD 3014, had separate record and playback head assemblies. The third, an Aiwa AD-M700, had a single head with separate record and playback gaps. In the Aiwa deck, therefore, the position of the record-head azimuth was automatically determined by the playback alignment with the test tape.

In the first tests, I made a recording of pink noise on the Tandberg TCD 3014 using Maxell XL II-S at 20 dB below Dolby level after the record head had been adjusted for the best response. The playback on the TCD 3014 was displayed and stored with a ⅓-octave real-time analyzer (Fig. 1, top). This same tape was then played back on the Nakamichi 582 (middle) and the Aiwa AD-M700 (bottom). All of the playback responses are quite flat over most of the range, but there are differences. The response on the Tandberg, which recorded the tape (top), is the flattest—as would be expected. The Nakamichi 582 playback (middle), however, displays a rising output above 2 kHz or so, which raises a question I'll answer a bit later. The response on the Aiwa AD-M700 deck would be considered disappointing, particularly in comparison with the other two decks. Immediately, it might be concluded that there was considerable alignment error, even though the AD-M700's head had been adjusted using the same test tape.

The second set of tests used the Nakamichi 582 as the recording deck, with a change to TDK SA-X tape, just to see if the tape skewing characteristics of a different cassette would affect the results. Figure 2 shows the very flat response at −20 dB with playback on the 582 deck (top). With playback on the Tandberg TCD 3014 (middle), however, the high-frequency rolloff is quite evident. The drooping response with the Aiwa deck is quite similar to that in the earlier figure, with a dB or so more rolloff.

Before drawing any conclusions, I ran a similar test set using the Aiwa AD-M700 as the recording deck, using XLII-S. The response in playback on the Aiwa deck is quite flat overall. The 8-dB drop at 20 kHz is a characteristic of this medium-performance deck. Playing this tape, the response on the Nakamichi 582 (middle) showed a rise above 1 kHz, while that on the Tandberg TCD 3014 (bottom) showed closer correspondence to the Aiwa deck results. Note that there is some vertical broadening of the 12.5-, 16- and 20-kHz filter-level indications for both the 582 and the TCD 3014—especially the 582. A separate check showed that the effect was caused by slight skewing of the tape on the Aiwa during recording. Because of the very short distance between the record and playback gaps of the Aiwa combination head, there were no skew-caused level variations in its playback.

An examination of the responses in

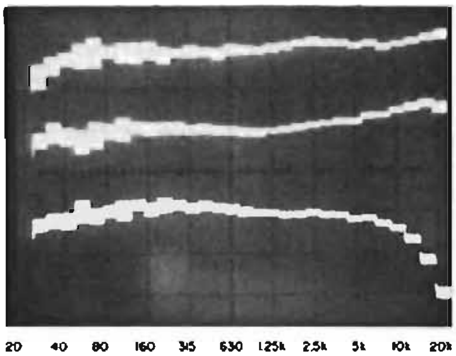


Fig. 1—Playback responses with recording made on Tandberg TCD 3014 cassette deck. Top, Tandberg; middle, Nakamichi 582; bottom, Aiwa AD-M700. (Vertical scale: 5 dB/div.)

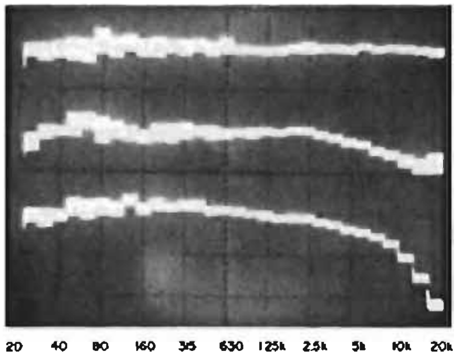


Fig. 2—Playback responses with recording made on Nakamichi 582. Top, Nakamichi; middle, Tandberg TCD 3014; bottom, Aiwa AD-M700. (Vertical scale: 5 dB/div.)

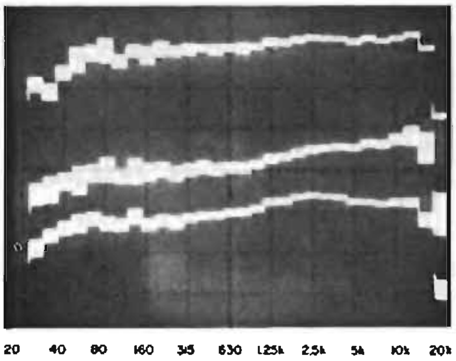


Fig. 3—Playback responses with recording made on Aiwa AD-M700. Top, Aiwa; middle, Nakamichi 582; bottom, Tandberg TCD 3014. (Vertical scale: 5 dB/div.)

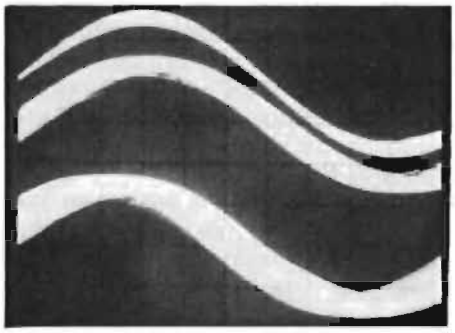


Fig. 4—Misalignment and jitter with 5-kHz test tone. Top, playback of reference track A on recording Nakamichi 582; middle, playback of trace B of 582, illustrating relative jitter; bottom, playback of same tape on Tandberg TCD 3014, showing alignment shift and jitter of track B relative to track A of TCD 3014. See text. (Horizontal scale: 30°/div.)

Figs. 1 to 3 along with some analysis, including reference to past experience, led me to these conclusions:

(1) The playback equalization of the Nakamichi 582 is relatively elevated at the higher frequencies compared to that of the Tandberg TCD 3014;

(2) The playback equalization and/or head response of the Aiwa AD-M700 rolls off compared to both of the other decks;

(3) The Aiwa deck uses "extra" record equalization to compensate for the roll-off in playback; and

(4) The alignment match between decks adjusted with the same alignment tape was fairly good, despite the equalization-caused response differences which might make things appear otherwise.

In confirmation of what some readers are already thinking, I must state that the playback equalization (70 µS for Type II), as given mathematically in international standards, is not followed exactly by the great majority of manufacturers. I won't take up space here to discuss the whole issue, but it can be noted that the Nakamichi playback is close to that for an ideal head, while the Tandberg's is closer to that for the IEC reference playback head.

Figures 1 and 2 reflect this relationship, with the Nakamichi recording rolled off on the Tandberg, and the Tandberg recording boosted on the Nakamichi. These response differences relate to equalization and *not* to azimuth error. Figure 4 illustrates what azimuth error might be found between these two decks. I recorded a 5-kHz tone on the Nakamichi 582, both tracks. In the playback, track A (top) was the 'scope sweep reference, and the broadening of the track B trace (middle) shows the jitter of B versus A. There was some amplitude variation on both tracks. The average position of its trace shows that B lags A by about 6° at 5 kHz, with 30° of total jitter. The phase error of 6° is equivalent to an alignment error of less than 1 minute of arc. The bottom trace is that of track-B playback of the same tape on the TCD 3014, with the 'scope locked to its track A. Here the jitter is higher, and B leads A by 20°, still just 2 minutes of alignment difference. So, a final word: Align heads carefully, but be aware of possible equalization discrepancies between decks.

(The distance between centerlines of the stereo tracks is 0.035 inch for cassette and 0.125 for quarter-track open-reel.)

To illustrate, an *Audio* review of a top-notch cassette deck stated that phase error measured 15° at a test frequency of 12,500 Hz. Thus P = 15, S = 1.875, D = 0.035, F = 12,500, so that

$$\alpha = \arc\sin \frac{15 \times 1.875}{360 \times 0.035 \times 12{,}500}$$

$$= \arc\sin 0.0001786$$

$$= 0.0102 \text{ degrees (or 0.61 minutes)}$$

So small an azimuth error would cause very minute azimuth loss, for example merely 0.03 dB loss at 20,000 Hz.

How Azimuth Loss Occurs

Assume that a sine-wave signal is recorded at 1⅞ ips on cassette tape. The recorded signal is equivalent to a series of bar magnets end to end, as in Fig. 6—north poles adjacent to north poles, and south poles to south poles. Each bar represents a half wavelength, and its poles correspond to the positive and negative peaks of the half wavelength. The higher the frequency, the shorter are the bar magnets because more of them must fit into each inch of tape. At the upper end of the treble range, say above 10,000 Hz, they become extremely short, on the order of less than 0.0001 inch. (At 10,000 Hz, a half-wavelength = 1.875/20,000 = 0.0000938 inch.)

At any given instant of playback, each edge of the head's gap contacts a given intensity and polarity of magnetic field produced by the bar magnets. Because the two gap edges contact different parts of a bar or of adjacent bars, most of the time each edge is at a different field intensity. Therefore most of the time a magnetic potential exists between the two edges. The potential constantly changes in intensity and polarity as the gap traverses the bar magnets, and the *changing* potential induces a voltage in the coil of the playback head.

As frequency rises and the bars become shorter, the difference in field intensity and polarity at the two edges of the gap increases, thereby increasing head output. Maximum output occurs when the distance between gap edges equals a half wavelength (one bar), with one gap edge contacting a north pole while the other contacts a south pole, resulting in maximum potential across the gap. Output falls rapidly as frequency increases further, reaching zero when an entire wavelength equals the distance across the head gap, since the potential will then always be equal at each gap edge. Hence, the importance of a very narrow playback gap for extended high-frequency response.

The greater the field intensity seen by each gap edge, the greater can be the magnetic potential between edges at various instants, and the greater can be the changes in potential, thus increasing head output. Contrariwise, if anything reduces the field intensity seen by the gap edges, this reduces head output.

Azimuth error performs such a reduction. When there is no azimuth error, in playback all parts of the left edge are in contact with the same polarity and intensity of magnetic field. That is, the upper and lower sections of the edge see the same field intensity as the center of the edge. But azimuth error tilts the edge with respect to the magnetic bar, so that the central, upper, and lower sections contact different magnetic intensities, which partially cancel each other. The same is true, of course, for the right edge.

To help visualize this, assume that the center of a gap edge is at the north pole of a bar magnet and thus sees maximum field intensity. But if the edge tilts, its top and bottom sections are no longer at the north pole and therefore are at points of reduced intensity. Accordingly the intensity seen by the edge as a whole is reduced.

This process doesn't work in reverse; that is, tilting the gap edge doesn't increase the field intensity seen by it. To illustrate, assume that the center of the edge is at the middle of a bar, where intensity is minimum. Now the top and bottom of the edge do see higher intensities than does the center of the edge. But the top and bottom incline in opposite directions, toward opposite polarities, so that these higher intensities cancel each other and leave the edge as a whole at minimum intensity.

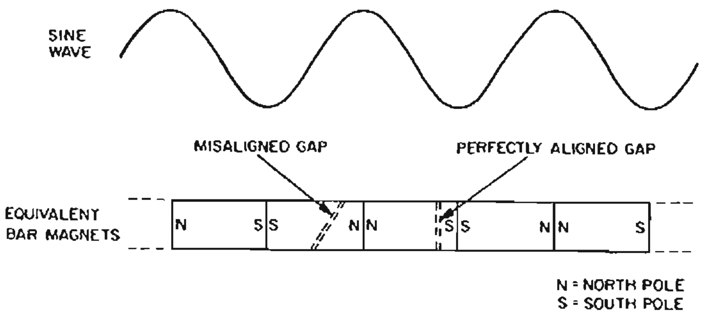


Fig. 6—How azimuth loss occurs: Recorded signals are essentially bar magnets of varying length. Head output depends on the difference in magnetic field bridged by the head gap—a difference greatly reduced when the gap is tilted to bridge areas of different flux.

Considering how short are the bar magnets at high frequencies, very slight gap tilt (azimuth error) will cause the top and bottom of a gap edge to ride very different field intensities, with substantial cancelling effect. To illustrate, assume a cassette player with an azimuth error of 9′, tilting the top of an edge to the left and the bottom to the right. Therefore the bottom of the edge is displaced 0.000063 inch to the right with respect to the top of the edge. (Displacement = sin α × gap length = 0.002618 × 0.024 inch = 0.000063 inch.) Next assume a frequency of 15,000 Hz, so that a half-wavelength, or one magnetic bar, = 1.875/30,000 = 0.000063 inch. Now assume that the center of the gap edge contacts a bar's north pole, where field intensity is maximum. Because the tilted edge in our example spans an entire magnetic bar, the top and bottom of the edge must be at midpoints of magnetic bars, where field intensity is minimal. Thus, the field intensity seen by the gap edge as a whole is quite substantially reduced.

At lower frequencies, where the recorded magnetic bars are longer, a given gap tilt causes the top and bottom of a gap edge to be displaced only a fraction of a bar relative to the center of the edge. Therefore the cancelling effect on field intensity seen by the edge is reduced, and azimuth loss is less.

The above discussion paves the way to our understanding why, for a given azimuth error and tape speed, the azimuth loss increases with track width. As the track width, and therefore gap height, is increased, the top and bottom of the tilted gap edge are displaced a greater distance along the tape path from the center of the gap. To illustrate: We noted earlier that at 1.875 ips an azimuth error of 9′ results in a displacement of the edge bottom relative to the edge top of 0.000063 inch when track width is 0.024 inch (cassette). When track width is 0.043 inch (open-reel), the displacement increases to 0.00011 inch. As already noted, the greater the displacement, the greater is the reduction in field intensity seen by the gap edge as a whole, increasing azimuth loss.

Minimizing Azimuth Error

From the foregoing it is obvious that great care must be exercised by all parties to the tape recording process in order to minimize azimuth error.

The cassette deck manufacturer must pay close attention to proper azimuth alignment of the head or heads used for recording and playback. This requires an accurate test tape on which a high frequency, such as 10,000 Hz, has been recorded with the gap of the record head (in the deck that produces the tape) exactly at a 90° angle relative to the direction of tape travel. The playback or record-playback head of the deck being aligned is then oriented for maximum output from the test tape. If the deck being aligned has a separate record head, it is adjusted, while recording and monitoring a high-frequency signal, for maximum output from the previously aligned playback head.

If head gaps are not colinear, so that correct alignment of one gap necessitates azimuth error of the other, an optimum position has to be found that achieves equal azimuth loss in both stereo channels. If there are separate heads for record and playback, optimization becomes more complex. And it is even more complex in the case of reversible decks which use heads with four gaps.

Production of an accurate azimuth alignment tape is not an easy matter and requires precise laboratory procedures. Even for test tapes made by companies of high reputation, it has been noted that somewhat different results may be obtained from tapes of different companies. However, these differences tend to be slight and are becoming slighter as new azimuth alignment tapes appear.

Care must be exercised by the deck manufacturer to properly adjust tape tension and thereby minimize tape slewing (a change in the angle of the tape with respect to the head). Toward the same end, the cassette deck must be designed so that the cassette is uniformly and securely locked in place.

Equal care applies to the cassette itself. It may appear to be a disarmingly simply affair but it is really a very sophisticated device that must be built with a high degree of precision in order to operate properly. Guides must be accurate and true, and the tape must be slit very accurately, in order to minimize slewing.

Reverse cassette operation presents an extra azimuth problem because the tape tends to slew differently when running from left to right than from right to left. One solution is to use a head with two gaps instead of four and rotate the heads 180°, with a separate azimuth adjustment (a stop screw) for each direction. Another solution is to turn the cassette over, as one turns the page of this magazine, so that the tape always runs in the same direction with respect to the heads. A third solution, used in the Nakamichi Dragon, is to continuously adjust the playback head azimuth *during operation.* This is achieved by dividing the gap for one of the playback tracks into two sections; as a tape is played, the phase difference between gap sections is constantly monitored, and azimuth is adjusted by a motor to minimize the phase difference and thereby minimize azimuth loss.

Build A High-Performance Noise Reducer

JOHN H. ROBERTS

Taping from high-quality disc pressings routinely tests the dynamic range of even the highest-tech cassette decks, and CDs are even more of a challenge to tape adequately. All the new, top-line decks come with such powerful noise-reduction (NR) systems as dbx or Dolby C, which help recordists handle today's very dynamic program material. But for those whose decks have no noise reduction or only Dolby B NR, I've devised a low-cost, add-on companding noise-reduction system with which their decks can handle the hottest signals, just like the best new equipment.

The theory behind companding NR stems from the observation that tape hiss is effectively masked by loud music, and is only audible during quiet passages. Companding NR (like dbx and the two Dolby systems, which work on similar principles) is a two-part, encode/decode process. A compressor circuit in the record encoder reduces gain for signals above a certain level (the circuit's unity-gain threshold, or 0-dB level), and increases it for signals below that level. This brings loud passages down below the limits of the tape deck's headroom and lifts quiet passages above the deck's noise floor. A complementary circuit in the playback decoder expands the signal back to normal, boosting the loud passages and cutting back on the soft ones. In the process, it also cuts back noise (Fig. 1). Thus, companding not only reduces tape noise but increases headroom, for cleaner and louder musical peaks.

John H. Roberts is President of Phoenix Systems, Manchester, Conn.

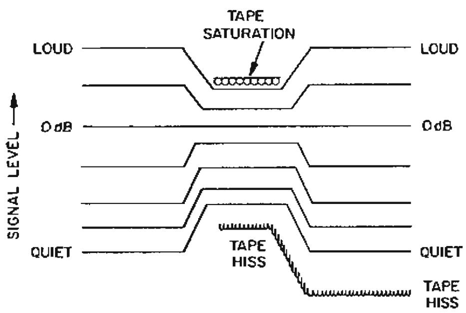
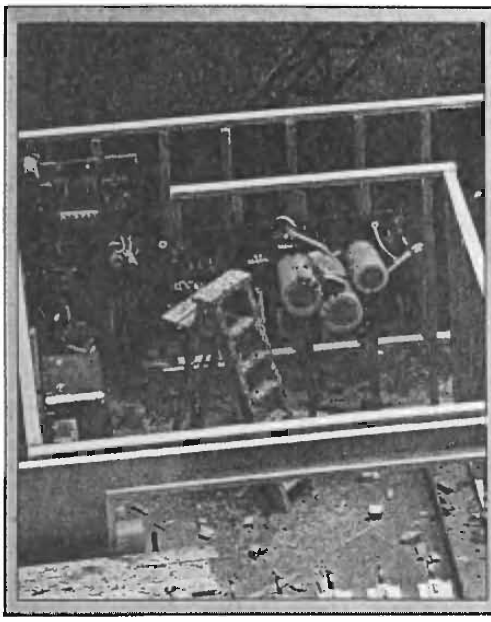


Fig. 1—How companding works.

The P-522 Noise Reducer

The Phoenix Model P-522 noise reducer, for which plans are given here, contains two simultaneous encode/decode channels (Figs. 2 and 3), allowing off-the-tape monitoring with natural dynamics when recording with three-head decks. Its operation is based on wide-band, 2:1 compression/expansion, and will deliver the same order of S/N improvement as the dbx systems (although no attempt has been made to make the two compatible).

Most companding NR systems rely on identical but inverse distortion in the playback expander to cancel out the third-harmonic distortion caused in the compressor by the audio signal modulating the gain-control voltage. The P-522 electronically cancels out this modulation at the compressor. Reducing this distortion at the source is advantageous, since imperfections in the tape record/playback process can affect the ability of conventional systems



By building this low-cost, companding NR system, you'll get more dynamic range in your recordings.

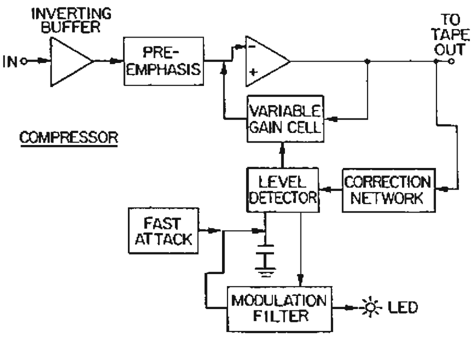


Fig. 2—Block diagram of the compressor section.

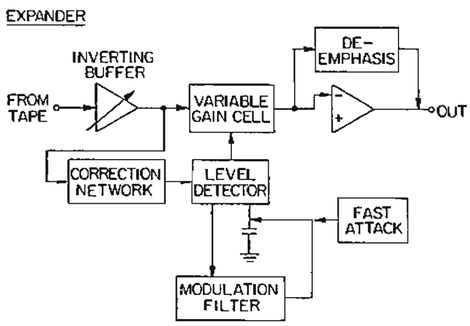


Fig. 3—Block diagram of the expander section.

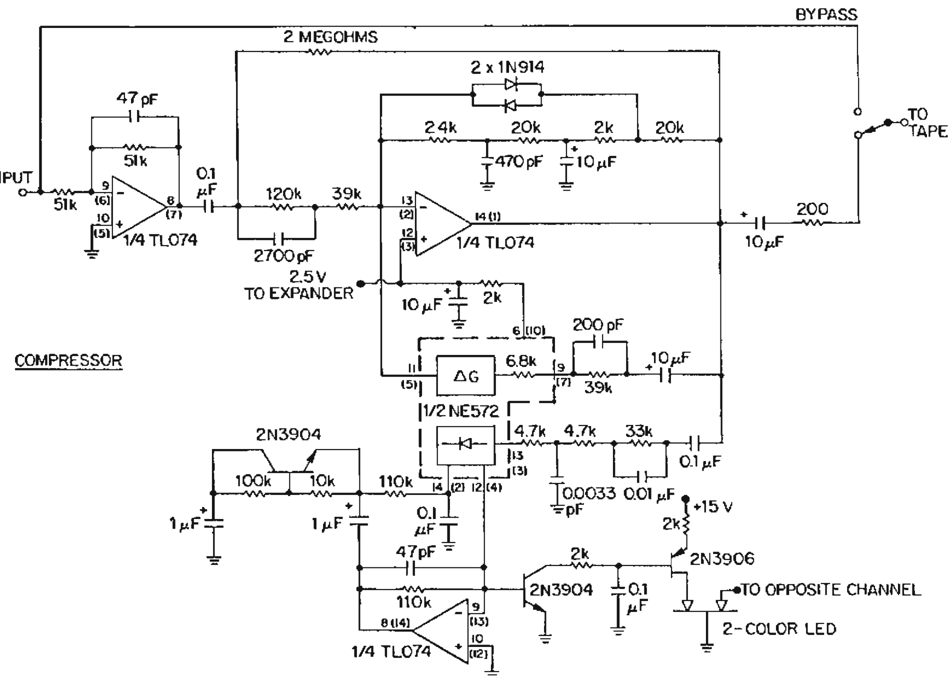


Fig. 4—Schematic diagram of the compressor section. Pin numbers shown on ICs here and in Figs. 5 and 6 are for left channel; numbers in parentheses are for right channel.

Power connections (not shown) are: +15 V to each NE572 pin 16 and TL074 pin 4, −15 V to each TL074 pin 11, and ground to each NE572 pin 8.

to cancel out this distortion. However, crossing the P-522 with a conventional compander, by using one system to decode tapes made with the other, will yield higher distortion than will either system decoding its own tapes. Also, this new approach to modulation-distortion control uses different optimum attack/release time constants, so some dynamic mistracking could also occur.

For anything but noncritical applications, I do not suggest mixing different types of noise reduction. If you have an older-type NR (such as Dolby B) in your deck, just switch it out when using the new compander. Do not use them both at once, as you will get no further improvement in S/N, but you may overcompress the highs and produce a loss of frequency response.

I have, however, found that playback of P-522 tapes in the car with no decoding at all can be quite acceptable, becoming less acceptable as the car system's frequency response and dynamic range begin to approach those of a good home system.

The P-522 uses high-frequency pre- and de-emphasis to reduce the audibility of tape hiss when only low-frequency signals are present. A correction network at the input to the level sensing port prevents solo high-frequency signals from being overrecorded. In fact, the network intentionally overcorrects, so high-frequency signals *when present alone* will be recorded slightly cooler than low-frequency signals. This reduces tape saturation and frequency response problems commonly experienced with unprocessed cassette recordings. Dynamic tracking errors can occur if the frequency content at the input to the playback expander is much different than the output of the record compressor.

An adaptive high-pass filter in the compressor attenuates low-level, low-frequency signals caused by warped record and rumble, before they can get encoded. This is important because tape recorders generally have limited frequency response in the low bass and would not reproduce the warp signal, which has peak energy in the 5-Hz region. To further reduce this sensitivity to signals that the tape recorder can't reproduce, the input to the gain-control detector is rolled off below 38 Hz and above 20 kHz. (Note: As

Dolby NR does not reduce noise at low frequencies, it will not be tricked by low-frequency response errors; however, it shows the same sensitivity to high-frequency errors as do all companding systems.)

Using the P-522

The P-522 is designed to hook up between a control preamp and a tape deck. Its only front-panel controls are on/off and in/out (bypass) switches.

Without noise reduction, setting record level is a juggling act between distortion, high-frequency saturation, and tape hiss. As the noise reduction gives you more than 20 dB of hiss reduction, the optimum 0-dB record point moves down quite a bit below meter 0 dB. I have found −6 dB to be a good setting on a few two-head machines I've worked with, but it's better (and fairly easy) to find the best recording level settings for your tape and machine by ear.

To do this, use a familiar disc with a good, clean top end (cymbals, snare drum, etc.) and record at progressively higher levels until you notice a loss of sheen or edge. Back off your recording level until full sound quality returns. You then have your optimum 0-dB level. Interstation FM hiss can also be used as a test signal for these comparisons, but it may have a bit more high-frequency content than typical discs.

When making these listening tests, be sure you are playing back at similar loudness levels. Your ear is more sensitive to high frequencies at louder levels, so don't be tricked by simple volume mismatches. Also, with two-head decks be sure to listen to the playback from the tape because the deck's output during recording will not reveal tape saturation losses.

Once you've optimized your record level, you can calibrate the P-522's internal playback gain trim. With a three-head machine, which allows monitoring off the tape, you can make this adjustment while recording. With a two-head deck, you'll have to first record a track, then make the gain-trim adjustment while listening to playback. Adjust the gain trim so there is no loudness change between the original source material and the tape playback when you toggle the tape-monitor switch on your control preamp. As a

PARTS LIST

Resistors, 1/4-Watt, 5%

Value, Ohms	Quantity
10	8
200	5
2k	6
2.4k	2
4.7k	8
10k	4
20k	6
33k	4
39k	10
51k	4
100k	4
110k	8
120k	4
2M	2

Miscellaneous

DPDT push-push switch	1
4PDT push-push switch	1
Phono jacks	8
50-kilohm trim pot, 3/8-inch, square, single-turn	1
28-V center-tapped transformer	1
Line cord and miscellaneous hardware	

Semiconductors

1N914 diode	6
1N4002 diode	4
2N3904 NPN transistor	6
2N3906 PNP transistor	2
Two-color LED	1
78L15, +15 V regulator	1
79L15, −15 V regulator	1
TL074CN op-amp	3
NE572 compander IC	2

Capacitors

Value	Quantity
47 pF, 10%, ceramic disc	8
470 pF, 10%, ceramic disc	2
0.1 µF, 20%, ceramic disc	9
0.02 µF, 500 V, ceramic disc	1
200 pF, 5%, polystyrene	4
2700 pF, 5%, polystyrene	4
0.0033 µF, 5%, polyester	4
0.01 µF, 5%, polyester	4
0.1 µF, 5%, Mylar	12
1 µF, 25 V, aluminum electrolytic	8
10 µF, 35 V, aluminum electrolytic	18
1000 µF, 35 V, aluminum electrolytic	2

The following are available from Phoenix Systems, P.O. Box 628, Manchester, Conn. 06040. Prices in effect through April 15, 1985. (Connecticut residents must add 7.5% sales tax.)

Complete kit of parts with instructions, P-522-NR	$79.00
Etched and drilled p.c. board, P-522-B	$9.00
TL074 quad low-noise, high-speed op-amp (three required), P-TL074	@ $2.50
NE572 dual compander IC (two required), P-NE572	@ $4.50
28-V, c.t. transformer, P-10-T	$6.50
Instruction set (including stat of p.c. board), P-522-INST	$2.50

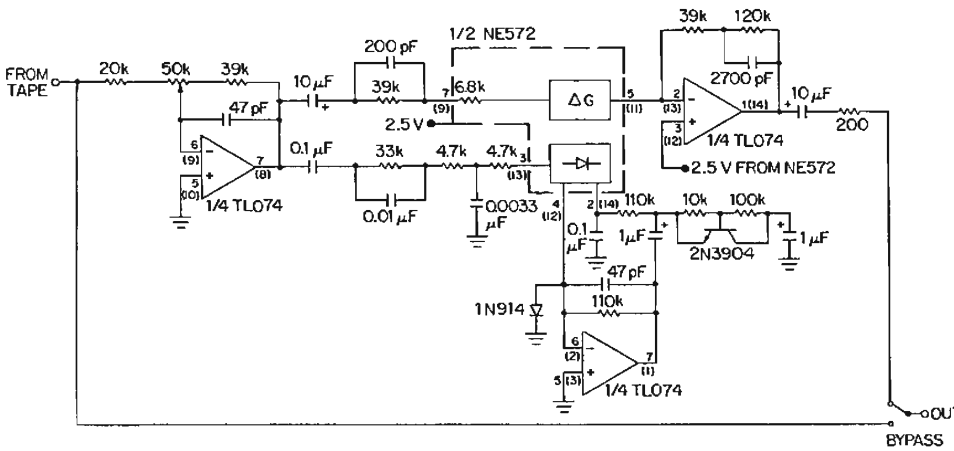


Fig. 5—Schematic diagram of the expander section.

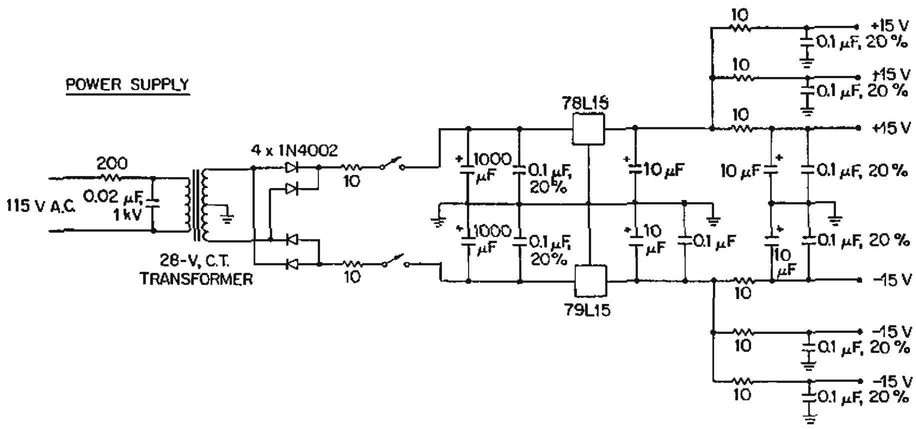
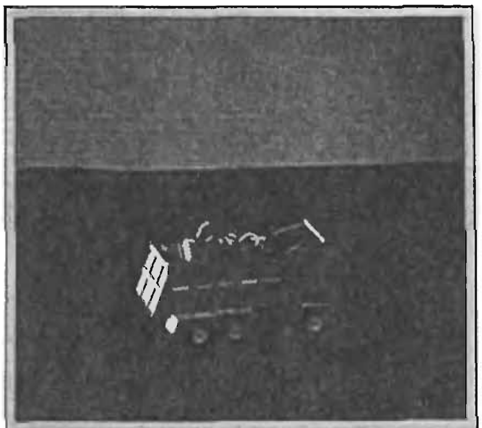


Fig. 6—Schematic diagram of the power supply. Note separate power feeds for each channel, to reduce crosstalk.



This noise-reduction system takes away more than 20 dB of tape hiss and noise.

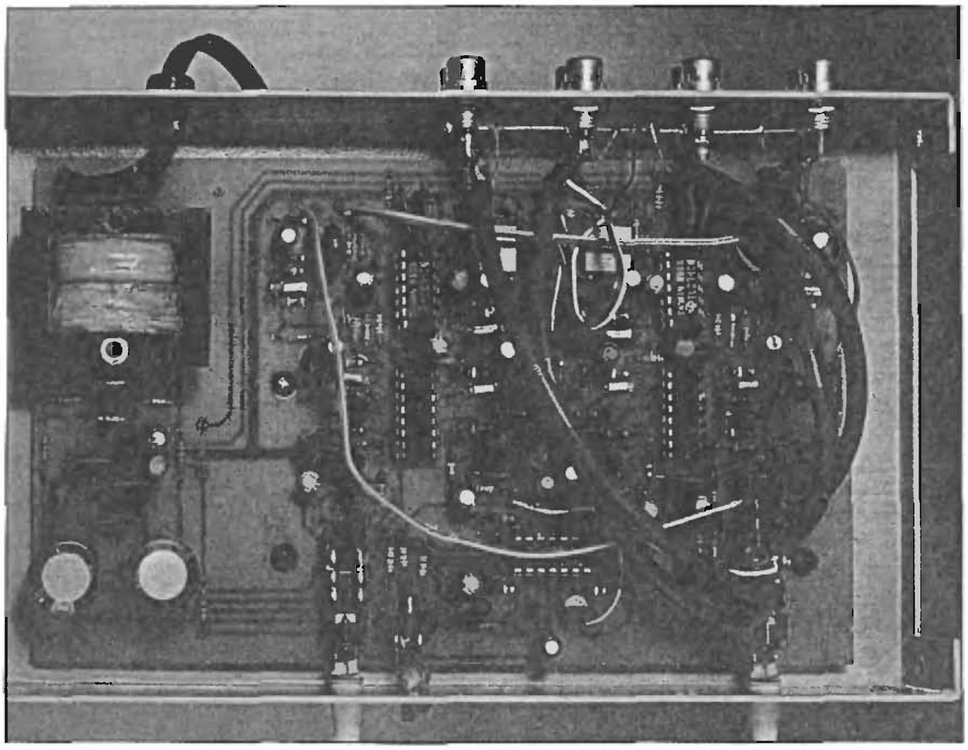


Fig. 7—Interior of P-522 noise-reduction unit, showing parts layout

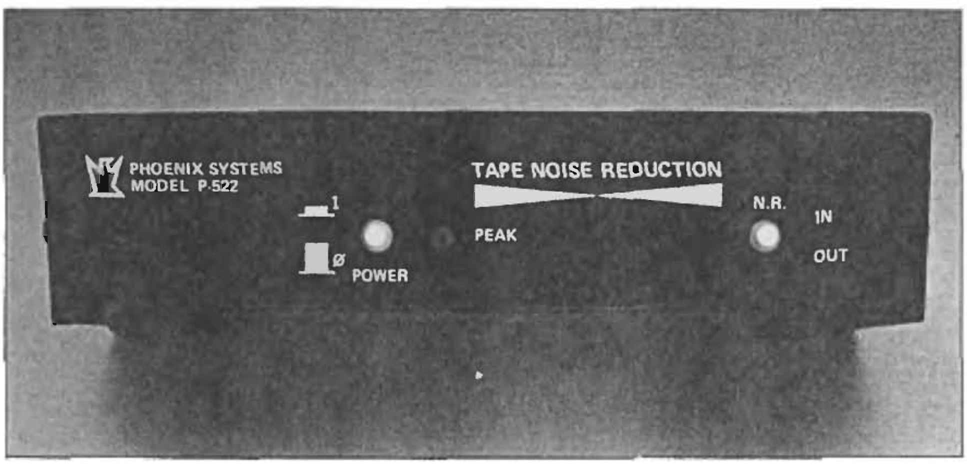


Fig. 8—Exterior of the completed unit.

result of the new, reduced record level, some two-head decks may return a louder signal when monitoring record than during actual playback.

The P-522 has a transient-overload indicator, a two-color LED that flashes red when the right record-channel overloads and green when the left one does. It is normal for this LED to flash on momentary overloads, but it should not stay lit or flash more than a few times per second.

To record or play back tapes without companding, just switch the P-522 into bypass, and the tape deck will be connected to your control preamp. Although the P-522 uses a hard-wire bypass, the inputs to the compressor are always connected—so it may be a good idea to leave the unit powered up whenever signal is flowing in the tape-monitor circuit. Otherwise, if your preamp's tape outputs aren't buffered, the slight resulting change in impedance might introduce distortion in your main signal path. (The same is true of tape recorders, which should also be turned on whenever connected to unbuffered tape outs.)

Building the P-522

Schematic diagrams for the P-522's compressor, expander, and power-supply sections are shown in Figs. 4, 5 and 6, respectively. While I recommend use of the p.c. board I've made available (see Parts List), you should be able to realize acceptable performance with other construction techniques. Do try to follow the layout in the photo (Fig. 7) as closely as you can, especially the grounding scheme. Use care when substituting parts; wider-tolerance parts can affect frequency response and dynamic tracking errors. Substituting tighter-tolerance parts won't hurt anything but your pocketbook.

Leonard Feldman and R. Aryana

THE DNR How it works

Companding noise-reduction systems, such as Dolby and dbx, may be the best-known noise reducers, but they're not the only ones. There is another type, referred to as noncomplementary or single-ended, with unique advantages of its own.

In companding NR systems, the signal must be encoded before transmission or recording, and decoded in reception or playback. These systems can reduce the amount of noise a signal picks up within the encode/decode loop, but they can't reduce noise already present in the signal.

Leonard Feldman is a Senior Editor of this magazine. R. Aryana is Chief Engineer of Advanced Audio Systems International in San Jose, California. Portions of this article have been adapted from "A Dynamic Noise Reduction System," an article by Mr. Feldman which originally appeared in the October 1981 issue of Radio-Electronics.

Companding NR systems (including Dolby B and C, dbx, and the one described by John Roberts elsewhere in this issue) have the most powerful noise-reduction ability, but they must trade power for compatibility. For instance, Dolby B-encoded tapes are compatible with systems lacking Dolby decoders—you need only turn the treble down to listen pleasurably—but this is the least powerful of the popular companding systems, with only 10 dB of noise reduction. Dolby C NR, with 20 dB of noise reduction, produces tapes which can be listened to reasonably well on systems equipped with Dolby B decoders, but it is barely listenable on systems without Dolby B (again, with the treble turned down in each case). Tapes made with dbx encoding are the least compatible—they're unlistenable on systems without dbx decoders—but provide the most noise reduction (35 dB or more).

Single-ended NR systems work in playback only, requiring no special encoding of the signal. While they cannot prevent noise pickup like the companding types, they can reduce noise already present in the signal. And they can be used with any program source: AM, FM, tape, disc, VCR audio tracks, telephone, or even such nonaudio applications as medical electronics and data transmission.

Perhaps the best-known of these systems is the Dynamic Noise Filter,

NOISE REDUCER:

and How to build it

developed by Richard Burwen. A single-chip noise-reduction system based on Burwen's, called DNR, is manufactured by National Semiconductor. (See "National's New Noise-Reduction Chip" by Ralph Hodges, *Audio*, November 1981.)

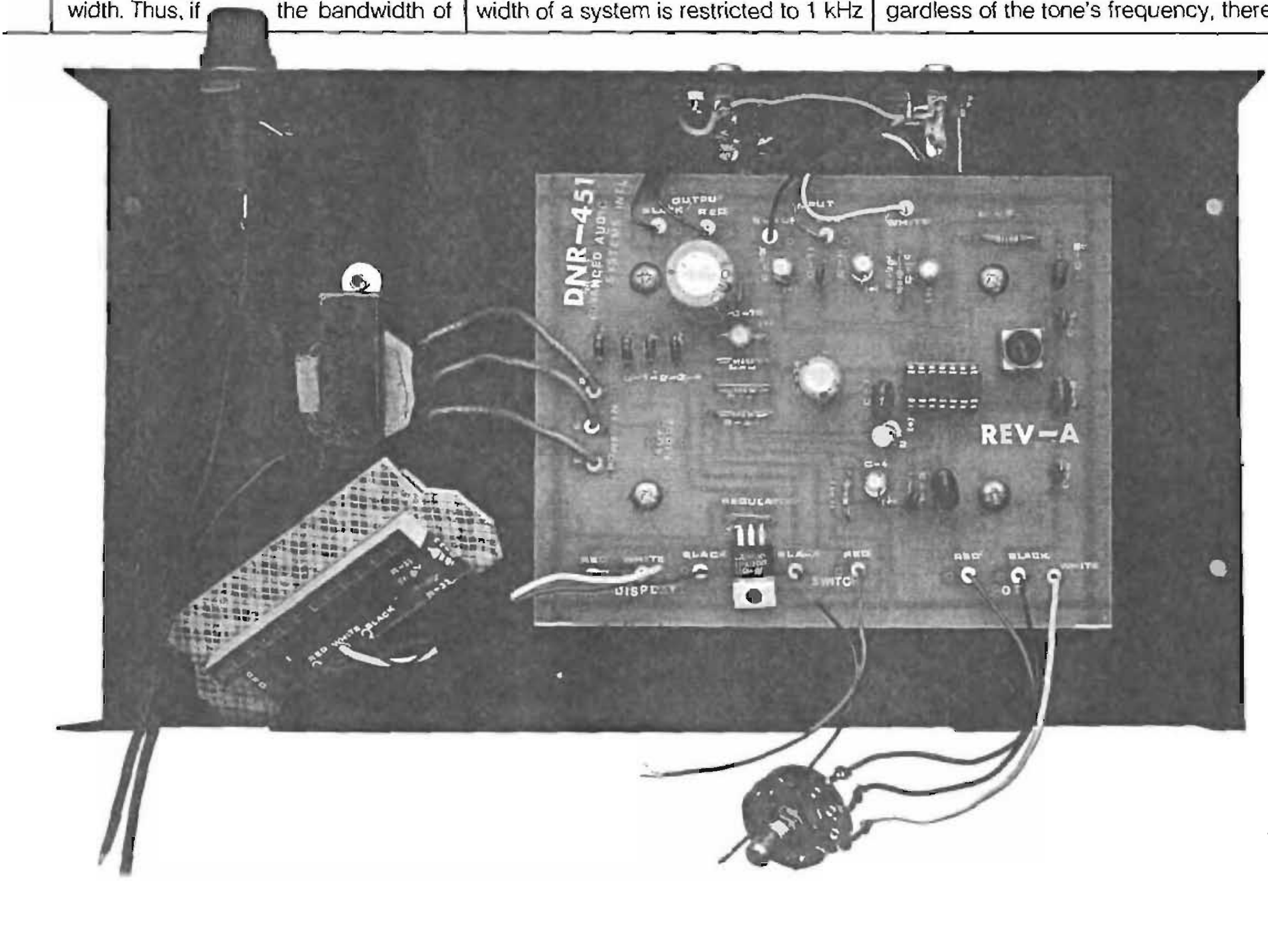
How DNR Works

The National Semiconductor noise-reduction system can provide up to 14 dB of noise reduction in stereo program material and is based upon two principles. The first of those states that noise output is proportional to system bandwidth. Suppose system noise is caused solely by resistive noise (noise added by the circuit resistors). In such a system, noise amplitude is uniform over the system's frequency bandwidth. Thus, if the bandwidth of the system is reduced, the noise content is also reduced.

Unfortunately, there isn't a simple correlation between the amplitude of the noise signal and the amplitude of the noise perceived by the listener. As shown in Fig. 1, the ear is most sensitive to noise in the 600 Hz to 6 kHz frequency range. For this reason, when measuring noise content in a system, a weighting filter is usually inserted in the measuring instrument to give better correlation between the measured signal-to-noise ratio and the subjective impression of noise. When a CCIR/ARM weighting filter (commonly used when measuring signal-to-noise ratios of cassette tape and decks) is used, it will yield noise-reduction numbers of between 14 and 18 dB when the bandwidth of a system is restricted to 1 kHz with single-pole and two-pole low-pass filters, as shown in the curves of Fig. 2.

Auditory Noise Masking

The second basic principle behind DNR is masking—that hearing one sound decreases our ability to hear another. For example, white noise (random noise containing all the audible frequencies at equal amplitude) raises the threshold of hearing a pure tone by an amount that depends on the frequency of that tone, as shown in Fig. 3. The curve shows a general trend. At a higher frequency, a tone has to be increased in amplitude (compared to a 1-kHz tone) to be heard. That is because a wider range of noise frequencies contributes to masking as the tone's frequency increases. But regardless of the tone's frequency, there



NR systems like dbx and Dolby only work on specially encoded recordings; the DNR system works on anything.

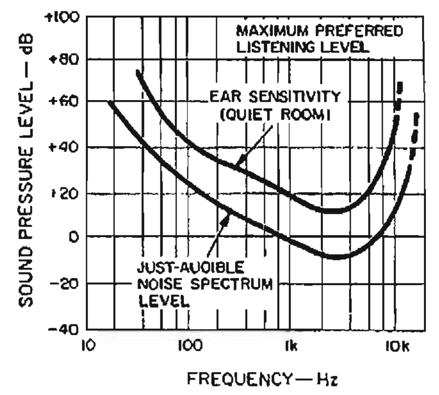


Fig. 1—The ear's sensitivity to noise varies with frequency.

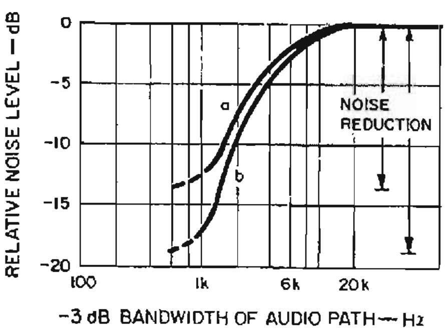


Fig. 2—How decreasing bandwidth affects noise reduction, with DNR configured as a single-pole (curve a) or double-pole (curve b) low-pass filter.

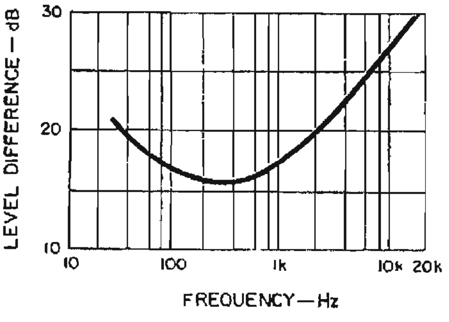


Fig. 3—Hearing thresholds above white noise for pure tone.

will be some range of noise frequencies that will be capable of masking that tone.

The results are not quite the same when we measure the ability of a single tone to mask undesired noise. Experimental results show that extremely high sound pressure levels of a single tone are required to provide masking. Even at the most effective frequencies (between 700 Hz and 1 kHz, near the natural resonance of the ear), sound pressure levels in excess of 75 dB are required to mask noise at a very low 16 dB SPL. Fortunately, those results apply only to *pure* single tones. With the complex signals that are a characteristic of music and speech, masking effects are much better. Most musical instruments produce broadband spectral components and a high concentration of energy around 1 kHz, which improve the noise-masking ability by more than 30 dB over a pure, 1-kHz tone.

From all of that, the designers at National Semiconductor concluded that if source material is at least 29 dB above the "noise floor," adequate masking can usually be obtained. Therefore, any noise-reduction system that dynamically restricts audio bandwidth will (by virtue of its previously calculated 14-dB improvement) insure a minimum perceived signal-to-noise ratio of 43 dB (29 dB + 14 dB) without audibly degrading the music program. A cassette tape recorded at a mean signal level of around −10 VU (volume units, as on a VU meter)—40 to 45 dB above the noise floor of the tape/system—will, with the aid of a bandwidth-varying noise-reduction system, be improved to a perceived signal-to-noise level of between 55 and 60 dB. If the recording was made at 0 VU, the improvement can be expected to provide an S/N ratio of better than 65 dB.

The DNR Audio Filters and Control Path

The general arrangement of the DNR system is shown in the block diagram of Fig. 4. Two low-pass filters (one for each stereo channel) are placed in the audio-signal path, their −3 dB bandwidths controlled by the amplitude and frequency of the incoming signals. Each filter response is flat below its cutoff frequency, with a smooth, single-pole roll-off above its corner frequency for any control setting. The resulting −6 dB/octave slope produces the most satisfactory results with modern and classical music having a wide frequency range. Steeper slopes can produce greater amounts of noise reduction for a given bandwidth, but are more suited to program material that does not have substantial high-frequency content. Cascading two filters will give a −12 dB/octave slope, with noise reduction as great as 18 dB (see Fig. 2).

As Fig. 4 shows, the LM1894 has three signal paths—right- and left-channel audio signal paths and a common bandwidth-control path. The main paths include audio low-pass filters whose cutoff frequencies vary in accordance with the control signal. A single control signal is used for both channels to keep the stereo image from wandering. That signal, in turn, is derived from the output of the summing amplifier, which is then filtered and rectified.

Since the spectra of musical instruments and the ear-sensitivity curve (Fig. 1) imply that masking is most effective at relatively *low* frequencies, you might assume that a low-pass filter would be good for the control path shown in Fig. 4. However, that turns out not to be the case. Figure 5 shows the frequency versus amplitude response of the DNR IC control path. The DNR system uses a *high-pass* filter with a −3 dB corner frequency of 6 kHz and −12 dB/octave roll-off slope. An optional notch at 19 kHz is for when the source material contains a stereo-FM pilot signal that might tend to increase minimum bandwidth above 800 Hz when the detector threshold is set at the noise floor.

The control-path frequency response is weighted in that manner because program material varies substantially in harmonic content, depending both on relative loudness and on the particular instruments being played. As an example, consider the case of a French horn. Most of the energy produced by that instrument is below 1 kHz. If a low-pass filter were used in the control path, it would respond to that energy and open up the filters to full bandwidth, unmasking noise in the 2-kHz and above region.

DNR works because output is proportional to bandwidth, and because hearing one sound decreases our ability to hear another.

To avoid that, the system looks for *high*-frequency energy in the music source, and, not finding any higher harmonics, in the case of the French horn, the noise remains filtered out and bandwidth remains restricted. Multiple instruments or a solo instrument such as a violin, for example, may have significant high-frequency energy that will not only provide good noise masking but will require a wider system bandwidth. To summarize, then, the detection of high frequencies in the system's control path indicates that large levels of energy must be present in the critical masking-frequency range. This means that the audio bandwidth can be safely increased to prevent audible degradation of the music, since the noise will remain masked. To make up for the relatively fast decrease in spectral energy with increasing frequency, the control-path response is increased at a 12 dB/octave rate.

Attack and Decay Times

If the detector of the DNR system were allowed to respond instantaneously to any input signal, ticks or noise bursts (of short duration but with rapid rise-times) would be able to open up the bandwidth of the system without simultaneous program masking. Also, different instruments have widely differing rise-time characteristics. With that in mind, the DNR system was designed with an attack time of 0.5 mS to

Fig. 4—Block diagram of the DNR system.

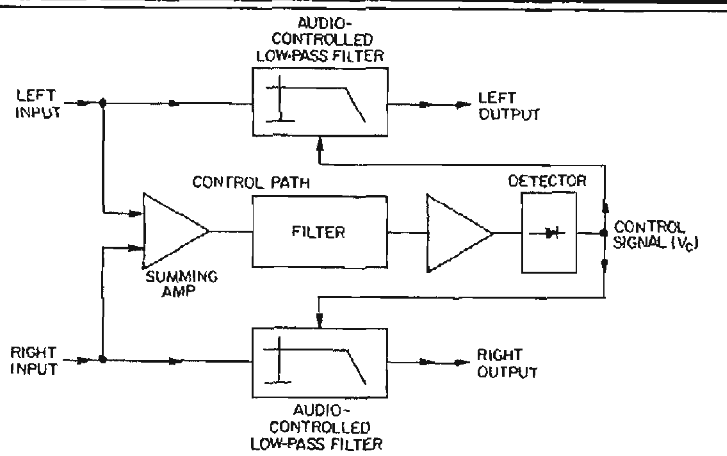
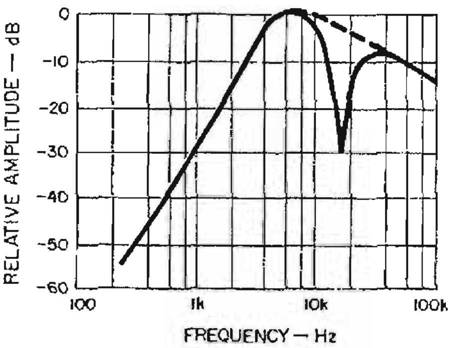


Fig. 5—Response curve for the DNR control path. Notch at 19 kHz is optional, to diminish response to FM pilot tone.



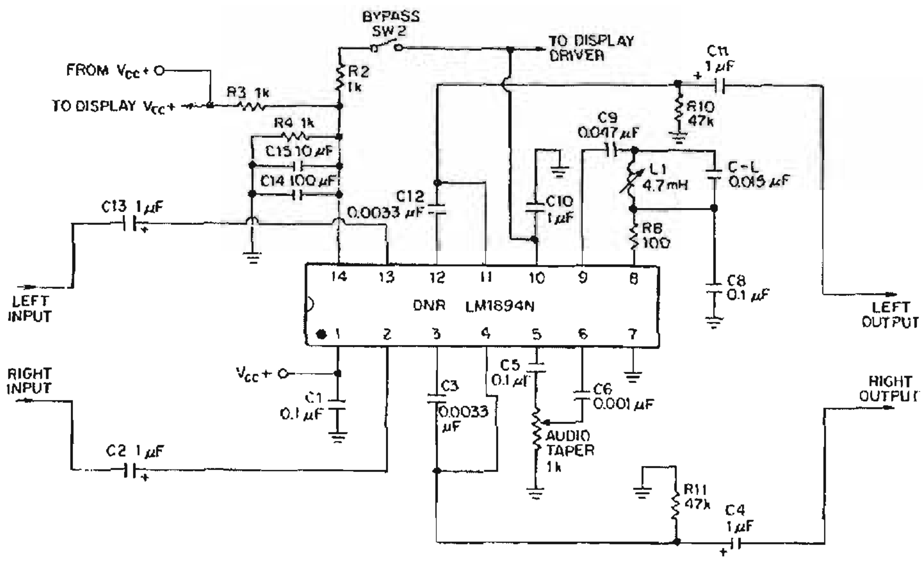
minimize potential loss of high-frequency transients. That does constitute a trade-off in that the system is susceptible to impulse-noise interference. Impulse noise, having fast rise- and decay times and quite a bit of high-frequency energy, must be eliminated using other techniques.

Once the detector has responded to a given musical transient, it must decay back to its inactive level when that transient is over. Once again, a compromise in parameters was required for the DNR system. Too slow a decay time would mean that system bandwidth would remain "wide open" for some period after the decay of the transient. A noise burst would be heard at the end of each musical transient since there would be nothing to mask it. If the decay was too rapid, on the other hand, a loss in apparent ambience would occur because harmonics occurring at the end of a transient would be suppressed. The DNR system decays to within 10% of final value in 50 mS. The ear's inability to recover sensitivity for 100 to 150 mS following a loud sound prevents the noise burst that is present at the end of each transient from being heard.

Using the DNR System

The DNR system is designed to be placed before a system's tone and volume controls. This is because any adjustment of these controls would alter the noise floor seen by the DNR control path. A sensitivity-adjustment pot is provided, which may need to be adjusted for the noise floors of different sources (e.g., tape, FM or phono). This control should therefore be left readily accessible.

Fig. 6—Schematic of the DNR system, main section.



A weighting curve adapts the control signal to the varying harmonic content of musical instruments and program material.

The system incorporates a display to assist in the proper setting of the sensitivity control. The display shows the instantaneous bandwidth of the two filters, *not* signal level (though signal level does affect it indirectly), and is logarithmic to best indicate the filters' audible effects. A bar-graph display is used instead of a meter because of the control signal's millisecond response time. The LM3915 display is recommended, as it requires only a few external parts and contains all the necessary circuitry for a 10-point, logarithmic display.

The left-hand LED corresponds to an 800-Hz bandwidth; the right-hand LED corresponds to a 30-kHz cutoff. The LEDs between these extremes each represent steps of approximately 1.5 times the frequency represented by the preceding step.

Using the Filter

The DNR unit can now be connected in the tape monitor loop of a receiver or amplifier. The sensitivity control should be turned down completely, and source material should be chosen that does not have musical content (the groove between cuts on a record, for instance).

Under these conditions, all but the first LED should be off. The sensitivity control is then advanced until the next LED just begins to flicker. This is an indication that the filter is barely opening on the noise floor and is capable of reaching full bandwidth on musical information above this level. Alternatively, the control may be advanced until there is a barely perceptible increase in the noise level and then backed off very slightly.

The bypass switch can be toggled between the bypass and active positions to compare the action of DNR with that of a full system response. The difference should be quite dramatic, giving a subjective improvement in S/N of 12 to 14 dB. The action of the filter is most apparent between record cuts, where it removes nearly all of the annoying hiss.

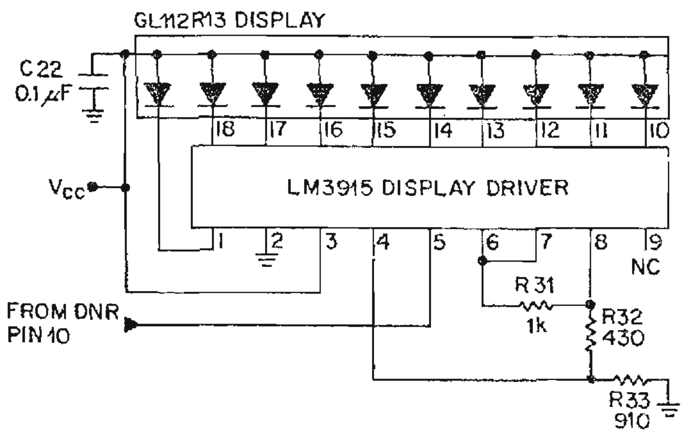


Fig. 7—Schematic of the display section.

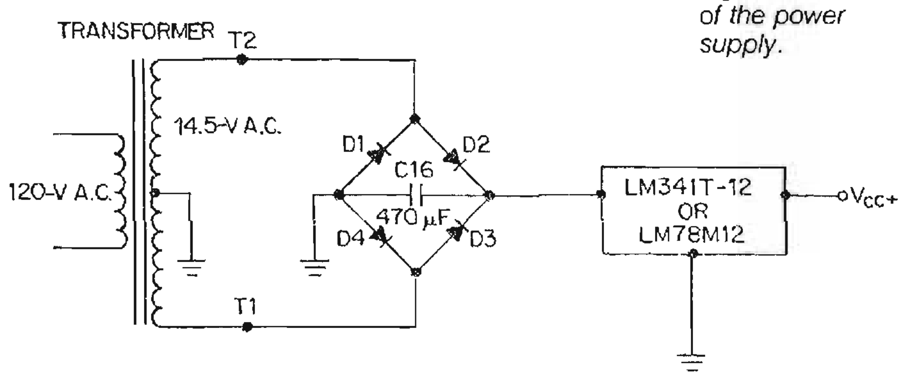


Fig. 8—Schematic of the power supply.

Parts List

Resistors

(All 5%, ¼-watt, except R1.)
R1—1-kilohm miniature pot, audio taper.
R2 through R4, R31—1 kilohm.
R8—100 ohms.
R10, R11—47 kilohms.
R32—430 ohms.
R33—910 ohms.

Capacitors

C1, C5, C8, C22—0.1-µF, 50-V Mylar.
C2, C4, C10, C11, C13—1-µF, 50-V electrolytic.
C3, C12—0.0033 µF, 50-V polystyrene, axial-lead.
C6—0.001-µF, 50-V Mylar.
C9—0.047-µF, 50-V Mylar.
C14—100-µF, 50-V electrolytic.
C15—10-µF, 16-V electrolytic.
C16—470-µF, 25-V electrolytic.
C-L—0.015-µF, 50-V Mylar.

Semiconductors

D1 through D4—1N4002 diodes.
IC2—LM1894N DNR.
IC3—LM341T-12 or LM78M12 regulator.
IC4—LM3915N display driver.
IC5—GL112-R13 12-element display.

Miscellaneous

SW1, SW2—SPST rocker switches.
L1—4.7-mH adjustable inductor, Q=35 at 19 kHz (Toko CLN20-740HM).
PH1, PH2—Dual phono jacks.
Transformer—14.5-V, 250-mA, center-tapped (Triad F-112X).
Fuse—¼-amp, slo-blow.
Fuse-holder, line cord, p.c. boards, control knob, and miscellaneous hardware.

The following are available from Advanced Audio Systems International, 4010 Moorpark Ave., Suite 105, San Jose, Cal. 95117. (California residents, add 4.5% tax.)
Complete kit, including silk-screened enclosure (DNR-200X) $122.50
Listed semiconductors and coil L1 (DNR-240X) $35.95
Main and display p.c. boards (DNR-280X) $22.50

The DNR system must precede the sound system's tone and volume controls, lest they alter the noise floor it senses.

You should be aware of a psychoacoustic effect that is common to all noise-reduction systems. The addition of high-frequency noise (such as tape hiss) to a music signal will seem to increase the high-frequency content of the music. Thus, upon first auditioning DNR using noisy source material, the user will seem to hear a degradation of the music's high-frequency content. The system's actual effect on the high-frequency information can be observed by using a quiet source and switching the filter in and out.

It should be noted that the filter is designed for an average input level of 750 mV rms. Some tape decks are capable of much larger output levels at "0 VU," and they should be attenuated accordingly to prevent overloading the filter inputs.

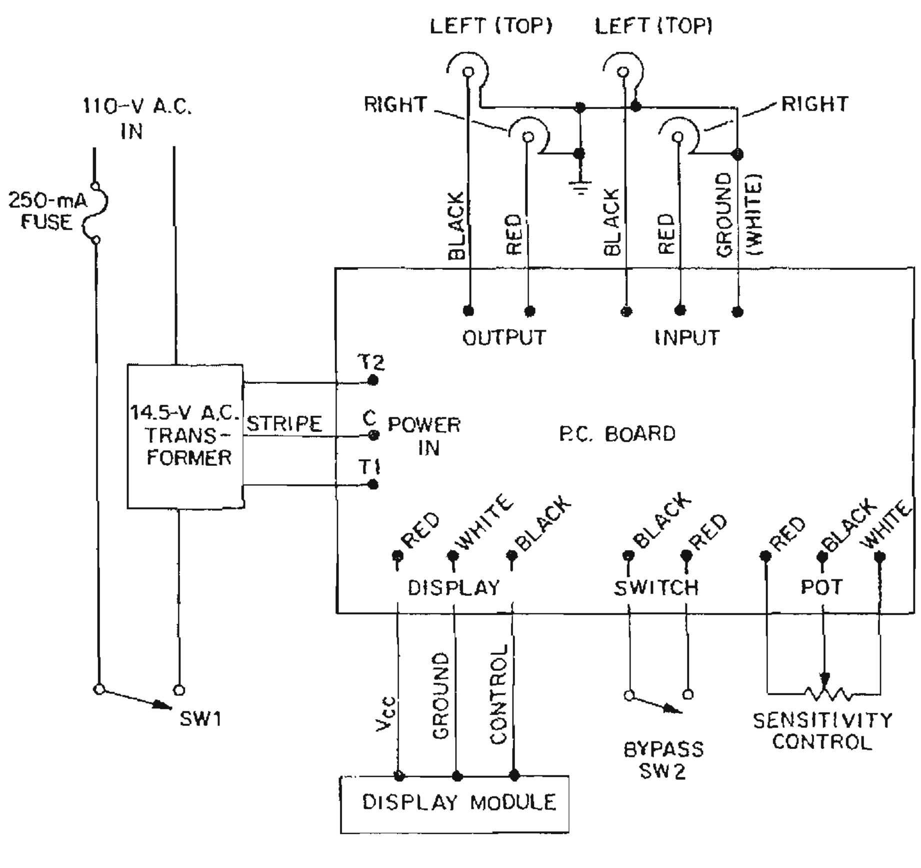
Building the DNR System

Construction is fairly simple, as the bulk of the circuitry is on two ICs, the LM1894N DNR chip and the LM3915 display (Figs. 6 and 7), plus a power supply (Fig. 8).

The 19-kHz multiplex pilot tone present in all stereo FM broadcasts is attenuated by L1 and C-L. The presence of this pilot tone will limit the noise-reduction capability, since the noise filter will sense the level of the pilot tone rather than the level of the noise source.

The inductor provided with the kit of parts described in the Parts List is pretuned, and its adjustment should not be altered. If, however, you purchase coil L1 separately, then it must be tuned to within about ±20 Hz of the 19-kHz pilot tone. The simplest way to obtain this reference frequency is to get it directly from the FM broadcaster. Tune your FM receiver and wait for a quiet interlude when there is no audio signal. Tune L1 for minimum noise-filter bandwidth as monitored on the front panel's LED display.

Fig. 9—Assembly diagram, for use with p.c. board.



Video and TV sound can create similar problems due to the presence of strong line-scan components at 15.734 kHz. This can be accommodated by substituting a capacitor of 0.018 to 0.022 μF for the 0.015-μF value indicated for capacitor C-L, and readjusting L1. If both FM pilot and video line frequencies cause problems, it might be advisable to build two traps, with a selector switch. On the other hand, if neither FM nor video-sound signals are to be processed, then choke L1, capacitors C-L and C8, and resistor R8 (which forms a frequency roll-off with C8) may be eliminated.

Printed circuit boards, silk-screened to show component placement and polarity, are available separately or as part of a complete parts kit. All components are mounted on these boards, except for the sensitivity control, switches, jacks, the power transformer and the fuse-holder. We recommend using IC sockets or Molex strips instead of soldering ICs in directly, to prevent possible damage to the ICs.

The component designations on the p.c. board and their explanation in the Parts List should provide all the information needed for successful board assembly. A schematic diagram (Fig. 6) is provided for technicians who feel more comfortable working from that, and for help in understanding and trouble-shooting the circuit. The display and power-supply circuits are shown in Figs. 7 and 8.

Figure 9 shows how the boards and other components are wired together. Wire functions and colors are also silk-screened on the boards.

Nonetheless, mistakes are still possible. Our experience has shown that the most common assembly errors are as follows: ICs inserted backward, electrolytics installed with incorrect polarity, power diodes reversed, bad solder connections, no fuse in the fuse-holder, and failure to wire-connect one or more phono-jack grounds to a common ground on the p.c. board. But with a little care (or, failing that, a little after-the-fact trouble-shooting), you should have an addition to your system that makes a worthwhile, audible difference. Get out your old, pre-Dolby tapes and surface-noisy records, and prepare to listen to them—and enjoy them—once again.

CASSETTE QUALI

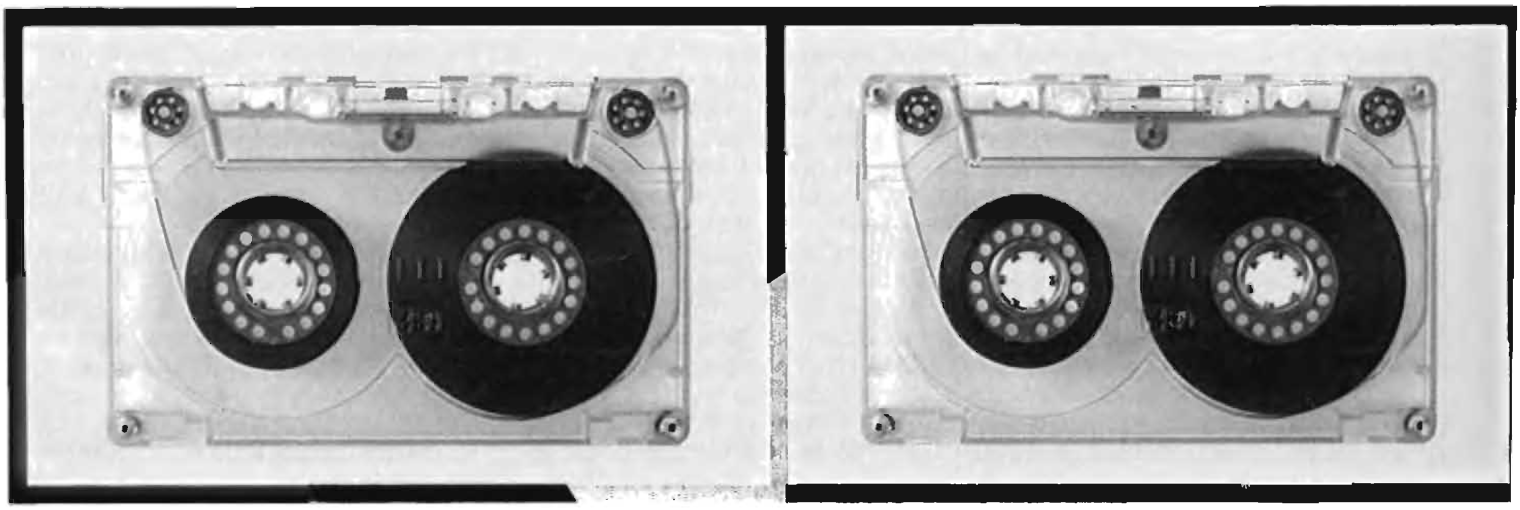
WHAT IS THE INDUSTRY

MIKE JONES

There was a time, not long ago, when the cassettes you recorded at home by copying LPs would sound superior to any of the music cassettes you could buy from your local record store. The same cannot be said today. The current music cassette, duplicated at high speed using Dolby HX Pro (see *Audio*, August 1984) and the latest tape formulation, is capable of superb fidelity. Its quality can compare favorably to mass-produced vinyl records—particularly those with more than 22 minutes recorded on each side.

Dolby HX Professional and new tapes are just part of the story, for in truth the music cassette has undergone major development during the last few years, benefiting from better cassette housings, improved mastering techniques, and new duplicating equipment. Another important factor has been the attitude of the duplicators themselves: They are newly and increasingly determined that their cassettes will be the best they can produce, comparable to those that can be produced at home. Indeed, the leading duplicators are so committed to improving quality that many of them attended a seminar organized by Electro Sound in California last year, following a similar event in London, to discuss various ways in which the quality of music cassettes could be improved even further.

Strange as it may seem, the largest obstacle to progress on quality is not a technical problem for duplicators but one of persuading the record companies to make full use of current technology—let alone what might be developed in the future. Despite the large profit margins involved, some record companies are willing to use only the cheapest materials for cassettes, in order to achieve maximum profits. They say that the consumers who buy their products do not care about quality, and the evidence given to support this claim is that they receive very few complaints. Boldly speaking, they have a captive market of 10- to 25-year-old music listeners who, if they wish to buy cassettes of their favorite groups, have to accept whatever quality the record company decides to produce. Why some record companies do not aim to give good value is quite beyond most analysts, especially



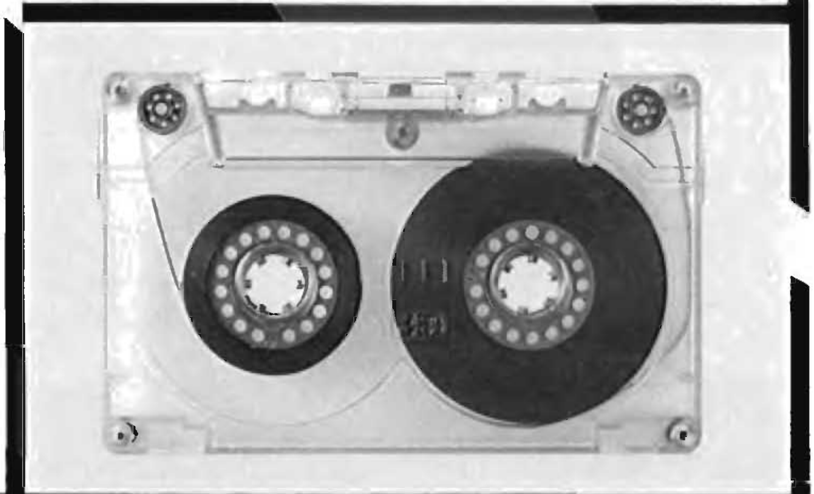
TY: DOING?

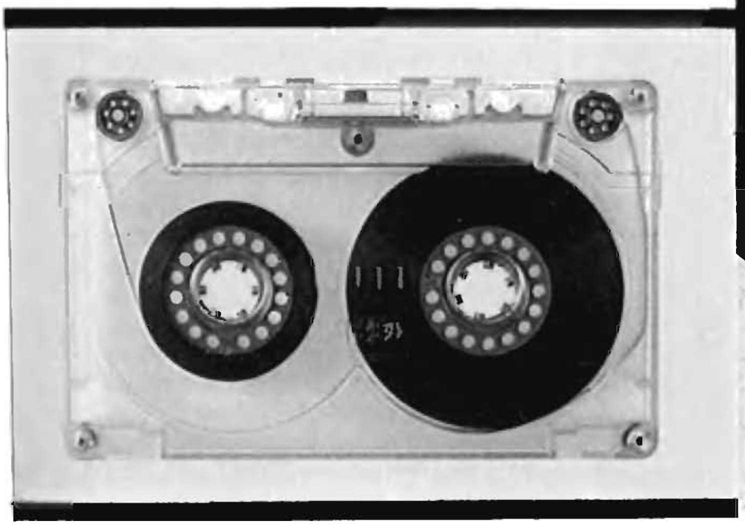
since they could obtain enormous improvements in quality using their existing equipment by paying more attention to detail during the transfer and manufacturing stages.

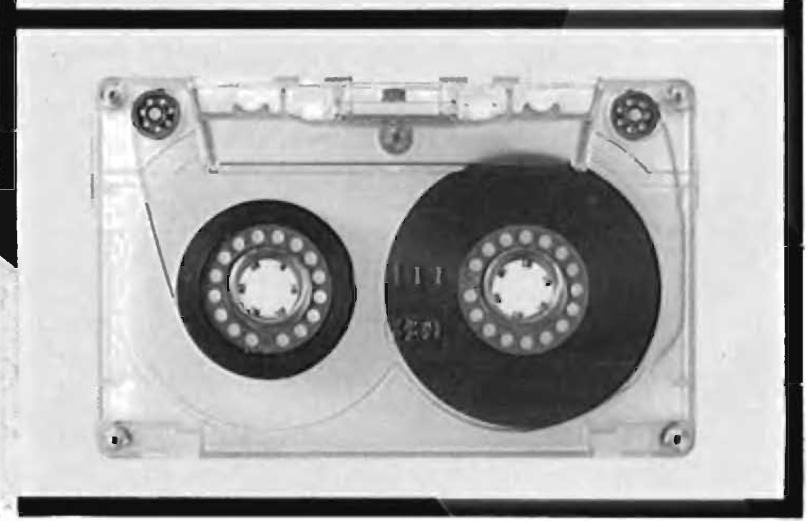
I believe that it is incumbent on the record producer to insist on good quality throughout; indeed, he can help make the necessary funds available by not allowing his group to waste valuable studio time, as is often the case.

Quality from the Beginning

Even when the best materials are used, the sound quality of a cassette can be easily ruined by an inferior master, and many of the speakers and delegates who attended the California seminar expressed concern over this. Ideally, the source master from which cassettes are duplicated should be either a digital or a first-generation analog copy of the stereo master. If subsequent-generation analog copies are used, the signal-to-noise ratio and distortion will deteriorate, thus restricting the dynamic range. Unfortunately, many record companies do not seem to realize







Even when the best materials are used, a music cassette's quality can be ruined by an inferior master.

or care about this, often sending fourth- or fifth-generation tapes for duplication. Under these conditions, it is this running master—not the cassette tape or the duplication process—that will limit the sound quality. A far better alternative would be to use digital transfer, which avoids many of these problems.

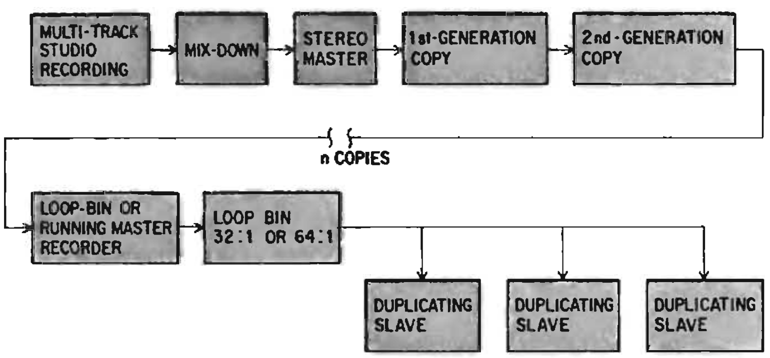
Before I go any further, I should explain why digital transfer is superior to analog with respect to deterioration of the audio signal. Once a recording has been converted to digital format, and providing that any subsequent editing, mixing or transfer is executed in the digital domain, avoiding D/A and A/D converters, it will not matter how many copies you make from the initial digital recording, or how many generations you produce—they will all sound identical to each other and will undergo no loss of quality.

The use of chrome tape and the advent of Dolby HX Professional have increased the dynamic range of cassettes, so that the use of high-quality masters is more crucial than ever. This subject was considered so important by the seminar organizers that an entire session was devoted to it. Ken Gundry of Dolby Laboratories gave an extremely interesting presentation on how Dolby HX Pro can be used to extend the dynamic range of music cassettes at high frequency, up to 10 dB at 15 kHz. When you listen to a cassette recorded with HX Pro, the improvement can be quite dramatic, with ferric tape sounding like chrome and the latter like metal. Some material similar to what Ken presented is shown here. In addition to demonstrating the beneficial effects of HX Pro on cassette tape, he also showed how easy it is for the performance of the running master to restrict the quality of the cassette, especially when the cassettes are duplicated at up to 64 times normal speed. Ideally, what we need is a method of extending the dynamic range of the running master as well as that of the cassette; this is exactly what is achieved by fitting Dolby HX Pro to the recorder on which the running master is recorded.

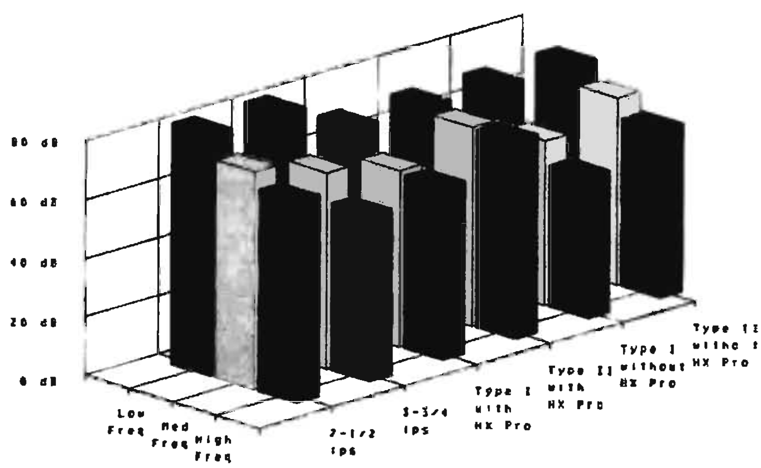
Two manufacturers, Otari and Studer, currently produce the new type of mastering machines. Studer has also improved the phase response and tape guidance of their recorder to increase the stability of the tape as it passes across the heads. The result is a cleanup of the phase response at the top end of the spectrum and an improved stereo image. This has proved necessary because the masters for duplicating cassettes at 64:1 are normally recorded at 3¾ ips.

Upgrading or replacing existing equipment, to enable a duplicator to take advantage of Dolby HX Pro, is an expensive business. While there is no doubt in my mind that it is worthwhile, some duplicators say that they cannot tell the difference between cassettes which have been recorded with HX and those which have not. If this is the case, then they should look very closely at their system or their duplicating methods and masters, for something is definitely wrong.

Any doubters should try listening to the prerecorded cassettes, currently being produced by Capitol and EMI Records, which carry the "XDR" logo. Not only were Capitol and EMI the first companies to use Dolby HX Pro on both sides of the Atlantic, they were also the first to develop a system whereby they could control the consistency and quality of the cassettes they produce on an international basis. The XDR program is one of continual devel-



Block diagram of cassette duplication.



S/N in three frequency ranges, for master tapes and duplicated cassettes. From left to right, the first two sets of bars represent 1-inch, 4-track tape with Dolby B NR. The remaining four sets of bars are for cassettes duplicated at 32:1 speed ratios, using Dolby B NR. Frequency ranges are: Low, 1 kHz for open reel, 375 Hz for cassette; medium, 1 to 10 kHz; high, 15 to 20 kHz.

The key to cassette quality is getting record companies to make full use of current and future technology.

opment and one that benefits the consumer a great deal. Unlike the "Chrome for Quality" logo, the XDR symbol means that the cassette has been produced to exacting standards which cover the entire recording chain from studio to final product. If just one part of the chain falls below specification, then the XDR logo is not used.

I hope that Dolby Laboratories will control the use of the HX Pro logo in a similar way so it, too, will become a symbol of quality. There is no doubt in my mind that BASF missed a golden opportunity to improve the standard of prerecorded cassettes when they allowed the industry to use the "Chrome for Quality" logo without any effective control as to how or when it was used. Now all it really states is that chrome tape has been used, which doesn't say anything about the quality of the recording—unlike XDR or Teldec's Direct Metal Mastering (DMM) logo, both of which do.

The Basic Particle

Having decided to produce the highest quality cassettes possible, what tapes are available to the cassette duplicator? The most common type is coated with gamma iron oxide, more commonly known as ferric, and it is similar to many of the Type I blank cassettes that you can buy. While gamma oxide tape has been around since the late '50s, the tape we use today is nothing like the product that was produced at that time. As Frank Diaz of Columbia Magnetics explained at the seminar, a new ferric particle is developed every five years or so. Modern ferric tapes are quieter, offer more dynamic range, and are far more stable than their predecessors.

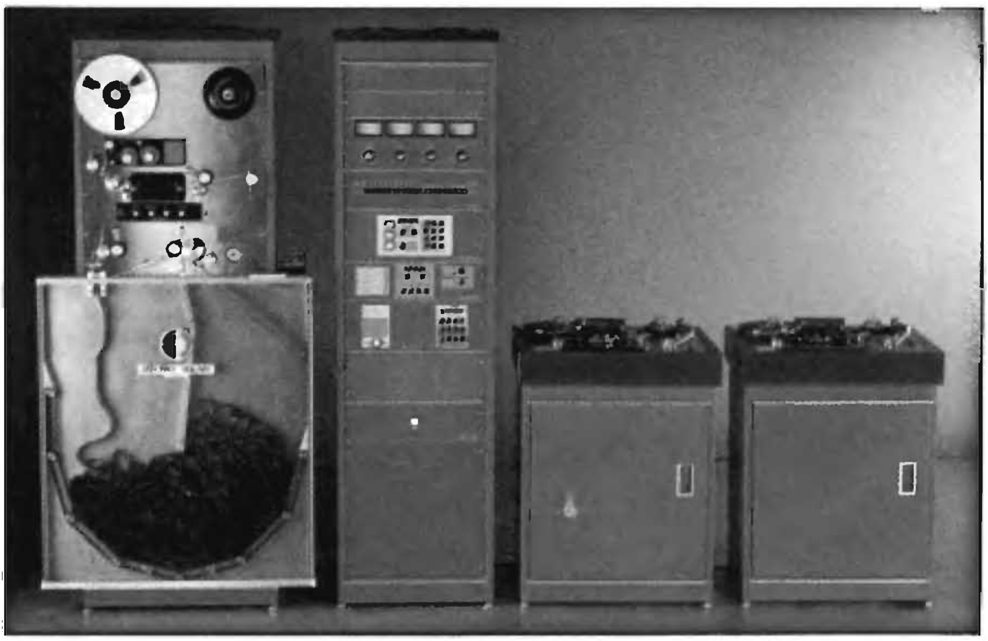
But is ferric good enough, or is there something better? Judging the issue on purely technical grounds, I have to say there is a better product, chrome tape, and I believe that the majority of the experts who attended the seminar would agree with me. Chrome tape's superiority is largely due to the efficiency and shape of the chromium dioxide particle and the way in which it outperforms pure ferric formulations for most applications, including audio, video and data. Indeed, chromium dioxide could be regarded as the Rolls-Royce of magnetic particles.

In audio recording, chrome's output level is similar to ferric's at low frequencies, but it provides a higher output at short wavelengths, which enables it to handle synthesized music and digital recordings with minimum compression. Because the chromium dioxide's particle size is smaller and more uniform than its ferric counterpart's, chrome can be packed onto the tape more evenly and with greater density, thus reducing bias noise and tape hiss and increasing output. Of particular interest to the critical listener is the substantial reduction in modulation noise that is obtained with chrome; this feature, in my view, makes its universal use for music cassettes well worthwhile. In addition, an old problem has been overcome by the introduction of new manufacturing methods; print-through has been reduced to a point where the figures obtained for chrome are now lower than those for many cobalt-doped tapes.

So, in pure audio terms, chrome has a lot going for it, and those recordings duplicated on chrome have a clarity and transparency that would be hard to obtain from any ferric tape. But because of commercial interests, chrome has not received the universal support from the industry that it should have obtained on purely technical grounds. Like Dolby HX Pro, some duplicators cannot hear any improvement when they use chrome, but again, this is likely due to problems with their equipment or low-quality masters. What they must realize is that to achieve the excellent results that chrome is capable of, the duplicator must pay a great deal of attention to detail if there's to be an audible difference in the finished product.

Because of the impact chrome has had on the industry, other tape manufacturers who do not have licenses from DuPont to produce chrome have produced alternatives such as Magnetite (from Agfa) or CS-1 (from Capitol). The initial results obtained with these new products are very promising, and in subjective listening tests they come close to those obtained with chrome. The important thing to remember is that all of them are a vast improvement over standard ferric formulations, *providing* they have been duplicated correctly and from a good-quality master.

High-speed cassette duplication frequently uses a continuous-loop master tape running at high speed through a loop bin. In this Electro Sound 8000 system, the master runs at 240 ips, while slave units record onto large pancakes of cassette tape at 240, 120 or 60 ips. Quality increases as the slave speed and speed ratio to the final cassette go down.



Cassettes recorded with HX Pro improve dramatically, with ferric sounding like chrome and the latter like metal.

Cassette Housings

Another seminar session dealt with cassette housings, and the majority of those attending agreed that, while the tracking accuracy of cassette housings has improved a great deal in recent years, there is still room for improvement. In truth, higher quality recordings place a greater demand than ever on the cassette housing (or shell), which has to be made more precisely so it can guide the tape across the playback head more accurately. Poor tracking will have the same effect as a playback head which is in need of azimuth adjustment.

Tape-tracking errors due to cassette-shell defects, such as the slanted pins shown in the lower drawing, have the same effects as playback-head azimuth alignment errors.

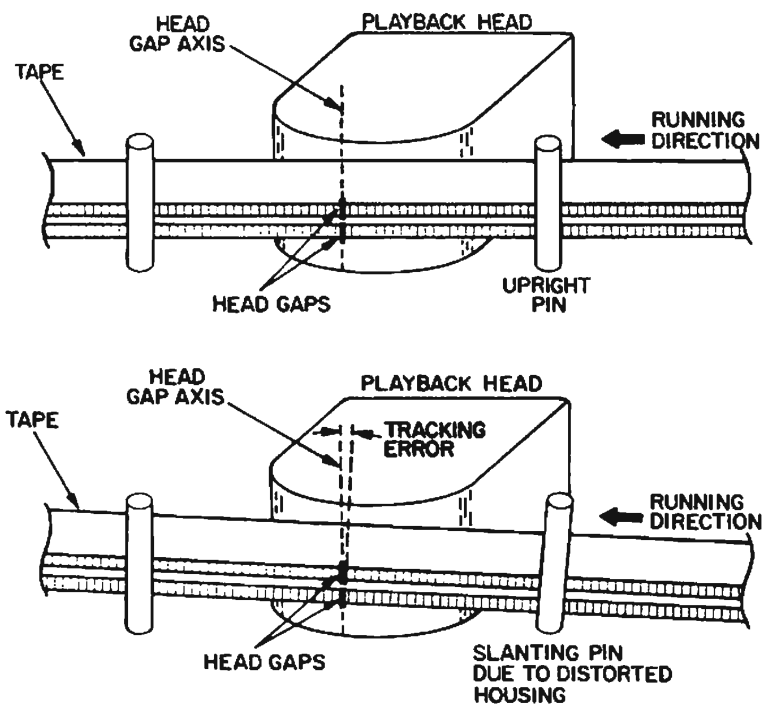


Table I—Tape speeds in various duplicating systems.

Electro Sound and Gauss Duplicating Systems

Duplicating Ratio	Running Master Speed	Real-Time Master Speed	Slave Speed
32:1	240 ips	7½ ips	60 ips
64:1	240 ips	3¾ ips	120 ips
128:1	240 ips	1⅞ ips	240 ips

New Otari System, ½-Inch Only

Duplicating Ratio	Running Master Speed	Real-Time Master Speed	Slave Speed
32:1	480 ips	15 ips	60 ips
64:1	480 ips	7½ ips	120 ips
128:1	480 ips	3¾ ips	240 ips

Concern was expressed at the seminar over the way some consumers store their cassettes, particularly those who use cassettes in cars, where temperatures can easily climb above the softening point of plastic. The point to remember is that the plastic body of the cassette will distort long before it melts. Distorted housings will not track well, nor will they sound right.

I had the pleasure of moderating the final session of the seminar, where the attendees took a look into the crystal ball at the future of the industry. Of course, the conversation soon came 'round to the Compact Disc and alternative digital systems, such as the digital cassette. Most agreed that the Compact Disc will soon become the major competitor to cassettes, especially for the classical market, and that vinyl would eventually fade away. Although digital cassette systems are being developed in Japan, manufacturers have not yet agreed on a single common standard. Because of the public's general awareness of the Compact Disc and of improved recording technology, a great deal of pressure will be put on the duplicating industry to improve the quality of cassettes even further. Pressure will also be put on the record industry to make use of current technology and future improvements. The danger is that the record companies may, in actual practice, do the opposite: Master from low-quality or noisy masters, as happened with the early Compact Disc releases.

As technology pushes forward we should take advantage of it in order to enrich our enjoyment of music. Under the leadership of companies such as Electro Sound, the duplicating industry has demonstrated that the quality of prerecorded cassettes can be very high indeed. However, it will only be through increased and continued public pressure for quality recordings that the market will more widely exhibit such quality. So long as sales and profits do not suffer because of low quality, record companies can hardly be blamed if they let quality slide by not investing in newer and better equipment and materials. They are, after all, in business to make money, and they need to be shown that Gresham's Law (that bad money drives out good) does not apply to music cassettes.

THE WHYS AND HOWS OF CASSETTE EQUALIZATION

HERMAN BURSTEIN

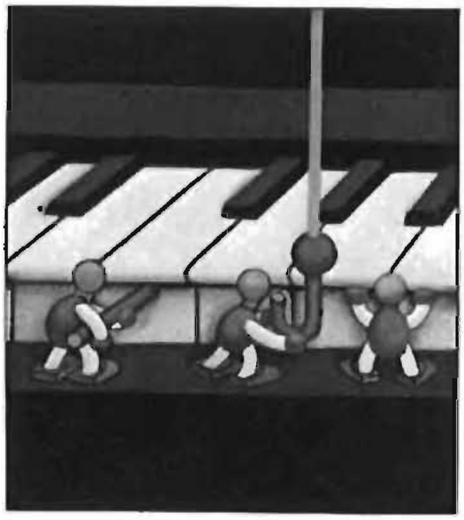
To get flat, wide-range response from tapes requires equalization, which is frequency alteration in recording and playback to overcome the tape system's inherent frequency deviations. The choice of equalization characteristics interrelates with problems of noise and distortion, and varies among tape types and even among tapes of the same type.

A tape playback head is a "velocity" device, whose output increases as the signal frequency increases. That is, for a given signal level on the tape, the output of the head doubles if frequency doubles, goes up tenfold if frequency increases by a factor of 10, and so on.

It is more convenient to express this relationship in decibels. For a constant signal level presented to the playback head, as frequency rises output rises just about 6 dB/octave, or exactly 20 dB/decade. This is true for an "ideal" head, one without losses or other aberrations, as distinguished from actual heads. Today's high-quality heads come quite close to the ideal, but there are still differences of some consequence in the extreme bass and, particularly, in the extreme treble.

Momentarily, let's assume we have both an ideal, lossless tape system which produces a flat recorded signal on the tape and an ideal playback head. Unequalized record-playback response is then the same as the output of an ideal head, shown by curve ABC in Fig. 1. To achieve flat response, it is merely necessary to employ playback equalization which mirrors head output. This is curve DBE, declining 6 dB/octave as frequency rises. (A simple R-C [resistance-capacitance] circuit could readily take care of the matter.) Curve FBG is the resulting flat response.

Figure 1 is not entirely fanciful. At a high tape speed such as 30 ips, the unequalized record-playback response and the required equalization would be very close to curves ABC and DBE. And life would be simple.



In playback, treble loss is due mainly to the width of the head's gap: The wider the gap, the greater the loss.

But as tape speed is reduced, significant losses appear, particularly in recording. By the time we get down to 1⅞ ips, these losses are profound. Figure 2 takes us from the ideal world into the real world, showing the typical unequalized record-playback response of a high-quality, two-head cassette deck—in this case, a Harman/Kardon CD391—when using Type II (chromium dioxide or ferricobalt) tape. (I am indebted to Peter Philips of Harman/Kardon for supplying measured data on the CD391's equalization and frequency response, which I used to produce Fig. 2 and several other figures.)

In Fig. 2 we principally note the huge treble loss, amounting to about 1 dB at 1 kHz and reaching about 41 dB by the time we get out to 20 kHz. This loss is the difference between the ideal, 6-dB/octave rising response (curve ABC in Fig. 1) and the actual treble response shown in Fig. 2. About 36 dB of the loss at 20 kHz occurs in recording, and about 5 dB in playback.

Treble loss in recording is due mainly to demagnetization. The recorded signal consists, in effect, of a series of bar magnets; as frequency rises they grow shorter (more cycles per second entail more—and necessarily shorter—magnets in a 1-S span of tape), so that their opposite poles get closer together and tend increasingly to cancel each other. The erasing effect of bias current is also substantial; this effect increases as frequency rises because high frequencies are not as deeply embedded in the tape as others. In addition, slight treble loss occurs in the record head due to winding capacitance, eddy currents, and hysteresis.

In playback, treble loss is due mainly to the width of the playback head's gap. The wider the gap, the greater the loss. As frequency increases and the recorded bar magnets grow shorter, gap width approaches magnet width and the resolving power of the gap begins to fail. The record-playback head in a two-head deck ordinarily has a wider gap, and incurs greater treble loss in playback, than the separate playback head in a three-head deck. Slight additional treble losses occur in the head due to winding capacitance, eddy currents, and hysteresis.

There is some further playback aberration below 40 Hz, too. First, there are several bumps in response, on the order of 0.5 to 1.5 dB. Second, there is a slight uptilt with respect to the ideal 6-dB/octave response, as shown by the smoothed version of the deck's response. The uptilt reaches about 3 dB at 20 Hz and makes a slight contribution to the bass boost needed in playback. The foregoing phenomena are due to the "contour effect," whereby the entire head, not merely its gap, reacts to the magnetic flux on the tape.

With the exception of the slight head losses due to winding capacitance, eddy currents and hysteresis, all the losses described above become increasingly severe as tape speed is reduced. It takes just as many flux changes ("bar magnets") to record a given frequency at a slow tape speed as at a high one. But as speed is reduced, the amount of tape that passes the head per second is also reduced, and the magnets must become shorter in order to fit into the allotted length of tape. As we have already noted, the major record and playback losses increase as the recorded magnets become shorter. In technical terms, these major losses increase as the recorded wavelength decreases, with wavelength in inches being tape speed in ips divided by frequency in Hz.

How Equalization Is Achieved

Figure 2 makes it obvious that, in general terms, bass boost and treble boost are needed to restore flat response. Not obvious is the amount of boost required and whether each kind of boost should be provided in recording or playback. Fortunately we have industry standards, which in broad terms call for the following:

- Bass boost is to occur largely in playback; if applied in recording, the vast amount of bass boost needed would overload the tape.
- Treble boost is to occur largely in recording. Substantial treble boost in playback would heavily accentuate noise, because great amplification is required for the tiny signal produced by the playback head.
- Record-head losses are to be compensated for in recording, playback-head losses in playback.
- A specific playback equalization curve is to be followed, depending on tape speed and tape type. This is fundamentally a bass-boost curve, modified (in accordance with the above principle) to compensate for each individual deck's playback-head losses. Stated differently, the *combination* of playback amplifier equalization *plus* head losses must conform to the standard playback curve.
- The record equalization is to be such that, in conjunction with standard playback equalization, flat response is achieved. This is largely a treble boost, some of which (usually very little) is to compensate for record-head losses.

We now turn to specifics for cassette-deck equalization.

Two standard playback equalization curves are provided for cassettes, as shown in Fig. 3. Curve ABC shows standard playback equalization for Type II (chromium dioxide or ferricobalt), Type III (ferrichrome—now little used), and Type IV (metal) tapes. Curve DBE shows standard playback equalization for Type I (ferric oxide) tapes. It is customary to show these curves using 400 Hz as the reference frequency.

The standards express these curves in terms of turnover (also called transition) frequencies, or in terms of time constants. The relationship between turnover frequency and time constant is f equals 159,155 divided by t, where

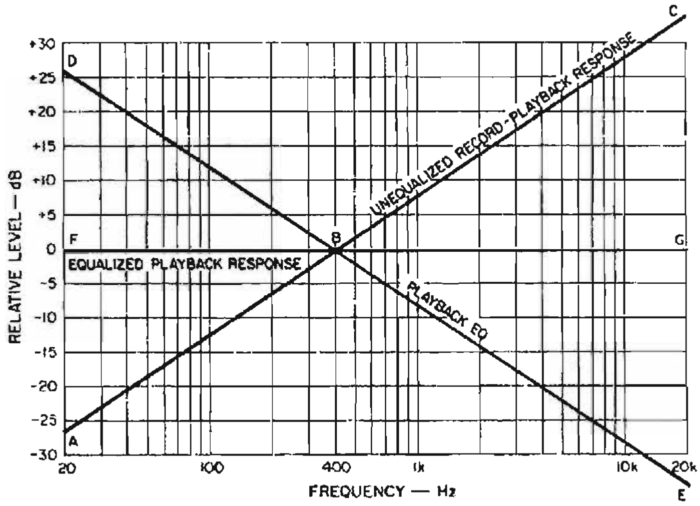


Fig. 1—Response and equalization in an ideal (lossless) tape recording system.

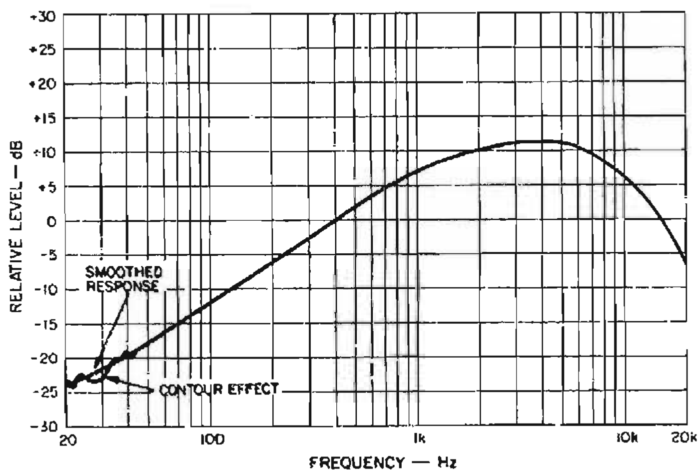


Fig. 2—Unequalized record-playback response of a two-head cassette deck (Harman/Kardon CD391), with Type II tape and appropriate bias.

f is turnover frequency in Hz and t is a time constant given in microseconds. Correspondingly, t equals 159,155 divided by f.

Curve ABC has designated time constants of 70 and 3,180 µS. Accordingly, the turnover frequencies are 2,274 and 50 Hz. This signifies that bass boost commences at 2,274 Hz (where it is up 3 dB) and levels off at 50 Hz (where it is 3 dB below maximum). Curve DBE has time constants of 120 and 3,180 µS, or turnover frequencies of 1,326 Hz and 50 Hz. Total bass boost—from above 20 kHz to below 20 Hz—is 33.1 dB for the 70-µS curve and 28.5 dB for the 120-µS curve.

Depending on tape type used, a cassette deck is supposed to conform to one of the two playback curves in Fig. 3. It bears repeating that the deck's *total* response—the combination of its playback-amplifier equalization and playback-head losses, not its equalization alone—must conform to these curves.

To see how this works in practice, let's return to the Harman/Kardon CD391 unit, whose unequalized response is shown in Fig. 2. The equalization supplied by that deck's record and playback amplifiers, and the resulting record-playback response, appear in Fig. 4. Curve ABC, consisting chiefly of bass boost, is provided by the playback amplifier, and curve DBE, consisting chiefly of treble boost, is supplied by the record amplifier. When the equalizations of ABC and DBE are applied to the unequalized record-playback response of Fig. 2, they produce the record-playback response of FBG in Fig. 4, which is substantially flat through the audio range.

It is to be noted in Fig. 4 that segment DB of record curve DBE includes a mild bass boost. This partly compensates for the fact that segment AB of playback curve ABC does not extend linearly all the way to the lowest frequencies but starts to level off (3 dB below maximum) at about 50 Hz—consistent with standard equalization. Further boost at the low bass end is supplied by the playback head's slight uptilt, as observed in Fig. 2. The net result of AB in playback, DB in recording, and playback uptilt is to maintain bass response a little short of flat. As shown by FB, bass response drops slightly below 35 Hz; it is about 1.5 dB down at 20 Hz.

At the extreme high end, we may note in Fig. 4 that this deck's record-playback response exhibits a trivial rise of about 1 dB. This is either because of slightly excessive treble boost due to component tolerances in the record equalization, or because the tape to which the record equalization was applied is a bit "hot" at the high end.

How close does actual playback-amplifier equalization (curve ABC in Fig. 4) come to the standard equalization (curve ABC in Fig. 3)? Figure 5 compares the two. In Fig. 5, DBE is the actual equalization and ABC is the standard. The two curves are very close. Below 400 Hz, segment DB of the actual curve is no more than 1 dB away from segment AB of the standard curve. This minuscule difference is probably due to component tolerances. Above 400 Hz, it appears at first sight that the actual and standard segments BE and BC part company too much. But it must be remembered that the industry standard requires the actual curve to include compensation

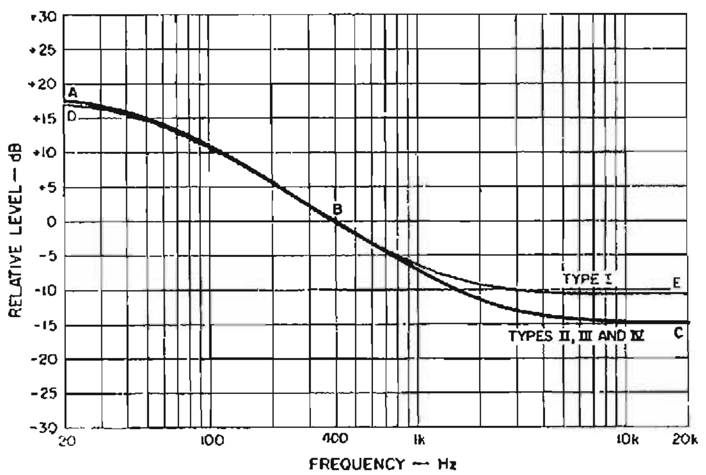


Fig. 3—Standard playback equalization curves for cassette, including both amplifier equalization and playback-head losses. Curve ABC, for Type II, III and IV tapes, has turnovers of 70 µS (2,274 Hz) and 3,180 µS (50 Hz); curve DBE, for Type I tape, has turnovers of 120 µS (1,326 Hz) and 3,180 µS.

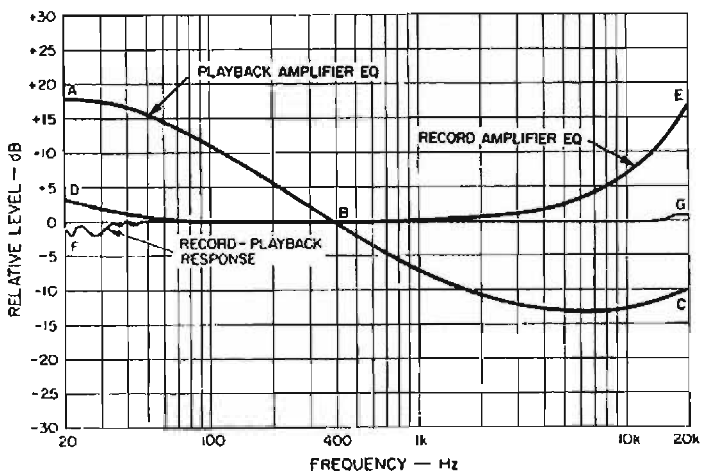


Fig. 4—Record and playback amplifier equalization of an actual deck (Harman/Kardon CD391 for Type II tape). Applying playback equalization ABC and record equalization DBE to the unequalized response of Fig. 2 yields the actual record-playback response, curve FBG.

for losses of the playback head. Therefore BE includes treble boost, reaching about 5 dB at 20 kHz, which essentially accounts for the difference between BE and BC. If one adds playback-head losses to BE, this brings the curve down so that it becomes BC.

Standardizing Playback Curves

As we shall bring out later, for a given cassette type, it is feasible to use more than one kind of playback curve in terms of the amount of bass boost provided. So why has the amount of bass boost been standardized? And why are there two standard curves for cassettes?

One reason is to extract the most from the tape in terms of extended treble response, minimum noise, and minimum distortion. These three desirables are conflicting; that is, an improvement in one respect necessitates a sacrifice in one or both of the others. Proper choice of turnover frequencies, particularly the upper one, can produce an optimum compromise among the conflicting requirements.

We can look at the standard playback curves of Fig. 3 in a different light than previously: We can view them as providing treble cut above 400 Hz. The greater the descent of the curve (the greater the treble cut), the more it reduces noise in the playback portion of the tape system. Therefore, the 70-µS curve initially appears preferable to the 120-µS curve.

However, things are not that neat. As we shall explain later, the amount of bass boost (or treble cut) employed in playback governs the amount of treble boost needed in recording. The greater the bass boost in playback, the greater the treble boost needed in recording. Thus, the 70-µS playback curve necessitates more record treble boost than does the 120-µS curve. But with increased record treble boost, there is increased risk of tape saturation, which results in distortion and impaired treble response. One could reduce the risk of tape saturation by lowering the signal level recorded on the tape, but this would reduce the signal-to-noise ratio. Alternatively, one could reduce the amount of treble boost needed by lowering the amount of bias current employed in recording. However, less bias entails more distortion. It is therefore necessary to look for a specific playback equalization, and thus a specific record equalization, which together permit an optimum compromise among the conflicting requirements for extended treble response, low noise, and low distortion.

The risk of tape saturation due to record treble boost varies with cassette type; hence, so does optimum playback equalization. The risk is greater for Type I cassettes than for the three other types. Therefore, the industry has taken the position—with which some disagree—that 120-µS playback equalization is optimum for Type I cassettes, and 70-µS for the other three types.

The second reason for standard playback equalization is, of course, to provide compatibility among tape decks. It is highly desirable for a cassette recorded on one deck to provide flat response when played on another. This is possible only if all decks use the same—that is, standard—playback equalization.

Record Equalization

We have seen that for a given cassette type there is a standard playback equalization curve. It is logical to ask whether there is also a standard record equalization curve. Strictly speaking, the answer is no. The industry standard calls for each deck to supply whatever record equalization is required to produce substantially flat response when the deck incorporates standard playback equalization; hence, a standard record curve is not needed. Moreover, it could also be troublesome. For a given cassette type, equalization supplied by the record amplifier tends to differ somewhat from one manufacturer's deck to another's for the following reasons:

- Manufacturers may have different concepts of "substantially flat" response. Those who elect to maintain response to 20 kHz or beyond tend to use more treble boost than those who choose to go only to 16 kHz or so. Returning to Fig. 4, we see that very substantial treble boost is needed to maintain response to 20 kHz; in this example, treble boost reaches about 17 dB at 20 kHz. But the greater the treble boost, the greater the likelihood of tape saturation, with undesirable consequences for treble response, distortion, and noise, as discussed earlier. Therefore, a manufacturer may decide to forgo flat response past 16 kHz or so in exchange for lower risk of tape saturation; accordingly, he will use less record treble boost.
- The next reason has to do with width of the playback head's gap. The narrower the gap, the more extended the treble response in playback. A separate playback head can have a narrower gap, and therefore better treble response, than will a head used for both record and playback. If a deck is capable of extended playback response, it becomes desirable to extend treble boost in recording.

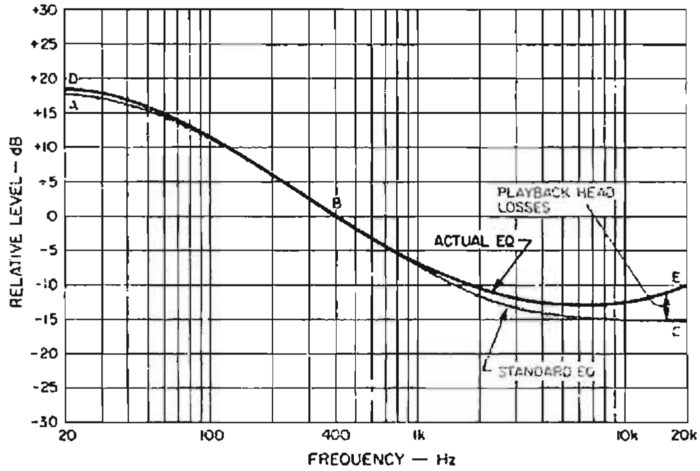


Fig. 5—Comparison of actual Type II playback-amplifier equalization of the CD391 with standard playback equalization, showing slight treble boost to compensate for playback-head losses.

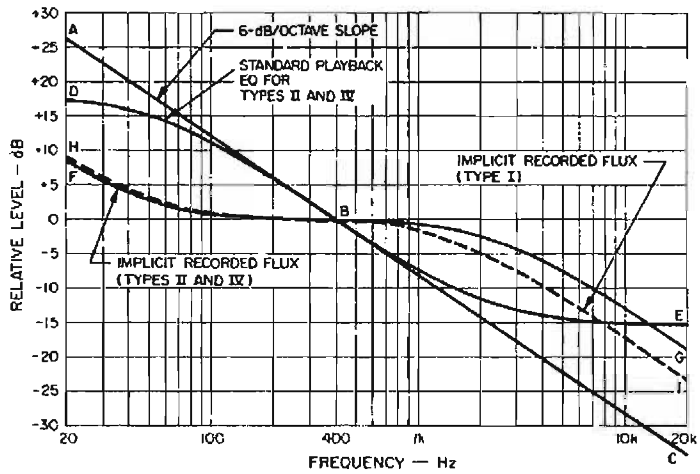
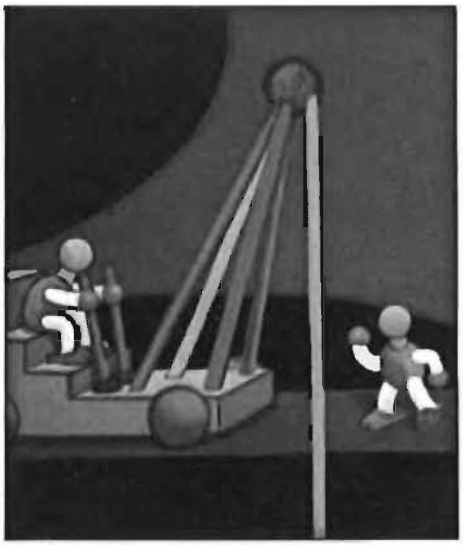


Fig. 6—For any tape type, recorded flux standards are implicit in the difference between the slope of an ideal playback head's equalization requirements (ABC) and standard playback equalization for that tape type. Curve FBG, the implicit standard flux for Type II and IV tapes, is derived from ABC and the 70-µS EQ curve (DBE). Curve HBI, for Type I tapes, is derived from ABC and the 120-µS curve shown in Fig. 3.



There is no standard record EQ, as each deck will supply whatever is required for flat response using standard playback EQ.

• Record equalization will also vary with the amount of bias current each deck manufacturer uses in recording. Bias current affects frequency response, noise, and distortion. As bias is increased, distortion is reduced, but treble response is also reduced. The reduction in treble could be offset by greater treble boost in recording, but this would increase the risk of tape saturation. Such risk could be reduced by lowering the recording level, but this would worsen the signal-to-noise ratio. Thus, the manufacturer seeks that bias current which, in his opinion, achieves an optimum compromise among the conflicting requirements for extended treble response, low noise, and low distortion. The decision as to optimum bias may differ a bit from one deck manufacturer to another; accordingly, so will the record treble boost each manufacturer uses.

• Deck manufacturers choose record equalization with respect to a specific tape. Since tapes of the same type but of different make or variety may differ somewhat in their treble characteristics, record treble boost will also vary.

• Some decks use Dolby HX Pro, which employs the high-frequency content of the audio signal as part of the total bias current. As the treble signal increases, the amount of bias current taken from the deck's bias oscillator is correspondingly reduced, to maintain a constant total bias. Such reduction in oscillator bias somewhat lessens the amount of treble boost needed.

Manufacturers may also differ in their concept of the amount of bass boost to be used in recording. We noted earlier that a slight amount of bass boost in recording is desirable to offset the levelling off of bass boost in playback from 50 Hz down. One manufacturer may elect a substantially full offset in order to maintain essentially flat record-playback response down to 20 Hz or so. Another may decide to maintain essentially flat response less far down, say to 40 or 50 Hz, and therefore will provide less bass boost in recording.

Standard Recorded Flux

Although there is no standard equalization curve with respect to the deck's record amplifier, there is an *implicit* standard with respect to the amount of magnetic flux (signal) recorded on the tape. As we saw in Fig. 1, if a signal were recorded flat (constant magnetic flux on the tape), the response of an ideal head (or a real head with amplifier compensation for its deviation from ideal) would rise 6 dB/octave. In this case, playback equalization would consist simply of a 6-dB/octave treble roll-off (or bass boost).

In practice, however, the playback system's equalization does not fall 6 dB/octave, but follows a standard equalization curve. This implies that the magnetic flux recorded on the tape cannot be constant if flat record-playback response is to be maintained: Magnetic flux must be altered in accordance with the difference between a line falling at a rate of 6 dB/octave with increasing frequency (ABC in Fig. 6) and the standard playback curve (DBE).

Subtracting the standard, 70-µS playback curve (DBE) from ABC, we get curve FBG as the implicit standard recorded flux for 70-µS cassettes. For flat response to be maintained from 20 Hz to 20 kHz, recorded flux must include a bass boost that reaches about 8.5 dB at 20 Hz and a treble drop that reaches nearly 19 dB at 20 kHz.

If this treble loss is permissible (and it is, in fact, mandatory), then it is not necessary to supply enough treble boost in recording to compensate for the entire 36 dB of record losses shown for a typical cassette deck in Fig. 2. Only 17 dB of treble boost need now be supplied by the recording amplifier—and, from Fig. 4, we see that the treble boost supplied by the deck in question reaches 17 dB at 20 kHz. In keeping with the principle that recording losses should be compensated in recording, and playback losses in playback, the remaining 5 dB of the 41-dB record-play treble loss (Fig. 2) is compensated for in playback equalization (Fig. 3).

The dashed curve HBI in Fig. 6 shows the implicit standard recorded flux for Type I (120-µS) cassettes, derived in the same manner as FBG. We've already seen the standard playback curve for Type I, curve DBE in Fig. 3. It is significant to note that in the treble range the recorded flux for Type I cassettes is about 4.4 dB less at 20 kHz than for the other types. Therefore, less record treble boost is needed for Type I.

The reason that recorded flux is only an implicit standard is that measuring it is a difficult laboratory procedure. Therefore, standards are based on playback equalization characteristics, which are more readily measured—and which, as we have just seen, imply standard recorded-flux curves.

Playback EQ and Record Treble Boost

Figure 6 has suggested that less record treble boost is required for Type I tapes than for the other types because of different playback equalization and, therefore, different recorded flux. Figure 7 shows specifically how choice of playback equalization, either 70- or 120-µS, affects the required amount of record treble boost for an actual cassette deck. Curve ABC repeats the unequalized record-playback response of the deck represented in Fig. 2, using Type II tape. In. Fig. 7, curve DBG shows the playback response if 70-µS playback equalization (ABC in Fig. 3) is applied to ABC. The difference between segments BI and BG is the required treble boost when 70-μS playback equalization is used. For example, at 10 kHz the playback response after playback equalization is about −9 dB. Hence, about 9 dB of treble boost is needed at 10 kHz to achieve flat record-playback response. As Figs. 4 and 5 indicate, 7 dB of this treble boost is supplied in recording, and the other 2 dB in playback.

Curve FBE in Fig. 7 shows the playback response that would result if 120-μS playback equalization were applied instead of 70-μS equalization. Less treble loss is now evident in playback so that less treble boost is needed in recording. For example, at 10 kHz the playback response is now only about 4.75 dB down, instead of 9 dB as before. (To achieve utterly flat response, record equalization would have to produce a very slight drop, less than 1 dB at most, between approximately 500 Hz and 5 kHz.)

Looking at the bass end, if 70-μS playback equalization is used, and if flat response is to be achieved, the required bass boost in recording is the difference between segments HB and DB. Such boost would reach about 5.5 dB at 20 Hz. Very similar record bass boost would be needed if 120-μS playback equalization were used instead.

(There is a seeming discrepancy: Fig. 6 indicates that, for flat response all the way down, recorded flux should be about 8.5 dB up at 20 Hz, while Fig. 7 indicates that bass boost of only about 5.5 dB is needed. However, we must recall from our discussion of Fig. 2 above that the playback head in question exhibits about 3 dB boost at 20 Hz, owing to the contour effect. This brings the required amount of record bass boost down to 5.5 dB at 20 Hz.)

Figure 7 shows us something important: How the choice of playback equalization affects the required amount of record treble boost. The greater the playback bass boost, the greater must be the record treble boost. Since Type I tape cannot safely accept as much record treble boost as the other types, 120-μS playback equalization (curve DBE in Fig. 3) is used for Type I, while 70-μS playback equalization (curve ABC in Fig. 3) is reserved for the other types.

Figure 7 reveals another important fact—that the unequalized record-playback response of a deck for a given type of tape does not precisely dictate the equalization needed for essentially flat response. We see in Fig. 7 that two kinds of playback equalization could be used—either 70- or 120-μS. By this token, other kinds could also be used, such as 90 μS, 100 μS, etc. And for each playback equalization characteristic, there would be an appropriate record treble boost.

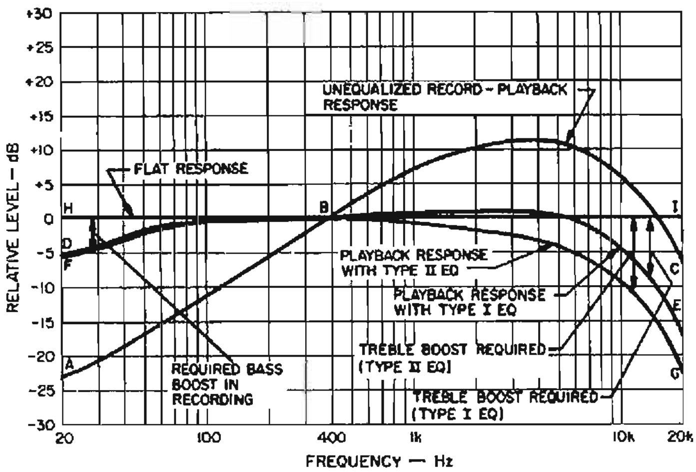


Fig. 7—How choice of playback equalization affects treble-boost requirements. This boost is primarily supplied in recording.

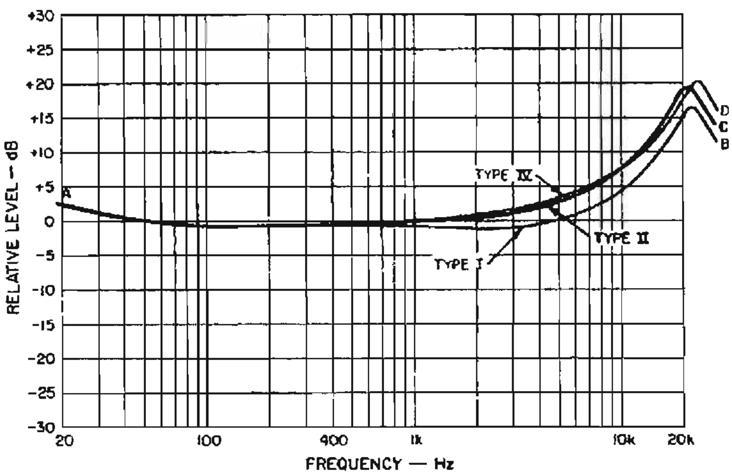


Fig. 8—Record equalization characteristics for an actual cassette deck (the Harman/Kardon CD391).

For the purpose of illustration, Fig. 8 shows the record equalization of an actual cassette deck for Types I, II, and IV tapes. Record treble boost is significantly less for Type I, in large part because of the difference in playback equalization between Type I and the others. (The difference between Type I treble boost and that of the others reaches about 3 dB instead of the 4.4 dB that we might expect on the basis of the difference between segments BE and BG in Fig. 7. This 1.4-dB variance is due to the different characteristics of the various tapes for which the deck in question is adjusted.)

A Controversy About Type II EQ

Some in the industry have questioned whether 70-μS playback equalization is the wisest choice for Type II tape. They would prefer a curve with an appreciably higher time constant than 70 μS, such as 120 μS. In other words, they would like less bass boost in playback and, therefore, less treble boost in recording. The greater treble boost entailed in 70-μS equalization

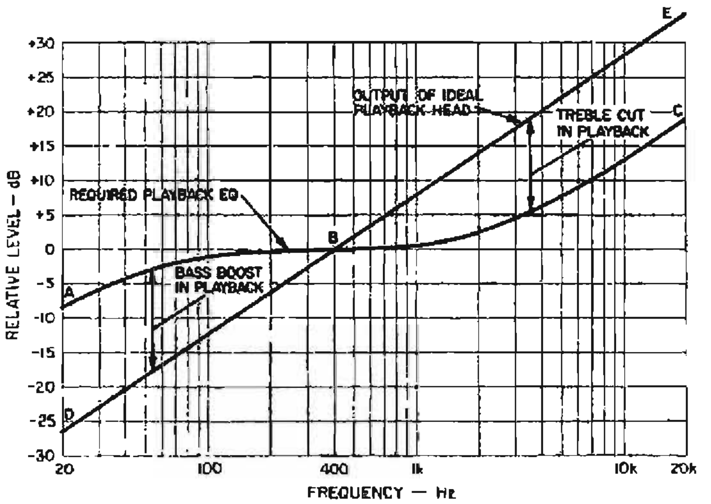


Fig. 9—The industry's revised (post-1965) way of expressing standard playback equalization. Curve ABC is not the equalization to be supplied by the playback system, but rather the response of a properly equalized playback system to the output of an ideal (lossless) head playing a tape recorded with constant magnetic flux. The difference between the ideal head's output (DBE) and curve ABC represents the equalization which the playback system must actually supply Compare curve ABC of Fig. 3.

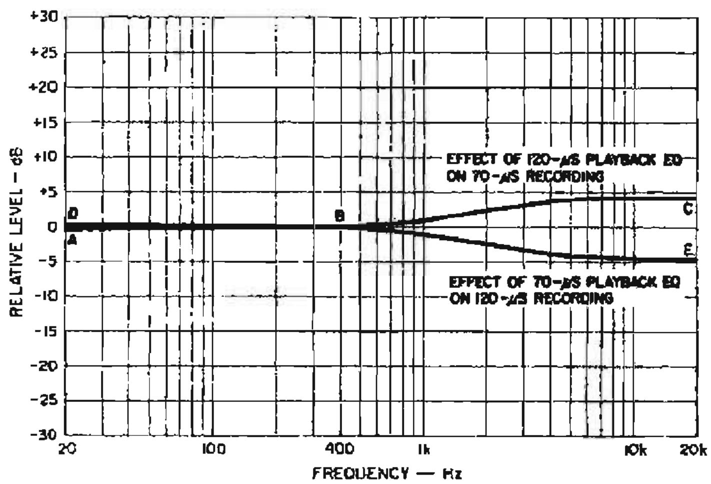


Fig. 10—Effect of playing 120-μS recordings with 70-μS playback EQ, and vice versa.

affords less recording headroom—i.e., less protection against tape saturation and its concomitant distortion and treble loss. They cite the increasing abundance of program material with strong high-frequency content, particularly on Compact Discs and premium phono discs, and the resultant need for adequate headroom. They would be willing to give up several dB of S/N (better with 70-μS equalization) in exchange for several dB more headroom (better with 120-μS equalization).

The controversy may be settled by continuing advances in the state of the art. Modern noise-reduction systems—principally Dolby C and dbx—afford signal-to-noise ratios in the 70- and 80-dB range. The recordist blessed with such S/N and seeking to avoid tape saturation can afford to lower the recording level several dB—say, by about 4.5 dB, which is the maximum difference between the record treble boost entailed in 70- and 120-μS equalization—and still have a very good signal-to-noise ratio. Developments such as Dolby HX Pro and the Dolby C treble recording characteristic reduce the risk of tape saturation.

Confusion About Playback EQ

Until about 1965, the industry expressed standard playback equalization in terms of the frequency contour the playback system should supply—primarily a bass-boost curve, as shown straightforwardly in Fig. 3. However, since 1965, playback EQ has been expressed in terms of the deck's desired output when playing a tape which produces constant flux in the core of the playback head at all frequencies, as shown in Fig. 9. Thus, the actual equalization is *not* curve ABC of Fig. 9, but the *difference* between this curve and the output of an ideal head (the 6-dB/octave slope of curve DBE). A graph of this difference would reproduce curve ABC of Fig. 3.

Effectively, the curve labelled "required playback EQ" in Fig. 9 is the inverse of that in Fig. 3, tilted to reflect the fact that it is referenced to a 6-dB/octave slope instead of the horizontal line of flat response. This has led to a frequent misunderstanding—that playback equalization consists largely of treble boost instead of bass boost.

Interchanging Equalization

Many readers have inquired about the effects of using equalization settings other than those normally recommended for specific tape types. The most obvious effects are slight alterations of treble response when playback and record equalization are mismatched. If a tape intended for 70-μS playback is played back with 120-µS EQ, a slight treble boost will be heard. If a recording intended for 120-µS playback is played back with 70-µS EQ, there will be a slight treble cut.

In theory, this cut or boost will reach 4.4 dB at 20 kHz, as shown in Fig. 10. In practice, however, the difference may be more on the order of 2.5 to 3 dB, as indicated by the record equalization curves of Fig. 8. The differences between actual and theoretical treble response could be explained in terms of differences in actual equalization curves supplied by the deck in question, or differences in the treble response and saturation characteristics of various tape formulations, especially if the tape in use is not the tape which the deck manufacturer used in adjusting record equalization characteristics.

What if Type I tape were both recorded and played back with Type II or IV equalization, or Type II or IV tape were recorded and played back with Type I EQ? In each case, the treble changes in recording would just about balance those in playback, so overall record-playback response would be only slightly affected.

However, we must keep in mind that signal-to-noise ratio would always be affected, with 70-µS playback equalization producing less noise (higher S/N). After all, that is the reason for using 70-µS instead of 120-µS playback equalization.

It should further be kept in mind that use of Type II or IV record equalization with a Type I tape will increase the risk of tape saturation, and therefore of distortion and loss of extreme treble, unless the recordist deliberately reduces the recording level by several dB.

How Open-Reel Decks Compare

To round out the discussion of our subject, let's compare open-reel tape deck equalization with cassette equalization. Standard playback equalization for open-reel decks follows the five basic principles listed near the beginning of this article. The playback curves for open-reel look much the same as for cassette, save for differences in the upper turnover frequency and reduced need for playback treble boost to compensate for playback-head losses. In all cases, the lower turnover frequency remains 50 Hz (3,180 µS). For speeds of 1⅞, 3¾, 7½ and 15 ips, there are official standards for conventional (ferric oxide) tape and de facto standards for EE (extra efficiency—akin to Type II) tape.

Table I—Upper turnover frequencies and total bass boost for standard playback curves. The lower turnover frequency is 50 Hz (3,180 µS) in all cases.

	Upper Turnover Frequency (f_2)	Total Bass Boost
Cassettes, Type I (1⅞ ips)	1,326 Hz (120 µS)	28.5 dB
Cassettes, Types II, III, IV (1⅞ ips)	2,274 Hz (70 µS)	33.1 dB
Open Reel, Conventional Tape		
1⅞ ips	1,326 Hz (120 µS)	28.5 dB
3¾ ips	1,768 Hz (90 µS)	31.0 dB
7½ and 15 ips	3,183 Hz (50 µS)	36.1 dB
Open Reel, EE Tape		
1⅞ ips	2,274 Hz (70 µS)	33.1 dB
3¾ ips	3,183 Hz (50 µS)	36.1 dB
7½ and 15 ips	4,547 Hz (35 µS)	39.2 dB

As stated earlier, record and treble losses become less severe as tape speed is increased. Therefore, less record treble boost is needed at higher speeds in order to achieve a given amount of recorded magnetic flux. Or, for the same treble boost as before, one can achieve more recorded flux; this in turn entails greater playback bass boost, with a consequent improvement in signal-to-noise ratio (see Fig. 6 for the relationship between playback bass boost and recorded flux). In practice, the upper turnover frequency is chosen to afford some of each of the advantages gained from higher tape speed: Somewhat less treble boost, reducing the risk of tape saturation, and somewhat more playback bass boost, resulting in a higher signal-to-noise ratio.

Both for cassette and open-reel decks, and for the various speeds and tape types commonly used, Table I shows the upper turnover frequencies of the standard playback curves. It also shows, for each curve, the total amount of equalization (bass boost) from frequencies above 20 kHz down to below 20 Hz. Total bass boost (equalization) is given by 20 log (f_2 divided by f_1), where f_2 is the upper turnover frequency and f_1 is the lower turnover frequency. If you want to calculate the amount of bass boost at a given frequency, use this equation:

$$B_f = 10 \log \frac{f_2^2 + f^2}{f_1^2 + f^2}$$

where B_f is bass boost at the frequency of interest, f is frequency of interest, f_1 is the lower turnover frequency (always 50 Hz), and f_2 is the upper turnover frequency.

For example, assume we want to know the bass boost at 1 kHz for the playback equalization curve with time constants of 70 and 3,180 µS. First we convert time constants into turnover frequencies by dividing the constants into 159,155 so that f_1 equals 50 Hz and f_2 equals 2,274 Hz. Then:

$$B_{1000} = 10 \log \frac{2274^2 + 1000^2}{50^2 + 1000^2}$$

$$= 10 \log 6.156 = 7.9 \text{ dB}.$$

If you wish to use 400 Hz as the 0-dB reference, calculate the boost for 400 Hz and subtract this from the boost for the frequency of interest, yielding B'_f. For example, B_{400} equals 15.2 dB; subtracting 15.2 dB from 7.9 dB shows that B'_{1000} equals −7.3 dB when 400 Hz is the 0-dB reference. That is, the equalization curve at 1 kHz is 7.3 dB below its level at 400 Hz.

The principles of tape equalization, and their implementation in cassette decks, are complex. Luckily for the tape user, however, one can make excellent recordings without grasping these principles in detail. It is necessary only to grasp the deck's equalization switch, and set it to match the tape that is being used.

DECK-TO-DECK MATCHING AND NR:

STRAIGHTENING THE MIRROR

Every person who has a tape recorder does some editing, and every cassette deck used by a serious recordist includes some form of noise reduction. Substantially all decks have Dolby B NR, and many of the current models have Dolby C NR as well. About 10% of the models listed in *Audio*'s Annual Equipment Directory (October 1985) include dbx NR. The probability is thus very high that an original recording and any edited copy made of it will have some form of noise reduction.

Because the frequency responses of the mastering, the rerecording and the final playback deck all contribute to the end result, it is important that response deviations be kept as small as possible, when they can be controlled. This is especially true when noise reduction is used. The material that follows provides guidelines on how to pinpoint what errors actually exist, determine their effects, and correct or compensate for them.

Real Frequency Response

For a long time, the only way to measure frequency response (more correctly, amplitude response versus frequency) was to set an oscillator to a number of separate frequencies, one by one, reading the output amplitudes on some sort of meter. The development of the VCO-based function generator led to the availability of sources that could be swept electronically from 20 Hz to 20 kHz. Because the better units had very stable amplitude over the entire frequency range, they were popular for evaluating all types of products. Relatively recent test instruments have introduced a noncontinuous type of

Howard A. Roberson

It might not seem logical to decode NR and then re-encode it on retaping, but my tests showed the best results when this method is used.

sweeping: A stepping through many frequencies under the control of a microprocessor.

Unfortunately, for all the appeal and popularity of the current sweeping/stepping sine-wave sources, they should not be used for evaluating the responses of tape recorders, especially if any form of noise reduction is used. The interrelationships among bias, spectral content of the signal, and self-demagnetization effects are such that the response at 10 kHz, for example, will vary with the amount of energy at lower frequencies.

Without noise reduction, these changes are not large, and the response errors are quite small. With noise reduction, however, other factors are introduced which cause much larger response deviations to appear *in the process of evaluating* single decks or an editing system. In past articles and "Equipment Profiles," I have commented that the dbx NR system *must* be tested with a broad-band signal, such as pink noise, so the dbx broad-band detector(s) can sense recording and playback levels properly and set the complementary encoder/decoder gains correctly. Use of a sweeping or stepping signal would generate level variations related to the deck's frequency response errors. These variations would be doubled by the decoder expansion, but the deviations resulting would *not* demonstrate a tracking error in the NR system—they would show the results of using the wrong test stimulus for the device under test. As the computer freaks say, GIGO (garbage in, garbage out).

Because the error that appeared with swept-sine testing of Dolby B NR was small, this system seemed to be immune to any similar sort of false response indication. With the introduction of Dolby C NR, there was—all of a sudden, it seemed—evidence of severe response and mistracking errors with the use of a swept/stepped sine-wave source. The new NR system seemed further damned when it evidenced mistracking at 10 to 15 dB below Dolby reference level (200 nWb/m), even though the response was quite good at −20 dB.

A review of the basic characteristics of both the Dolby B and C NR systems convinced me that a broad-band test

signal was a must for accurate response testing of both versions. However, a serious question remained on the large deviations that appeared when Dolby C NR was tested with pink noise. Figure 1 shows playback responses of three decks, each playing a tape it didn't record. None of the results would be considered acceptable. Was there some sort of measurement error?

After rereading Ray Dolby's paper on the Dolby C NR system, I concluded that I should take a better look at the spectral characteristics of music. Dr. Dolby's paper clearly stated that filtering was included to eliminate the problem of differences in high-frequency responses, but normal pink noise is not that close to music, spectrally. Figure 2 shows the range of maximum levels measured over a period of time in the 10 standard octave bands during a portion of an audiophile recording of "Night on Bald Mountain" by Mussorgsky. In this 'scope photo, take particular note of the fact that the highest levels in the upper frequencies roll off at about 5 dB/octave, from the 1-kHz band on up. The pattern shown is matched closely by most types of music, whether recorded or live.

There does exist, therefore, a very valid reason for shaping the pink-noise signal to make it more music-like. This will yield a more accurate indication of the actual mistracking between the Dolby sliding-band compressing encoder and expanding decoder in the real world of music and tape recorders. The first version of what I call "PN/Music" has a spectrum as shown in the bottom trace of Fig. 3. The roll-off was obtained with the combination of a 1½-octave filter having a 14-dB cut centered on 19 kHz, and a low-pass cutoff filter at 20 kHz. For convenient display analysis, the playback from the deck was given a complementary boost (top trace) by trimming the filter's shape and position to get the flat total response shown in the center. The roll-off frequency chosen is somewhat higher than that of the music sampled in Fig. 2, which makes the test stimulus a bit more challenging than the music sample and closer to what some synthesizers might do. The energy in the 20-kHz band, however, is 15 dB below what it would be without the roll-off—*much* closer to the actual relative levels in music.

Figure 4 shows the results of four test conditions with a high-quality, three-head deck. The top trace is the measured output with the monitor switch on "Source." The second trace is the record/playback response with Dolby C NR at −20 dB, with the pink noise extending to 50 kHz. It would be a mistake to regard what is shown as severe tracking error, for, as seen in the third trace, cutting the noise off at 25 kHz improved the response dramatically. There were relatively minor deviations in response over a range of levels from −25 dB to Dolby level. The bottom trace was secured using PN/Music. Its shape stayed *exactly* the same from the bottom-level noise limit to almost "+10" on the deck's meters, showing that the tracking was actually outstanding—something that would have been obscured if a non-music-like test signal had been used.

Record on One Deck, Play on Another

For the following investigation, a total of seven cassette decks were used. For convenient reference, I assigned numbers to each: Nakamichi BX-300,

#1; Revox B215, #2; Akai GXR99, #3; Technics RS-B48R, #4; TEAC V-500X, #5; Aiwa AD-M700, #6; and Nakamichi 582, #7. This collection represented a wide range of price and performance. All of the decks have Dolby B NR, decks #1 to #5 have Dolby C NR as well, and decks #4 and #5 also have dbx NR. To designate recording on one deck with playback on another, I will use abbreviations; recording on deck #2 with playback on deck #4, for example, will be shown as "R2/P4."

In general, tests were run using the NR systems built into the decks. To pin down the cause of some of the effects observed, I used two outboard processors, a Nakamichi NR-200 for Dolby NR tests and a dbx 224 for dbx checks.

The first task was to ascertain the record/playback responses without NR. Figure 5 shows the results with decks #1 to #5: The responses are all quite flat, but some differences exist. Figure 6 shows what happens when recordings made on deck #1 are subsequently played on decks #1 to #5 (R1/P1 to R1/P5). Similar checks were made of R2/P1 to R2/P5, R3/P1 to R3/P5, and R4/P1 to R4/P5. Collectively, they told this story: Deck #1 showed a high-end boost when playing back tapes from other decks, particularly #2 and #3. Deck #2 played the tape from #1 quite flat, but showed a similar boost with the tape from #3 and a general high-end rise with the tape from #4. Deck #3 played tapes from #2 and #3 very flat, but had a slight droop with the tape from #1 and a roll-off with the tape from #4. Deck #4 showed a sharp roll-off at the high end with all tapes, and had a noticeable droop with the tape from #1.

The differences described here are primarily from the various versions of playback equalization as actually used in the decks. The boosts and roll-offs are *not* from the very small discrepancies in azimuth alignment that existed during the tests. This is the first area of concern to be introduced: Even without NR, the flatness of the playback can be affected by differences in playback equalization, and perhaps azimuth alignment, from one deck to another. With NR, these response deviations can be expanded by action of the decoder. To minimize such effects

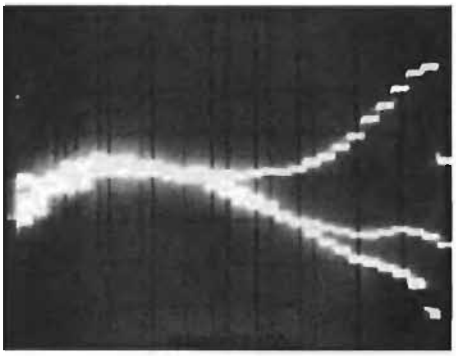


Fig. 1—Record/playback responses using pink noise with Dolby C NR. Deck #1 playback of tape made on deck #4 shows large high-end boost (top trace). Deck #3 playback of a tape from deck #2 has large droop at high end (middle trace). Deck #4 playback of a tape from deck #5 has even more droop, plus roll-off (bottom trace). (Vertical scale for this and all other figures: 5 dB/div.)

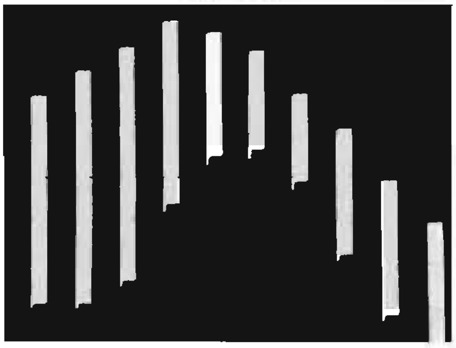


Fig. 2—Range of octave-band maximum levels during a portion of a recording of Mussorgsky's "Night on Bald Mountain."

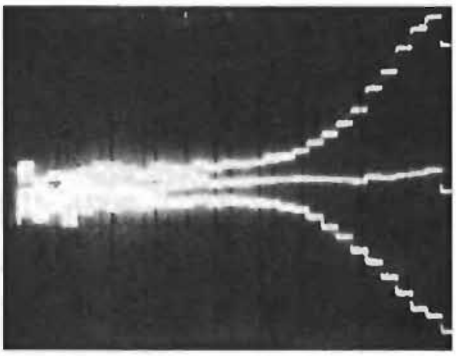


Fig. 3—Shaped pink-noise spectrum to make "PN/Music" (bottom trace), equalization added after tape recorder's playback to make flat RTA display with flat record/playback responses (top trace), and result without recorder in loop (middle trace).

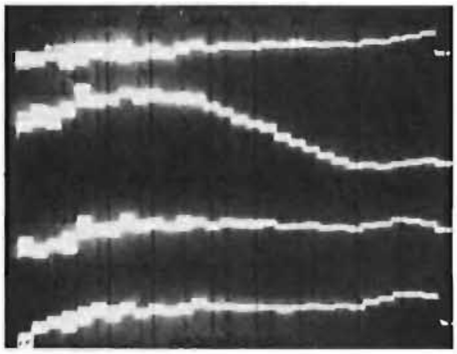


Fig. 4—Four test conditions using a high-quality, three-head deck. From top to bottom: Monitor switch on "Source," record/playback response at −20 dB with pink noise from generator extending to 50 kHz, record/playback with 25-kHz low-pass filter on noise generator's output, and record/playback with PN/Music. All record/playback responses were made with Dolby C NR.

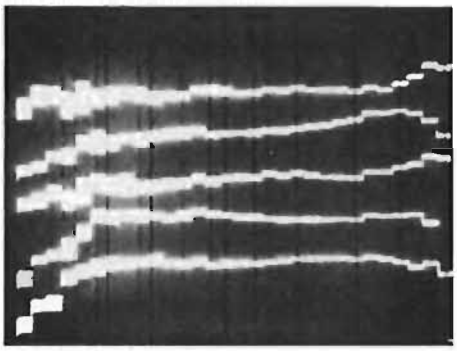


Fig. 5—Record/playback responses without NR for decks #1 through #5 (top to bottom); see text.

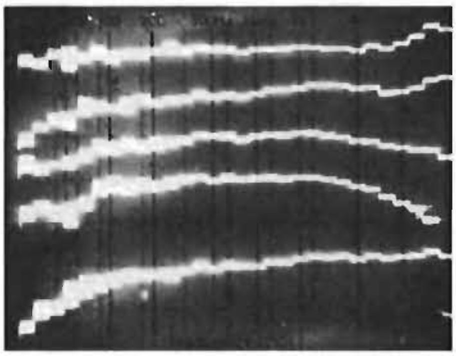


Fig. 6—Responses of tape recorded on deck #1 played back on decks #1 through #5 (top to bottom).

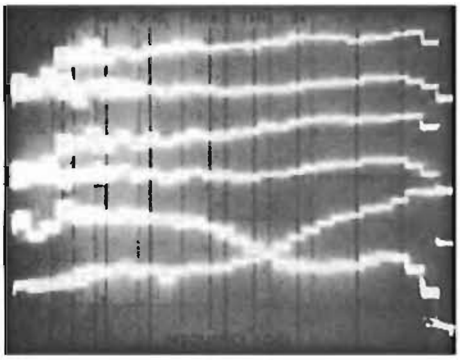


Fig. 7—Record/playback responses with decks #6 and #7. From top to bottom: R6/P6 without NR, R7/P7 without NR, R6/P6 with Dolby B NR, R7/P7 with Dolby B NR, R6/P7 with Dolby B NR showing a rising high end, and R7/P6 with Dolby B NR showing a high-end droop.

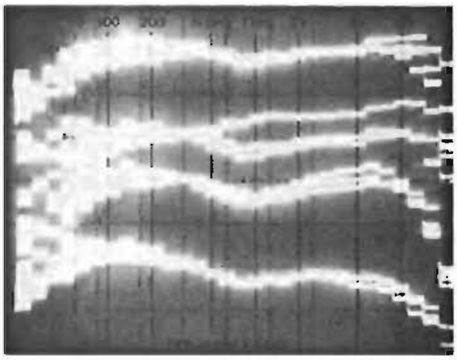


Fig. 10—Dolby B NR responses using Nakamichi NR-200 processor. From top to bottom: R3/P3 to R6/P6, overlaid; R3/P4 to P6; R4/P3, P5, and P6; R5/P3, P4, and P6.

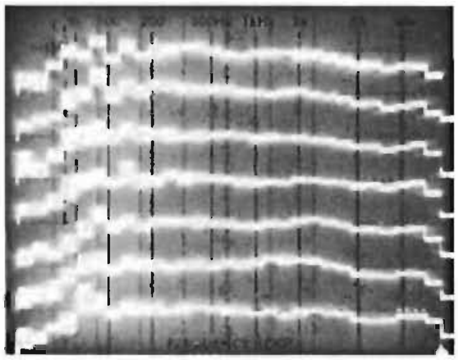


Fig. 13—Effects of decoding and re-encoding using Dolby B NR. Traces are of R7/P7 feeding R6/P6 at levels from −25 to +5 dB. See text.

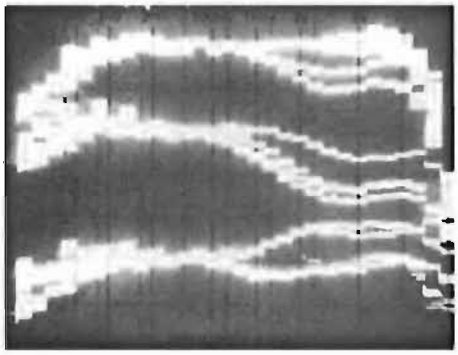


Fig. 8—Responses, from top to bottom, of R4/P3, P5, P6, and P7; R5/P3, P4, P6, and P7; R3/P4 to P7. See text.

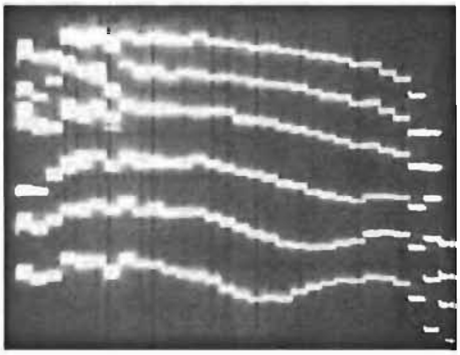


Fig. 11—R7/P6 with Dolby B NR, using Nakamichi NR-200, from −25 to +5 dB.

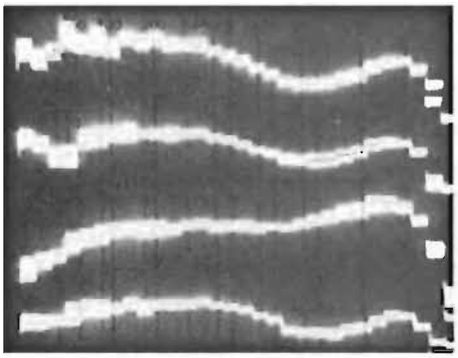


Fig. 9—Effects of multiplex filter on Dolby NR mistracking. Each trace shows overlaid responses of tapes recorded on deck #7, with and without MPX filter, played back on decks #6, #4, #5, and #3 (top to bottom).

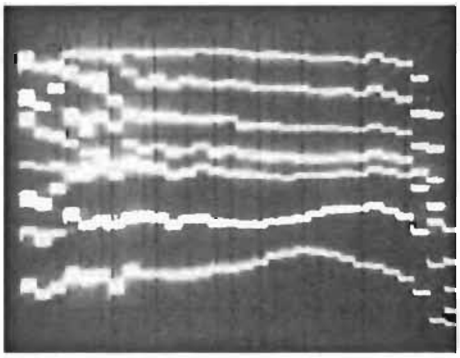


Fig. 12—R7/P6 with Dolby B NR, using Nakamichi NR-200, from about −20 to +5 dB with +4 dB play calibration offset.

when editing tapes, careful deck selection can well be in order.

What Happens with Dolby B NR

Figure 7 shows the results of interchanging tapes between decks #6 and #7. Both decks' record/play responses (R6/P6 and R7/P7) are very good, with or without Dolby B NR, but the R6/P7 and R7/P6 responses (with NR) are very poor. I should point out that all responses were measured with PN/Music, which would ensure the best test results. With deck #7, it was possible to adjust record sensitivity to match Dolby level, but that was not possible to do with deck #6. The latter also has a drooping playback response compared to deck #7. This does appear to be a bad combination for Dolby B NR recording on one deck and later playback on the other.

Figure 8 shows the results for several deck-to-deck crosses: A tape recorded on deck #4 played back on decks #3, #5, #6, and #7; a tape from deck #5 played back on decks #3, #4, #6, and #7; and a tape from deck #3 played back on deck #4 through deck #7. The tape from #4 worked well in decks #5 and #7, and the tape from #3 had just a slight saddle in the mid-highs when played on decks #4 and #6. The rest would be considered marginal or worse.

The tape from deck #7 was then played back on decks #3 to #6 (Fig.

Even without NR, playback flatness can be affected by differences in playback equalization, and perhaps azimuth alignment, from one deck to another.

9), after recording with and without the multiplex filter. The very minor differences demonstrate that use of the filter will not solve the problem.

The next tests used the Nakamichi NR-200 as the Dolby B NR processor for decks #3 to #6. This permitted setting up each deck to match the same NR unit, thus removing the question of Dolby NR differences from deck to deck. I must emphasize that this procedure does *not* eliminate all possible errors in making play- and record-level calibrations. They *are* minimized, but the two-head decks (#4 and #5) and the 1-dB meter steps of the NR-200 made exact level matching among decks close to impossible.

The topmost trace of Fig. 10 shows the overlaid record/playback responses of decks #3 to #6. There is some spreading at the low end and differences at the high end, but they are quite close in general. The other traces, from top to bottom, show results for R3/P4 to P6; R4/P3, P5, and P6; and R5/P3, P4, and P6. These results *are* better than those in Fig. 8, where the individual decks' NR systems were used, but the responses are disappointing in showing that effects remained from level and/or response differences.

I mentioned earlier that I could not be certain how well I had matched calibration levels, even when using the NR-200 as the common NR system. Figure 11 shows the Dolby B NR responses for R7/P6 over a range of levels from about −25 to +5 dB. Mistracking is certainly in evidence, and there is a general roll-off across a good part of the band at all levels. I then adjusted "Play Cal" on the NR-200 to make the −20 dB response of R7/P6 look the flattest, and reran the range-of-levels test. Figure 12 demonstrates the great improvement in responses over the entire level range with the "Play Cal" offset. A separate check showed that I had increased the play level by +4 dB. I had not expected that this adjustment would be so effective. In addition, I was surprised at the large adjustment needed to compensate for both calibration errors and response effects on overall level to make deck #6 track properly when playing tapes made on deck #7.

All of the above has been predicated

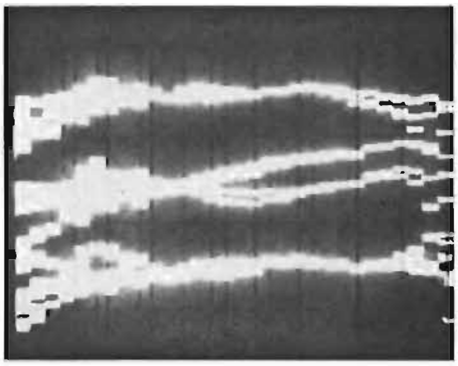


Fig. 14—Overall record/playback responses using each deck's Dolby C NR circuits. From top to bottom: R1/P2 to P5; R2/P1, P3, P4, and P5; R3/P1, P2, P4, and P5.

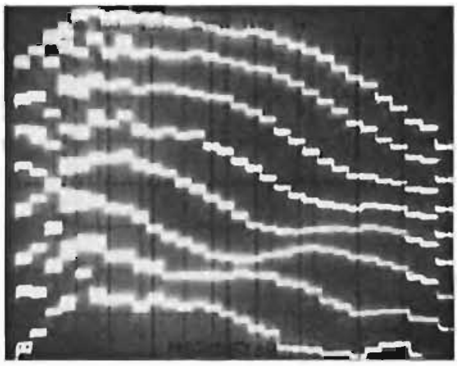


Fig. 16—R5/P6 with Dolby C NR, using Nakamichi NR-200, from −30 to +5 dB.

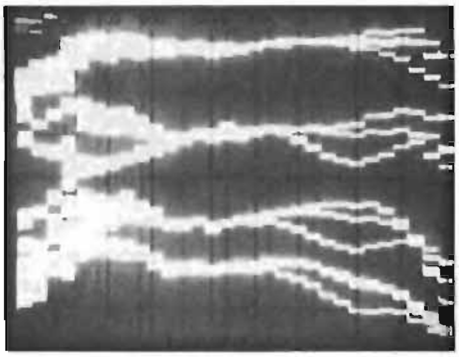


Fig. 15—Record/playback responses with Dolby C NR using Nakamichi NR-200 processor. From top to bottom: R3/P3 to R6/P6, overlaid; R3/P4 to P6; R4/P3, P5, and P6; R5/P3, P4, and P6.

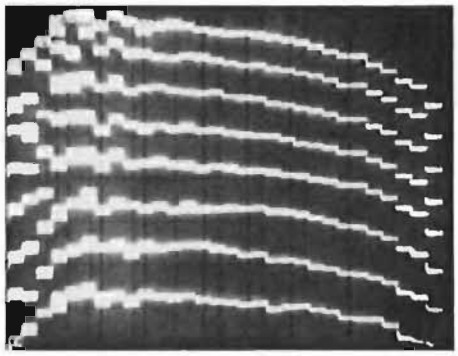


Fig. 17—Same as Fig. 16 but with play calibration offset +5 dB.

on the condition that the ultimate playback deck is not available for the editing and on the assumption that there is no advantage in decoding a Dolby B NR tape just to encode it again in making a copy. In other words, we should copy the Dolby B NR tape with NR out—or should we? It seems to make sense to avoid decoding and re-encoding, but the previous tests have shown many poor results from doing just that. Figure 13 shows the result of R7/P7 as the source for R6/P6, all with Dolby B NR. With this combination, P7 was decoded and R6 was re-encoded; final decoding was done with P6. The range in levels is from −25 to +5 dB, with excellent tracking in all respects. A comparison with Fig. 7 demonstrates

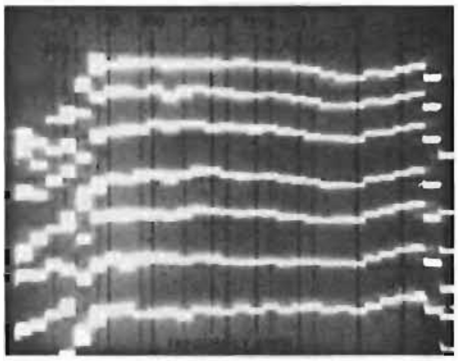


Fig. 18—R4/P4 with that playback feeding R3/P3, with both decks using Dolby C NR, over a range from −25 to +5 dB.

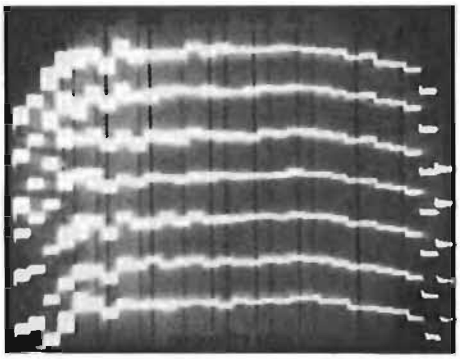


Fig. 19—R6/P6 with dbx NR, using dbx 224 processor, from −25 to +5 dB.

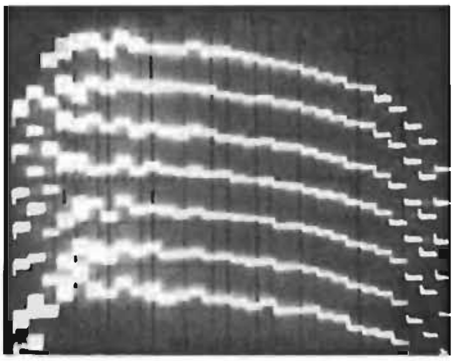


Fig. 20—Same as Fig. 19 but for R5/P6.

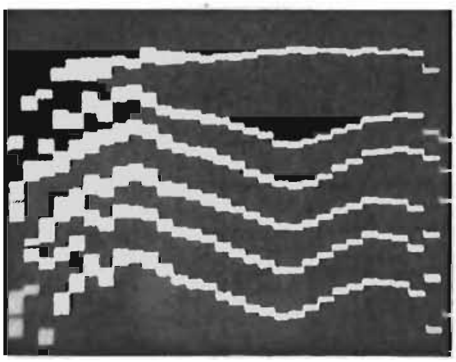


Fig. 21—From top to bottom: R6/P6 using dbx 224, R6/P6 with non-flat equalization before the dbx encoder, R7/P7 with P6 as the input, P7 back to R6/P6, P6 back to R7/P7, and then P7 back to R6/P6 again, making a fifth-generation copy.

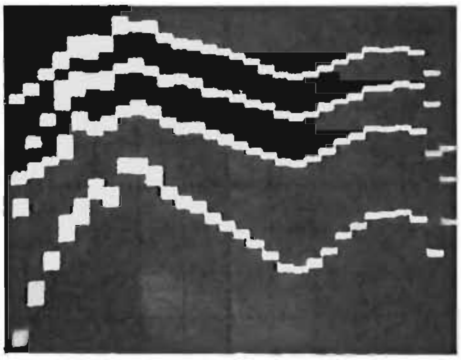


Fig. 22—Buildup of recorder response errors. From top to bottom: Test signal with same equalization as in Fig. 21, R6/P6 with this equalization inserted after the dbx 224 encoder and before the deck's input to simulate recorder response errors, R7/P7 without further equalization and using P6 as the source, and R6/P6 with additional equalization between encoder output and deck input using P7 as the source.

the great improvement gained by doing what seemed unnecessary.

If at all possible, you should make your edited copy on the ultimate playback deck with NR switched in for both decks. Even if the final-play deck is not available to you, follow the same plan, using NR on both decks involved in making the copy. If you are using a double-well editing deck, make certain you know what you can and can't do as far as NR is concerned. The probability is that you will *not* be able to decode the tape you're playing and re-encode the copy being made, which would be preferable. You may have no choice but to copy the tape in its encoded form, in which case you can try different tape formulations to see which gives the best final result. It would probably also help to match recording levels so that copies of Dolby-level tones would also be at Dolby level; this may take some experimenting.

To summarize what has been shown: Because of differences in calibration levels and recorder responses, both playback and record/playback, there is likely to be some Dolby B NR mistracking when a tape made on one recorder is played back on another. The best plan when making an edited copy is to decode the original on the playing deck and re-encode it on the recording deck. It is possible to make considerable improvements in the playback of a Dolby B NR tape recorded on another deck by using an outboard processor and adjusting the play calibration level. An increase in this level will bring up the drooping mids and highs; a decrease will pull down a high-end exaggeration or reduce extra presence. This will be discussed further, after a look at Dolby C and dbx NR.

And Then Came Dolby C NR

Considering some of the poor results I've obtained from testing in months past, I was really interested to see how Dolby C NR would perform using PN/Music. Figure 14 shows R1/P2 to P5; R2/P1, P3, P4, and P5; and R3/P1, P2, P4, and P5. The majority of these responses are really quite good, with the exceptions of R1/P4, which had a sharp roll-off, and R2/P1 and P5, which showed elevated mids and highs. Next, the NR-200 was inserted, with Dolby C NR selected, to replace the NR systems for decks #3 to #6. The record/play responses of each deck were quite good (Fig. 15, top trace), but the playbacks on other decks were not to be applauded.

To get a better feel for what was happening, I used a range of levels from −30 to +5 dB for R5/P6 using the NR-200. Figure 16 displays obvious mistracking and also a severe roll-off at the high end. I went back to "Play Cal" and increased the play level by 5 dB. There was still a general roll-off with increasing frequency (Fig. 17), but the response shape was very consistent over the range of levels. Fairly simple equalization, even a little treble from a tone control, would bring the response to within ±1 or ±2 dB over most of the audio band. Further comments on level calibration, responses, and tracking will be made later.

Figure 18 demonstrates the improvement possible with Dolby C NR if there is decoding with the play deck and re-encoding with the editing deck. The combination used was R4/P4 with that playback feeding R3/P3, over a range from −25 to +5 dB. Compare the flatness of these results with the third set of traces in Fig. 15, where the same combination of recorders had a

If at all possible, make an edited copy on the deck which you ultimately will be using for playback, with noise reduction switched in for both decks involved.

boost at 5 kHz and a severe roll-off above that when there was no decoding and subsequent re-encoding between recorders.

The above results illustrate that with Dolby C NR, as with Dolby B NR, the final playback will most likely have the best response if the copy is made on the ultimate play deck. Level calibration adjustments with Dolby C NR also showed beneficial corrections of response deviations from mistracking.

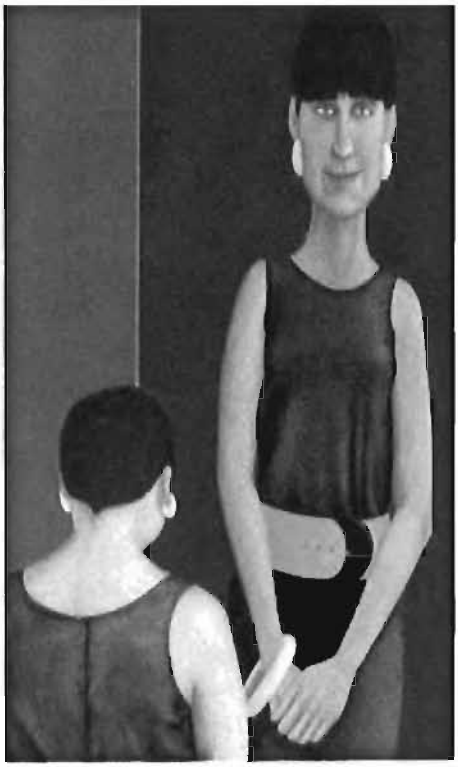
Editing with dbx NR

Although most decks do not have dbx NR, a fair percentage of audiophiles who have more than one deck probably have that capability, and some prefer to use it when editing. Also, among people who have small studios or who are involved in multitracking for other reasons, dbx is the NR system in widest use, especially if TEAC equipment is employed.

Figure 19 shows R6/P6 using the dbx 224 processor over a range of levels from −25 to +5 dB. It is obvious that the response droops somewhat at the high end, but there is no change in response shape over the entire range. A slight reduction in bias or a small boost with external equalization would make all of these responses very flat. Figure 20 covers the same range of levels but for R5/P6, also using the dbx 224 processor. The response shapes are very consistent, but the roll-off from the low to the high end is too large to be acceptable. It is true, however, that external equalization would gain a flat response at all levels.

Other checks showed that the dbx system of deck #5 tracked the 224 within ±1 dB across the band over the entire range of levels—in fact, it was usually within ±0.5 dB. The dbx circuits of deck #4, a less expensive unit, did not match the 224 as closely, but all points at all levels stayed within ±1.5 dB. The fundamental cause of the falling responses in Fig. 20 was the lower playback equalization of deck #6 compared to deck #5. This demonstrates the importance of flat recorder responses *with the same playback equalization* for best results when moving tapes from one deck to another.

I would like to emphasize that the dbx system does *not* multiply frequency response deviations in what is being recorded. Figure 21 provides proof of this assertion. At the top is R6/P6 using the 224 processor. Next is R6/P6 with equalization inserted *before* the encoder; in other words, a non-flat response was recorded. The third trace shows R7/P7 with the playback output from deck #6 as the input. In the fourth trace, deck #7's playback is the input for R6/P6 again, which goes to R7/P7 again (fifth trace), and finally to R6/P6 (bottom trace). The external equalizer, of course, was used only for the initial recording. There are some slight changes in the responses, but the *fifth*-generation playback using dbx NR was *very* close to the first playback, and careful examination is required to find any differences.



Deviations in the response of the recorder, however, *do* cause errors. The deviations are multiplied in rerecording, such as when bouncing tracks, but they are *not* multiplied by dbx NR. This is shown in Fig. 22. The topmost trace is of the test signal, with the same equalization as used in Fig. 21. The second trace is of R6/P6; the equalization was applied after the dbx 224 encoder—that is, within the encode/decode loop. Note that this signal is, nevertheless, still very similar to the topmost response. The third trace is R7/P7, with the second trace (decoded output of P6) as the input signal; no NR or equalization was inserted. Up to this point, the playback looks like the original input, with the addition of the decks' response variations.

In the bottom trace, deck #7's output (the third trace) was rerecorded on deck #6 with dbx NR, and the equalization was inserted once again between the encoder and the input of the deck. As the trace shows, the deviations are double those in the response itself—but this is due to the double use of the equalization. The purpose of the equalization was to simulate a recorder having poor response; the dbx NR has added nothing further to the deviation. When Dolby NR is used, however, response discrepancies could cause some mistracking due to overall level shifts.

The point here is that flat recorder response is even more important in multi-tracking with repeated playbacks and rerecording, where response errors will build up. This is true with or without NR.

Guidelines for Good Editing

Considering the chamber of horrors presented above, the recordist could be quite convinced that avoiding any form of NR is the best course. In fact, if a high signal/noise ratio is not needed, such as with a fair amount of rock music that exhibits considerable compression, editing without NR (except as needed to decode an encoded original) will ensure that there cannot be any response-error multiplication in playback. I suggest marking any tapes recorded without NR to make it clear that NR should not be used in playback.

In your general approach to making recordings, especially originals, make certain that all heads are aligned accurately. Select recording tape that gives the flattest response at −15 to −20 dB both with and without NR, and trim bias, if possible, for the best overall flatness. If you do not have test equipment, rely on listening tests. Checking the "before" and "after" sound quality using FM interstation noise is a good method if you don't have a pink-noise generator. Very few decks these days have record-sensitivity adjustments; consider yourself fortunate if your deck

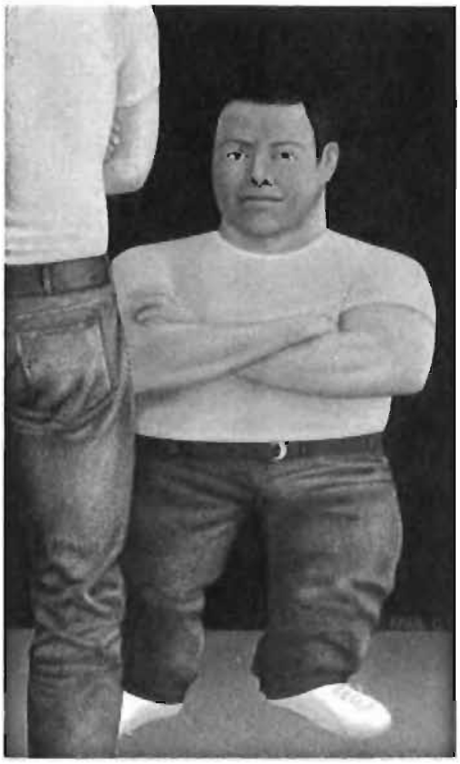
Accurate head alignment, proper bias adjustment, and matching the tape's record sensitivity are required for original, playback, and editing decks.

has that facility, either manual or automatic. Matching record sensitivity is essential for getting good Dolby NR responses, but the matching is not important with dbx NR.

Some manufacturers state which specific tape formulation was used to set up a particular deck, and that is usually a reliable choice for making an original recording. If a long list of tapes is given in the manual, the information is virtually worthless unless the deck has automatic calibration. One way to select a tape for correct record sensitivity is to record a tone of 400 Hz or so at zero level on the meters, rewind the tape, and check the meter levels in playback. Many decks are set up quite well, and there is a good chance that the tape which shows the same level in record and in playback is the one with the correct sensitivity. This test cannot be run with a noise source at a high level because saturation effects at the high frequencies will cause level discrepancies to appear in playback.

The requirements for the editing deck are the same as for the original/playback deck: Aligned heads, adjustment of bias for the best response, and record sensitivity to match the tape used. Tape selection, of course, can secure excellent bias and sensitivity matching in quite a few cases. If your source is not another tape, then the deck you are recording on is the original as far as taping is concerned. In any event, the recordist is faced with the challenge of making a tape that will perform well on another deck, without that deck being available for checking such things as head alignment or Dolby play-level calibration. And this says nothing about the playback response of that deck.

If the deck that will ultimately be used for playback *is* available, record your final, edited tape on it. If your source is another tape deck, it should be set for whatever NR decoding is appropriate to the original tape. The signal should be re-encoded by the deck you are recording on, even if the same NR system is used. As the tests described above have emphasized, flat responses are a must for all encode/decode NR systems, and level calibration is a further requirement for the two Dolby NR systems. If the deck used for final playback is not available



and it is a premium deck of recent vintage, the editing deck should be similar, primarily to match the playback equalization. If the final-use deck is of medium quality, it is less significant what the editing deck is. If the quality of the final-play deck is unknown, or you will be editing for decks with different requirements, consider acquiring a deck of recent vintage and of medium to medium-high cost.

If you must make copies of tapes that are already NR-encoded, I recommend that you get a separate processor (or two) with record and playback level adjustments—useful with dbx and close to essential with Dolby NR for the best results. Purchasing a Dolby-level calibration tape will be necessary to get the most out of the processor, and that's a good thing to do even if you just want to check the play/meter calibration of your decks.

If the tapes to be edited have Dolby NR, consider the possibility that you may improve the overall sound by changing play calibration to get better flatness and tracking. If possible, have the person who supplied the original tape put a bit of pink noise on one end of it. The noise should be limited to 20 kHz and recorded at −20 dB, using the same NR as the rest of the tape. By ear or—even better—with an RTA, you will be able to use the processor's play calibration to flatten the response and then equalize it to make it really flat. Verify your choices by listening to the recorded material: Raise the play calibration level to get rid of dullness and to fill in the middle, and reduce the play calibration level to bring down exaggerated highs and excessive presence. If everything sounds fine, don't change anything.

It may be helpful to remember this fact: With a single recording and subsequent playback on any other deck, none of the NR systems discussed here will generate uncorrectable response errors—at least if level (for Dolby NR) and equalization adjustments can be made between the processor and the deck(s). For example, if record sensitivity is too high with Dolby NR on the recording deck, it can be compensated for by lowering the play calibration level of the processor fed by the playback deck.

Nakamichi and Tandberg decks include sharp cut-off filters above 25 kHz to help remove the possibility that above-band energy will cause Dolby NR tracking problems. With other decks, the user might want to add a low-pass filter if there's any doubt, such as when recording a synthesizer. Because the multiplex filter's response rises between 19 and 38 kHz, it might not prove to be effective enough for this purpose.

Summary

With all of the decks under your control, try to have record and playback heads accurately aligned. Accurate tape-to-deck matching is essential to make the responses as flat as possible for both Dolby and dbx NR and to get the right record sensitivity for good Dolby NR tracking. When copying a tape with NR, decode it on the playing deck and then re-encode it on the editing deck for the best overall results. Remember that differences in playback equalization can have a noticeable effect on the response in final playback. Flat record/playback responses are especially important when rerecording a number of times. Be wary of possible measurement errors if you make your own tests. And finally, good luck and good listening!

EQ & NR: STRIKING A BALANCE

Most recordists stick to standard equalization settings to match the tape type being used: 120 µS for Type I (normal) tapes, and 70 µS for Type II (CrO_2) and Type IV (metal) tapes. In fact, the majority of tape decks set EQ automatically, decoding the holes in the edge of the cassette. Some decks do, however, permit the user to switch-select the EQ, sometimes independently from the bias level. There are possible advantages in being able to do this: Using 120-µS EQ for higher high-frequency MOLs (but higher noise) or using 70-µS EQ for lower noise (but lower high-frequency MOLs). This relationship is based, of course, upon using the same EQ setting for both recording and playback.

You can't necessarily assume that tapes made with noise reduction will always be played back with that same NR mode. Actually, it's rather common for a tape made with NR, especially Dolby B NR, to be played back on another recorder, such as an inexpensive portable, that does not have Dolby B NR. The sound usually will be a bit too bright, but that's not bad, particularly when compared to what happens when tapes encoded with Dolby C or dbx NR are played back without decoding. And if the NR systems are different, what then? And with an EQ change?

120- vs. 70-µS Equalization

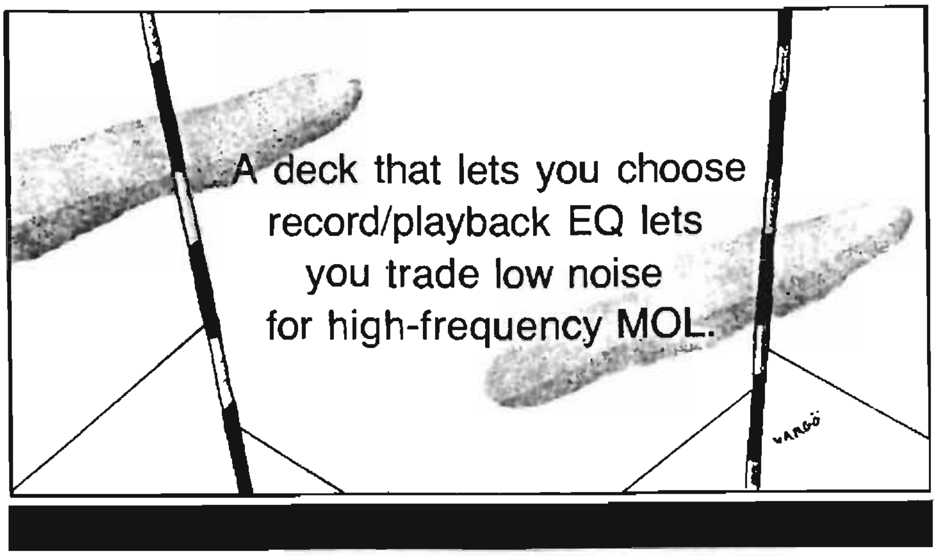
Using the same equalization for recording and playback will yield a flat frequency response, provided the deck is well matched to the tape. This is true for either 120- or 70-µS EQ, as can be seen in Fig. 1. When 70-µS playback equalization is used with a tape made for 120-µS playback EQ, however, there is a large droop in the response above 1 kHz (top trace). From the same relationship, there is a large boost (middle trace) when 120-µS playback EQ is used with a tape needing 70-µS EQ. The addition of Dolby C NR in the first case (bottom trace) causes a doubling of the droop, and the great loss in high-frequency content is very audible.

The conclusion at this point is that the playback EQ might be changed on purpose to bring down excessive brightness (120 to 70) or to bring up a dull high end (70 to 120), *but* if Dolby NR is involved, the effect will be much more exaggerated. The change with Dolby B NR would be less than what is shown in the figure, but the recordist should be aware of the possible effects. The boosts or droops from playback EQ changes would *not* be increased by using dbx NR, however.

Dolby B vs. Dolby C NR

As stated earlier, tapes encoded with Dolby B NR may seem overly bright if not decoded, but will still be acceptable for noncritical playback. Similar statements have been made concerning the playback of Dolby C NR tapes with the use of Dolby B NR decoding. Figure 2 illustrates that there is not a large difference in the playback response when using Dolby B NR decoding on a tape made with Dolby C NR. At meter-zero level (top trace), the response with Dolby C NR decoding is very flat, but response with Dolby B NR decoding falls slowly above 2 kHz, with a rapid roll-off above 10 kHz. This occurs simply because the Dolby B NR

HOWARD A. ROBERSON



decoder does not compensate for the Dolby C NR anti-saturation and spectral-skewing encoding. At −15 dB (middle trace), there is very close correspondence in the responses over the great majority of the entire band. At −30 dB (bottom trace), there is some elevation in the response above 500 Hz with Dolby B NR decoding. This is to be expected, because Dolby B NR does not have as much encoding boost or decoding cut as Dolby C NR at this low level.

Figure 3 presents the other side of this comparison: Dolby C NR decoding on a tape encoded with Dolby B NR. The reverse of the results in Fig. 2 is shown: A high-frequency boost, especially at meter zero (top trace), and a reduction in level above 500 Hz at −30 dB (bottom trace), except for the peak at 20 kHz.

Figures 2 and 3 demonstrate that, over a range of important levels, relatively small response differences result

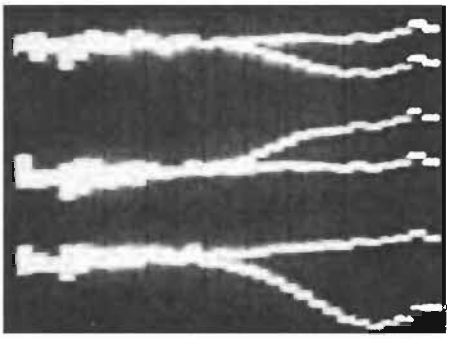


Fig. 1—Wrong equalization in playback. Top traces were recorded with 120-µS EQ and played back with 120- and 70-µS EQ. Middle traces were recorded with 70-µS EQ and played back with 70- and 120-µS EQ. Bottom traces are same as top set but recorded and played back using Dolby C NR. (Vertical scale: 5 dB/div.)

when using Dolby B NR decoding on tapes made with Dolby C NR, or vice versa. The high-frequency boost with Dolby C NR decoding at meter zero (Fig. 3) was a bit much, however. All this demonstrates that Dolby B NR should be selected for playback of Dolby B NR encoded tapes, as is possible with any cassette deck that has Dolby C NR.

Dolby C vs. dbx NR

Every so often, someone makes a statement or asks a question indicating a belief that the dbx and Dolby NR systems are pretty much alike and that encoding and decoding differences are minor. Figure 4 should help resolve this misunderstanding. Recordings were made at five meter levels, in 10-dB steps from +10 (top trace) to −30 (bottom trace). Dolby C NR was used for the recording, but dbx NR was used in playback. Two basic effects resulted from this combination: Severe response deviations across the band, and nonlinear changes in level at all frequencies.

Three factors contribute to the results shown in Fig. 4. First, dbx decoding includes expansion even at the highest signal levels, but Dolby C NR decoding does not expand when the signal amplitude rises above Dolby level. Second, dbx decoders have high-frequency cut even at very high levels, but Dolby C NR decoders actually boost high frequencies. Finally, the dbx decoder's response is the same at all levels, but the Dolby C decoder applies a broad high-frequency cut that deepens and extends lower in frequency as the level is reduced. Notice how the 10-dB steps in the overall record level are rendered into changes of 4 to 16 dB in playback, depending on the frequency and overall level. The negative results are very obvious and most unsatisfactory with any sort of music.

Next, I recorded with dbx NR and then used Dolby C NR in playback. Record levels ranged from +10 (top trace) to −40 dB (bottom trace), with the zero-meter level set without NR switched in. Figure 5 illustrates the very poor responses from this combination. Because Dolby C NR does not have expansion or high-frequency cut at higher levels, the 10-dB steps have been reduced to 5 dB or even less, and the high end is elevated 10 dB or more. Only at −40 dB (relative record level) does the response become roughly flat. This is another hopeless case of bad sonics that emphasizes the lack of compatibility between dbx and Dolby C NR.

Dolby C with dbx NR

Some time ago, Technical Editor Ivan Berger ("Spectrum," August 1985) commented on using both Dolby and dbx noise-reduction systems. The aim is to get a large reduction in noise with less susceptibility to low-level noise pumping. The combination is set up using the deck's built-in Dolby NR system and an external dbx processor. I tried the combination, using a Nakamichi CR-7A deck with Dolby C NR and an external dbx 224 processor. Figure 6 shows the record/playback responses from +10 (top trace) to −30 dB (bottom trace), relative to meter zero. (The −20 and −30 responses were purposely shifted up to put them in the same figure.) Except for a little roll-off at the high end, the responses are very flat for most levels. To show how little response is affected by adding dbx to Dolby C NR, the traces for the combined NR systems are overlaid on traces made with Dolby C NR alone.

Figure 7 shows what happened with further reductions in level, from −30 to −60 dB, all with Dolby C NR alone and also with dbx NR added. The results remained really quite good, although the Dolby C NR response at −60 dB has a rise in noise at both the low and high ends.

I was pleasantly surprised by how good the responses remained over the 70-dB range shown in Figs. 6 and 7. They emphatically confirm that the two NR systems will not adversely affect

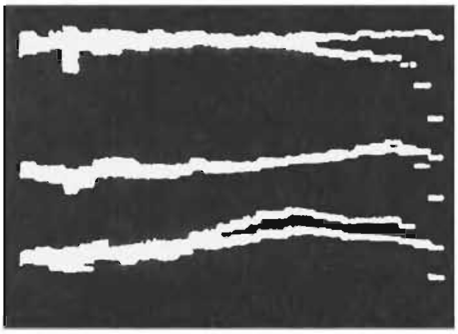


Fig. 2—Recorded with Dolby C NR and played back with Dolby B and C NR. Top pair at zero meter level, middle at −15 dB, and bottom at −30 dB. See text. (Vertical scale: 5 dB/div.)

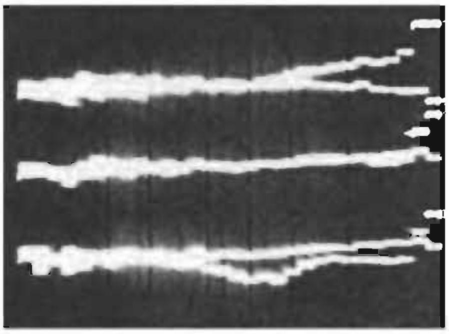


Fig. 3—Same as Fig. 2 but with Dolby B NR during recording. Note the bottom trace's midrange deviation at −30 dB; see text.

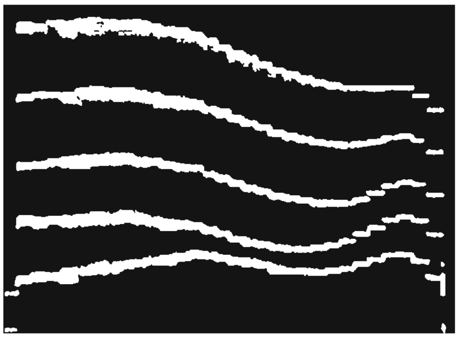


Fig. 4—Effects of recording with Dolby C NR and playing back using dbx NR. Recording levels, from top: +10, 0, −10, −20, and −30 dB. See text. (Vertical scale: 10 dB/div.)

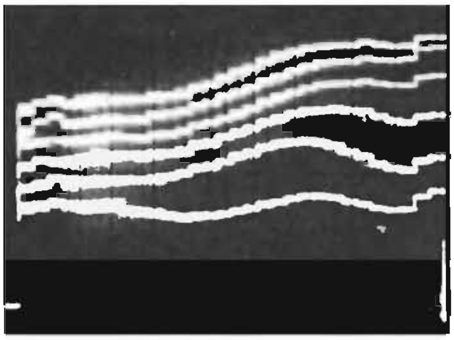


Fig. 5—Reverse procedure of Fig. 4, here using dbx NR during recording and Dolby C NR during playback. See text. (All other test conditions same as Fig. 4.)

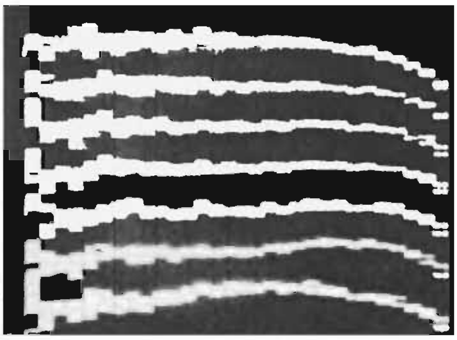


Fig. 6—Effects of using Dolby C NR alone and Dolby C NR in conjunction with dbx NR, in both recording and playback; see text. Traces made using both NR systems are overlaid on traces made with Dolby C NR alone; note the close match at most levels. Recording levels, from top: +10, +5, 0, −5, −10, −20, and −30 dB. (Vertical scale: 5 dB/div., with traces at −20 and −30 dB shifted up for clarity.)

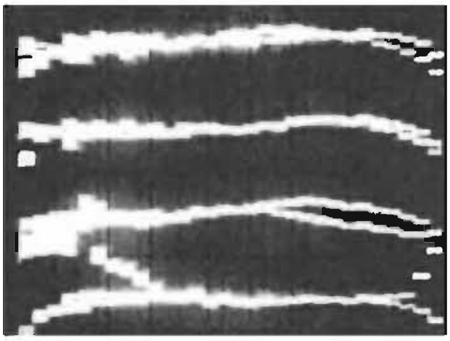


Fig. 7—Same as Fig. 6 but for recording levels of (from top) −30, −40, −50, and −60 dB.

flat recorder response, such as that provided by the Nakamichi CR-7A used in these tests. Adding a bit of equalization after the NR decoders—a boost that increased from 1 dB at 12 kHz to 2 or 3 dB at 20 kHz—would make all the responses fit within ±2 dB from 30 Hz to 20 kHz. To keep the equalizer filter from adding unwanted noise, I prefer to keep the slight roll-off.

Summary

Equalization changes are valid options, when the deck permits, for getting better record/playback high-frequency MOLs (120 µS) or lower noise (70 µS). If the equalization is changed just for playback, a noticeable change will occur in the level of the higher frequencies, which may or may not be desirable. If Dolby NR is used, the change in playback EQ causes exaggerated effects, usually unwanted.

Changes from Dolby B NR encoding to no NR or Dolby C NR in playback will be noticeable, but perhaps acceptable in noncritical listening. Changes from Dolby C NR to Dolby B NR in playback may be similarly acceptable. Adding or taking out Dolby C NR decoding when listening will cause gross and unacceptable sonic effects.

The effects would be most unwanted with dbx NR as well, if the playback does not have the needed decoding or if decoding is added when the recording had no encoding. Using Dolby NR decoding with dbx encoding, or dbx decoding with Dolby NR encoding, produces severe response and level distortions of the original signal, and the combinations have no place in normal listening.

It is possible to use both dbx and Dolby C NR together, by using an external dbx encoder/decoder and a deck with Dolby C NR built in; this will have minimal effects on response and level linearities. It is up to the individual user to decide whether the complexity and cost is worth the possible reduction of low-level noise pumping.

As a generality, I can state that EQ and NR modes should not be switched between recording and playback. This discussion, however, may help the careful user understand the relationships better and choose certain combinations to gain the most satisfactory overall results.

Oberlin Smith

Centenary of Magnetic Recording

JON R. SANK

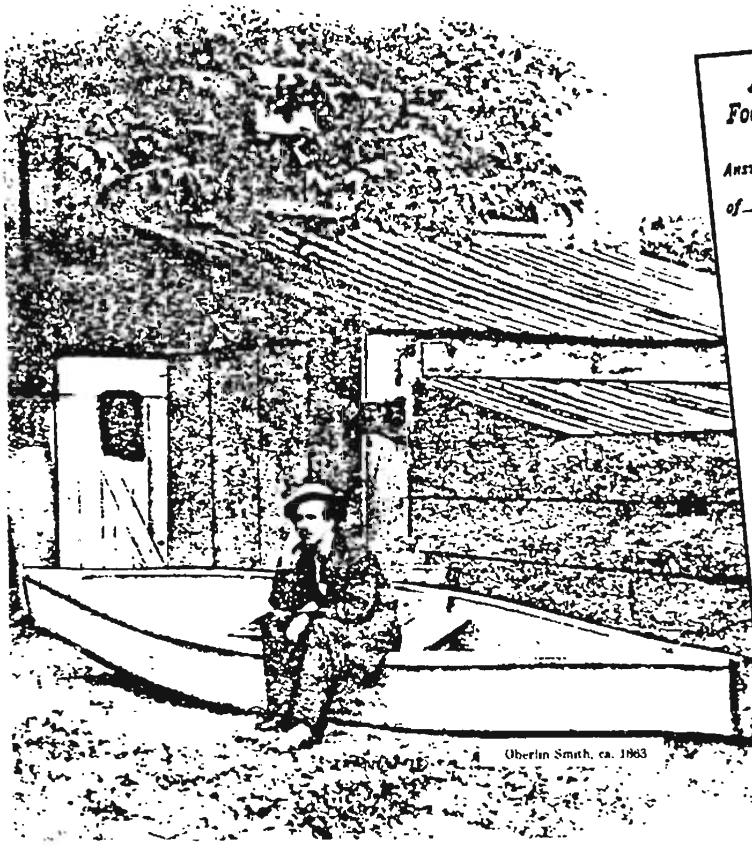
Audio history was revised by a paper in the March 1988 *Journal of the Audio Engineering Society*. Until then, Valdemar Poulsen of Denmark was thought to have invented magnetic recording in 1898, with his Telegraphone machine. In the *Journal* article, "1888-1988: A Hundred Years of Magnetic Sound Recording," author Friedrich Karl Engel, of the German firm BASF, revealed an 1888 paper by Oberlin Smith of Bridgeton, N.J., proving that Smith had spawned the idea before Poulsen. It is remarkable that Smith's article remained unknown all these years, despite the fact that it is readily available on microfilm—even at the Philadelphia Free Library near my home. Fortunately, Engel found Oberlin Smith's writings in time for magnetic recording's centennial year.

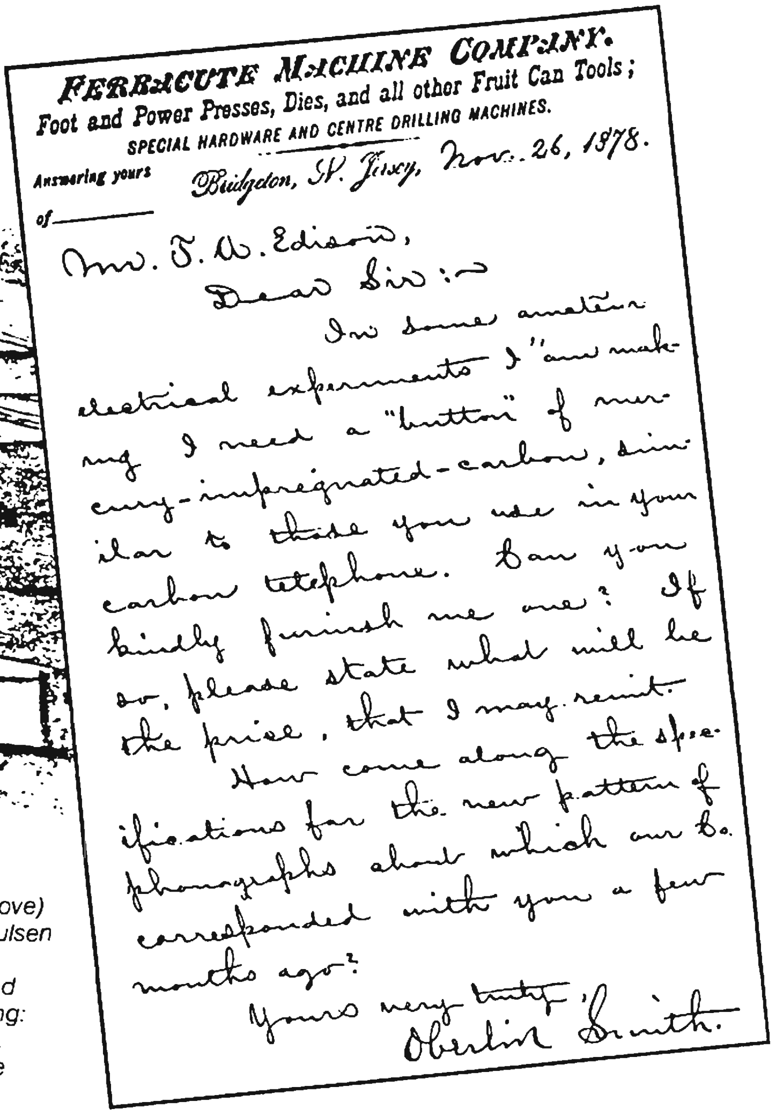
When I told this story to one of the junior audiophiles in my family, he found it hard to believe that the art was so ancient. This is understandable because magnetic recording was not introduced in the U.S. until after World War II. My AES colleague and audio-museum curator, Jack Mullin, discovered the Magnetophon tape recorder in Germany after the war and brought it to the U.S. He recorded Bing Crosby's radio shows on that machine, using the German tapes. Via a complex chain of events, Ampex and 3M became involved in magnetic recording and developed the recorders and tapes that led to the products of today.

The first magnetic recorder I can recall was a Webcor wire recorder; it reproduced speech reasonably well, but music sounded bad. My present antique collection includes an RCA wire recorder which, instead of having open spools like those on the Webcor, has a cartridge resembling an overgrown 8-track. In about 1950, I purchased a TapeMaster tape recorder which had only one motor. This recorder never ran at the correct speed, but early on the morning of June 2, 1953, I managed to record the coronation of Queen Elizabeth II via shortwave. I still have the recording, now dubbed to modern tape. (One reel of the master had been on paper-backed tape.) Later, in my acoustic lab at RCA, I had a monstrous RCA RT-11 full-track, ¼-inch tape recorder which I hardly used because the capstan motor sounded like a railroad car. Finally, some 20 years ago, I obtained the solid-state successor to the RT-11, the RT-21. It is a 2-track, ¼-inch machine with three big motors and has performed reliably

Photograph: Robert Lewis

Photograph: Jon R. Sank





FERRACUTE MACHINE COMPANY.
Foot and Power Presses, Dies, and all other Fruit Can Tools;
SPECIAL HARDWARE AND CENTRE DRILLING MACHINES.

Answering yours of ________ Bridgeton, N. Jersey, Nov. 26, 1878.

Mr. T. A. Edison,
Dear Sir:—
In some amateur electrical experiments I am making I need a "button" of mercury-impregnated-carbon, similar to those you use in your carbon telephone. Can you kindly furnish me one? If so, please state what will be the price, that I may remit.
How come along the specifications for the new pattern of phonographs about which our Co. corresponded with you a few months ago?
Yours very truly,
Oberlin Smith.

Fig. 1—Oberlin Smith's letter to Thomas Edison. Photo of Smith and this letter are reprinted from The Antique Phonograph Monthly, *February 1975.*

Oberlin Smith (above) and Valdemar Poulsen can each be said to have discovered magnetic recording: Smith has priority, but Poulsen made the first working models.

to the present. It is very useful for editing and dubbing to cassettes.

Poulsen's wire recorder was demonstrated at the Paris Exposition of 1900. The lack of amplification prevented it from being a commercial success. Since it was a purely electrical system, amplification was needed. The very popular mechanical cylinder and disc recorders of the early part of the century could drive a reproducing horn directly, and so they did not need amplification. Nevertheless, Poulsen is reported to have received the Exposition's Grand Prize. His machine seems to have been the first working magnetic recorder. Later versions included magnetic disc and cylinder units. There is no evidence that Smith ever completed a working recorder, and his models perished in a factory fire in 1903.

Oberlin Smith visited Thomas Edison and heard a demonstration of his cylinder phonograph, apparently in 1878. Afterwards, Smith developed ideas on alternate forms of the mechanical phonograph and invented the magnetic recording and playback systems. He documented these ideas by filing papers with the Clerk of Cumberland County, N.J. on September 24, 1878. Then, on October 4 of that year, he filed a "caveat" paper with the U.S. Patent Office. His experiments at the time were witnessed, but no description of the hardware is available. Shortly after filing the caveat, he wrote a letter to Edison (Fig. 1), requesting some electrical parts. This letter, discovered in the Edison Museum, indicates Smith's developmental activities in the fall of 1878. He was, however, also busy managing a growing tool company, and didn't find time to do any more experiments in the following years. Ten years later, in 1888, he decided to make public his invention by means of a short article, "Some Possible Forms of Phonograph." The article was published in *The Electrical World*, and in it, Smith encouraged others to continue his work.

I compared the documents of 1878 with the 1888 article and concluded

The recording medium favored by Smith was a string impregnated with magnetic particles, similar in principle to modern plastic tape coated with magnetic powder.

that the systems described are identical. In 10 years, Smith had not made any discoveries which would cause any changes in his invention. His 1911 letter to the *Journal of the Franklin Institute* (which we'll discuss later) indicated that he did no more work on magnetic recording after 1878, and the models destroyed in the fire were those from his early work. (Both sets of papers also show some proposed variations in Edison's cylinder phonograph.) Two of Smith's sketches from the September 8, 1888 issue of *The Electrical World* are reproduced here (Figs. 2 and 3). This newspaper-style periodical, though printed in New York and regionally oriented, was widely read by those interested in electricity. Shortly before the turn of the century, there were many individual inventors working in all areas of electricity. Examining the 1888 issue of *The Electrical World* showed that one person had built an electric generating plant in his home, while others were inventing telephone gadgets. More search may reveal some long-forgotten device that needs re-inventing.

Figure 2 shows Smith's recorder. A magnetic wire or string travels from one spool to the other. The direct current flowing through the telephone transmitter (carbon microphone) is modulated by the audio signal, and both act to magnetize the moving medium by means of the induction coil. The d.c. bias in a carbon microphone is relatively stronger than the signal current. This could have functioned similarly to the a.c. bias of modern analog recording, which moves the magnetic operating point to the linear region of the medium.

Figure 3 shows the magnetic recorder in playback mode. A voltage is induced in the pickup coil as the lines of flux from the magnetized media cut the turns of the coil. This audio signal would be heard in the telephone receiver if an amplifier were inserted at point X. Smith recognized that the voltage would be too low to drive the earphone directly, and therefore he perceived that an "intensifier" was needed. His only thought was a battery; the vacuum tube amplifier was not invented until the next century.

In the September 29, 1888 issue of *The Electrical World*, a one-paragraph note quotes a letter from Smith describing improvements to recording and playback systems. With reference to our Fig. 2, he wrote that the recording transducer should be a coil wrapped around an electromagnet, with the core against the magnetic medium. This is clearly a forerunner of the magnetic head. The playback pickup helix in Fig. 3 would be shortened to one turn to "localize the magnetism," which fits in with Smith's correct notion of tiny magnets along the string. This modification leads to the modern idea of a very small gap in the playback head core. The playing or reading device in any system which scans a moving medium must be small compared to the details in that particular medium, be it magnetic tape, phonograph record, digital Compact Disc, or optical film soundtrack.

Smith also had some accurate and prophetic thoughts about the recording medium. He thought that it was

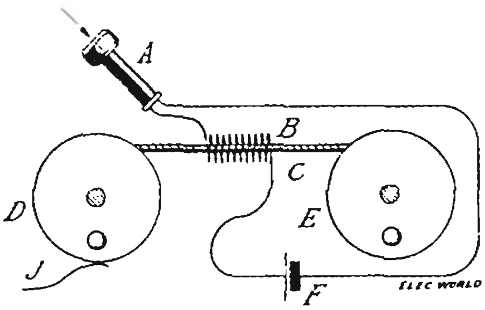


Fig. 2—Smith's magnetic recorder in record mode. Reproduced from the September 8, 1888 issue of The Electrical World.

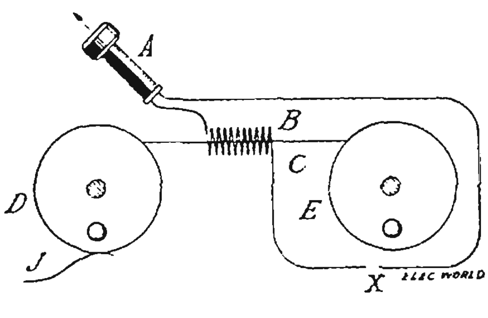


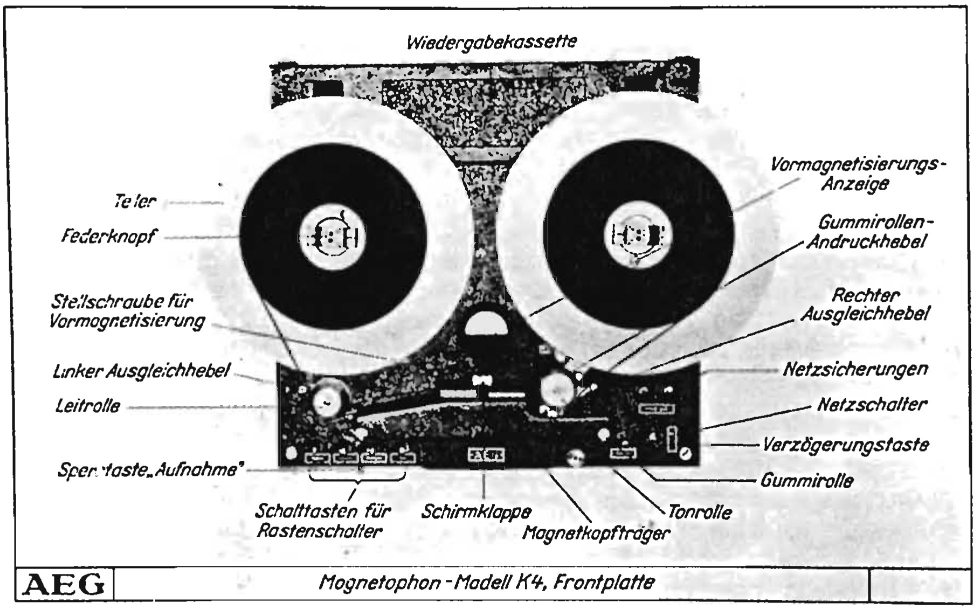
Fig. 3—Smith's magnetic recorder in playback mode. Reproduced from the September 8, 1888 issue of The Electrical World

incorrect to use a magnetic wire, because the little recorded magnets would demagnetize each other, resulting in low output. Therefore, his favored medium was a string or, more specifically, a cotton sewing thread filled with magnetic particles. This was, in principle, the same as our modern plastic tape, which is coated with magnetic powder and held together with some "glue," the binder. Now we know that a magnetic wire will give ample output compared to tape, but the high-frequency response of magnetic wire is poorer if both are running at the same speed. Oliver Read has stated that recording wire must be run at 24 ips to equal the quality of tape at 7½ ips. (See "Suggested Reading" at the end of this article.)

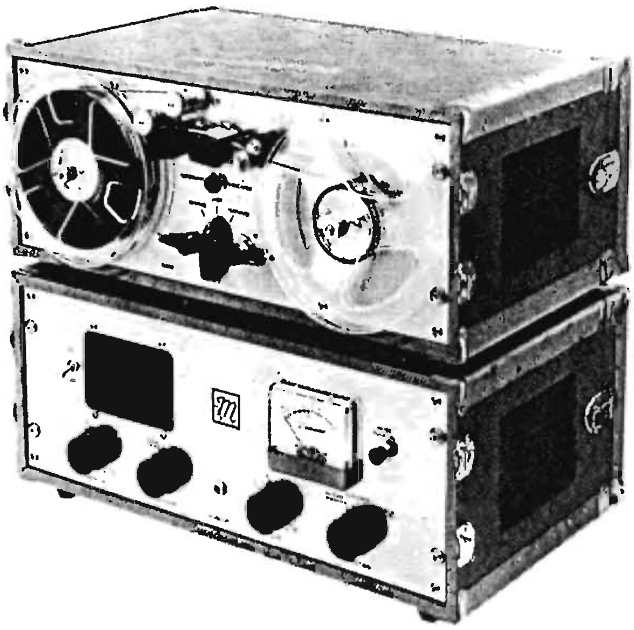
The solid wire proved to be good enough for the telephone quality standards of the day. Poulsen was successful, and Smith was surprised. In Smith's 1911 letter to the *Journal of the Franklin Institute*, he stated that Poulsen's machines were designed exactly the same as in his own earlier descriptions, but he gave Poulsen credit for discovering that the little recorded magnets in a solid medium do not demagnetize each other. Smith mentioned solid wire as a medium in his 1878 papers filed with the Cumberland County Clerk and in the 1888 article, but he "accidentally" omitted mention of wire in his filing with the U.S. Patent Office. If only Smith could have known that some 40 years in the future, magnetic wire would give way to particle-coated tape.

Who was Oberlin Smith? This interested me, because I have spent much time in Bridgeton, N.J. in recent years, helping the hospital with its paging systems and with noise control of mechanical equipment. I have enjoyed the town's abundance of Victorian architecture and its many historic places. I contacted William Chestnut of the Bridgeton Antiquarian League, who put me in touch with Arthur Cox, a high school art teacher who is an expert on Oberlin Smith, one of Bridgeton's most famous citizens. Cox had just completed a book titled *Ferracute: The History of an American Enterprise*, which chronicles Smith's company as well as his spare-time inventions. These included not only magnetic recording

Smith recognized that an "intensifier" was needed to drive his recorder's earphone, but the vacuum tube amp wasn't invented until the next century.



Magnetophon-Modell K4, Frontplatte



The first recorders to use a ribbon-like tape medium were produced as a joint venture by AEG and BASF in Germany in the early 1930s. Recorders such as the Magnecord PT-6 (right) appeared in the U.S. in the late 1940s.

but also his Autofono, the first jukebox. I am indebted to Cox for his book as well as for copies of many old documents which he has collected.

Smith was born in Ohio in 1840 and died in Bridgeton in 1926. He was a prolific inventor and held 70 patents. In 1890, he was president of the American Society of Mechanical Engineers. Smith never seemed to be short of cash and, at age 23, opened a machine shop in Bridgeton which later became the Ferracute Machine Company. Ferracute (which means "sharp iron") made presses for cutting and forming metals, and it grew rapidly because of the need for tools to make tin cans quickly for the food industry. In the early part of the 20th century, the Ford Motor Company became Ferracute's largest customer, and during both World Wars, the plant produced presses for making small-arms ammunition. After World War II, business slowed, and in 1968, the company was sold to the Fulton Iron Works of St. Louis. The Bridgeton plant was closed and remains empty to this day.

One of my hobbies is taking aerial photographs while piloting, and Ferracute's location on a lake made a nice picture (see first page of article). The plant, as shown, was built in 1905, after the 1903 fire destroyed the original, along with Smith's recorders. The Smith mansion, Lochwold, was located on the lake a few hundred yards from the plant. It was destroyed by fire in 1934, and I could see no traces. The plant offices were in the little building with the conical roof, adjacent to the factory building. Smith's office was in the room below the cone.

His most valuable contributions were in the mechanical realm, and his 1896 book, *Pressworking of Metals*, remains a classic in its field. Oberlin Smith loved to play phonograph records, and, incidentally, he was a very good dancer. ▲

Author's Acknowledgments

Arthur Cox, in addition to furnishing documents, supplied the formal portrait of Smith at the beginning of this article, for which I am grateful. I would also like to acknowledge the assistance of Allen Koenigsberg, publisher of *The Antique Phonograph Monthly*, who provided a copy of Smith's letter to Edison, which originally came from the archives of the Edison National Historic Site in West Orange, N.J.

Suggested Reading

Benson, K. B., ed., *Audio Engineering Handbook*, McGraw-Hill, New York, 1988. Benson's *Handbook* has a good history of magnetic recording by Busby of Ampex; you may also wish to read my chapter on microphones.

Cox and Malim, *Ferracute: The History of an American Enterprise*, Bridgeton, N.J., 1985. (Available for $32 postpaid from Arthur Cox, P.O. Box 411, Bridgeton, N.J. 08302.)

Read, Oliver, *The Recording and Reproduction of Sound*, Howard W. Sams, Indianapolis, 1952. Read's old book contains an excellent description of wire recorders and how they magnetize the wire, as well as a comparison to tape recording.

The Antique Phonograph Monthly, 502 East 17th St., Brooklyn, N.Y. 11226. Allen Koenigsberg's digest-sized publication is for those avidly interested in the older means of recording, not necessarily collectors. Articles on unusual subjects and the classified advertising section are both interesting.

HOW HOT ARE CDs?

New measurements of CDs show peak levels that challenge both tape and recordists.

Howard A. Roberson

To get the best analog tape/recorder performance, recordists should know the distribution of the music's spectral peaks throughout the audio band. In the article "Basics of Tape Performance" (September 1982), I referred to two sources of information on musical spectra. A figure in that article showed the levels rolling off at about 6 dB per octave above 1 or 2 kHz. This appeared to be fortuitous, because the MOL limit (maximum output level for 3% distortion) of cassette tape fell at about the same rate, and the spectral roll-off started from the same frequency.

Although the spectral shape was a good match for analog recording, the figure was in error. I concluded that the original sources did not include a wide enough range of musical material in one case and used a poor microphone location in the other. Considering all types of music, I could see that the high-frequency roll-offs in the spectra I was using were unrealistic. Where, then, could I get good musical spectra without spending hours making measurements of actual performances?

I decided it would be worthwhile to measure third-octave spectra from a good assortment of CDs. I selected a total of 40, most of which had received enthusiastic reviews for performance and sound quality (Table I). Music from the baroque, classical, Romantic, and other periods was selected, with an assortment of overtures, arias, and organ music. Some pop/rock and movie music was included as well. For the most part, the classical CDs are listed in the Table in order of composition.

Test Procedures

My test plan was to find the highest momentary peak levels in 30 third-octave bands (25 Hz to 20 kHz) over the duration of each piece. Peak levels of any varying musical waveform are continually changing. I wanted to get and hold the highest peak level that occurred anywhere in the music in each of the 30 third-octave bands. I used an Ivie IE-30A RTA in its "Accumulate" mode to do this. The musical transients were long enough for the RTA to capture the actual peaks within a dB or so. Maximum peak levels at the highest frequencies were caused by cymbal crashes. The highest bass levels were from organ or bass drum, except for the cannon in the "1812" Overture.

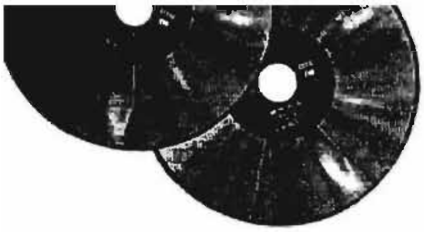
I plotted the accumulated peak levels of the CDs and of four FM pop/rock stations in each third-octave band; I then tabulated my readings. Because of variations in level from CD to CD and some changes in the measurement chain, the plotted band levels did not have a common reference. My examination of all band levels revealed that all CDs were relatively flat in the region from 200 Hz to 1 kHz. I tabulated the band levels for each CD referred to the average of its levels in the 200-, 250-, 315-, 400-, 500-, 630-, 800-, and 1000-Hz bands. Then I was able to make direct comparisons on the spectral shapes from one disc to the next.

Figure 1 shows the range of peak levels in all bands of the 40 CDs, relative to their average levels in the reference region from 200 Hz to 1 kHz. The top curve shows the highest peak levels recorded from each band, relative to the average reference-region levels of all the CDs tested. This curve shows how high the relative level might be in

TABLE I

Compact Discs evaluated for peak levels.

CD No.	Label & Catalog No.	Composer & Artist	Title & Orchestra	Portion Tested
1	Argo 411613-2-ZH	Mozart Brown	Sinfonia Concertante, K.364 Academy of St. Martin	All
2	Telarc CD-80108	Mozart Mackerras	Eine Kleine Nachtmusik, K.525 Prague Chamber Orchestra	All
3	Telarc CD-80070	Vivaldi Ozawa	The Four Seasons Boston Symphony Orchestra	"Spring"
4	Deutsche Gram 410024-2-GH	Schubert	"Death and the Maiden" Quartet Amadeus Quartet	All
5	Orfeo C-045901-A	Mozart Jochum	Symphony No. 39 Bamberg Symphony Orchestra	All
6	Telarc CD-80114	Mozart Previn	Quintet for Piano and Winds, K.452 Vienna Wind Soloists	All
7	Telarc CD-80114	Beethoven Previn	Quintet for Piano and Winds, Op.16 Vienna Wind Soloists	All
8	Philips 411471-2-PH	Tchaikovsky Marriner	Serenade in C for String Orchestra Academy of St. Martin	All
9	London 414203-2-LH	Berlioz Dutoit	Symphonie Fantastique Montreal Symphony Orchestra	Mvts 4 & 5
10	Telarc CD-80047	Tchaikovsky Maazel	Symphony No. 4 Cleveland Symphony Orchestra	All
11	London 410116-2-LH	Dvorak Solti	Symphony No. 9 Chicago Symphony Orchestra	All
12	Telarc CD-80051	Saint-Saens Ormandy	Symphony No. 3 ("Organ") Murray, Philadelphia Orchestra	All
13	Telarc CD-80071	Debussy Slatkin	La Mer St. Louis Symphony Orchestra	All
14	Telarc CD-80039	Stravinsky Shaw	"Firebird" Suite Atlanta Symphony Orchestra	All
15	Chandos CHAN-8309	Elgar Gibson	Froissart Overture Scottish National Orchestra	All
16	Telarc CD-80041	Tchaikovsky Kunzel	"1812" Overture Cincinnati Symphony Orchestra	All
17	Telarc CD-80088	Bach Murray	Toccata and Fugue in D Minor	All
18	Telarc CD-80123	Bach/Dorsey Dorsey	Italian Concerto (from *Bachbusters*)	All
19	Philips 412790-2-PH2	Bach	Brandenburg Concerto No. 2 I Musici	All
20	Philips 412790-2-PH2	Bach	Brandenburg Concerto No. 5 I Musici	All
21	Telarc CD-80116	Von Suppe Kunzel	Light Cavalry Overture Cincinnati Pops Orchestra	All
22	Telarc CD-80116	Reznicek Kunzel	"Donna Diana" Overture Cincinnati Pops Orchestra	All
23	Telarc CD-80116	Offenbach Kunzel	Orpheus in the Underworld Cincinnati Pops Orchestra	All
24	Telarc CD-80116	Rossini Kunzel	William Tell Overture Cincinnati Pops Orchestra	All
25	Harmonia Mundi HMC-901149	Charpentier	Motets (for one or two voices) Nelson, Jacobs, Concerto Vocale	All
26	Philips 412631-2-PH	Simon Estes	Spirituals	All
27	Philips 411148-2-PH	Mozart Te Kanawa	Arias Davis, London Symphony Orchestra	1-5
28	Archiv 410647-2	Handel Pinnock/ Preston	Dettingen Te Deum English Concert/ Westminster Abbey Choir	All
29	Telarc CD-80083	Wagner Marriner	Die Meistersinger, Act I Minnesota Orchestra	Prelude
30	MMG MCD-10025	Waldteufel Kunzel	Music of Waldteufel Cincinnati Pops Orchestra	All
31	MMG MCD-10005	Sousa Kunzel	Peaches and Cream Cincinnati Pops Orchestra	All
32	Telarc CD-80094	Various Kunzel	Star Tracks Cincinnati Pops Orchestra	All
33	Warner Bros 25264-2	Dire Straits	Brothers in Arms	All
34	Fantasy FCD623CCR2	Creedence Clearwater Revival	Chronicle, Vol. I	All
35	A&M CD-3735	The Police	Synchronicity	All
36	Columbia CK-35047	Air Supply	Love & Other Bruises	All
37	Phila. Intl. ZK-38539	Patti LaBelle	I'm in Love Again	All
38	Sparrow CDP-41039	Deniece Williams	So Glad I Know	All
39	Epic E2K-37037	The Clash	Sandinista	1st of 2 CDs
40	Telarc CD-80106	Various Kunzel	Time Warp Cincinnati Pops Orchestra	All



Measuring peak spectra of 40 CDs showed me that each type of music presents a different challenge.

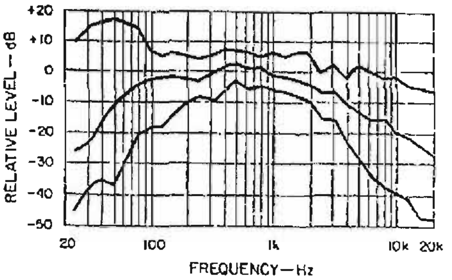


Fig. 1—Range of peak levels, in all bands, for 40 CDs. Highest recorded peak levels for any CD are shown at top, followed by average peak levels for all 40 discs (middle), and minimum peak levels for any CD (bottom). Levels are shown relative to average level, from 200 Hz to 1 kHz, of all 40 discs.

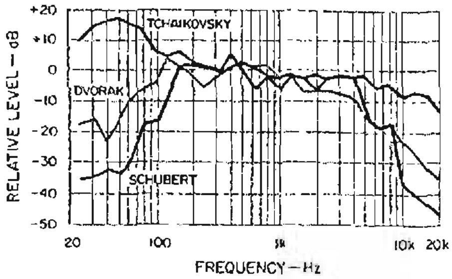


Fig. 2—Spectra of three CDs, ranked by bass content. Tchaikovsky's "1812" Overture had the highest content of the discs surveyed; Dvořák's Symphony No. 9 had median bass levels, and Schubert's "Death and the Maiden" Quartet had least bass. Plots are positioned to put their average levels (from 200 Hz to 1 kHz) on the 0-dB reference line.

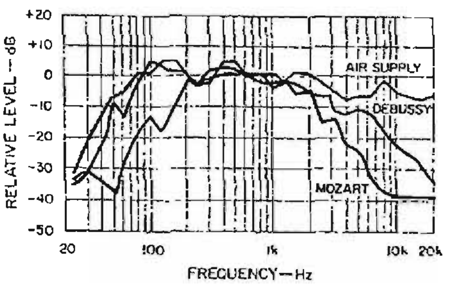


Fig. 3—Spectra of three CDs ranked by treble content, showing highest treble content (Air Supply, Love and Other Bruises*), median treble content (Debussy, "La Mer"), and least treble* content *(Mozart, Quintet for Piano and Winds).*

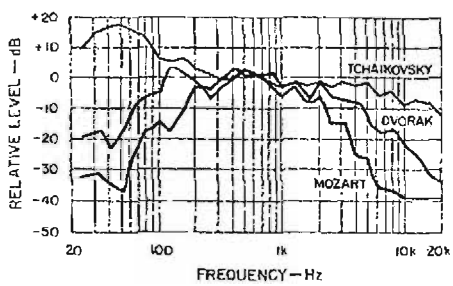


Fig. 4—Spectra of three CDs ranked by bass and treble content, showing highest bass and treble content (Tchaikovsky, "1812" Overture), median content (Dvořák, Symphony No. 9), and least bass and treble (Mozart, Quintet for Piano and Winds).

any one or more of the bands. The middle curve shows the average of all peak levels measured in each band; note that this curve rolls off noticeably above and below the reference region. The bottom curve shows the lowest peak levels measured from any of the 40 CDs in each third-octave region. This minimum-peak-level curve shows that there is some music with little bass or high-treble energy.

Figure 2 shows the spectra of three specific CDs, ranked by bass content. The cannon in Tchaikovsky's "1812" Overture definitely generated the most bass of any CD in my test group. I judged the Dvořák Symphony No. 9 CD to have the median bass level of the group and Schubert's "Death and the Maiden" Quartet to be the CD with the least bass. A similar process was used to generate Fig. 3—for the maximum-, median-, and minimum-treble CDs—and Fig. 4, covering the same rankings for both bass and treble. The figures detail something which we all know to be true, if we consider all types of music: There is a wide range in the amount of energy in both the low- and high-frequency ends of the total music spectrum.

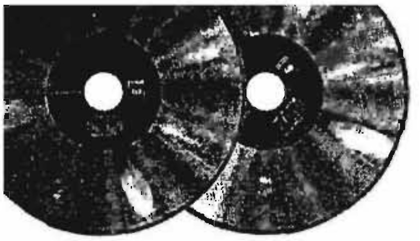
Spectral Envelopes

After generating these figures, I decided to try classifying the 40 peak-level spectra on hand. Using transillumination, I traced the original spectrum plots onto graph paper. I started with the plot for CD No. 1 (Mozart's "Sinfonia Concertante") and plotted its spectral envelope. The envelopes for all other CDs were checked for possible matches. I found a number that were very similar to the Mozart disc and a few which corresponded satisfactorily, if I was willing to allow differences of a dB or two. All this music was from the baroque and classical periods, except for some vocal works.

Figure 5 shows this spectral envelope. (The CD numbers are given in the caption, so you can refer to Table I.) To get a better feel for how to meter such music, I measured the average (VU) and peak levels for pink noise, with a response shaped to match the spectral envelope. I aligned the highest level in this envelope exactly with the zero reference to show that the highest levels would be indicated accurately with peak-responding meters. The envelope curve in Fig. 6 covers seven CDs and quite a variety of music. This envelope matched peak third-octave levels for Berlioz, Dvořák, and Tchaikovsky Symphonies, Reznicek's "Donna Diana" Overture, Debussy's "La Mer," Dorsey's *Bachbusters*, and four FM pop/rock stations. This envelope has more low- and high-frequency energy above 2 kHz than the one in Fig. 5.

The envelope in Fig. 7, for three works with organ, shows the major influence of the low organ notes, the highest levels of which are centered around 50 Hz. The organ in *Time Warp* is excerpted from "Also Sprach Zarathustra" by Strauss. Although the levels were highest below 70 Hz, the low-frequency peak levels were easy to meter correctly. The envelope of Fig. 8 covers Stravinsky's "Firebird" Suite and four overtures. The highest levels were from the bass drum, but there was considerable energy across the entire audio band. Some care was needed to catch the peak levels from the drum beats, but it was a relatively small adjustment and easily made.

Figure 9 shows the spectral envelope generated by the music of Waldteufel and Sousa. Both of these CDs have sudden bass drum peaks (visible on the envelope). My checks indicated that if a recordist were setting input gain based on signal levels elsewhere in these recordings, his levels would probably be about 2 dB high on these drum peaks. The music from *Star Tracks* and *Brothers in Arms* (Fig. 10)



When the spectrum of peaks in the music matches the tape's MOL curve, record levels can safely be high.

had a wide and quite flat peak spectrum before a roll-off at around 10 kHz. Accurate metering was easy with these sources.

Figure 11 revealed a close correspondence in peak spectra among six pop/rock CDs. Because of the general steadiness in the peak level and the wide high-level spectrum, the recordist would probably set the level *slightly* low. On the other hand, the cannon shots in Tchaikovsky's "1812" Overture (Fig. 12) would be very hard to meter correctly, and thus overrecording would be very likely.

Metering, EQ, and MOLs

Peak-responding meters on most decks are fast enough (less than 20 mS) to show typical musical transients, but they read the signals before record equalization, with its high-frequency boost, is applied. As a result, some high-frequency distortion and saturation might be occurring without the meters giving any warning of it. Because of the general level roll-off in the higher frequencies for all music, however, there usually would be little change in the peak meter indications even if the meters were reading the equalized signal. Indications for average-responding (VU-type) meters were usually 8 to 10 dB below those for typical recorder peak-responding meters.

When the levels were measured with an absolute-peak meter, which can display fast transients, they were at least 14 to 15 dB above the indications on the VU meters. Because of the very short duration of these instantaneous peaks (less than 200 µS), distortion would probably not be heard if they were slightly above the MOL limit. These figures, however, should help emphasize that when using a VU-type meter, the recordist must allow at least 10 dB of headroom to get low-distortion recording.

The fall-off in tape MOLs above 1 kHz is well known. If there was no record (or playback) equalization, the high-frequency MOLs would drop relatively little—but the noise would go up. The cassette format is locked into 70- and 120-µS equalizations. A small number of decks allow switching equalization separately, regardless of tape type; this can be helpful, as I will show later. Open-reel recorders, at tape speeds of 3¾ and 7½ ips, may

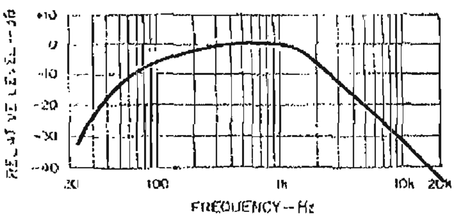


Fig. 5—Peak-level spectral envelope, with highest level used as 0-dB reference, for classical and baroque music plus vocal solos and choral works. The discs are listed in Table I as CDs 1 to 8, 19, 20, and 25 to 28.

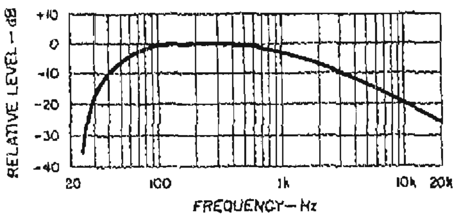


Fig. 6—Peak-level spectral envelope for symphonies, overtures, and other classical works (CDs 9, 10, 11, 13, 18, 22, and 29) plus pop/rock FM stations; see text.

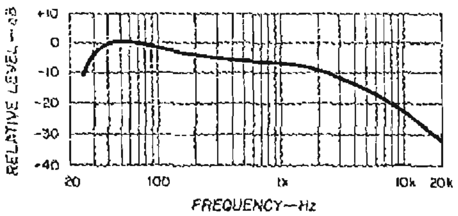


Fig. 7—Peak-level spectral envelope for works with organ (Saint-Saëns, Symphony No. 3; Bach, Toccata and Fugue in D Minor, and Strauss, "Also Sprach Zarathustra"), listed in Table I as CDs 12, 17, and 40.

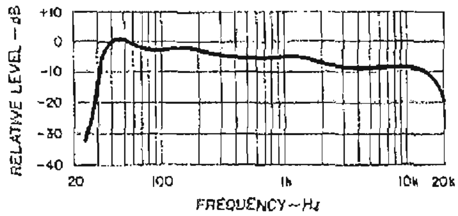


Fig. 8—Peak-level spectral envelope for overtures (CDs 15, 21, 23, and 24) and Stravinsky's "Firebird" Suite (CD 14).

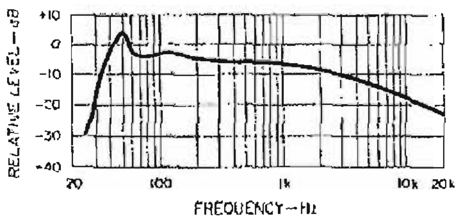


Fig. 9—Peak-level spectral envelope for music of Waldteufel and Sousa (CDs 30 and 31). The peak at 50 Hz is from bass drum beats; see text.

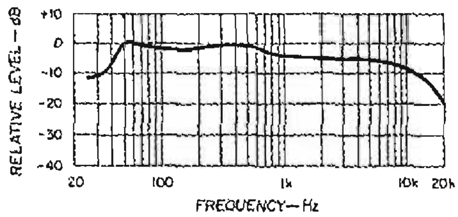


Fig. 10—Peak-level spectral envelope for music from Star Tracks *and* Brothers in Arms *(CDs 32 and 33).*

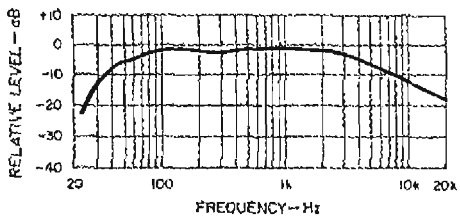


Fig. 11—Peak-level spectral envelope for pop/rock music (CDs 34 to 39).

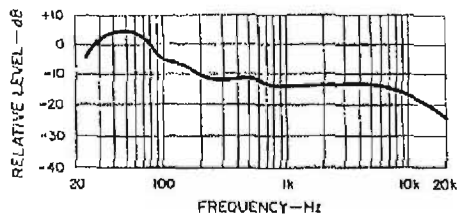


Fig. 12—Peak-level spectral envelope for Tchaikovsky's "1812" Overture (CD 16).

use as much record-equalization boost as do cassette recorders. At the higher tape speeds, particularly 30 ips, the required boost for open-reel recording is significantly less.

Figures 13, 14, and 15 show the range of MOLs, without noise reduction, using a Nakamichi CR-7A deck for the 35 tapes covered in my last tape survey (November 1987). The dashed lines at the low-frequency ends of the curves show the drop in MOLs from 100 to 50 Hz for the typical deck. The best decks, in this regard, have about a 2-dB drop, and the worst decks have about a 10-dB reduction in MOLs from 100 to 50 Hz. At 40 Hz, the reduction in performance for all decks is even greater. The Type I MOLs are shown in Fig. 13, the Type IIs are in Fig. 14, and those for Type IV are in Fig. 15. In these figures, the top curve for each tape type is for the best MOLs, and the bottom curve is for the worst MOLs.

When I replotted some of this MOL data, I had to conclude that there was too great a discontinuity between the distortion levels shown at frequencies up to 1 kHz (where I measure third-order harmonic distortion) and from 2 kHz up (where the tape's frequency limits force me to use third-order twin-tone IM measurements; see sidebar). The reference level I had been using for the twin-tone IM tests was based on the rms levels of those tones, as specified by an IEC Standard I was using. However, DIN Standard 45 403, for nonlinear distortion measurements, clearly states that peak level is the proper reference. Careful examination of distortion products, with a spectrum analyzer, provided real-world confirmation. The peak level of twin tones, with each one at the same level, is 6 dB higher than a single tone and 3 dB higher than the twin-tone rms level.

Figures 16, 17, and 18 show the effect of this necessary adjustment with the twin-tone IM MOL curves raised 3 dB. (Twin-tone IM data, in the earlier survey, should be increased 3 dB as well. Relative tape rankings remain the same.) The little jog that remains with Type II (middle) and IV (bottom) tapes is correct for the standard 70-µS equalization. The dashed line above the Type II and IV MOL curves, above 1 kHz, shows the MOL increase that 120-µS equalization would yield.

DISTORTION TESTS & ANALOG RECORDING

In general, measuring harmonic distortion is straightforward. There are many low-distortion sources these days and analyzers are available in a number of formats. Tape noise in analog recording, however, makes it difficult to get reliable distortion data unless the recorded flux level is above 50 nWb/m or so. If 3% distortion is the criterion, the levels will be much higher and harmonic distortion is easily measured—particularly at lower frequencies.

The distortion in analog tape recording is primarily third-order, and the third harmonic is the most prominent in the playback of a single recorded tone. Accurate assessment of third-order distortion, in general, requires flat record/playback response out to the frequency of the third harmonic. When the record levels are high enough to cause 3% distortion, however, the response on a typical deck begins falling off by 1 kHz. Data taken at higher frequencies is valid for assessing *harmonic* distortion, but third-order distortion involves more than simple harmonics. Complex musical energy around the same frequency and at the same level would cause many combinations of sum- and difference-frequency distortion products. In other words, measuring the harmonic distortion of a single tone is not adequate for assessing nonlinear performance with music—except at lower frequencies.

A two-tone test signal can be used for tests out to the response limits of the deck, as its difference-distortion products remain in band. The third-order difference products for two tones, f_1 and f_2, are:

$f_3 = 2f_1 - f_2$ and

$f_4 = 2f_2 - f_1$.

Another way of stating these frequency relationships is:

$f_3 = f_1 - (f_2 - f_1)$ and

$f_4 = f_2 + (f_2 - f_1)$.

There are also two sum products:

$2f_1 + f_2 = 3f_1 + (f_2 - f_1)$ and

$2f_2 + f_1 = 3f_2 - (f_2 - f_1)$,

which show their frequency grouping with the third harmonics.

Figure B1 shows the waveform of a 500- and 600-Hz twin-tone signal. Each individual tone was four divisions high, peak to peak, but the combination is almost eight divisions high. This doubling of peak levels holds true for all combinations of frequencies. Figure B2 is the spectral display of the fundamentals and the distortion in the playback from a Nakamichi CR-7A deck. The level reference is the peak level of the two tones, which is 6 dB above the level of each fundamental.

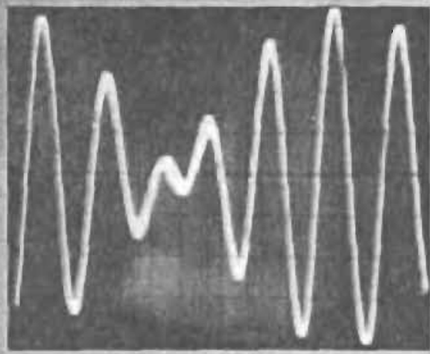


Fig. B1—Waveform of 500- and 600-Hz test signal.

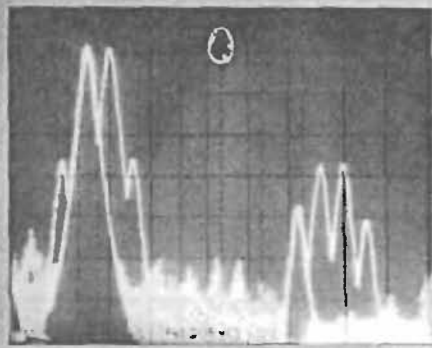
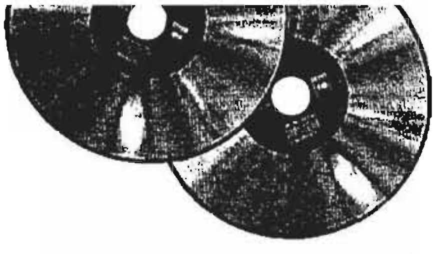


Fig. B2—Third-order distortion products. At left are 500- and 600-Hz fundamentals, flanked by third-order difference products. At right are third harmonic of 500-Hz fundamental (1,500 Hz), third-order sum products of 500 and 600 Hz, and third harmonic of 600 Hz (1,800 Hz). Scales: Horizontal, 200 Hz/div.; vertical, 10 dB/div.

Two sweeps were actually made with the analyzer. One with just the 500-Hz signal and the other with both tones. Notice that the 1,500-Hz third-harmonic distortion product stays the same in level when the second tone is added. Also note how much higher the difference and sum distortion products are in comparison to the harmonic distortion product. The great difference between the amplitudes of the harmonic and IM distortion products is due to the 6-dB increase in the effective recording level when the second tone is added. When single-tone and twin-tone *peak* levels are the same, the overall third-order distortion levels are the same. As the test-tone frequency increases, the levels of the third-order harmonic and sum distortion products are reduced more and more, relative to the level of the difference-frequency products.

With close spacing between the fundamentals, f_1 and f_2, their output levels will be almost identical, as will the levels of f_3 and f_4, the third-order difference frequencies. As a result, distortion tests are possible almost to the very response limit of the deck. Music is made up of many more frequencies than just two, but twin-tone testing allows exercising a deck's entire audio band. H.A.R.

To avoid audible distortion, even from single peaks, the record level reductions listed here are a must.



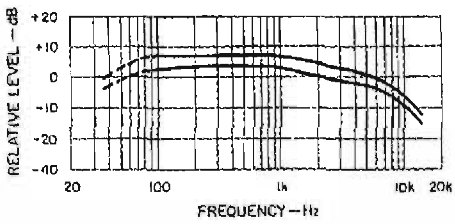


Fig. 13—Range of MOLs for 3% distortion for 13 IEC Type I tapes. High-frequency MOLs shown are referred to rms level of twin-tone signal re: Dolby level; see text.

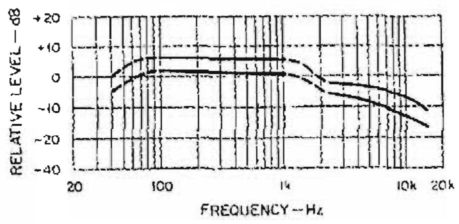


Fig. 14—Same as Fig. 13 but for 15 Type II tapes.

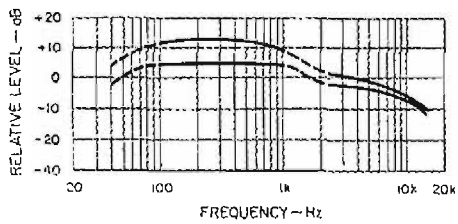


Fig. 15—Same as Fig. 13 but for seven Type IV tapes.

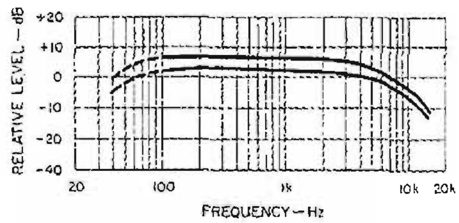


Fig. 16—Range of MOLs for 3% distortion for 13 IEC Type I tapes. High-frequency MOLs shown are referred to peak level of twin-tone signal re: Dolby level; see text.

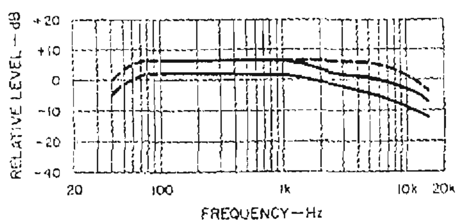


Fig. 17—Same as Fig. 16 but for 15 Type II tapes.

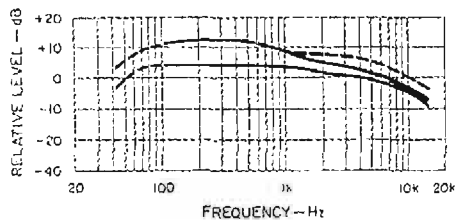


Fig. 18—Same as Fig. 16 but for seven Type IV tapes.

I thought that the MOL curves should provide a level-limiting curve for the spectral envelopes. I did have some question about what happened at all frequencies with broadband signals. I used a Type I tape with the Nakamichi CR-7A deck and ran a compression test with pink noise. The first test here used flat pink noise (20-kHz roll-off), with the input level adjusted in 1-dB steps from −10 to +20 dB relative to meter zero. I used a special dual attenuator, with one section increasing input level and the other section decreasing playback level. In this way, the level to the RTA was constant except for effects from compression.

The bottom trace of Fig. 19 shows the result of this test. The flat upper edge of the trace shows the flat frequency response obtained when recording at −10 dB; the bottom edge shows the response when recording at +20, where output was actually lowered due to compression/saturation. Notice how much greater this effect was at the highest frequencies than at 2 kHz and below.

For the next test, I used equalization to shape the pink noise to match the MOLs for the particular tape and recorder in use. The top trace of Fig. 19 shows the result of adjusting the input over the same meter range, from −10 to +20 dB relative to meter zero. With this shaping of the "music" to match the MOL limit across the whole band, the compression was very much the same in each of the 30 third-octave bands, as shown by the close parallel between the upper and lower edges of the trace.

The next step was to overlay the MOL curves on the eight spectral envelopes (from Figs. 5 through 12) for guidance on what record levels would be possible. The guidelines given here assume the meters are peak responding, with the level first set to match the 400-Hz MOL figure. In this case, there would be a downward level adjustment to prevent *any* part of the envelope from protruding above the MOL limit.

Figures 20 and 21 demonstrate how I did this. The top curve in each of these two figures is the best MOL curve for Type I tapes from Fig. 16. The bottom curve in Fig. 20 is the peak-level spectral envelope from Fig. 7. Notice how low the recording level would have to be set to prevent distortion at 40 Hz. Figure 21 shows that there is a better match to the deck/tape's MOLs with the envelope from Fig. 11. The actual limit occurs slightly above 15 kHz. Pushing the level higher, until this envelope intersected the MOL curve at 15 kHz, would not generate much in-band *harmonic* distortion products. But there would be high-level, third-order, difference-tone *IM* distortion products that would be in band.

Table II lists the required dB corrections in maximum meter readings relative to the 400-Hz MOL limit, to prevent noticeable distortion anywhere else in the entire band. The Table shows where the MOL limit occurs and gives the results for both the best and the worst MOLs shown in Figs. 16, 17, and 18. The envelopes in Figs. 5 and 6 show a concentration of energy in the middle of the music spectrum. The required adjustments, in most cases, are relatively small. The high level of the low organ notes (Fig. 7) requires a sizable reduction in the overall recording level to limit distortion. The spectrum is flatter in Fig. 8, but the required reductions are also large in all cases.

The high-level drum beats at 50 Hz (Fig. 9) would be hard to catch, and a large level reduction is required here also. The reductions are somewhat less with the envelope shown in Fig. 10, but they are important, reaching limits at one end of the spectrum or the other, depending on the shape of the MOL curve. The pop/rock CDs (Fig.

11) have peak band levels that are flat for most of the audio band but roll off at the extremes. For this type of music, the recordist needs to make some downward adjustment, but less than would be necessary for a number of other types of music. The required compensation for the cannon shots in the "1812" Overture (Fig. 12) cannot be defined as accurately as for the other envelopes. But it appears that the maximum meter level should be about 10 dB lower than the 400-Hz MOL limit for the tape/recorder used.

If the audiophile sets his goal to be the prevention of 3% distortion at *any* time, *even on a single peak or two*, the adjustments of level listed in Table II are *musts*. For lower quality decks, the reductions should actually be greater. If we consider how great some of the reductions should be, it is quite discouraging. It just won't seem right to set the levels to less than meter zero on peak-responding meters, and even lower on VU-type meters. Some of the envelopes require this, however, and the distortion *will* be low. Let me suggest that the envelopes and Table II provide guidelines which point out where distortion will start with increasing record levels. By all means, the recordist should listen carefully, for it may be difficult to accept noise that goes with really low distortion.

Interesting details can be gleaned from the Table. For example, notice that with Fig. 11 the limit for Type II and IV tapes is *not* at the frequency extremes but at 2.5 or 3.0 kHz. Notice also that in many cases—and especially when using the Type IV tapes—a greater adjustment is needed for the tapes with the *best*, as opposed to the worst, MOLs. This is the result of the shape of the MOL curves: The best MOL curves drop off noticeably from low to high frequencies, but the worst MOL curves are relatively flat.

The use of 120-µS equalization with Type II and IV tapes (see Figs. 17 and 18) did not provide as much of an advantage as I had thought it would. There was really no advantage in an equalization change for the envelopes in Figs. 5, 6, 7, 9, and 12. The improvement in record level for high-MOL Type IVs was 3 dB for the envelope of Fig. 8 and 3.5 dB for the envelope of Fig. 10. The improvement was about 2 dB for both Type II and IV tapes with the envelope of Fig. 11.

All of the preceding material relating to tape/recorder performance has been for operation without noise reduction. What will noise reduction do to help remove at least some of the limitations discussed here? First, let's take a look at what Dolby C NR might do. In the past, I have shown many high-level

TABLE II

Required corrections to maximum record level allowable (400-Hz MOL reference) to limit distortion.

		TYPE I		TYPE II		TYPE IV	
Fig. No.	MOLs	Freq. of Limit	Corr., dB	Freq. of Limit	Corr., dB	Freq. of Limit	Corr., dB
5	Best	1.0 kHz	−0.5	1.0 kHz	0.0	1.5 kHz	−3.0
5	Worst	1.0 kHz	−1.0	1.0 kHz	0.0	1.2 kHz	−0.5
6	Best	200 Hz	−0.3	200 Hz	0.0	1.5 kHz	−1.5
6	Worst	100 Hz	−1.0	200 Hz	0.0	200 Hz	0.0
7	Best	40 Hz	−8.0	40 Hz	−7.5	40 Hz	−8.5
7	Worst	40 Hz	−8.5	40 Hz	−7.5	40 Hz	−7.5
8	Best	15 kHz	−8.5	40 Hz	−7.5	13 kHz	−10.0
8	Worst	40 Hz	−8.5	40 Hz	−7.0	40 Hz	−7.5
9	Best	50 Hz	−7.5	50 Hz	−7.5	50 Hz	−9.0
9	Worst	50 Hz	−7.0	50 Hz	−7.0	50 Hz	−7.0
10	Best	15 kHz	−6.0	50 Hz	−3.5	11 kHz	−7.5
10	Worst	15 kHz	−5.0	50 Hz	−3.5	50 Hz	−3.5
11	Best	15 kHz	−4.5	2.5 kHz	−2.5	3.0 kHz	−7.0
11	Worst	15 kHz	−2.0	3.0 kHz	−1.5	2.5 kHz	−1.5
12	Best	40 Hz	−10.0	40 Hz	−10.0	40 Hz	−10.0
12	Worst	40 Hz	−10.0	40 Hz	−10.0	40 Hz	−10.0

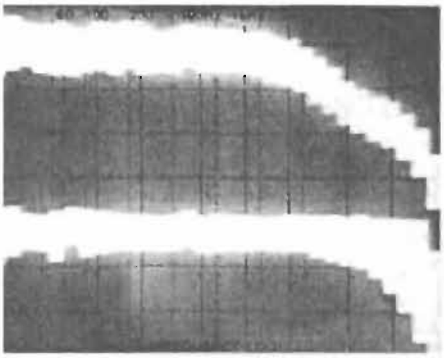
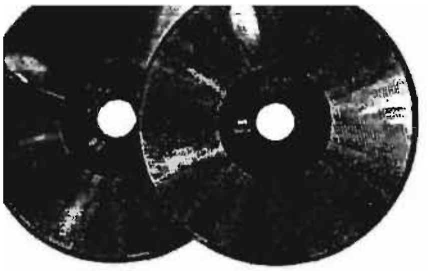


Fig. 19—Compression test using Type I tape on Nakamichi CR-7A. Record level was varied in 1-dB steps from −10 to +20 dB. With normal pink noise (bottom trace), response was flat at low recording levels (upper edge of trace) but compressed at higher levels (lower edge), especially at higher frequencies. With pink noise shaped to match the MOL curve (top trace), frequency response of uncompressed signals (upper edge) and saturation-compressed signals (lower edge) was almost the same. Vertical scale: 5 dB/div.

responses, especially for swept sine waves, that evidenced obvious and worthwhile headroom extension with Dolby C NR. After learning more about the limitations of falling high-frequency MOLs, I had to wonder about the extent of the effects of this NR in actual recording. The first thing I did was to check what the meter indications were for a Dolby-level calibration tape and then for pink noise at the same rms output voltage. The Dolby level from the tape read correctly at meter zero, but the deck's peak-responding meters bounced between +4 and +7 dB with the pink noise.

I then looked at record/playback response—with Dolby C NR, over a range of levels with pink noise—both flat and with response shelved at −5 dB above 3 kHz. While looking at playback after rewinding, I switched the NR in and out, both to see the result with NR and to check what the encoder responses were at these levels. There was some anti-saturation shaping of the encoder response, starting just above Dolby level.

Figure 22 shows record/playback responses using pink noise with a very high record level. The responses shown in the top trace resulted from using flat pink noise. The response without NR is the middle portion of this trace. Dolby B NR caused a somewhat greater roll-off at the highest frequencies, but Dolby C NR secured a very obvious reduction of the roll-off. The responses shown in the bottom trace



Pop/rock proves fairly easy to record well, while the "1812" Overture would be hard to meter properly.

are from using pink noise rolled off to match the tape/recorder MOL curve. The record level was increased to match that for flat pink noise. Even with this rolled-off signal, switching to Dolby C NR is somewhat helpful here. I should note that these tests do not recreate any particular music condition and just indicate the benefit of antisaturation.

In a tape deck having dbx II NR, the compansion system's zero-gain point should be set at about −10 dB or perhaps slightly lower. When this is done, dbx II NR would help in preventing high distortion with most of the spectral envelopes shown. For some recorders in particular, however, dbx II NR has shown rather high distortion and rolled-off response in the region of low organ notes and bass drum beats. Post-playback equalization might help correct the roll-off in some of these cases.

Some readers might like to find where the 400-Hz, 3% distortion point is just by looking at the playback waveform. Figure 23 shows how this can be done. There are two overlaid waveforms: The good sine wave is from the signal source, and the squashed one is from the deck playback. To show this, you need a signal source (around 400 Hz is best) and a two-channel oscilloscope. The oscilloscope should have combination synchronization so that both the source and playback waveforms (which are read at slightly different times) can be aligned.

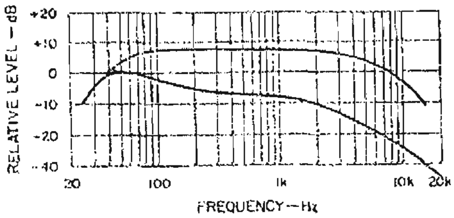


Fig. 20—Comparison of peak-spectrum envelope of organ music (Fig. 7) with Type I, best MOL curve shows that low-distortion recording limit occurs at 40 Hz.

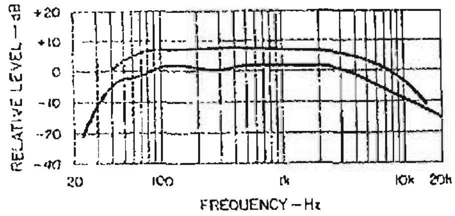


Fig. 21—Comparison of peak-spectrum envelope for pop/rock music (Fig. 11) with Type I, best MOL curve shows low-distortion limit at 15 kHz.

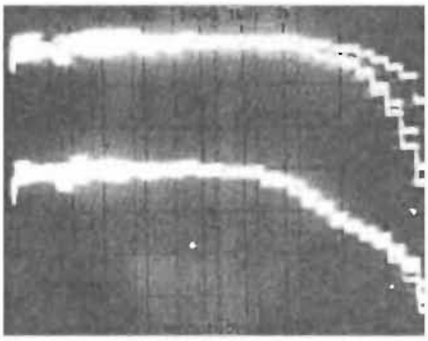


Fig. 22—Effects of Dolby NR on response at +20 dB recording level for flat pink-noise input (top trace) and MOL-shaped pink noise (bottom trace). Where traces split, at right, upper portion is with Dolby C NR, middle portion is without NR, and lower portion is with Dolby B NR; see text. Vertical scale: 5 dB/div.

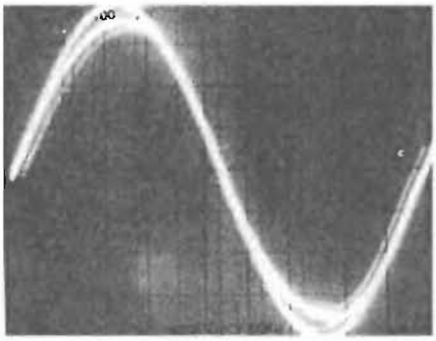


Fig. 23—Shape of playback sine wave with low distortion (larger trace) and with 3% distortion (smaller trace); see text.

The source-waveform gain is set to make it fill the full eight divisions. The gain for playback is adjusted to make its waveform match the source waveform as exactly as possible over the straighter portions, with some vertical positioning perhaps needed for accurate alignment. The distortion is quite close to 3% when the playback waveform is squashed half a division at both top and bottom. This is not an *exact* indication, but it will help to define reference levels.

It is good for the recordist to know where on the meters the low-frequency (250 to 400 Hz), 3% distortion limit is for each of the tape formulations used. This article is a first try at obtaining a better understanding of the range of peak-level spectral envelopes for many types of music. With continuing examination of additional CDs, I may find a need for other envelopes. On the other hand, the envelope in Fig. 7 is somewhat similar to that of Fig. 9, and the levels of the envelope in Fig. 12 are close to matching those at the frequency extremes in both Figs. 7 and 9.

The MOL curves which have accompanied my *Audio* tape tests are based on a 3%, third-order distortion limit, measuring harmonic distortion for the lower frequencies and twin-tone difference distortion for the higher frequencies. Of course, 3% is certainly *not* a low level of distortion. In some of my own listening, I thought that the results were much better when I limited the record level to what would amount to about 1% distortion, as indicated on peak-responding meters; this level limit would be 5 dB lower than the MOL criterion. Now that I have looked at the various spectral envelopes, I will be more discerning about what to expect. What I thought was my 1% distortion, peak-meter limit may really be close to a 3% limit for momentary peaks at any point in the band.

I do hope this discussion will help recordists become more aware of the challenges provided by various types of music. In digital recording with either R-DAT or PCM, the exact spectral shapes are less important, but good recording practice is still required to ensure that sudden peaks do not drive the system into overload. Primarily, these guidelines should help you find the best record-level settings for the various possible analog tape/recorder combinations. ◢

Whither the Stereo LP?

JOHN EARGLE

From its introduction in 1957 until 1983, the vinyl LP was the major medium for stereo sound in the home. It was in 1983 that the Philips cassette eclipsed the LP in unit sales.

You may recall that the cassette was introduced around 1966, and at that time was of little more than dictating-machine quality. Through a series of developments in tape technology, electronics design, and noise reduction, the cassette emerged as a remarkable medium. During the 1970s, it effectively knocked open-reel tape out of the consumer marketplace.

As a home recording medium, the cassette eventually reached quality levels high enough to satisfy even fussy users. And as a carrier of recorded program material, it survived the rigors of high-speed duplication. In time, the base of cassette players in the home, and especially in the automobile, increased to the point where tape decks rivalled turntables, and the cassette became the dominant medium for recorded music. The important thing to note is that it took some 17 years for this to happen.

The CD was introduced in 1982, and its phenomenal growth is something we have all witnessed in recent years. In 1982, no one could have foreseen that it would eclipse the LP in dollar sales in 1987 and overtake it in unit sales in 1988. The CD has accomplished in five years what it took 17 years for the cassette to do.

The big question is this: Is the LP truly doomed? Or are we looking at a realignment of marketplace tastes and priorities, in which the LP will settle into a new but lower volume plateau? There are many commercial factors to be considered here, and I will try to sort them out.

First, we must realize that the record industry thrives as much on new technology as it does on new artists and new music. The record industry reached a peak in 1978, and the slump

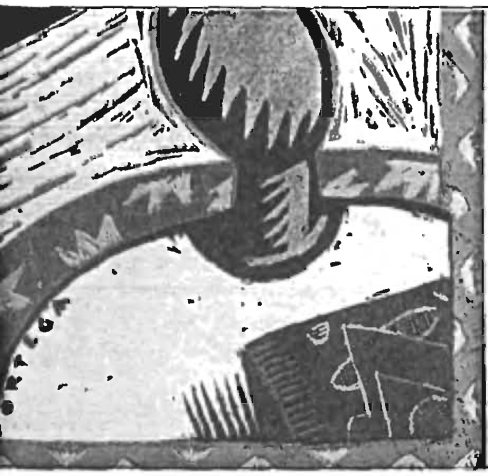
that followed was devastating. The last five years have seen new growth in the industry, and it has been largely fueled by enthusiasm for the CD. Last year, in fact, was the banner year in the history of the record business.

CD mania has carried through into consumer electronics and record retailing as well, the high prices of discs notwithstanding. In an effort to make room for new CD releases, many large retail chains have cut back on LP bin space, and many record companies have adopted a CD-only policy for classical releases. Record retailers want to maximize the yield of every square foot of store space, and record manufacturers are always looking for ways to delete slow-moving product and reduce their catalog size.

A related factor, at least in the United States, is the tooling down of LP manufacturing. In the last two decades, RCA and Capitol/EMI have between them closed six LP pressing plants, leaving the business largely to CBS, WEA (Warner/Elektra/Atlantic), and a group of relatively small, independent pressing houses. Ultimately, it may be the independents who keep the LP art going.

It would appear that economic factors are hastening the demise of the LP when, in fact, there may be a market

ILLUSTRATION: JOSÉ ORTEGA



ANALOG TAPE RECORDING NEED NOT SUFFER FROM COMPARISON WITH ANY DIGITAL RECORDING SYSTEM BASED UPON TODAY'S RECORDING STANDARDS.

which is not quite ready to die. Given a stronger dollar, there would be plenty of high-quality imported LP product to fill some of this need. Many American record manufacturers, however, feel that between cassettes and CDs, they are pretty well covering the important retail bases.

If there is to be an ongoing market for the LP, it will be that which is fueled largely by high-end audiophile tastes. To many, this must seem paradoxical in an age of rapidly improving digital technology, but it is the case nonetheless. A trip through any high-end hi-fi show will reveal a large number of expensive turntable/cartridge combinations, most reproducing superb sound. (The same, I might add, holds for CD demonstrations which use later generation players and well-recorded program material.) In time, many of the LP's proponents may adopt the CD, but for the present, they are deeply committed to the older medium.

What, then, are the characteristics of the LP which make it such a favorite? And how good can the LP really be? In order to answer these questions, let's take a short look at the history of the medium.

Alan Blumlein cut the first 45/45 stereo disc in 1931, but it couldn't be played at that time. Many in the industry felt it would never be practical, and it was more or less hidden in EMI's archives for many years.

It wasn't until 1947 that the mono LP became a reality, and with it came a new approach to phonograph cartridge engineering. This sparked a reexamination of stereo disc problems, and by the mid-1950s, stereo had become a fact in the laboratory. By 1957, there was product on the market. It didn't always sound good, though, and there remained much work to be done. One by one, the significant problems were solved.

In the early 1960s, the discrepancy between vertical cutting and playback angles was identified and the adjustment made. About that same time, the problems of tracing distortion were addressed by "predistorting" the groove electronically. Later, this technique was abandoned in favor of elliptical playback stylus design, which further alleviated high-frequency loss at inner grooves. All along, there were improvements in vinyl formulations for pressing, resulting in lower noise. The American industry, however, was never able to maintain the generally high levels in pressing quality routinely met by Japanese, German, and Dutch manufacturers.

By the late 1960s, direct-to-disc recording had established new limits for the stereo LP by circumventing the tape recorder altogether and simplifying the overall recording chain. Then came the golden era of $18 LPs. Throughout the 1970s, many small, audiophile-oriented companies regularly turned out superior disc product, with most of the pressing done in Japan. Denon and JVC releases were imported from Japan, and domestic labels such as Telarc and Delos were putting out superior product, via Soundstream digital sources, long before the majors realized that a market for a "super record" existed. Mobile Fidelity took the

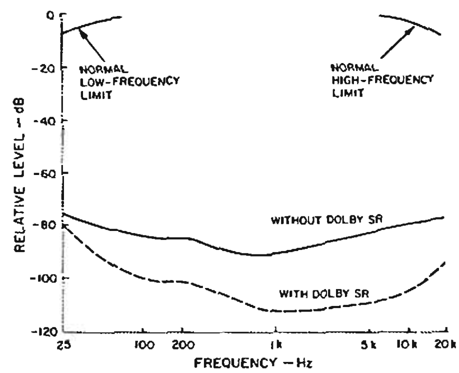


Fig. 1*—Signal space available for standard analog recording on 15-ips tape, with and without Dolby SR. The top solid curve shows the normal limits in analog tape systems due to equalization requirements at low frequencies and the risk of tape saturation at high frequencies. Dolby SR effectively compensates for these problems, creating overall flat power-bandwidth capability at full signal level. Upper frequency boundaries, with or without SR, are not absolute limits. Low-level noise readings were made in third-octave bands; thus, the data indicates the effective dynamic range of the system.*

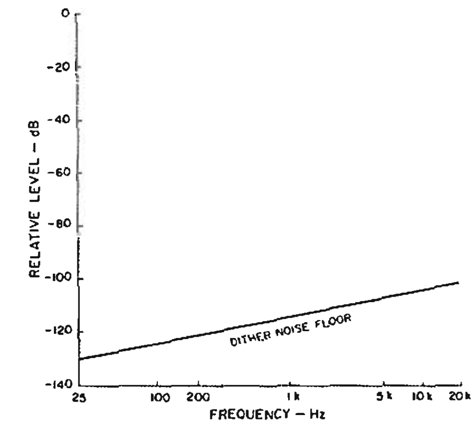


Fig. 2*—Signal space for a 16-bit digital system operating with a 44.1-kHz sampling rate. Once converted to the digital domain, these characteristics can be carried through to the end product—in this case, the CD itself. A dithered noise floor has been added to the digital input signal to linearize the system's low-level performance. Low-level noise readings were made in third-octave bands.*

boldest stand of all in leasing master tapes from the majors, transferring them to disc at half speed with loving care, and making much better product than the larger record companies knew was possible.

But back at the studio and the laboratory, there were still many nonlinearities in the overall disc transfer process remaining to be solved. Some of these could be dealt with and compensated for by creative "adjustment" of transfer processes and even of studio techniques themselves. (See my discussion of this in "Do CDs Sound Different?" in the November 1987 issue.)

Digital recorders had come into the studio during the mid-1970s. While solving the knotty problems of alignment and response stability, these machines did not provide the natural cushion at high recording levels which the analog machines did. It was at this point that many audiophiles developed their first doubts about digital recording technology in general.

When the earliest CDs hit the market, these doubts were reinforced. Many of the new discs did sound strident. It has only been in recent years that the problems with digital have been solved, again through adjustments in recording philosophy as well as improvements in recording and playback processes.

How good can the stereo LP be, given today's state of the art? It can be a superlative medium. An all-analog recording chain of the highest quality would certainly begin in the studio with Dolby Spectral Recording and analog tape running at 15 ips. Dolby SR is that company's latest generation in complementary pre- and post-processing for noise and distortion reduction at the initial recording stage, and its overall dynamic range capability is shown in Fig. 1. It effectively adds about 25 dB of dynamic range to the normal performance of a tape recording channel. (The zero reference level in Fig. 1 is established as the normal maximum operating level the engineer wishes to reach. Normally, with current tape formulations, this is set at around 200 nWb/m. Any modulation over this reference level will fall into the analog cushion range. It is best to avoid this, but there would be no catastrophe if occasional forays into this area were unavoidable.)

Figure 2 shows the signal space of a 16-bit digital system operating at a sampling frequency of 44.1 kHz. The

Dolby Spectral Recording

Most readers of *Audio* probably have some familiarity with the basics of noise reduction, if for no other reason than that they have had these functions on their cassette machines for years. The heart of most noise-reduction systems is signal compression during recording and complementary signal expansion during playback. Complementary equalization is also an important part of the noise-reduction process, as is the selection of proper attack- and release-time characteristics for the various gain manipulations.

Dolby Laboratories introduced its A-type noise-reduction system for professional use in the late 1960s, with the consumer B- and C-type systems following later. Nothing is free, and the price paid for the dynamic range extension of the earlier noise-reduction systems was occasional audibility of the compression/expansion actions when those actions were "unmasked" by the specific nature of the input signal. Dolby Laboratories analyzed the audibility problems of the earlier systems and, in 1986, introduced yet another generation, called Spectral Recording, which embodies many of the advantages of the earlier A-type system plus the sliding-band techniques introduced in the B- and C-type systems.

The action of Dolby SR is basically to analyze the spectral composition of the input signal on a continuous basis and to define a protective dynamic "gain envelope" for the signal, such that no part of it will drop below the audible noise threshold of the recording medium. On playback, the inverse action is carried out, and the original signal dynamics are restored. The overall improvement in dynamic range over the non-Dolby SR recording channel is about 24 dB, from 2 to 8 kHz.

At the lowest recording levels, the SR system is in its full boost mode in recording and full attenuation mode in playback. What this means is that the inherent tape noise floor of the recorder is lowered by as much as 25 dB in the range where the ear is most sensitive to noise. The long solid curve in Fig. 1 shows the normal limit for tape without Dolby SR, while the dashed curve shows the effect of Dolby SR. The spectral characteristics have been plotted based on third-octave noise measurements and, thus, are some 14 to 15 dB lower than wideband noise measurements. To the extent that we can hear pure tones in the midband which are some 12 to 15 dB below a wideband noise level, the total spread between upper and lower bounds in Fig. 1 gives an accurate indication of the subjective dynamic range capability of the recording system. The gradually fading boundaries indicate that the limits are not firmly fixed and that recorded modulation may exceed them slightly.

In Fig. 1, spectral data on the analog recording system is from Camras [1, page 351], modified by recent tape improvements which I have described [3, page 305]. The Dolby SR data is from Dolby [2].

As with all Dolby NR systems, a full-level signal goes through the system almost unimpeded, with little action by the signal-analysis and gain-change circuitry. It is only when the signal drops to lower levels that the circuit complexities come into play.

With good analog tape recording channels already pushing a dynamic range of 68 to 70 dB, the additional 25 dB afforded by Dolby SR gets it into the range of 90 to 95 dB, which is comparable to digital recorders. Some engineers say it is even better.

Is there a price for all this improvement? Not as far as I can tell. I auditioned a pair of Dolby SR retrofit cards with my Dolby A units, and I could not make the action audible—no matter how hard I tried. *J.E.*

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3. Eargle, J., *Handbook of Recording Engineering*, Van Nostrand Reinhold, New York, 1986.
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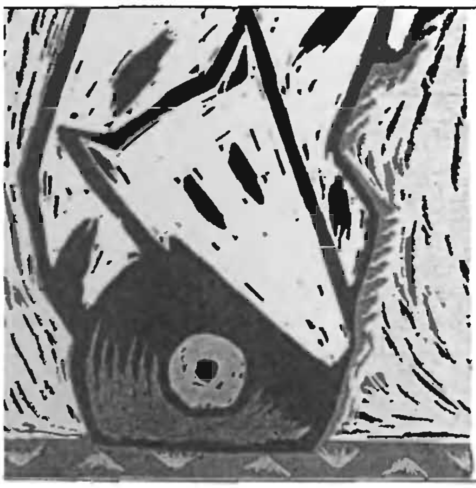


WILL THE IMPROVEMENTS IN ANALOG RECORDING TECHNIQUES, SUCH AS DOLBY SR AND DMM, GIVE THE LP A NEEDED REPRIEVE? I SUSPECT THEY WILL.

upper boundary is firmly established, due to the hard clipping of the system once full modulation is reached. Also, the upper frequency limit of the system is firmly established at 20 kHz, due to the filter demands of anti-aliasing.

A comparison of Figs. 1 and 2 shows that, at some higher frequencies, the signal space of the Dolby SR-equipped tape recorder actually exceeds that of the digital system. Furthermore, the upper frequency limit of the Dolby SR analog system is not limited to 20 kHz. Actually, Dolby SR could be used with 30-ips tape recorders, with overall system response well beyond 20 kHz.

Analog tape recording, then, given the dynamic code/decode action of Dolby SR, is effectively a match for current digital standards and does not take a back seat at all in any listening comparisons. However, subsequent analog transfers will show some degree of degradation, while subsequent transfers of digital recordings can be virtual clones of the original.

Moving on to the stereo LP itself, the quality improvements inherent in Direct Metal Mastering are substantial. One of the best papers covering DMM was published in *Audio* (April 1987). In that article, authors B. V. Pisha and George Alexandrovich covered the technology in detail. Figures 3 and 4 represent measurements presented in this article and data regarding the performance of disc systems at high frequencies. Note that DMM provides a lower noise floor for the system as well as extended high-frequency response.

DMM accomplishes these improvements through use of a precision cutting tool which has no burnishing facets, cutting directly into a copper surface instead of the conventional lacquer-coated disc. Lacquer is a complex organic mixture and introduces its own distortion and noise into the cutting process. By comparison, the copper is dimensionally stable and produces a more accurate signal, virtually free of bothersome groove echo. Further, the number of replicated generations between the master and the finished product can be reduced, with consequent improvement in noise.

While the superb low-noise characteristics of Dolby SR are not carried through into the finished disc product, the resulting noise floor is quite acceptably low. Partisans of extended frequency response will be quick to point out that the response of the finished discs, at outer and middle diameters, can easily exceed 20 kHz.

Time will tell if the improvements in analog processes, both tape and disc, will be able to forestall the demise of the LP and give it a reprieve. I suspect they will. Like so many other fans of the CD, I have a vast collection of LPs going back to the beginning of that art. Many items in my collection will never see the light of day on CD, so I do a good bit of listening to both formats. I am comfortable with both mediums. However, given the choice of buying Sheffield's Moscow Sessions on CD (made from digital masters) or LP (made from analog masters), I opted for the CDs. But I might, at least some of the time, be fooled in a blind A/B comparison!

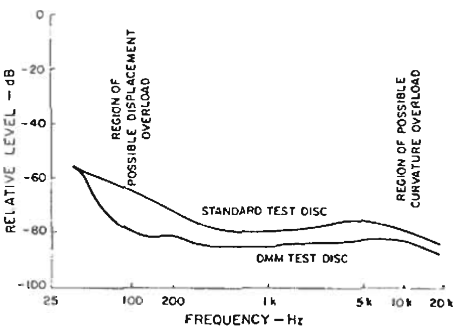


Fig. 3*—Signal space for DMM and standard stereo LPs. The upper limit shown is determined by a maximum recorded stylus velocity of 10 cm/S, as modified by RIAA pre-emphasis. Low-level noise readings were made in third-octave bands. (After Pisha and Alexandrovich.)*

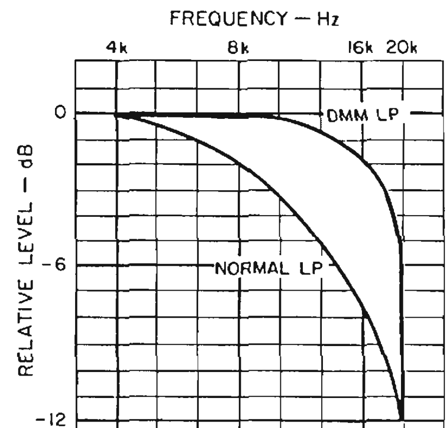


Fig. 4*—Reduction in high-frequency losses at moderate playback levels when DMM is used. The losses are minimized due to the absence of burnishing facets on the DMM cutting stylus. Curves shown are from playback of actual pressings. (From Pisha and Alexandrovich.)*

INTRODUCING DOLBY S-TYPE NOISE REDUCTION

KENNETH GUNDRY AND JOSEPH HULL

Since the introduction of the Dolby SR (spectral recording) process in 1986, the professional music recording, broadcast, and cinema industries have equipped more than 35,000 analog audio channels worldwide with this recording enhancement system. Dolby S-type, a new noise-reduction system from Dolby Laboratories for improving the popular home audio cassette, uses several of Dolby SR's proven techniques to achieve similar ends—increased headroom, lowered distortion, and greatly reduced noise. Dolby S-type is less complex and less costly than Dolby SR. Among other reasons, this is possible because home listening levels are lower than those in studios, and because the spectral content of cassette noise differs from that of open-reel tape. However, both systems share such objectives as freedom from audible side effects, and are based upon similar principles of operation.

Kenneth Gundry, Dolby Laboratories' Principal Staff Engineer, has been with the company since 1971 and is currently in charge of developing a Dolby S-type encoder for use in production of prerecorded cassettes. He is also the inventor of the Dolby HX (headroom-extension) system and codeveloper of the Dolby ADM (adaptive delta modulation) digital system.

Joseph Hull, resident wordsmith at Dolby Laboratories, originally joined the company in 1978, after working as Director of Communications at Advent. As a freelancer, he wrote manuals and product literature for Boston Acoustics, Cambridge SoundWorks, Kloss Video, Lucasfilm, and Mitsubishi.

The authors are indebted to their colleagues Stan Cossette, who has spearheaded the development of Dolby S-type, and Ray Dolby, who conceived the principles upon which it is based.

Principle of Least Treatment

Complementary noise-reduction systems, such as those developed by Dolby Laboratories, boost the signal as it is recorded (compression) and then reduce the boosted signal by the same amount (expansion) in playback; tape noise is also reduced by the same amount. The original signal theoretically survives the complementary process unscathed, as opposed to playback-only NR systems, where the original signal is inevitably damaged in the attempt to remove noise retroactively.

The companding action need not be the same at all frequencies. For example, in cassette recording, tape hiss so predominates that it is desirable to concentrate the noise-reduction action at higher frequencies. Yet whatever the spectral effect of the action, it must be confined to lower level signals to prevent overloading the medium (e.g., saturating magnetic tape) when high-level signals occur.

Dolby A-type NR uses four fixed-action bands; Dolby B and C use one sliding band. Dolby S uses both sliding and fixed bands.

At first glance, designing a system to operate only at lower levels makes sense, because high-level signals are assumed to mask noise. Wide-band companders, which boost the entire frequency spectrum at low levels upon recording and lower it at playback, work on this assumption. Unfortunately, it is not entirely valid.

On quiet signals, noise reduction is indeed effective, because full recording boost is applied (Fig. 1A). However, because it is necessary to prevent overload by not boosting higher level signals as they are recorded, these systems allow the noise level to go up upon playback as signals get louder, an effect called noise modulation (Fig. 1B). The higher noise can be heard under certain circumstances, because it is only at and near a signal's frequency that masking occurs. If the music is loud in only part of the spectrum, noise will be heard in the other parts, where there is neither masking nor boost (i.e., no NR effect). The result is annoying changes in noise level concurrent with changes in signal level.

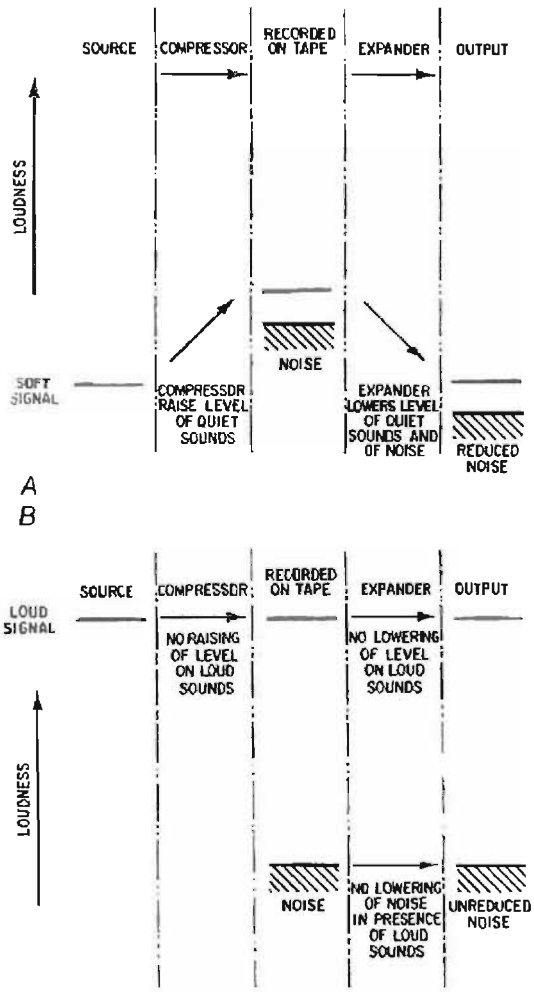


Fig. 1—With a wide-band compander noise is reduced in the presence of soft signals (A) but is allowed to rise back to its original level in the presence of loud signals (B)

The ideal noise-reduction system, on the other hand, would act *wherever* signals fall below a certain threshold, even when there are loud signals elsewhere in the spectrum. Ideally, with a loud rap on a bass drum, there would be no record boost on the drum itself, to prevent overloading the recording medium. But there would be full boost, and therefore effective noise reduction upon playback, over the rest of the spectrum.

We call the application of constant gain wherever there are no high-level signals, even in the presence of such signals elsewhere in the spectrum, "the principle of least treatment." Reductions in record boost to prevent overload should be confined just to those parts of the spectrum where loud signals, and thus natural masking, occur (Fig. 2A). This results in audibly consistent noise reduction. By contrast, in the presence of any loud signal, a wide-band compander reduces the record boost *throughout* the spectrum, resulting in audible changes in the NR effect (Fig. 2B).

The Dolby A-type professional noise-reduction system strives for the ideal by splitting the spectrum into four bands with independently acting companders. Thus, with a loud bass drum, Dolby A-type NR does not operate in the low-frequency band where masking occurs, but does act in the higher frequency bands where there is no masking. In Dolby B-type and C-type NR, a single companding band of frequencies slides up out of the way of the bass drum, keeping the NR effect active at higher frequencies where tape hiss is audible. In Dolby S-type and Dolby SR, a combination of fixed and sliding bands, along with other new developments, results in the closest adherence yet to the principle of least treatment.

Benefits of Least Treatment

The major benefit of adhering to the principle of least treatment is a better recording system, virtually free of such side effects as noise modulation. However, the fact that high-level signals have little effect on low-level signals has significant additional advantages in the real world, where encoded recordings are subject to decoding errors.

Decoding errors can be divided into two categories. The first we might call "inadvertent," that is, errors resulting from frequency response or level changes introduced between original encoding and ultimate decoding with the same complementary system.

These are the kind of errors that occur, for example, when using a tape formulation for which the original recorder was not optimized. The other category we might call "deliberate," that is, errors that would result, say, from playing back Dolby S-type cassettes on a machine that is equipped only with Dolby B-type NR.

As a result of its adherence to the least-treatment principle, Dolby S-type is robust. If a tape is made on a recorder with less than perfect response, the listener is unlikely to notice anything wrong beyond the original imperfection itself (if indeed it is audible). If a tape is recorded with S-type and played with B-type, a critical listener may notice a reduction in dynamic range, which may even be desirable in a noisy environment such as an automobile. However, there is virtually no distracting "pumping" or other dynamic artifacts. Minimizing the effect of high-level signals on low-level signals, which eliminates the principal mechanism by which the ear detects the use of level-sensitive processing, may well become a factor in the software industry's consideration of releasing Dolby S-type prerecorded cassettes.

Action Substitution

"Action substitution," which was applied first in Dolby SR and is now being used in Dolby S-type, is a new development enabling closer adherence to the principle of least treatment. It results from combining both fixed-band and sliding-band techniques in a way that maintains their advantages while mitigating their disadvantages.

Figure 3A illustrates a sliding-band system. When high-level signals, if any, are relatively low in frequency, the band assumes the quiescent characteristic represented by the solid curve. If a higher frequency signal then comes along at a level high enough to require less boost, the band must slide up considerably, even to achieve only 2 dB less boost as shown (dashed curve). Thus, considerable noise reduction below the dominant signal's frequency is lost (as shown by the shaded area)—a disadvantage. However, above that frequency, the NR effect is essentially unchanged—an advantage.

Figure 3B illustrates a fixed-band system having the same quiescent characteristic, again represented by the solid curve. When, as above, the dominant signal is loud enough to require less boost, there is an overall reduction of boost (dashed curve) and an equivalent loss of NR effect (shaded area). This means that unlike the sliding band, the fixed-band system causes a loss of NR effect above the dominant frequency—a disadvantage. However, there is significantly less loss of noise-reduction effect *below* the dominant frequency than with the sliding band—an advantage.

At higher frequencies, where tape hiss predominates, Dolby S-type combines both sliding and fixed bands in a way which results in what we call "action substitution."

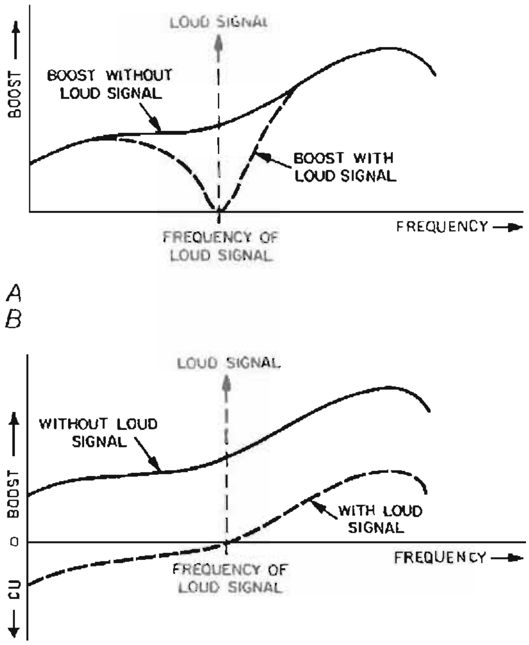


Fig. 2—The effect of high-level signals on cassette noise-reduction systems vary with the system used. In systems such as Dolby S-type, which follow the principle of least treatment (A), gain is reduced only at and near the frequency of a loud signal. Record boost, and thus noise reduction, is therefore unaffected except in a region masked by the signal itself. In a wide-band compander system (B), gain is reduced throughout the spectrum, regardless of the loud signal's frequency. Record boost, and thus NR, is therefore lowered or eliminated everywhere that masking does not occur

Figure 3C illustrates an action-substitution system having the same quiescent characteristic as the individual system discussed above (solid curve). When less boost is required (dashed curve), the action of the fixed band predominates below the dominant frequency, so that less NR effect is lost than when using a sliding band alone. Above the dominant frequency, the sliding band predominates, resulting in none of the NR loss which would occur with a fixed band alone. Thus, with action substitution the boost of low-level signals is more constant, as is the

NR effect, so that the system conforms more closely to the principle of least treatment.

Action substitution has an additional benefit. With a complementary NR system, changes in level introduced after the signal is initially processed can cause the playback processor to mistrack, that is, it may not act as a precise mirror image of the record processor. With a sliding-band system, a relatively small (and otherwise innocuous) level change introduced by the recorder at the dominant frequency can cause disproportionate, and potentially audible, decoder mistracking at lower frequencies. With Dolby S-type, however, the fixed band predominates at frequencies below the dominant frequency, reducing the potential for audible mistracking.

Modulation Control

In contrast to action substitution, modulation control, also developed originally for Dolby SR, deals with the effects of high-level signals outside the NR bands which need not, and should not, be boosted. With a sliding-band system, such high-level signals cause the band to slide up in frequency, out of their way. However, the higher the signal level at a given frequency, the further away the band slides. If left to its own devices, a sliding band can move so far away as to create a gap between its noise-reduction effect and the natural masking of the high-level signal, thereby causing a subtle form of noise modulation.

With a fixed-band system, dominant high-level signals nominally outside, but close to, the band can cause an undesirable reduction in the band's boost. This is because the filter used to create the bandpass cannot have an infinitely steep slope. If the dominant signal is strong enough, even quite far down the slope it will have the same effect as a lower-level dominant signal well within the bandpass. As with the sliding band, the high-level signal causes a reduction in NR effect where there should be none; that is, it causes noise modulation.

With Dolby S-type, a special technique called modulation control is applied to both the sliding and the fixed bands. It reduces the tendency of a sliding band to move further away from high-level signals than is necessary (Fig. 4A), and reduces a fixed band's tendency to react to high-level signals outside, but close to, the band (Fig. 4B). Thus, like action substitution, modulation control helps to keep all low-level signals in a more constantly boosted state in accordance with the principle of least treatment.

Spectral Skewing and Antisaturation

Spectral skewing and antisaturation techniques have been incorporated in Dolby S-type, as in Dolby SR. Spectral skewing consists of networks in the encoder which roll off the extreme low and high ends of the spectrum; complementary networks in the decoder restore flat response. The networks re-

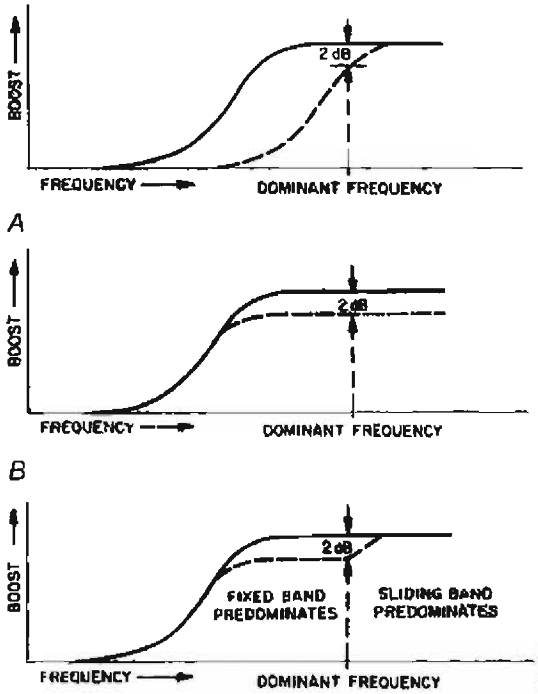


Fig 3—Effects of dominant signals within the NR band also vary with the system. To effect a given decrease in gain (2 dB shown) at the dominant frequency, the NR band of a sliding-band system (A) slides up, reducing NR at frequencies below the signal. In fixed-band system (B), boost (and thus NR) is reduced throughout the band by the amount of gain decrease required at the dominant frequency. Combining fixed and sliding bands (the "action substitution" system used in Dolby S-type NR) confines the gain decrease more closely to the dominant frequency, resulting in less loss of NR (C)

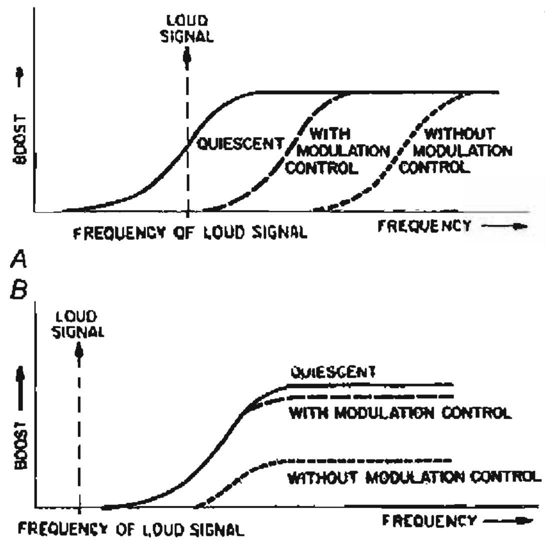


Fig. 4—Modulation control reduces the tendency of a sliding band to move further away from a high-level signal than necessary (A) and of a fixed-band system to reduce gain in the presence of a high-level signal at a nearby frequency (B).

If a tape is recorded with Dolby S-type NR and played back with B-type, you may only notice a reduction in dynamic range between the two

duce the dependency of the system's action on signals at the extreme ends of the spectrum, thereby reducing decoder mistracking as the result of response errors introduced by the recorder in those regions. Such errors include those at low frequencies caused by head bumps, and those at high frequencies caused by variations among tape formulations even within the same nominal category, and by head-azimuth variations between the machine on which a tape is recorded and those on which it is played back. While spectral skewing results in some loss of NR effect at those extremes, the ear is so insensitive to noise at the extremes that the benefits far outweigh the theoretical NR loss.

Antisaturation consists of high-frequency shelving networks which operate at high signal levels; complementary networks restore flat response at playback. The shelving significantly reduces the high-frequency losses and distortion caused by tape saturation, thereby significantly extending headroom and further reducing the likelihood of decoder mistracking. Antisaturation reaches lower in frequency than spectral skewing, so its effects are limited to high levels to prevent any audible loss of NR effect.

Antisaturation effects are also contributed by both the low- and high-frequency spectral skewing networks. The low-frequency network, for example, virtually cancels out the low-frequency boost imparted by standard 3,180-µS cassette equalization, resulting in a notable reduction in distortion on strong low-frequency signals. This improvement is possible because Dolby S-type provides noise reduction at low frequencies, without which eliminating the standard pre-emphasis would increase low-frequency noise. The combined effects of spectral skewing and antisaturation techniques at both low and high frequencies can be seen in Fig. 5, which illustrates Dolby S-type's overall encode characteristics.

Multi-Level Staggered Action

The principle of least treatment calls for a fixed gain, determined by the amount of noise reduction desired, wherever in the spectrum signals fall below some threshold. But there is also a need to fix the gain on signals which occur above a higher threshold, to reduce the effects of high-level overshoots. Overshoots are brief, exaggerated increases in level which occur during the time it takes for a compressor to react to a suddenly louder signal and start reducing the gain. At low signal levels, overshoots are of little concern: They are recorded onto the tape and then "undone" by the decoder at playback. But at high signal levels, overshoots from the encoder can get lost as a result of tape overload, resulting in distortion and decoder mistracking.

Therefore, changes in gain should occur only at intermediate levels, a characteristic we call bilinear compression and expansion (Fig. 6). To achieve this characteristic with

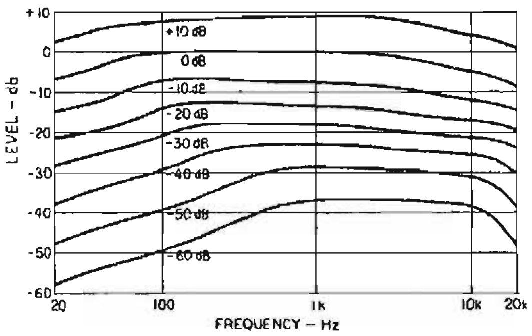


Fig. 5—Dolby S-type encode characteristics, showing the effects of spectral skewing and antisaturation at various input levels. The decode characteristics are the inverse of these curves.

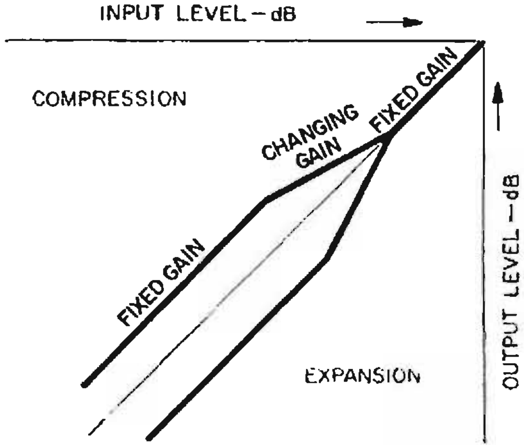


Fig 6—Idealized bilinear compression and expansion

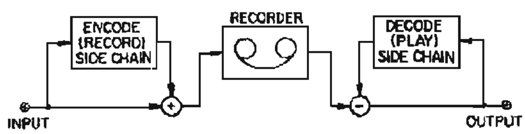


Fig. 7—In Dolby S-type, as in Dolby SR and the Dolby A, B, and C NR systems, processing takes place in a side chain, whose output is added to the main signal path for encoding and subtracted from it for decoding.

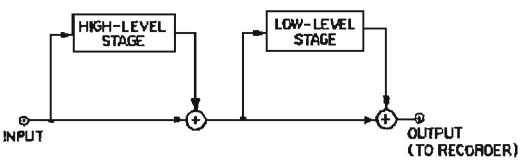


Fig 8—In Dolby S-type, two high-frequency stages, sensitive to different signal levels, are used in the side chain (encoder shown).

Combining fixed-band with sliding-band NR techniques maintains the advantages of both while mitigating disadvantages.

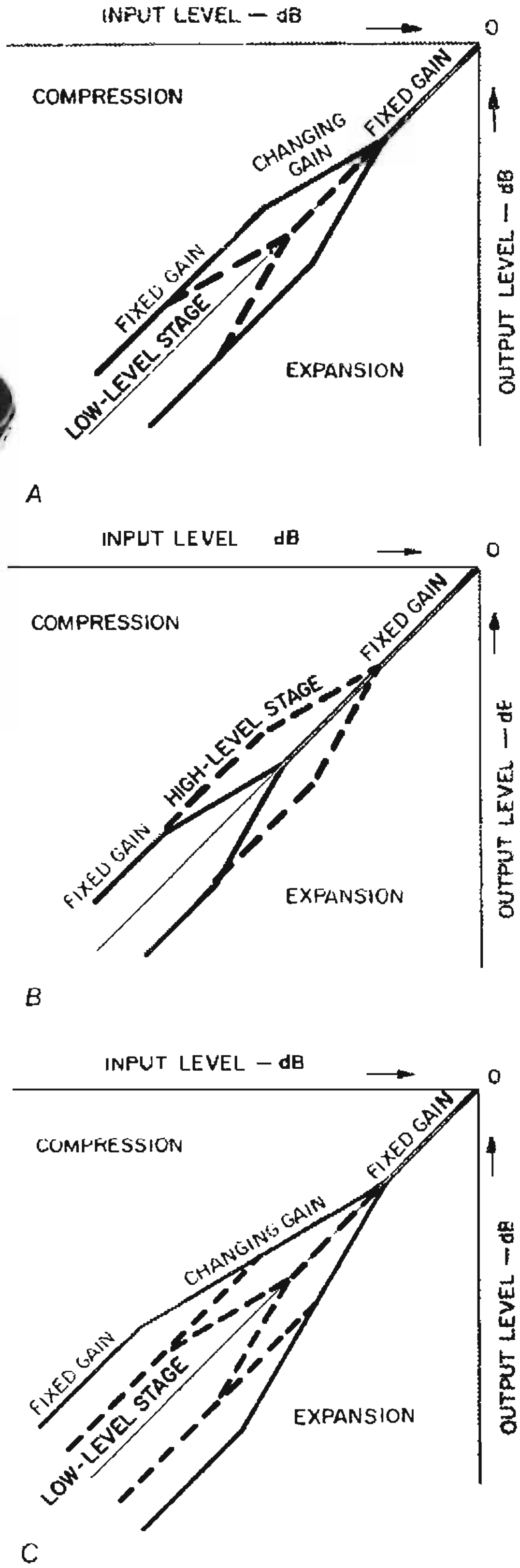


Fig 9 The Dolby S-type NR system prevents multiplication of compression ratios by staggering the signal levels at which gain changes occur, as shown for the high-level stage (A) and low-level stage (B). However, at low levels, the boosts of the two stages add (C) to provide more noise reduction.

Dolby S-type, as in all our systems, a dual-path circuit configuration is used whereby processing takes place only in a side chain, whose output is added to the main signal path for encoding and subtracted from the main signal for decoding (Fig. 7). At low input levels, the side chain's compressed output makes a significant contribution to the encoder's total output. Because the encoded signal is still comparatively low, the overshoots introduced are of little consequence, as described above. However, as the input signal level increases, the side chain's contribution lessens proportionally, so that the unprocessed main path predominates. Eventually a level is reached above which the side chain might as well not be there; changes in gain, and therefore overshoots, virtually cease.

However, as more boost is designed into the compressor to achieve more noise reduction, the levels below and above which gain is fixed tend to move lower and higher, respectively. Preventing the one from going too low and the other from going too high runs the risk of introducing too high a compression ratio, which could magnify response and level errors in the recorder and thereby increase the potential for decoder mistracking. Therefore, at the high frequencies, where cassette noise predominates, Dolby S-type uses *two* 12-dB companding stages connected in series to provide what we call "multi-level staggered action," a technique developed originally for Dolby C-type NR and refined for Dolby SR (Fig. 8). The use of two stages enables us to achieve more noise reduction and maintain the advantages of a bilinear characteristic, without introducing unduly high compression.

Each 12-dB stage has a bilinear compression/expansion characteristic. At the low signal levels where maximum noise reduction is desired, the boosts imparted by the two stages add to provide the desired 24 dB. However, the thresholds of the stages are staggered: In what we call the low-level stage, the levels above and below which gain is fixed are lower than those for what we call the high-level stage. As a result, their compression ratios do not multiply, and the signal is never subjected to a higher ratio than that of an individual stage (Fig. 9).

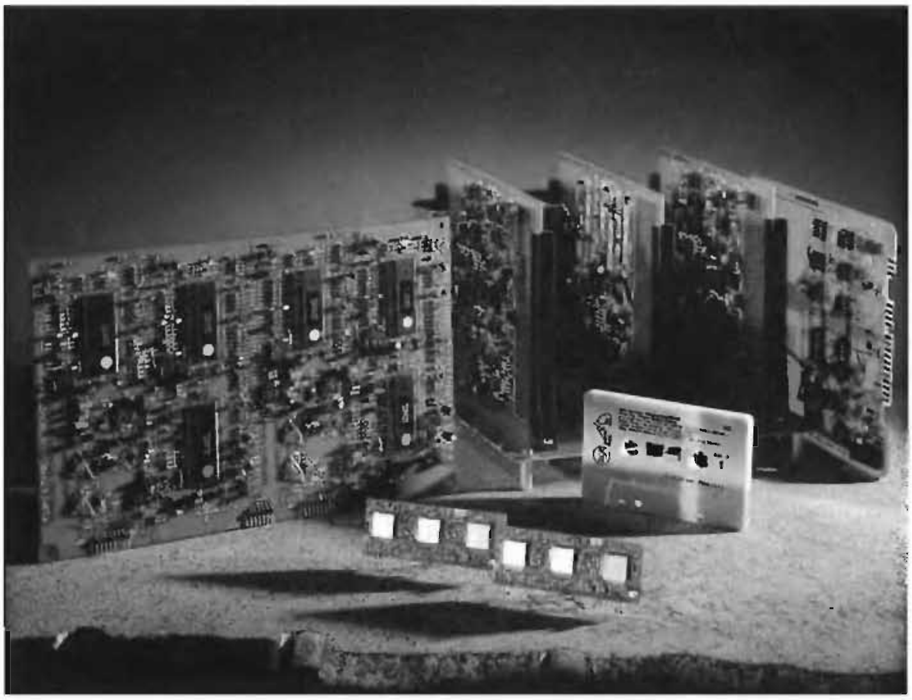
Multi-level staggered action has additional benefits. For example, the slopes of both stages' NR bands combine to provide steeper overall characteristics (Fig. 10), so dominant high-level signals can be that much closer in frequency to the bands without causing their gain, and thus the NR effect, to change. In addition, production tolerances for the individual stages of a multi-level configuration can be wider than for a single-level configuration with similar parameters, resulting in a system more readily mass-produced.

Final Elements

Figure 11 is a block diagram of a complete Dolby S-type encoder (the decoder is essen-

DOLBY S-TYPE AT A GLANCE

- New system for cassette recording derived from Dolby SR combines both fixed and sliding bands.
- 24 dB of noise reduction at higher frequencies, with 10 dB at lower frequencies.
- Increased headroom, particularly at frequency extremes.
- Minimal effect of high-level signals on low-level signals minimizes noise modulation as well as decoding errors.
- Encoded signal free of dynamic artifacts.
- Newly developed dedicated IC configuration for use in consumer products.
- New higher performance standards for products licensed to incorporate Dolby S-type.



Development stages of the new Dolby S NR system: Right background, multi-board configuration used to finalize Dolby S-type circuit design; left, two-channel card with new Dolby S-type three-IC sets suitable for production tape decks; center, space-saving single-channel hybrid circuits.

tially a mirror image of this). There is one element on the diagram that we have not yet discussed: A single-stage fixed band providing 10 dB of low-frequency noise reduction, in addition to the high-frequency stages' 24 dB. Because of the spectral content of cassette noise, the ear's reduced sensitivity to low-frequency noise, and the high-frequency stages' relatively low reach, the low-frequency band has to provide only modest noise reduction, and only below 200 Hz. For these reasons it was also judged that providing both fixed and sliding low-frequency bands was not subjectively worth the added cost. The low-frequency band also helps balance the encoded signal spectrally for playback without Dolby S-type decoding.

Another element in Dolby S-type is not on the block diagram at all: Dolby Laboratories requirement that recorders with Dolby S-type meet new, higher performance standards. These include extended high-frequency response, tighter overall response tolerances, a new standard ensuring head height accuracy, increased overload margin in the electronics, lower wow and flutter, and, for the first time in the cassette recorder industry, a head azimuth standard. These new standards will not only contribute to unprecedented cassette performance but will also help to ensure that tapes recorded on one machine—including prerecorded cassettes—will play back with unprecedented accuracy on any other.

Photograph: Scott Peterson

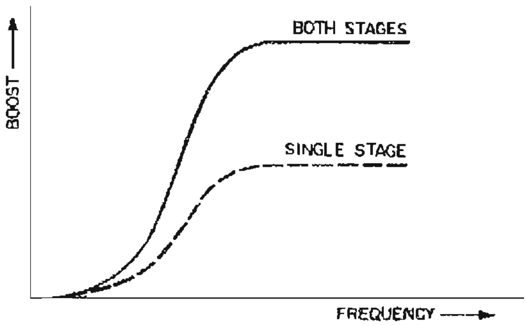


Fig. 10—Dolby S-type's multi-level configuration steepens the slopes of the NR band, further minimizing the effects of high-level signals on low-level ones.

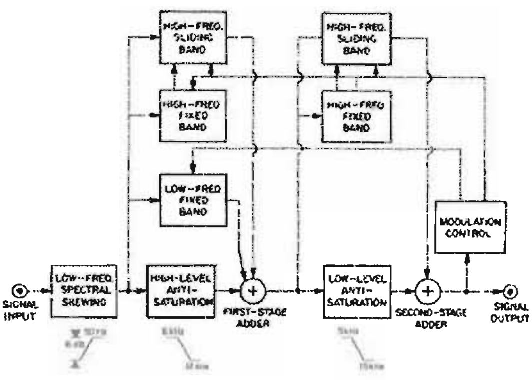


Fig. 11—Block diagram of the S-type encoder. The decoder diagram would be essentially a mirror image of this.

The first tape decks with Dolby S-type will use a new, dedicated three-IC set developed with our cooperation by Sony's IC division, which will be making them available to all Dolby licensees. Later this year, Sony expects to complete the development of a single-chip version having identical performance and other IC manufacturers have expressed interest in doing so as well. However, Dolby S-type is always likely to cost more than our current consumer systems. That higher cost, combined with the higher overall performance required of the machine, means that cassette recorders with Dolby S-type may remain at the higher end of the price range. The first models, expected later this year, definitely will be so.

Recorders licensed for Dolby S-type NR must meet new performance standards that would also help ensure better compatibility between tape decks

Dolby S-Type and The Future

We cannot predict how many home listeners might want better cassette performance, and how much more they will be willing to pay for it. However, the success of the CD indicates that higher quality sound is appealing to a significant market, and we have found that, at the highest playback levels likely to be encountered in the home, sophisticated listeners subjected to A/B comparisons of CDs and Dolby S-type cassettes are unable to identify which is which with any regularity. We are also unable to predict if the prerecorded cassette industry, unwilling to release titles in more than one format, will consider Dolby S-type cassettes sufficiently "compatible" with B-type playback to issue significant numbers in the new format. Be that as it may, the initial response to demonstrations we have conducted for the industry is generally favorable, and we are proceeding with the development of an appropriate professional encoder. Adding to these factors is the enormous investment in the cassette format by consumers, the music industry, and the audio industry (prerecorded cassettes significantly outsell CDs and LPs combined, and more than 280 million cassette machines with Dolby noise reduction alone have been sold). Therefore, there is a real possibility that Dolby S-type will extend and increase the returns on that investment, just as Dolby SR is already doing for professional analog formats.

This short evaluation of Dolby Laboratories' new Dolby S NR system was made using a Teac V-10000 Esoteric cassette deck, which also has Dolby B and C NR. In addition, it has signal generators which produce a 400-Hz tone for level calibration and a 10-kHz tone for bias setting. I used Fuji FR Metal tape for all my tests.

Pink noise, band-limited at 15 Hz and 25 kHz, was the source for the first tests. After deck calibration, a third-octave RTA showed a rising response above 10 kHz at −20 dB, so I increased the bias slightly to get flatter overall responses. Figure B1 shows record/playback responses from +5 dB (relative to meter zero, which is at Dolby level) down to −25 dB, in 5-dB steps. The highest levels show some saturation effects, but the curves are very flat, in general, over the wide range in levels.

One of the more interesting features of Dolby S NR is the spectral skewing at both low and high frequencies. (Dolby C NR has it at high frequencies only.) Both Dolby C and S NR have high-frequency antisaturation circuits. I measured the saturation caused by increasing levels (from −20 to +10 dB) for Dolby B, C, and S NR and without NR. The great improvement across the entire band with Dolby S NR, compared to Dolby B or no NR, was immediately very obvious. Resistance to saturation with Dolby C NR was close to that for Dolby S NR at the high frequencies but was clearly not as good as with Dolby S NR at the low frequencies. The low-frequency maximum output level (MOL) results with Dolby C NR were very slightly better than with Dolby B NR. With Dolby S NR, however, the MOL improvement over Dolby C NR was 0.7 dB at 1 kHz, in-

A FIRST TEST
DOLBY S-TYPE

creasing at lower frequencies to over 8 dB at 20 Hz, a significant change. The Dolby S saturation output level (SOL) results were better than those for Dolby C NR from 3 kHz to about 14 kHz, where they dropped just below those for Dolby C.

I recorded the band-limited pink noise at −20 dB with Dolby C NR and played it back using Dolby B (Fig. B2). There was some boost around 8 kHz and a roll-off above 10 kHz, but these were not bad, overall, for a change in mode at a level sensitive to errors. I also recorded with Dolby S and played this back with Dolby B NR. In this case, the changes in frequency response were more widespread and the level shifts were greater. I tried the same two combinations over a range of levels, and the basic results generally remained the same: For playback with Dolby B NR, the response deviations were less with the Dolby C recording than with the Dolby S recording.

I purposely misadjusted bias and level calibrations a few different ways and confirmed Dolby Laboratories' claim that Dolby S NR is more resistant than Dolby C NR to mistracking from calibration errors. Final conclusions awaited results from the listening tests.

Next, I ran signal-to-noise tests, referred to Dolby level, using all NR modes. With A-weighting, the ratios were 55.0, 63.4, 72.1, and 73.4 dBA for no NR, B, C, and S NR, respectively. With CCIR/ARM weighting, the figures were 52.0, 62.2, 71.8, and 71.8 dB, in the same order. Checking noise in third-octave bands, I confirmed that low-frequency noise (around 80 to 100 Hz) was 10 dB lower with Dolby S NR than with any other NR choice The noise with Dolby S NR was slightly higher than with Dolby C NR from 2.5 to 5 kHz, but noise with C NR was noticeably higher than with S NR from 20 Hz to 2 kHz and from 10 to 20 kHz. From the RTA display, the maximum reduction in third-octave noise with Dolby S NR was 20.5 dB at 1 and 1.6 kHz. Referred to the 400-Hz MOLs, the signal-to-noise ratios were 72.6, 81.6, and 84.2 dBA for Dolby B, C, and S NR, respectively.

The first CD I tried recording was *Bach: The Organs at First Congregational Church, Los Angeles*, with Michael Murray (Telarc CD-80088). I was quickly convinced that at high recording levels, Dolby S NR yielded superior results on the low pedal notes. Between tracks, I detected no difference in noise level between the CD source and the tape playback with Dolby S NR; with Dolby C NR, I had heard a slight difference.

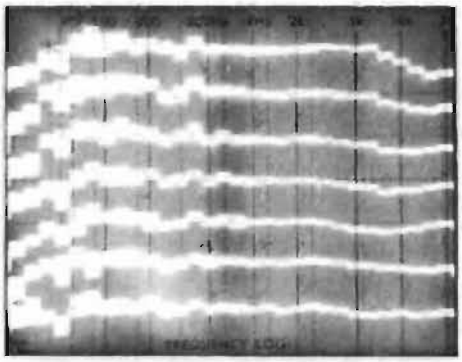


Fig. B1— Record/playback responses at (top to bottom) +5, 0, −5, −10, −15, −20, and −25 dB, on Fuji FR Metal tape, using Dolby S NR. (Vertical scale: 5 dB/div.)

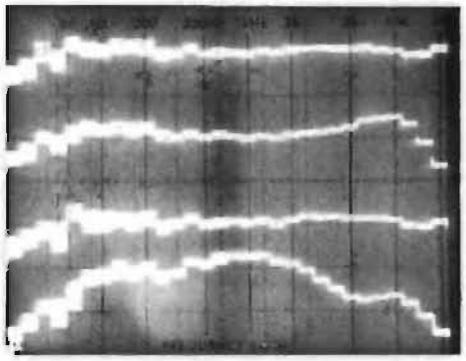


Fig. B2— Compatibility test between Dolby NR systems, using recordings made at −20 dB. Curves show: Recording made with Dolby C NR, played back with Dolby C (top) and Dolby B NR (second curve), and recording made with Dolby S NR played back with Dolby S (third curve) and Dolby B NR (bottom curve).

The next CD was Tchaikovsky's *1812 Overture* (Telarc CD-80041), performed by Kunzel and the Cincinnati Symphony Orchestra. I concentrated on recording and playing back the last minute of the overture. Very quickly I demonstrated the challenge of recording this CD: I thought I had set the input pots conservatively, but the cannons caused levels way above +10 and the sound was badly distorted, so I reduced level a bit. At that point, I got very acceptable results with Dolby S NR, even though momentary peaks still came up to +10. For the first time, I truly appreciated the cannon sound in the playback. Other NR modes were definitely not acceptable unless the level was lowered greatly.

Stravinsky's *Firebird Suite* (Telarc CD-80039), with Shaw and the Atlanta Symphony Orchestra, was very well recorded with Dolby S NR. The playback of the "Infernal Dance of King Kastchei" and the finale was low in distortion and well detailed from the bass drum to the cymbal crashes. I then tried recording other sections of this CD with Dolby S NR, switching back and forth in playback between this NR system and Dolby B or C NR. In the medium- and low-level passages, I could hear the response shifts which had shown up in the bench tests, and I certainly preferred the sound when playing this Dolby S-encoded tape through the Dolby S decoder. Yet when listening to this same tape through Dolby B and C decoders, though the spectral balance was no longer accurate, it did not change when the level of the signal changed even though the levels varied over a wide range. I did not detect any disturbing pumping or obvious shifts in spectral characteristics. Overall, the sonic compatibility with Dolby B and C NR was definitely better than I thought it would be.

Dolby S NR has established itself as my preferred noise-reduction process for its resistance to overload across the band, good signal-to-noise ratio at all levels, low distortion, and resistance to calibration errors. Let's hope the chip makers can bring the cost down to facilitate including Dolby S NR in more than just the top-of-the-line cassette decks.

Howard A. Roberson

Revival

Problems and Solutions In Long-Term Tape Performance

Several years ago, a friend told me that many of his prized tape recordings had suddenly started to deteriorate. As a collector of rare and historic recordings who was aware of their fragility, he had committed many to tape. A good number had been recorded only a few years previously. They were all made with great care, and often required hours of editing and equalizing to assemble on tape. Now, he found, the passage of the tapes over the tape heads and guides produced a squeal that was heard not only from the tape machine, but also through the loudspeaker. Some of the tapes would continuously leave head-clogging deposits and require him to stop the machine every few minutes for cleaning; others might even bind in the guides. He could offer no reason for this, but he indicated that it seemed to occur rather suddenly and he thought it was largely attributable to one brand of tape that he had been using.

Initially, I didn't register the story as really serious, and I assumed that he had probably purchased some cheap, used tape. Interestingly enough, a few other collectors began telling me similar tales of woe, about whole collections of recordings that were now largely useless. The tapes used included 3M, Ampex, low-cost Shamrock, and other brands. As these collectors were largely old-time radio collectors who are notorious for their use of extremely cheap tape and their disregard for high-quality audio, the problem still didn't register.

The problem finally reached me when I began to notice a short squeak, which would quickly go away, when playing some tapes in my own collection. If the squeak was any more serious, a quick cleaning of the heads and/or guides would usually eliminate the unpleasantness for the entire playing of the reel. Not many months later, though, these and other tapes would squeal almost continuously, leave deposits on heads and guides, and, in extreme cases, jam in the guides!

I brought up the subject with a tape supplier and user who categorically stated that he thought such problems only occur with cheaper, non-top-quality tapes. He suggested that I use a particular top-line tape, which, it turned out, had been among the very tapes whose squealing had sparked my investigation. In that instance, the tapes were from the transcription service of one of the richest radio stations in the U.S. They had been donated to the local broadcast museum, and I was transcribing them for the use of the museum's patrons.

Thinking the tape players might be at fault, I tried three different machines. These tapes squealed and jammed in each one. Tape transfer could only be obtained by repetitively stopping and cleaning the heads and making countless flying starts.

Michael N. Stosich

Over time, the lubricants in some tapes break down, leaving a white, mold-like, powdery residue on the tape's edges.

I began my inquiry into this problem by contacting dealers, professional recording engineers, record producers, tape-head manufacturers, and acquaintances and friends who engage in serious audio. Later, I contacted tape manufacturers, people I knew at various sound archives and government agencies involved with recordings, and even a developer of early tape recorders. Information on the subject of handling and storing tape is not too easy to come by, and a lot of it seems redundant. Much of the following is culled from technical papers and articles obtained from 3M and Ampex.

Tape Problems

Recording tape consists of a flexible backing material coated with a mixture of magnetic oxides in a flexible binder material that holds the oxides on the tape. Formerly, the backing material was acetate; now it is almost always a polyester film. Most high-quality tapes also have a textured, conductive back coating. The binder is usually implicated in tape squeal. Because of the proprietary nature of the processes and formulas involved in the manufacture of recording tape, it is not possible to determine the exact cause of any degradation that may be taking place.

My experience and that of others is that this problem is associated with top-quality, brand-name tape formulations which appeared in the later '60s and the '70s [1]. I know no cases of old-fashioned acetate tapes squealing or jamming, although I am told it can happen. Acetate tapes do tend to dry out and become wrinkled and crumbly. They are prone to constant breakage, but at least they play. Interestingly enough, some of the humbler quality, less polished, and less expensive tape formulations, such as Ampex 631/641 and the discontinued Scotch 150, never seem to have the problem.

To have a low-noise product, and to minimize head and tape guide wear, it is desirable to have a smooth, highly polished tape surface. This is especially important to maintain low noise in low-speed operation, such as cassette recording. However, tape can be *too* smooth, and tape manufacturers are now approaching the limit in this area. Tape's abrasiveness helps keep the path of tape travel clean. A tape with too smooth a surface will not be efficient at removing minor debris from the path. In addition, a minute air gap is required between the surfaces of heads and tape to maintain accurate tape motion. With too smooth a tape surface, the tape will tend to stick to the surface of tape heads and guides, alternately sticking to the head and breaking loose in a continuous cycle [2]. When this occurs at a very high rate, we identify it as squeal. The high polish of some quality tapes may exacerbate the potential for squeal.

Many of us are aware of cheaper "white box" tapes that were once commonly sold. They usually were regarded by purchasers as rejects, substandard or defective. The defects were often quite apparent ones, such as width problems, splices, and squealing. Some tape salespeople have suggested that bad batches of tape have sometimes been unintentionally delivered to stores as first-line, because they either did not get caught by quality control or were only later found to have problems. Substandard tapes have been known to show up in the marketplace, but this appears to be an insignificant problem. The problems that people experience are due both to the combination of basic materials from which most tapes are made and to inadequate storage.

I know of three explanations why tape squeal is now emerging so frequently. One reason is that over time, the lubricants in the tape have broken down. Tape with this problem exhibits a white, mold-like, powdery residue on the edges of a reel [3].

Another conjecture is that the binders of some back-coated tapes break down with time due to their particular chemical makeup, and that higher temperatures hasten this process [1]. Chemicals released from this degradation cause the friction that produces the familiar squeal.

The prevalent explanation of tape squeal is that the polyurethane binders used in most modern formulations of recording tape are hygroscopic—that is, they absorb moisture from the air to a degree that depends on the humidity [4]. These binders, depending on the length of their urethane molecules, have been shown to undergo a chemical change at high humidity and high temperature. Ampex has recently acknowledged that this problem can occur with the binders used in their Type 406 and 456 tapes manufactured between 1975 and 1984 [3]. This degradation, known as hydrolysis, can occur very rapidly—under laboratory conditions, in as little as four weeks at a relative humidity (RH) near 100% [5]. In fact, one suggestion for testing tape is subjecting it to 80% to 95% RH at 120° to 130° F for three days and examining it for problems [6]. Under less extreme conditions, degradation may take much longer to occur and may not affect all the molecules constituting the binder.

High temperature and humidity are looked on as the main causes of hydrolysis. Humidity is considered the basic culprit, with temperature merely hastening the process. Hydrolysis induces the molecular chains of the binder to break down, which causes the resultant chemicals and/or the tape's lubricants to arrive eventually at the surface of the tape. Tape that has undergone hydrolysis becomes sticky, thus adhering to and squealing against tape heads, and can even cause many layers of tape to stick together in a block on the reel. When the binder's breakdown is complete, the oxide layer may crack off the backing. A 3M publication cites cases of tape stored for extended periods of time at 80° F and 80% RH actually having the layers stick together after 15 years [4]. The effects of humidity (squealing, stickiness), once exhibited, often indicate permanent damage to the tape. Fortunately, the literature also indicates that hydrolysis is somewhat reversible. In mild cases it can be reversed by subjecting the tape to a very low humidity (11%) for a period of time [5].

Storage Environment

Table I summarizes the temperature and humidity ranges that have been recommended by a number of published sources on the subject for normal and archival storage environments and those cited as detrimental. Sometimes this had to be extrapolated from the articles. Papers on the subject usually do not state exactly what constitutes a detrimental environment for tape, but 3M [4] does refer to deterioration after storage at 80% RH and 80° F, and a recent article in the *Journal of the Audio Engineering Society* by an Ampex authority [7] suggests deterioration at conditions above 40% RH and 68° F. This latter assertion agrees with temperature-versus-humidity curves, published by Agfa (Fig. 1), for two levels of tape degradation caused by binder hydrolysis [8]. The curves indicate 40% RH at 68° F, for example, to be a marginal storage condition, and 70% RH at 80° F to be a detrimental storage condition.

Authorities on the subject are in general agreement that low temperatures and humidities are necessary for long-term storage of recording tapes. Typically, the recommended temperature range is from 60° to 75° F and the humidity from 30% to 50%. It should be noted from Table I that exceptionally low humidities, i.e., 25% to 38%, are recommended for archival storage by most recent sources. Many common laboratory and household hygrometers (instruments for measuring relative humidity) indicate the ideal humidity range for humans as being 45% to 65%. Someone who lives in Colorado, where the humidity seldom exceeds 30%, recently told me that he has never experienced any of the squeal problems using the same brands and types of tape.

Nowhere in the literature is there a definition of just what period of time constitutes archival. In fact, the longest times referred to in any of the papers is 35 years—that concerned acetate tape, and I have never heard a report of acetate tape squealing. Articles and papers usually use 10 or 20 years when referring to storage times of modern polyester formulations (which, after all, are not much older than that). One who is familiar with old recordings, however, would hardly refer to such comparatively short periods as archival. I would suggest that normal storage periods for other types of items might be 25 years and that archival

Table I—Tape storage conditions as defined by temperature and humidity. The figures represent the range of values given in various publications of Ampex, BKM Associates, European Broadcasting Union, and 3M, and in papers and books by Marvin Camras, Howard M. Tremaine, and Jim Wheeler.

	Temperature, Degrees F			Relative Humidity, %		
	Archival	Normal	Detrimental	Archival	Normal	Detrimental
Minimum	32	60	68	25	20	40
Mean	68	67.5	114.5	38.1	41.8	73.3
Maximum	81	80	167	60	60	100

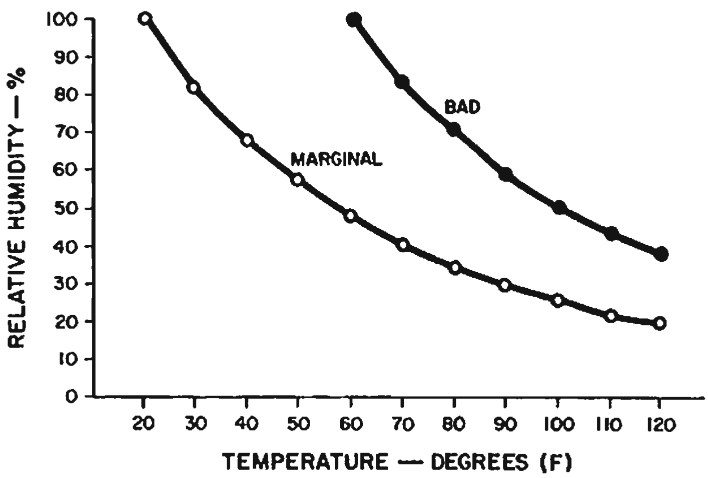


Fig. 1—Temperature and humidity conditions which cause marginal or bad tape degradation due to hydrolysis of the binder, as determined by Agfa.

Authorities agree that both humidity and temperature should be kept low for tape storage. They don't agree on how low they should be.

should refer to at least 50 years. Table I indicates that archival storage conditions are quite dry and cool by the standards of summers in central U.S.A. As the ranges given in the Table imply, the storage conditions considered normal by some authorities overlap the conditions that other authorities suggest as archival, and conditions recommended for archival storage by one source exceed those listed as detrimental by another! One could interpret the information to suggest that under normal storage conditions, the life of tape might be very short. In fact, Ampex now states only that their product should last at least 10 years under proper storage conditions [3].

What is disturbing about this is that many of these tapes may be recordings of important family events, master tapes of celebrated musical and dramatic performances, or recordings of important historical events. Recordings assumed to be permanent may be deterioriating on the shelves. I have carefully made many safety copies of crumbly, old acetate tapes only to find those safety copies squealing after only two years. It is frightening to think that the only way we can make a safety copy of a recording on an unstable medium is on another unstable medium. Much of what is written on the subject basically tells us how we might obtain a safety copy of a deteriorated tape. We must consider the fact that every time we introduce a generation of recording, as in safety dubbing, we audibly degrade the recording with distortion, noise, and speed irregularity.

Storage Problems

Special precautions should be taken when storing tapes in building basements. Being porous, concrete absorbs ground moisture and acts like a wick. Concrete floors and walls, therefore, are very effective humidifiers. A dehumidifier may not be sufficient to remove this moisture. My advice is to make sure no ground water is in contact with the floor. Most newer houses have sump pumps, but older ones often do not. I highly recommend the installation of one if your house is not so equipped. In my case, the house was 60 years old and a hole had to be dug into the floor. Crushed stone was put into the hole and then a sump pump was added, for a total cost of about $500. While it would be nice to have drain tile around the basement to connect to the pump pit, it is not absolutely necessary. Water will tend to collect at that spot, and the ejector pump will remove it. In many basements that are not too deep, a dehumidifier may suffice, but I recommend one in every case. If silica gel or other non-liquefying desiccants are available, place a small amount in the tape boxes and/or store reels in plastic bags [9].

People who store their collections of tapes above ground, but who don't mind the heat in summer, may also have problems. I am amazed at the number of people who don't have air conditioning even in the relatively hot, humid climate I live in. Some collectors may think that they don't need air conditioning for themselves, but according to the data, their tapes may not survive in the high humidity and temperature that summer often brings. The best way to reduce house or apartment humidity is to use air conditioning. In fact, much of the feeling of coolness associated with air conditioning is due to its lowering of ambient humidity.

Remedies

What can you do to tapes that won't play? One solution, suggested by a longtime record producer, is to dub the tapes at a minimum of twice the recording speed. The idea is to shift the tape's mechanical vibrations way up in frequency, possibly even to a frequency so high that the tape's mass will limit its vibration. Theoretically, the tape won't have the time to stick.

As mentioned before, some of the humidity damage can be reversed or reduced. Before throwing the reels out, here are a few hints: First, dehumidify the tapes. If a vacuum chamber is available, try leaving the affected tapes in it for an afternoon or a day. If the season is warm and the weather dry, try sticking them in an attic for an afternoon, with the boxes open (assuming the attic's temperature doesn't exceed 140° F). In winter, when the heat is on and the humidity (if you don't have a super humidifier) is low, open the tape boxes and let them dry out for a few weeks.

I have heard of several methods of rejuvenating tapes by baking them in ovens. However, the benefits may not last. One method is to bake them under precise conditions in a convection oven. This process requires regulation of the temperature within 3° F over a time and temperature cycle [1]. It is claimed that this allows the binder's broken chemical bonds to be remade. An alternate baking method has been described for drying out tapes which have been subjected either to excess moisture or to flooding [7]. For excess moisture, the process is to bake the tape in an oven at 120° F for 24 hours, cool, then rewind and fast forward the tape a couple times. For flooding, the process is repeated.

A laboratory oven is recommended for this process, as kitchen ovens can easily overheat and ruin a tape. Nevertheless, I have adapted the described processes to the home oven so that I can salvage otherwise unplayable tapes to make safety copies of them. The following method for rejuvenating tapes is similar and has been effective but must be followed with extreme care to avoid catastrophic mistakes: Heat a stove oven to approximately 125° F (probably the lowest setting on the dial). You must use an oven thermometer, as it is imperative that the tape not be heated above 150° F. After the oven's temperature has stabilized for 10 to 15 minutes, turn it off. If the oven will not stabilize at a temperature below 140° F, simply turn it off and wait until the temperature, as measured on the oven thermometer, drops below 140°. Quickly place the bad reel into the oven, and leave it there for at least 30 minutes—don't turn the oven back on! (When on, an oven produces pulses of high temperature, then cools. Thus the average temperature may be low, but the peaks can damage the plastic reels and tape itself.) After the reel has been lightly baked for 30 minutes, it should be removed and allowed to cool and stabilize for several hours or overnight. Do not rush the process. If necessary, repeat the process again. I have found the results to be cumulative. Tapes thus treated often play with little or no squeal.

As mentioned before, the benefits of baking may not be permanent with severely deteriorated tapes. Practice this with totally unimportant reels before subjecting a valuable reel to this potentially destructive treatment. In fact, this method may only allow you the ability to make a safety dub of the deteriorated tape. Paradoxically, it may be advisable to use one of the older formulations, with a rougher polish (such as Ampex 631 or 641), to ensure against the recurrence of squealing in the safety copy.

Very recently Agfa showed, by setting up a unit to recondition and copy such tapes, that they not only were aware of the problem but that they acknowledged some responsibilities to users of their products [10]. By its proprietary XT process, Agfa hopes to provide customers with a means of retrieving information "lost" on deteriorated tapes. Prices range between $280 and $350 for processing a 2,500-foot reel. Whether they will process other manufacturers' tape is not known at this time.

Machine Problems

You can minimize friction by removing oxides that deteriorated tapes have shed onto heads and guides. So keep these surfaces clean with either commercial tape-deck cleaning products or, as I do, with cotton swabs and isopropyl alcohol. Be careful when buying products labelled "rubbing alcohol." Avoid the many such products that include oils.

When dubbing a bad reel onto a good one, you may have to stop and clean machine surfaces several times before completing a dub. More often than not, even when I cannot see a buildup, cleaning these surfaces mysteriously stops the squeal—for a while. Make sure that all heads and guides are demagnetized, as magnetized surfaces tend to attract loose magnetic oxides.

If the squeal is very mild, it may be possible to alleviate it somewhat on machines equipped with manual holdback-tension switches (often labelled "reel size" or something similar), by reducing tape tension. If playing a 10-inch reel, set the switch to the tension recommended for a 7-inch reel. If you need to play a 7-inch reel, wind the tape onto a 10-inch reel. If you don't have a switch, and the machine is a 7-inch machine, it may be possible to wind the tape onto a 7-inch reel with a large hub, such as those once popular for prerecorded tapes. Bear in mind that altering the holdback tension can induce other tape motion problems, such as improper reel braking, flutter, or running off speed; so proceed with caution.

One collector, James L. Snyder, has suggested that using a bidirectional tape recorder may help [11]. These machines often have asymmetrical head and guide arrangements, so that the tape passes over the drive components in a different order for each direction of play. Using three different brands of machine, Snyder found that, 85% of the time, tapes would squeal in one direction but not in the other. This may also be due to a difference in holdback and take-up tensions in the two directions. While I haven't personally done this, I have used different machines with different head/guide arrangements and have obtained similar results.

Tape heads and guides with "flats" worn into them can aggravate the situation by enlarging the surface the tape passes over and can stick to, thus producing or increasing squeal. If the guides cannot be rotated to expose new surfaces, they should be replaced. Even relapped heads may lose their previous sharp profile and develop a larger contact area that aggravates sticking and squealing.

The design of some tape guides and tensioners can add to the problem. Some machines, such as old Revoxes and Vikings, have spring-metal tensioners which serve as "shock absorbers" over which the tape rides [12]. These have been known to whine as the tape passes in contact with them and impart squeal to the recording or playback. I found that the spring-loaded, adjustable-height tape guides on both of my Technics RS-1500s acted the same way. I replaced these two guides with a pair of ridged, nonadjustable ones identical to others employed elsewhere on the deck. This resulted in only a slight reduction in some tapes' tendency to squeal, but even such small effects may combine with the effects of other remedies to significantly reduce the problem.

If the need is to make a safety copy of a deteriorated tape, it may be advantageous to remove any and all unnecessary surfaces over which the tape passes and to which it might adhere. Such surfaces include unused

You can sometimes reduce squeal by resetting your deck's holdback tension or putting the tape on reels with bigger hubs.

erase and record heads, and even some guides. If the machine in question has a removable head-block system, a special head block can be contrived having only a play head and minimum number of guides. I have used this method with great success over the last several years.

Lubricants

In lieu of drying out tapes and cleaning and demagnetizing playing surfaces, few other things can be done. Applying some sort of lubricant either to the head or tape has been tried by many, but usually with limited results. I know of people who have tried spraying silicone lubricant on tape heads and reels, but to little avail. As an experiment, I tried applying a little of this lubricant to tape heads, and it worked for only the time it took the tape to wipe the head dry—about a minute. The collector mentioned earlier in this article went to the extreme of applying French horn valve oil to particularly unplayable tapes to make safety copies. Just recently, he contacted me to describe a method where he repeatedly fast-spooled the tape over the surface of a silicone-impregnated record/tape cleaning cloth like those made by Radio Shack and countless other manufacturers. He noted that by the time oxide ceased being deposited on the cloth, the tape was ready to play properly. One producer of historical and old radio broadcast records offered the following suggestion: Lubricate the oxide surface of troublesome tapes by rewinding each tape so it passes over the lead of a pencil, whose graphite will act as a dry lubricant. Another record producer suggested the application of motion-picture film lubricants such as Filmagic Pylon Blue Lubricant, Xekote, or Vitafilm Lubricant and Preservative. These motion-picture products are applied both to the projector mechanics and to the film directly. Their safety when used with recording tape has yet to be demonstrated, so I recommend using them only as a last resort.

Ampex Corporation was kind enough to provide me with a sample of "Topical Lubricant Solution—0.5% Fluorosilicone," a tape lubricant consisting basically of a small percentage of silicone of some kind, dissolved in Freon TF. To use it, one has to hold a lubricant-soaked applicator against the oxide side of the tape as it passes into the head/guide of concern. While it was somewhat effective, it proved extremely difficult to use, as it was necessary to repeatedly soak the applicator to get through a complete reel of tape. Tape squeal would often begin before this could be done. It was also very awkward to hold a cotton swab in precise alignment against the tape for extended periods. I also looked into Krytox, a fluorinated oil made by Du Pont, which is used by some manufacturers of magnetic memory discs as a surface lubricant [13]. Krytox has been indicated as useful in lubricating old tapes whose lubricants have actually migrated away. For this use, a 1% solution of Krytox 143AC in 99% Freon TF was suggested. It is applied directly to the tape with a soaked applicator while fast-spooling [7]. On the theory that this might also apply to tape affected by hydrolysis, I applied a 2% Krytox 143AC solution to a tape severely deteriorated by hydrolysis but observed little benefit. Furthermore, Krytox costs $176 per pound, and Freon TF cannot normally be purchased except as small bottles of tape head cleaner or in spray cans. (Freon TF is commonly available in Radio Shack stores as their professional tape head cleaner.) These products and chemicals are either not commercially available, difficult to locate, or very expensive. They are only mentioned here for the sake of completeness.

I was able, however, to find similar products expressly aimed at recording tape. I know of only a few commercial products presently available that address squeal. GC Electronics of Rockford, Illinois produces a "Tape Head Lubricant," Cat. No. 30-124-2. Amongst its listed benefits are a reduction of wow, flutter, and squeals. Radio Shack sells a tape care kit, Cat. No. 44-217, which includes cotton swabs, a head cleaner, and a head lubricant. Both of these lubricants appear to be a silicone in an alcohol base. As described previously, these products proved to have very limited benefits, as the tape quickly carried off the lubricant. Some people have had successful results at lubricating tape when they applied Radio Shack's head lubricant directly to the oxide surface of the tape with a cotton swab while fast-spooling [11]. I suggest the following for more uniform application: Start from one end of the reel, applying half of the intended volume of lubricant, and then follow up by applying the remainder from the other end of the reel. Be careful not to get lubricant all over tape drive surfaces like the rubber pinch roller and capstan, which would severely limit their ability to regulate tape speed.

Another product is Last Factory System Formula #9 Interlast Tape Head Treatment. The manufacturer claims that it is not a lubricant in the traditional sense, and that it does not introduce a film between head and tape. (Such a lubricant film might reduce high-frequency performance by introducing too large a gap, and also might contaminate the tape with potentially detrimental products.) The manufacturer's explanation of how the product reduces the "surface energy" of the head face is not fully comprehensible to me. My experience with the product is that it can reduce the squeal considerably. Often, it will work for the playing of an entire reel. With more seriously deteriorated tapes, it may provide only short-term squeal-free operation—usually, but not always, enough to get a safety copy.

Another curious product I found is not advertised as a tape lubricant, but its application is similar to other products I have described. Last Factory Formula #10 Tape Preservative comes with its own applicator and marker labels to indicate treated tapes. The applicator is a large plastic foam swab. Application is made by pouring a quantity of preservative into the applicator and fast-spooling the tape with the applicator in contact. I do doubt the uniformity of application via this method, but there are few alternatives. Last claims that this product can slow the process of hydrolysis and thus multiply the life of tape three to seven times [14]. However, I have not noticed from one application any reduction in squeal with tapes that have already degraded. If you've dried out some tapes, it has been suggested that with time they'll be more likely to exhibit the problem than new, or properly stored, tapes. It might be a very good idea to initially dry out a tape by one of the methods I have suggested, then apply Formula #10. A dried-out tape may better absorb the beneficial compounds, resulting in a more permanent restoration. The effectiveness of this process awaits long-term conclusive evidence. As the product is most likely to apply to recording tape as described, this process may alternately serve as a simple home method of cleaning tape of some of the gummy material and loose oxide produced by hydrolysis.

Going on the assumption that the sticky products of deterioration collect on the playing surface of the tape, I attempted to clean tapes that had recently shown a tendency to squeal. I cleaned samples of the tape with several fast-spooling passes, using a cleaning/lubricating cloth and also by pinching the tape lightly with a cloth dampened with Last Formula #10 and with straight Freon TF. In all cases, the tendency to squeal was reduced. However, I do not know the long-term effects of cleaning with such Freon-based solutions as Last or Radio Shack head cleaner, since such cleaning (as opposed to other methods such as baking) definitely removes some constituent of the binder.

Conclusions

I offer these specific recommendations that may help prevent premature deterioration of your tape collection. In temperate climates, store tapes in basements only as a last resort. If tape must be stored in a basement, use a dehumidifier, and ensure against ground water being in contact with the floor. Also, never store tape in non-temperature-controlled garages. In summer, use air conditioning, at least where the tape is stored. Never use a tape that exhibits even a slight, one-time squeak. And finally, consider using older, higher noise tape formulations, which are more abrasive.

The following techniques may prove helpful in improving playback of degraded tapes for dubbing: Demagnetize and clean heads frequently, clean the tape, reduce tape tension, use tape head/guide lubricants, dehumidify/bake tapes, remove unused heads and guides, and make sure guides do not resonate at a squeal frequency.

The techniques described in this article were largely developed while under pressure to maintain recording production. These methods are doubtlessly not the only ones available for dealing with deteriorated tape. Since tape recording as we know it is only about 40 years old, and the specific problems described here are far more recent, we are still in the preliminary stages of identifying the problems and the anecdotal and experimental stages of dealing with them. These techniques, however, have been used by knowledgeable people who required definite results.

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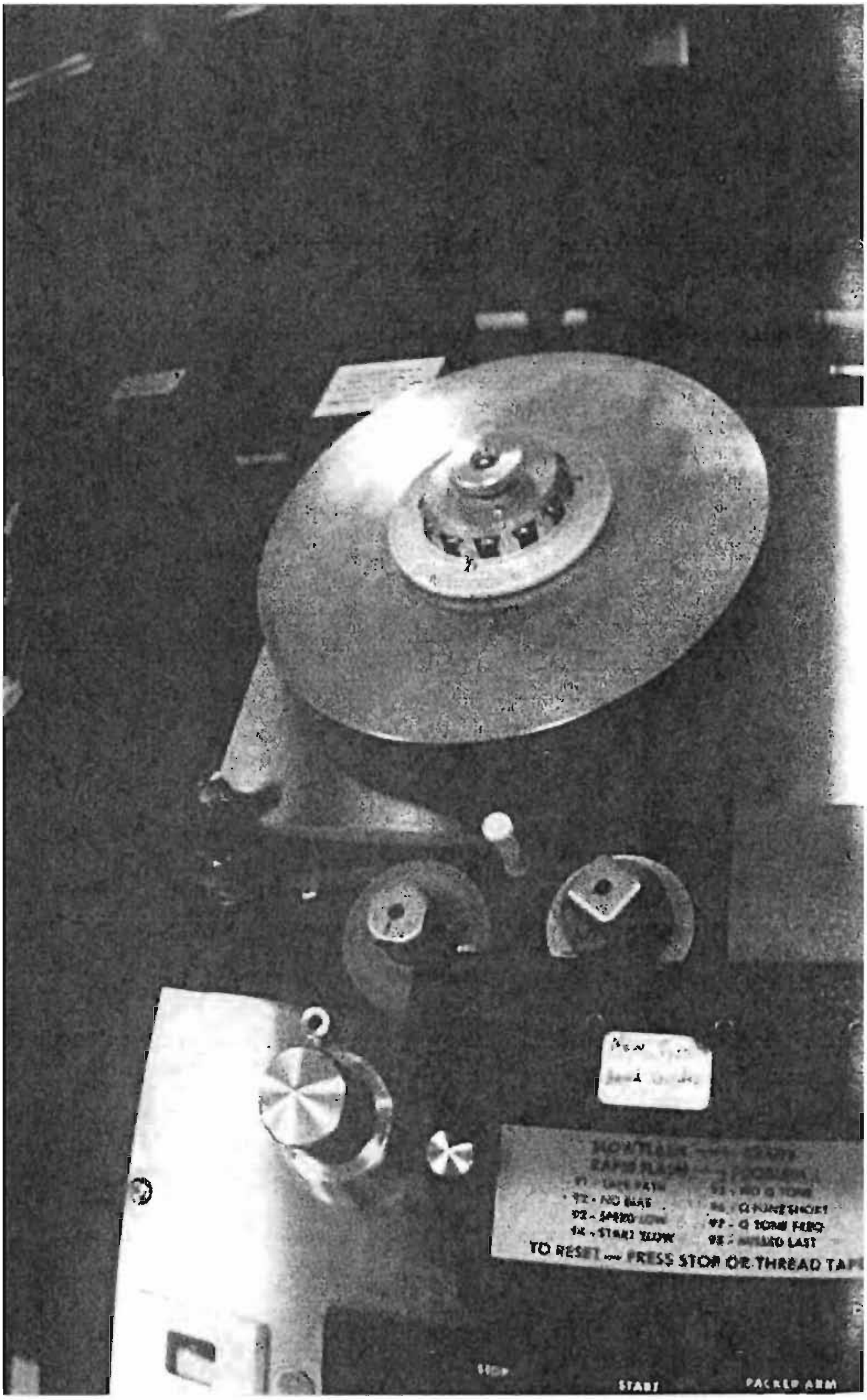
N E W L I F E I N

DIGITAL D

JOHN EARGLE



The Philips Compact Cassette is very likely the most universal carrier of recorded music the world has ever known. It is estimated that there are three cassette players per household in the United States; considering the popularity of auto stereo and various forms of personal stereo, this seems reasonable. Overall, the world's population of players is estimated to be greater than one billion.

The cassette was introduced into the United States in the mid-1960s, and its basic technical performance went little beyond the requirements of an office dictating machine. At the time, RCA, Ford Motor Company, and Learjet were promoting the Stereo-8 endless loop cartridge for stereo-on-the-go applications. With a tape speed of 1⅞ ips, the cassette suffered by comparison with the 3¾-ips Stereo-8 format. But in just a few years, the cassette won out, primarily because it allowed the consumer the flexibility of conveniently recording material at home for later replay in the automobile—something that Stereo-8 could not do.

In terms of basic quality objectives for home recording, the cassette was up against reel-to-reel stereo tape. In time, the improvements in tape, shell design, and recorder performance, and above all the addition of Dolby B noise reduction, helped the cassette mount a formidable assault on reel-to-reel; by the mid-1970s the cassette again became the clear winner. Record companies had seen all of this coming and

Group of Gauss 2400 high-speed recorders.

CASSETTES

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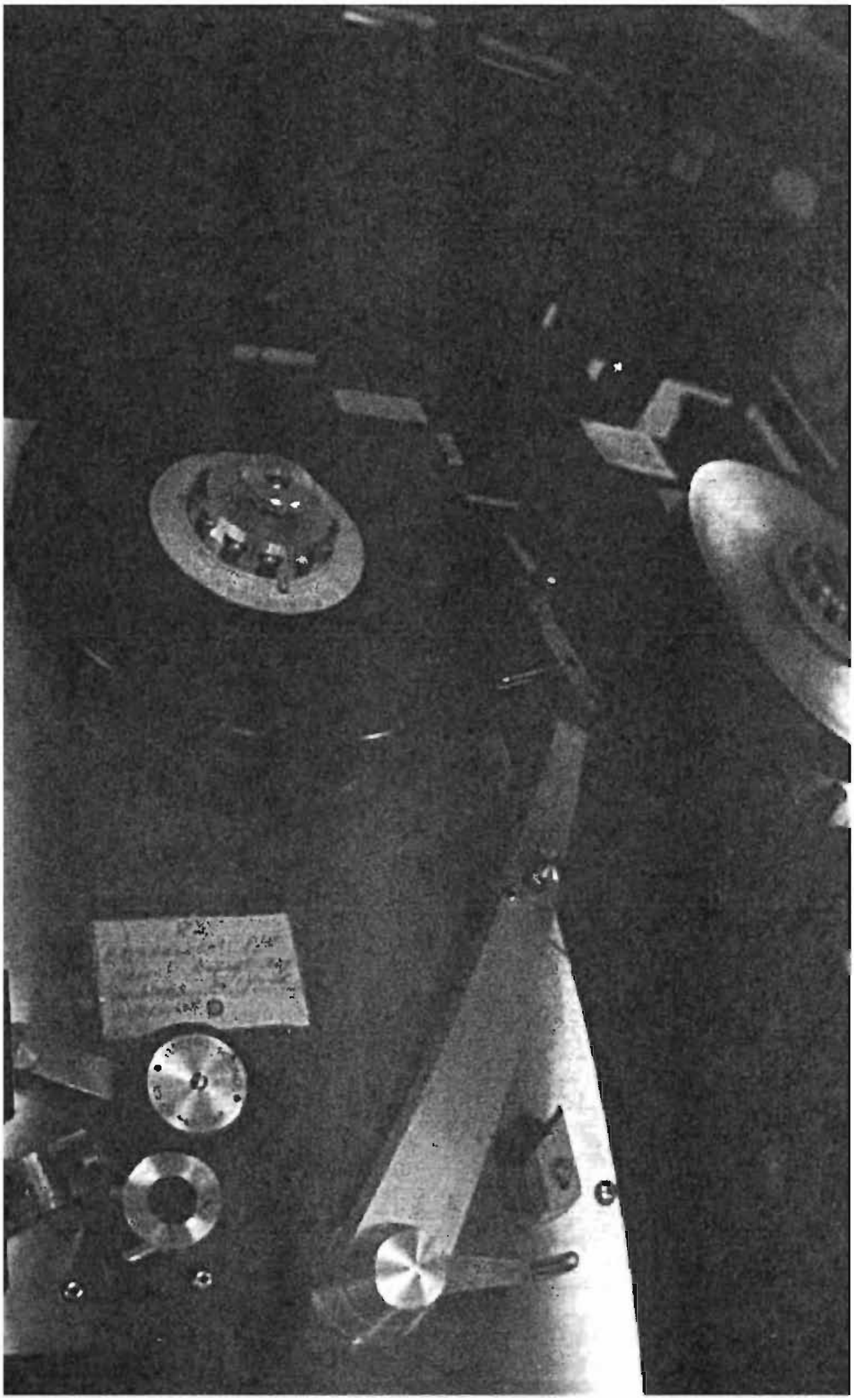

Electro Sound 8000-series high-speed master recorder.

had geared up to produce prerecorded cassettes on a large scale.

In 1983 the unit sales of cassettes in the United States exceeded those of the LP, and the later onslaught of the CD put the LP into further eclipse. Now, even as the CD gains in momentum, the cassette holds strong and promises to do so for many years to come.

The cassette is mass-duplicated in a manner which is physically identical to the consumer's in-home recording procedure. There are, of course, economies of scale, but essentially the tape to be copied is run on a master transport, and a number of "slaves" are fed directly from it. On a modest scale there may be no more than four slaves, and the duplicating speed ratio may be no more than 4:1. The slaves themselves may be standard cassette transports which have been modified for the increased speed of operation, and the "master" may simply be a cassette itself. This method is shown in Fig. 1 and would be applicable for very short runs of speech-quality program.

A substantial step forward in quality is shown in Fig. 2. Here, the master is a digital source, a DAT or possibly a Sony PCM-F1. During the mid-1980s, Nakamichi duplicated short runs of very high-quality cassettes using a method similar to this. The source was a digital F1 tape whose transfer quality was carefully monitored by noting the digital block-error rate as the tape was repeatedly played back. The slaves themselves were top-of-the-line Nakamichi recorders which were self-aligning before each run. The duplicating ratio was, of course, 1:1, or so-called *real time*. Tapes were duplicated with Dolby B or Dolby C noise reduction, and HX Pro headroom extension was routinely used. Each side had to be duplicated separately.

Nakamichi provided this service for owners of their machines who wanted, and were willing to pay for, audiophile-quality cassettes from such labels as Delos and Sheffield. Nakamichi could honestly say that these tapes "were first-generation copies of the original digital master" if all steps in the preparation of the running master had been carried out in the digital domain.

These duplicating methods are quite "labor intensive" in that the yield of product per man-hour is limited. However, they remain just about the only way for producing short runs.

For larger scale duplication, however, some fundamental changes must be made and other considerations enter into the picture. Many slaves may be used, and the duplicating ratio may be as high as 96:1 for music and 128:1 for speech quality. (See Fig. 3.) The production time required for each copy is essentially reduced and can be determined by dividing the playing time in one direction by the duplicating ratio multiplied by the number of slaves. The master itself is run as an endless loop with all four tracks copied at the same time. Figure 3 shows a modern duplicating system with four slaves and an endless-loop tape bin for the master. The cassettes are duplicated end-to-end on large "pancakes" of tape which are subsequently broken down and loaded into the cassette shells.

Many of the larger duplicators injection-mold their own cassette shells and operate large printing shops for paper inserts and labels. Their basic purchased operating materials are commodities, such as plastic, paper and tape—for which prices are relatively negotiable. Such "vertical integration" reduces operating costs considerably in a highly competitive marketplace. Attention to line scheduling and quality assurance maintains good "loading" of the plant, again in the interests of controlling operating costs and minimizing downtime.

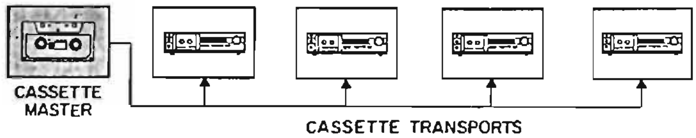

Fig. 1
Modest duplicating setup for limited runs of speech-quality cassettes. No more than four slaves are used, and speed ratio is 4:1 or less.

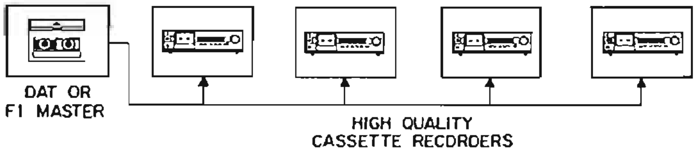

Fig. 2
Real-time (1:1) duplication from digital copy master, for high quality.

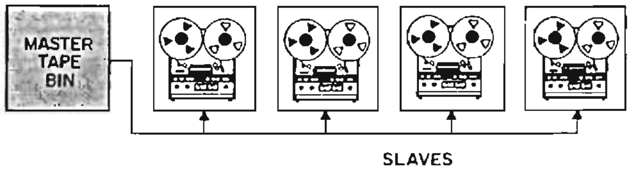

Fig. 3
Large-scale duplication, using endless-loop master and slave recording tape "pancakes." Setup shown here is for music, using 7½-ips master running at 240 ips and tape pancakes running at 60 ips, a 32:1 duplicating speed ratio. For speech-quality tapes, 3¾-ips masters and still higher speeds would be used, for ratios of 128:1 or more.

The duplicating ratios used today are a far cry from the 8:1 methods of the 1960s. The electronic challenges were relatively easy to meet; bias frequency has been raised to the range of 8 to 10 MHz, and equalization requirements have been simply transposed upward as required. The major problems to overcome were in the physical handling of the tape. Tape skew and air filming have been the major problems in ensuring positive tape-to-head contact at high speeds. The design of the endless-loop tape bin for the master was a considerable challenge, in that the requirements were to move the tape at a speed as high as 480 ips while handling it gently to minimize wear and damage.

At this point, we will follow a master tape through the entire duplicating cycle, noting the many procedures which ensure quality and process control.

Duplicating plants prefer to receive a master in the form of a digital clone of the final edited master, with A and B sides clearly indicated. The digital tapes are then carefully checked to determine if there are any potentially troublesome high frequencies present in the program. If not, then the digital masters are transferred to ½-inch tape running at 7½ ips. If the cassettes are to be in Dolby B (most of them are), then the 7½-ips tapes will be made as a Dolby B copy. In a single pass, side A is copied in one direction, while side

Even though the CD has gained in sales momentum, the cassette holds strong and promises to do so for many years.

B is copied in the other. The tape made at this point is referred to as the *running master*.

Should there have been any problems with the source tapes, these would be discussed with the client and adjustments made either at the client's studio or at the duplicating plant.

Before the actual production run, the master will be copied onto a cassette on a master-slave setup, which, in effect, duplicates all production conditions. It is transferred at the intended production level and on the intended tape stock. It is then sent to the client as a reference of what the finished product will sound like. The client usually has a technician check the tape and judge the quality of the transfer. Then, if all has gone well, the client will send an approval to the duplicating plant and production can then get underway.

The production control department takes over at this point and ensures that all paper components (labels, inserts) are on hand. Even though the entire order for cassettes may be quite large, the client may not want more than, say, 5,000 or 10,000 units at any one time. The run is scheduled accordingly, and the master tape is loaded into the endless-loop bin. For high-quality music duplication, a ratio of 96:1 would probably not be exceeded. At this duplication ratio, and assuming a total program length of about 40 minutes, a single slave can produce about 1,800 copies per eight-hour shift. Thus, a master transport with four or five slaves might easily handle the production run in about half a shift.

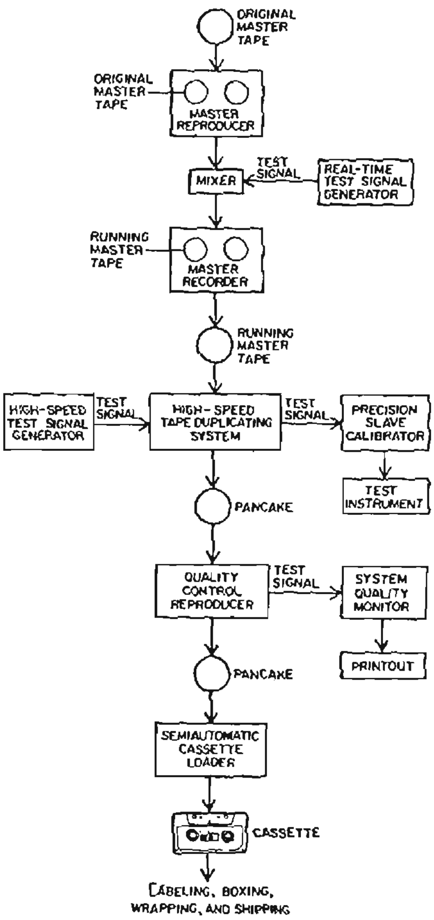

Fig. 4
Tape duplicating chain, with electronic quality-control stages.

Obviously, one cannot play every cassette to ensure that everything has gone properly. Cassettes are spot-checked according to established sampling procedures, but more to the point, all of the process and the incoming materials are carefully monitored. The duplicating chain itself is subject to several levels of maintenance, including those which are performed daily, such as cleaning, and those that are done at much wider intervals, such as replacement of heads and bearings. Between these extremes is routine electrical checkout of the systems.

While the incoming raw tape stock is sampled to make sure it meets magnetic requirements, each duplicated 8,400-foot pancake of tape is recorded with a diagnostic sequence of signals. This serves as a running check on outgoing quality, since no pancake will be loaded into cassette shells unless the pancake passes the test.

A pancake contains many passes of the master, and between each pass is a low-frequency cue tone which is used during the breakdown process. At this stage, the pancake is loaded onto the cassette winder, and individual cassettes are loaded at speeds up to 360 ips. The cue tone stops the winding process when each cassette is fully loaded. The operator then removes the loaded cassette and replaces it with a new shell.

The loaded cassettes then find their way to the labeling machine, which affixes the appropriate label to each side of the shell. Then, they move on to final packing and shrink-wrapping.

With recent developments in random access memory (RAM) capability and high-speed digital-to-analog (D/A) conversion, it is now possible to operate duplicating slaves directly from a digital source, thus bypassing the analog running master completely. The technology for this was developed by Concept Design, and one of the major installations of this is at the Sonopress manufacturing facility in Asheville, N.C.

Master tapes are received from clients, preferably in the Sony 1630 format, which stores the program on ¾-inch videocassettes. From this source, the plant's mastering department makes a pair of R-DAT clones, one for side A and one for side B. When a program is to be duplicated, the two R-DATs are loaded into RAM simultaneously, but on a real-time basis. Thus, it takes about 20 minutes to "load in" each new program. When duplication gets underway, the RAM and high-speed D/A converter function just like the running master and master transport in the typical duplicating setup. The duplicating ratio is 80:1, so this helps offset some of the rather long load-in time for new programs. In all other regards, the plant works very much like a traditional duplicator.

Injection-molding department, Sonopress, Weaverville, N.C.

Sonopress is owned by BMG, which is the parent company of RCA Records, and the technology can be heard on RCA cassettes as well as A & M, Delos, and Telarc product. Sonopress places great emphasis on quality-control procedures to ensure that the benefits of the new technology will clearly find their way into the consumer's home.

The process gets us one step nearer to the digital master, in that each cassette is a first-generation copy of the digital master, but the next major technical leap forward for the cassette will be the DCC, or Digital Compact Cassette. The DCC is designed to complement the standard cassette and not necessarily make it obsolete. Since all DCC players will also play standard cassettes, there will be no need for wholesale adoption of the new medium. In fact, it may be quite reasonable to make the conjecture that speech-only cassettes, a very large market indeed, may remain analog based for the foreseeable future.

Tape duplicators and manufacturers of duplicating equipment are overwhelmingly in favor of DCC because it reinforces their fundamental investment in technology. DCC is designed to be duplicated at a 64:1 ratio on traditional slaves which have been outfitted with new heads. A number of details have yet to be worked out in DCC, but it appears poised for introduction in 1992.

BEYOND

MARK WEAVERS

If you're like most discerning buyers of audio cassettes, you're concerned about output performance—and rightly so. And whether or not you consciously consider criteria such as sensitivity, output, and signal-to-noise ratio, you demand consistency in the cassettes you buy. That means you expect the quality and high performance of the cassette you purchase today to be consistent with the tapes you buy next month and the month after that. You also assume that your audio cassettes will deliver the same performance today as when you bought them, perhaps several years ago.

Output performance, or electromagnetic performance, can be one good criterion for your buying selection. But there are other criteria to consider as well, and over the life of the cassette, they may be even more important. Electromagnetic performance may *sell* cassettes, but it seldom is a reason that consumers return them. Instead, users are more likely to reject cassettes because they fail for mechanical or environmental reasons. I want to focus on the important issue of environmental stability—and how one very experienced tape manufacturer ensures that audio cassettes will provide long-term reliability for users.

If consumers can't easily assess the hidden factors that contribute to a quality audio cassette, how can they hope to make a wise selection? Part of the answer lies in the array of industry standards that serve as guidelines for audio cassette manufacturers. Manufacturers who are committed to quality pay heed to these standards because they represent a solid base line for consistency and long-term reliability. This attention has a payoff for consumers, who end up "buying" more than the audio cassette itself: They also gain the expertise of a cassette manufacturer who is aware of the complex criteria that must be built into a consistently high-quality audio product. In other words, if a consumer chooses cassettes from a reliable, quality-conscious

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OUTPUT

Environmental Stability of Audio Cassettes

ILLUSTRATIONS: BOB SCOTT

manufacturer, there's a good chance the manufacturer has already done much of the selection work by qualifying the materials and the manufacturing processes, and by carefully auditing products before they reach the shelf.

Figure 1 illustrates the electromagnetic performance of a Scotch XSII-s audio cassette, including its maximum output level (MOL) at both low and high frequencies, sensitivity, distortion, and the bias noise floor. These parameters are carefully specified and controlled by 3M along with additional criteria for component and assembly dimensions, visual appearance, functional performance, durability, packaging, and environmental stability.

If an audio cassette is not carefully designed for environmental stability, its other features, such as electromagnetic performance and durability, can rapidly become inconsequential. Among the most revealing assessments of cassette weakness are those associated with overall runnability after exposure to the environmental conditions encountered in shipping or use by consumers. A dashboard or trunk on an August day in Arizona, humid conditions along the southern seacoast, or a winter in Alaska are likely scenarios for environmental damage that can shorten the life expectancy of an audio cassette. And we have not yet considered dust, sand, or dirt.

To evaluate the short- and long-term effects of these scenarios, it is possible to recreate extreme environmental conditions in programmed environmental chambers. Based on years of experience, 3M engineers have designed test conditions beyond the environmental extremes expected in real life. For example, cassette samples might be subjected to storage conditions of 150° F (65° C) and 85% relative humidity for a week or more. Following this, the samples would be tested for performance, wow and flutter (the nonuniform movement of the tape over the playback heads), and dropouts (momentary losses of signal)—with the objective being the same performance quality after the exposure as before. Figure 2 illustrates the onset of wow and flutter as a result of exposure to high temperature and humidity.

An environmental test might call for placing a cassette recorder into a chamber at 104° F (40° C) and 85% relative humidity, then measuring the operating tape performance over an extended period. It's not unusual to have the recorder fail before some cassettes do in such an environment, because of corrosion or moisture absorption in the recorder's pinch rollers or brake pads.

Environmental testing of this type helps establish temperature and humidity ranges such as those in Fig. 3, where the areas designated "Storage" and "Operating" represent test conditions for Scotch cassettes. Data from environmental tests performed regularly verify that audio cassettes will operate within the temperature and humidity boundaries outlined in the figure, and that the cassettes can be stored under the indicated conditions for a limited period of time. By establishing and assessing these limits, 3M can tighten its material and manufacturing specifications to provide audio cassettes that are mechanically and functionally reliable.

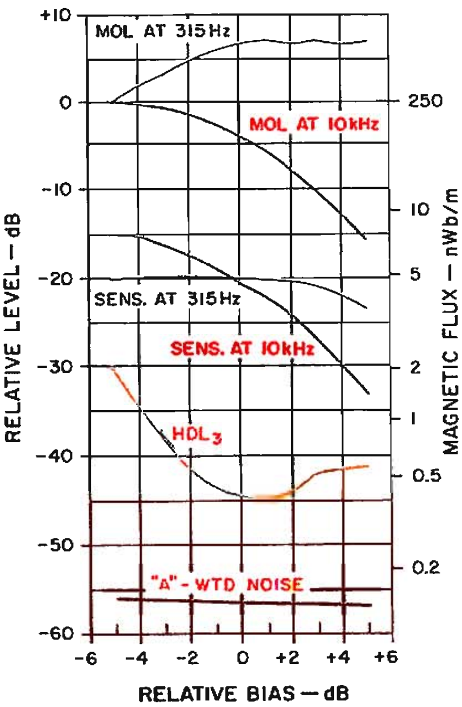


Fig. 1
Electromagnetic performance of Scotch XSII-s (Type II) cassette.

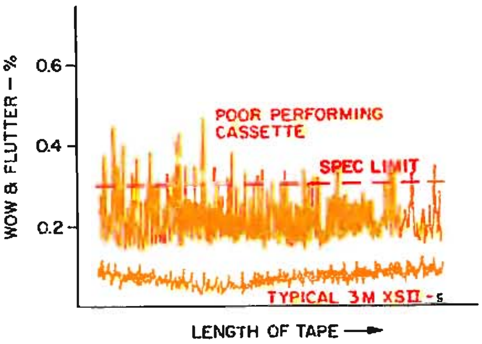


Fig. 2
Development of wow and flutter as a result of high temperature and humidity in storage.

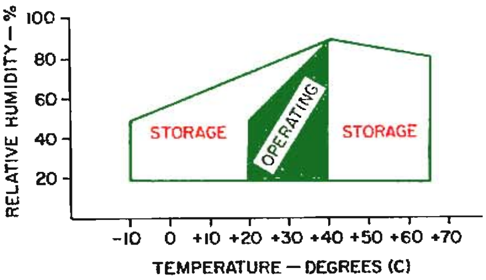


Fig. 3
Test conditions help define boundaries for optimum operation and storage of cassettes.

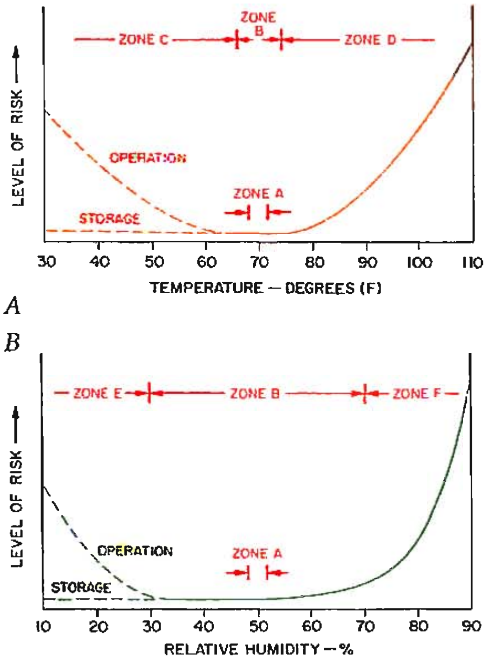


Fig. 4
Risk of performance loss due to extremes of temperature (A) and humidity (B) during operation or storage.

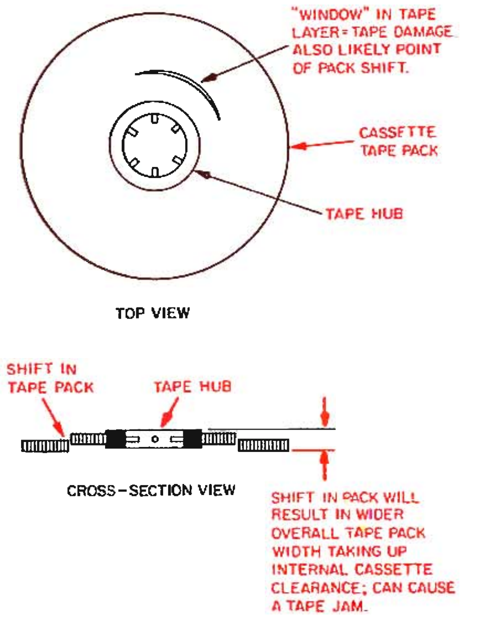


Fig. 5
Effects of cold-temperature storage with rough handling while cold.

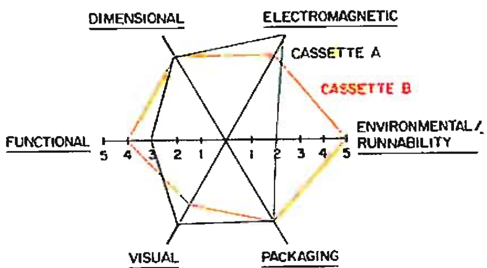


Fig. 6
Comparison of performance criteria for two hypothetical audio cassettes.

A cassette's output will be unimportant if it can't stand up to high temperatures and high humidity.

Collection of data from environmental tests helps engineers assess the boundaries beyond which audio cassettes may be likely to fail. For example, Fig. 4 depicts the risk of problems when exposing cassettes to extremes of temperature (Fig. 4A) or humidity (Fig. 4B) during operation or storage. In general terms, the greater the variation from comfortable room temperature and humidity, the greater the risk for damage to the cassette. This correlation is not meant to suggest that an individual audio cassette will inevitably fail under environmental extremes, but rather that there is more risk to the specific product because of the exposure.

In Fig. 4, zone A represents the ranges of temperature and humidity control used for establishing a reference; zone B represents the limits of ideal operating conditions or the ranges of temperature and humidity that minimize risk while still allowing for a practical degree of flexibility.

In Fig. 4A, at higher temperatures (represented by zone D on the graph) the risk level for problems rises. The tape pack may tighten, increasing the chance of dropouts from several sources: Captured or wound-in debris, impressions from a non-round tape hub lock, or impressions from the leader to the tape splice. In addition, at higher temperature, polymers used in the magnetic coatings may soften, causing tape layers to stick together. This condition can cause high torque, an excessive amount of force required to move the tape in the cassette. The result can be objectionable wow or flutter.

If temperature rises above 167° F (75° C), the halves of the plastic cassette shell might distort, destroying the fit of the cassette within the machine. At very high temperatures, the tensilized polyester backing used for audio cassette tapes could shrink, causing distortion of the tape and subsequent poor contact with the recorder heads—the result being poor output uniformity.

In contrast, the risk of damage when storing a typical audio cassette at low temperature is not great (zone C in Fig. 4A). Some loosening of the tape pack may occur, but if the cassette is allowed to acclimate to room temperature for 24 hours, the tape wind usually returns to its original tightness. If the tape is subjected to rough handling or used while very cold, loose tape layers could slip upon one another, causing "windows" to form in the cassette. Figure 5 illustrates an example of tape pack shift and the resulting windowing effect.

Figure 4B shows cassette risks associated with extremes of humidity. Studies by 3M indicate that low humidity (represented by zone E) presents no increased risks for either short- or long-term storage of audio cassettes. However, low humidity environments create the potential for greater static charge, and thereby increased attraction of airborne dust and debris. Storage in the protective plastic album case is, therefore, an important factor in long-term cassette reliability.

Environmental studies show that high humidity (zone F in Fig. 4B) can have a detrimental effect on audio cassette performance. Long-term storage in such an environment increases the risk of moisture absorption in the tape pack, resulting in jerky movement of the tape over the head surface; it is often heard in playback as high wow and flutter. If tape is not carefully formulated, increased absorption of moisture in the magnetic layers can increase the risk of clogged recorder heads.

Although the numerous criteria for assessing audio cassettes are based on varying units of measure, diagrams can be used to compare relative rankings of audio cassette characteristics. Figure 6 is an example of such a diagram based on dimensional, visual, functional, output, and environmental and runnability assessments of audio cassette performance.

Cassette A ranks high for output performance but poorly for functional and environmental reliability. Cassette B ranks high in the functional and environmental categories but somewhat lower for output performance. Cassette B would be the better choice in this example: It has good signal properties with excellent functional and stable environmental characteristics.

Clearly, output performance is the most obvious selling tool available to audio cassette retailers. But consumers should be concerned about other cassette features as well—and these less visible performance characteristics are the foundation of a tape manufacturer's expertise in technology. In addition to strong output performance, documented environmental performance helps ensure the consistency of audio cassettes and their long-term usability and reliability for consumers. A

RASSIC TAPE

HOW TO BE A GOOD AUDIO DINO

DAVID B. CAMERON

Despite our whiz-bang digital age, in which most, if not all, forms of audio, video, graphics, business, communication, medicine, and even sex seem to be wed to one chip or another, there are still many reasons to support the old ways of analog media. The audio dinosaurs, among which I proudly count myself, are in possession of a great wealth of obsolete but still quite useful equipment, records, and tapes (that's hardware, firmware, and software to ye uninitiated) that for one reason or another we wish to retain. Some of us still create new programs by using the time-honored methods in which "cut," "paste," and "program" are *not* computer terms.

As a reader of this magazine, you are likely to have some cherished analog audio, and we are here today to discuss the care and feeding of quarter-inch reel-to-reel and cassette tapes. There are several reasons you may need to do this, including replacing old splices and leaders, replacing sections of worn or damaged tape, replacing a damaged or low-quality cassette shell, and, for the more adventurous, editing/creating new arrangements.

CUTTING TAPE IS AN IRREVOCABLE ACT SO YOU MUST MAKE YOUR CUTS IN JUST THE RIGHT PLACES.

I edit audio professionally, but it makes up no more than 10% of my business. I need good quality but can't allocate a full-scale business budget to audio. Today's high-end consumer-level equipment affords many choices of recording gear, but even if you have been recording on reel-to-reel for a zillion years, you still may not have a machine that lends itself to repair or editing.

Any repair/editing deck needs some version of the following features: An open tape path (which offers easy access to head and quick threading), a quick-acting pause (both in engage and release), and the ability to shuffle the reels in pause mode with the amplifiers on.

For obvious reasons, some great deals can currently be found in used high-quality 7-inch open-reel decks. Pawn shops, flea markets, classified ads, and individuals are good sources, but make sure you get some kind of working guarantee, even if it's only 24 hours, so you'll have a chance to put a deck through its paces. A machine that has been used and maintained will be better than one that has been sitting around. By all means look inside for large accumulations of dust. Stay away from any deck that has lots of dust or that appears "too clean" on the outside. If the rubber pinch roller is glazed, attempt a proper cleaning with a suitable conditioner (such as Realistic Non-Slip Fluid, Radio Shack #44-1013, or a lint-free swab dipped in automotive belt dressing) or be sure you can get a replacement.

My system is built around quarter-inch reel-to-reel tape decks with 7½-ips capability. They are all 20+ years old, but I do my own maintenance and keep them up to spec. The main recording deck is the venerable Teac A6010, which has run (almost) flawlessly for 25 years. My favorite editing unit, a play-only Sony TC-155, has mechanical controls and the unique feature of retracting the rubber pinch roller in stop mode (which you can see in Fig. 3). This makes for totally open tape access, a "must" when editing. The Teac, with its Ampex-type transport, also provides good tape access, but it is easier for me to keep the Teac vertical and the Sony horizontal. The Sony's mechanical pause is quick and has a good feel (I replace the brake pad with fresh surgical tubing—every 10 years or so!), and the hubs do not have clutch pads to wear out.

ILLUSTRATION: BILL MAYER

FIG. 1- TAPE EDITING AND SPLICING TOOLS

I recently acquired two old Akai/Roberts decks for $75, and all they needed was a cleanup. On the down side, their two-handled controls are not convenient for editing; the space between a 7-inch reel and the rubber roller is less than the width of my thumb, and the tape path is circuitous. However, because these decks are excellent in playback, I use them for mixdowns and slow winding.

THE CUTTING EDGE

The highest quality way to do low-budget editing and repair is the old-fashioned way—by simply cutting the tape, rearranging the pieces, and splicing them back together. (The requisite tools are shown in Fig. 1.) Care has to be taken, both physically and audibly. You do not want to have scraps of odd lengths (or rolls) of tape lying around collecting dust and getting lost. And since cutting tape is an irrevocable act, you've got to be sure you cut in the right place.

BY STRESSING ACCURACY, YOU CAN DO AMAZING TRICKS WITH A GOOD EAR, PATIENCE, AND A RAZOR BLADE

Digital editing systems claim an accuracy of 0.0005 second. With an analog tape deck, the smallest distance you can shuffle the tape past the heads (Fig. 2) and still hear something significant is about four times the thickness of a razor blade. At $7^1/2$ ips, that amounts to 0.005 second. For a "normal" edit, I'd challenge anyone to hear the difference between 0.0005 and 0.005 second, but there are limitations. Generally you can edit between words, but not within words. It would take digital gear, for instance, to make an unrecorded word from an assortment of sounds. However, a razor cut can go very nicely between sounds. Voice may be the easiest to edit, but it can be tricky to disguise the actual sound change of an edit. Music, with its greater variety of sounds to choose from and hide within, may be a more difficult splice but is often a less noticeable one.

FIG. 2-THE SHUFFLE

For repair you can generally do or redo splices as you get to them. The same goes for gross rearranging. Just splice, and then roll your selections onto the take-up reel in the new order. For editing in which recording a "master" tape is involved, simply record the segments in order with at least 1 or 2 seconds of silence between them. If you're editing music and have any doubts about where you may be able to make a cut, give yourself room to find the perfect spot by including more of the segment than you think you'll need, such as the next (or last) verse or instrumental reprise.

FIG. 3-MARKING THE POINT OF EDIT

Your task is then to simply remove the unwanted blank segments by finding the perfect spot to cut your segments together. If repair is your goal, the decision of where to cut is often made for you.

TOOLS AND CUTS

The basic editing tool is a splicing block to hold the tape in place while you cut it and apply splicing tape. There are several types; I don't recommend the guillotine-type devices that trim the splicing tape or that require using $^1/_4$-inch-wide tabs of splicing tape. The tapes may not line up properly; even worse, the blades can get too dull and become too misaligned before you notice, and they are troublesome to replace. The best splicing tools are a single-edge razor blade, the old-fashioned EDITall splicing block of solid aluminum, and a pair of fine scissors.

FIG. 4-CUTTING BOTH TAPES AT ONCE

A diagonal splice is best for all non-critical unions, such as inserting leader, and the easy splices where there is actually a moment of silence. For a tight edit, the perpendicular splice is the only choice. You may have to listen to the "before" and "after" sounds several times before finding the perfect spot to make an edit. If there is an obvious beat or other right-on-the-spot sound, then your job is easy. Cut the tape *just before* that sound—the aforementioned four thicknesses of a razor blade. You may also have to listen at full speed, at half speed, or even at quarter speed if you can—and manually rock the reels to move the tape back and forth across the heads for a short distance. If there is not an

FIG. 5-REALIGNING THE CUT ENDS

easy, on-the-beat spot, look for a matching sound somewhere else nearby, like the odd cymbal crash between the second and third beats or some other very non-obvious place. You must make sure that the before-and-after sounds match in key, rhythm, and tonal quality. The splice itself can also make noise; a misplaced diagonal splice can make the sound "gurgle," and a carelessly placed vertical one can cause an abrupt change that is discomforting at best. Many vocal artists trail the voice on well after the last beat or even start before the first, so looking for an unexpected place to edit is frequently the only choice.

As vocalists or speakers open their mouths just before their vocal cords begin to vibrate, you will frequently hear the sound of their lips parting or smacking. It usually sounds more natural if the "smack" is left with its matching word, but between the two can be a great place to hide a splice. Do try to leave a smack in, even if it is not the natural mate of the next word; to remove it entirely can sound unnatural.

FIG. 6-INITIAL BURNISHING IN THE BLOCK

Watch out for key changes. Don't assume you can make a splice anywhere you want. Many tunes go up a half note for the last verse (or so) in order to brighten the end of the song. To take the music down a half note, even if the splice is perfectly timed, will sound dull and draggy. Don't do it.

Stress accuracy, and you will be amazed at the tricks you can do with a good ear, a little patience, and a razor blade.

Of course, the best edit is one that is not heard at all. It is one of my greatest pleasures to tell clients exactly where an edit is and watch their faces as it goes by unnoticed. If you happen to have tape machines that run at 15 ips, you can effectively double your accuracy to 0.0025 second, but remember, *all* your machines will need that speed. (In my case that would mean replacing four machines; 7½ ips does just fine for me.)

FIG. 7-REBURNISHING, ON A NONMAGNETIC SURFACE

EDITING WITH A RAZOR BLADE

The following steps detail the mechanics of cut-and-splice editing and tape repair. For visual clarity, the demo splice in the photographs was made with two types of leaders.

(1) For repair, get your tapes and sequences in order. For editing, record your sequences onto a master tape.

(2) For editing, find your stop-and-start points by listening repeatedly and at slower/shuffle speeds to make sure you have compatible before-and-after sounds. Mark them lightly with a sharp grease pencil directly over the head gap (Fig. 3). For repair you may not need to determine the splice location so precisely.

(3) If repairing, you may have to use an old splice location. Carefully peel off the old splicing tape by slipping the razor blade under a corner. It may break, but be consoled: Most situations can handle two layers of splicing tape.

(4) If editing, cut the tape about 2 inches *after* the first mark and 2 inches *before* the second mark. The 2 inches of tape is for handling, so you can avoid getting fingerprints on tape that is to be kept. The intervening tape may be rolled off on the floor or stashed on a spare reel for recycling.

FIG. 8-TRIMMING THE EDIT

(5) In your EDITall block, place both tapes on top of each other, with the two marks exactly aligned over the appropriate (vertical or diagonal) slot, and cut through both tapes at the same time (Fig. 4). Remove the unwanted tape ends. Notice the little tab sticking out from the roll of splicing tape in Figs. 1 and 6: *Always* leave a tab. I even bend it back, which cleans my finger and also leaves some skin oils on the tab so it won't stick to the roll. The reason is simple: Splicing tape is very thin, and if it adheres to the roll it could take six weeks to get it back. (Use the razor blade.) This is why many people prefer a tape dispenser, but using a dispenser is wasteful and slower than having the tape right there where you need it.

FIG. 9-CHECKING THE FINAL EDIT

(6) If the good tapes did not shift *at all*, you may apply the splicing tape. For better accuracy, I usually realign the good tapes to one side of the cutting slot so that the optical blackness of the slot doesn't hamper the perception of alignment (Fig. 5). Repairs, if already suitably cut, may be aligned without trimming.

(7) Pre-cut splicing tabs are applied in line with the audio tape, being almost the same width. A length of bulk splicing tape (about 2¼ inches long) should be applied at a 45° angle, with the excess flapping to each side equally. (The flaps become handles in step 9.)

(8) Burnish the splicing tape lightly with your fingernail (Fig. 6) or a soft plastic stylus; the corner of an old credit card will also do.

(9) Being careful to touch only your splicing-tape handles and the edges of the recording tape, remove the tapes from the editing block and relocate the splice to a clean, nonmetallic surface. Burnish the splice to the edges, checking that the splicing tape is sticking thoroughly (no white air bubbles) over its entire surface (Fig. 7).

NATURALLY, THE BEST TAPE EDIT IS ONE SO WELL DONE IT CANNOT BE HEARD.

(10) Trim the handles off with fine scissors (Fig. 8). The trim should scoop in slightly: If you cut exactly along the edge of the recording tape, tape guides and rollers will be exposed to the gooey edge of the splicing tape. Obviously, you must not scoop in so far as to cut into the recording tracks. Guillotine-type editors will usually perform this trim; pre-cut tabs do not require trimming, provided they have been well centered on the recording tape.

(11) Inspect the final edit (Fig. 9), and test by listening.

(12) Start saving for a digital system. (Just kidding—or not!)

If you need to redo a splice, do it right away: Splicing tape "sets" and gets stickier with time, and after a lot of time it gets somewhat brittle. At no time should you touch the front *or back* of any tape that will travel over the heads. Touch the edges when you must, but remember that salt, perspiration, oils, and other unnamed grungies associated with human skin can ruin the tape and the tape heads. Some people use "lintless" gloves, but I find them cumbersome (and the typical cotton laboratory gloves are far from lintless). I intend to try some of the new synthetic high-tech gloves on my next audio job, but until then the best advice for making clean splices is the same as for working in a darkroom: Work in a clean, cool, static-free, and dry location.

EDITING CASSETTES

Cassettes also can be edited with a razor blade, but since it's impossible to mark the edit point or shuffle the tape past the heads, such editing is limited to gross manipulations of sequences with abundant space in between. A more likely reason to physically cut and splice cassette tape is repair, either of the tape itself or to replace a broken or low-quality (e.g., bonded) housing. A dirty rubber pinch roller can "eat" your tape, possibly creating the need to cut out a crinkled section or add leader. This is done in exactly the same manner as with quarter-inch tape but requires extra care because of the thin and narrow material.

The sequence of steps will depend on what you're trying to accomplish: Saving the tape or switching to a better housing. Editing is best done while the tape is still in the housing, since that is the only environment where there is axial stability. There are no reels in a cassette, only hubs. The cassette housing or, in some cases, its lubricating sheets are used to control the tape axially.

If possible, do your editing, tape repair, and leader-adding first. Leave the short end of the tape free, but wind it fully onto a hub. A handy manual winder can be any appropriately sized hexagonal (six-sided) shaft such as a pencil, an Allen wrench, or (my personal favorite) a short section of the ubiquitous Bic pen with a collar and a round handle to twirl between your fingers. (See Fig. 10.) Figure 11 shows the cassette-width slot of a dual purpose editing block in use. The process is exactly the same as with quarter-inch tape, but you may want to use 3-mm-wide editing tabs instead of bulk editing tape because the narrow cassette tape is more difficult to handle. The tabs can be applied while the tape is still held in place by the arms of the editing block.

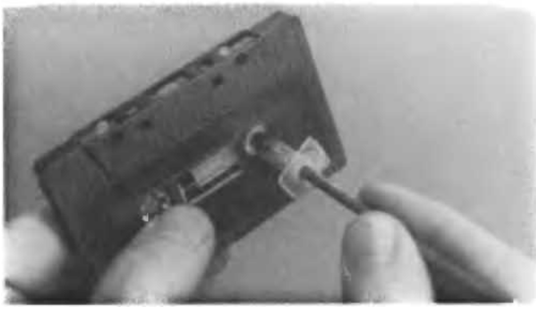


FIG. 10—MANUALLY WINDING A CASSETTE WITH A HOME-MADE TOOL

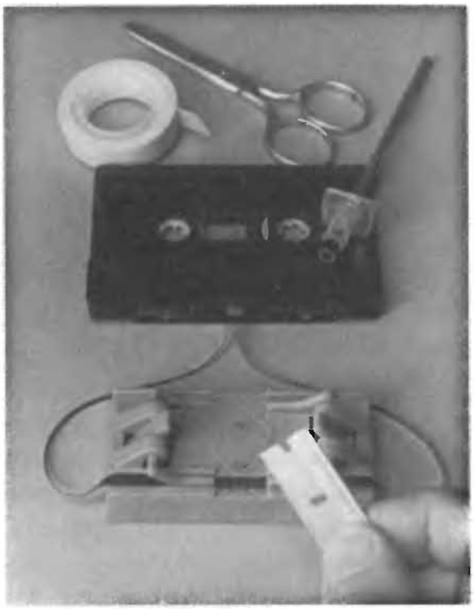


FIG. 11—CASSETTE EDITING

Opening the cassette doesn't have to be a difficult task. If it is a bonded housing, the only real care that must be taken is to preserve the tape. Do not hold the cassette up in free space to pry open the housing, as this can easily lead to instant magnetic spaghetti. Put the cassette on a firm surface, and, with your fingers or tools, pry the top shell off. Breakage is fine, but do consider saving the lubricating plastic sheets; you may need them again. If you have a screwed-together housing, put it on a firm surface and remove the screws with a tiny Phillips-head screwdriver. Carefully lift the upper shell off, leaving the corner rollers in place, and then remove the top lubricating sheet, if any.

Before removing the tape and hubs, prepare the receiving housing/bottom half with lubricating sheet, corner rollers, and pressure pad. To avoid touching the tape with your fingers, slip small pieces of card stock under the tape, like two spatulas, and lift the tape and hubs up and out. Place them directly into the new cassette. Using tweezers or a straightened paper clip, route the tape along its intended path: From the feed hub, out over a corner roller, over the pressure pad, over the other corner roller, and to the take-up hub. If you need to attach (new) tape to a hub, do not use splicing tape directly on the hub. Either splice to a short leader, or use the slots or snap fitting in the hub to secure the tape end.

I hope that the cliché "Getting there is half the fun" applies to our current transition between the two great mistresses of the day: Annie Log and Di Gital. Princess Di is undoubtedly the Queen Bee of the future, but ol' Annie has been a loyal companion and deserves to not be shuffled prematurely into dinosaur-like extinction. Here's hoping, and good luck. A

IT TOOK THE BEST HEADS IN THE INDUSTRY TO MAKE AKAI'S NEWEST PROFESSIONAL DECK.


½ Track
GX Record Head


½ Track
GX Playback Head


¼ Track
GX Playback Head


Erase Head

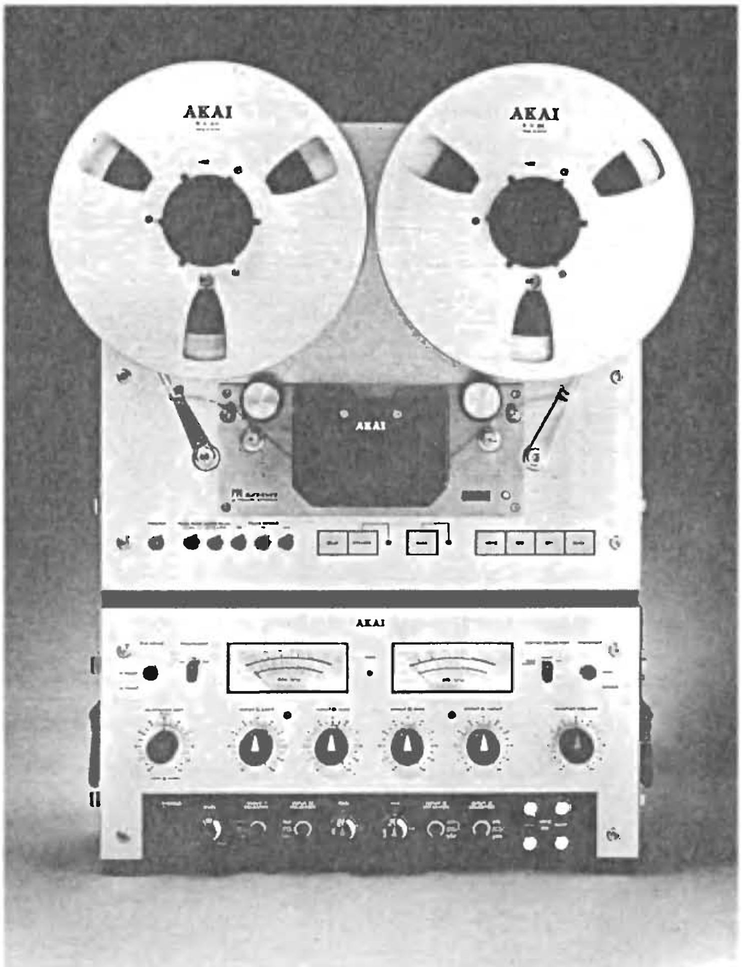
For years, AKAI's patented glass and crystal (GX) heads have been making recorded history. Not only for unsurpassed sound quality, but for unequalled wearability as well. Guaranteed, in fact, for 150,000 hours; the equivalent of playing 24 hours a day for almost 17½ years.

Now AKAI puts its heads together in one dynamite machine: the PRO-1000. The four head, 2-track mastering machine that's as much at home on location as it is in the studio.

See the PRO-1000 at your AKAI dealer or write us for information.

But do it soon. Because it's already turning the heads of a lot of people.

Features:
☐ 3 Motors ☐ 3 Speeds ☐ 4 Heads ☐ ½ Track Record/Play and ¼ Track Play ☐ 10½" Reel Capacity ☐ Large Illuminated 40 dB meters; read Peak, VU, and Bias ☐ Four In/Two Out Mixer, built-in ☐ Panpots ☐ 20 dB Microphone input attenuators ☐ Variable EQ and Bias controls ☐ Special inputs for outboard noise reduction units, i.e. "DOLBY,"* "DBX." ☐ Double Capstan, Closed Loop Drive System ☐ Remote Control operation (optional RC-17 and RC-18) ☐ Feather touch, full logic solenoid control system ☐ NAB playback standards ☐ Fade-in and fade-out controls ☐ Separate sections for tape transport and tape amplifier with heavy duty carrying handles on both sections ☐ Pre-set clutches on all input level controls.



Specifications:
Wow and Flutter 15 IPS: 0.025% WRMS, 7½ IPS: 0.04% WRMS, 3¾ IPS: 0.08% WRMS ☐ **Frequency Response** 15 IPS @ "0" VU: 50-20 kHz ± 1 dB, 7½ IPS @ "0" VU: 40-24 kHz ± 3 dB, 3¾ IPS @ "0" VU: 60-12 kHz ± 3 dB ☐ **Overall Distortion** Not more than 1% @ 1 kHz @ "0" VU for all speeds ☐ **Signal to Noise Ratio** 62 dB ☐ **Heads** (4), ½ Track GX Record, ½ Track GX Playback, ¼ Track GX Playback, Full Track Erase ☐ **Motors** (3), (1) AC Servo Capstan Drive Motor, oil circulating, center pole generated (CPG). (2) Eddy Current Motors, for fast forward and rewind, oil circulating ☐ **Inputs** Microphone (4), Line (4) ☐ **Outputs** Line (4), Mixer (2), Headphone (1).

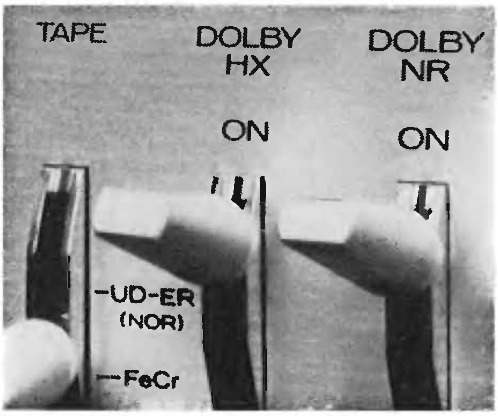
*TM of Dolby Labs., Inc.

AKAI

AKAI America, Ltd., 2139 E. Del Amo Blvd., P.O. Box 6010, Compton, CA 90224

DOLBY® HEADROOM EXTENSION: A Significant Advance in Cassette Recording

At higher frequencies, with even the best tape formulations, there are two major problems in cassette recording. The most familiar is hiss, background noise which is particularly annoying at higher frequencies. The other is tape saturation, the inability of tape to capture high frequencies at high levels. You may have noticed tape saturation as the dulling of highs on percussion, brass instruments, or other program material rich in high frequencies, as well as the distortion of closely-miked sibilant voices.



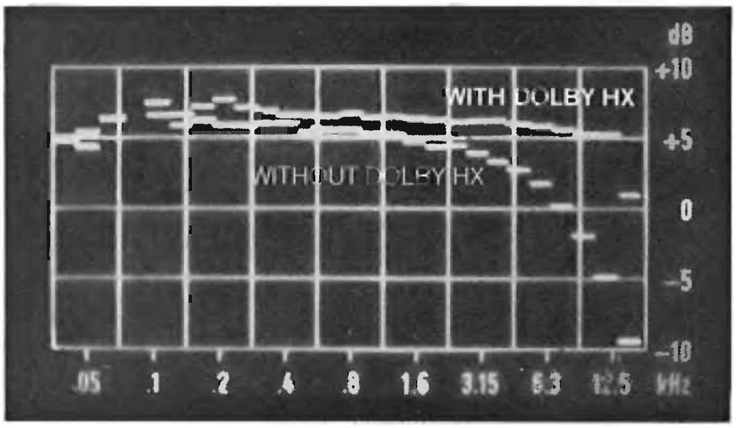


FIGURE 1. Pink noise recorded with and without Dolby HX on a low-cost iron oxide cassette tape, at an average level of −10 dB (referred to Dolby level).

For years Dolby B-type noise reduction has been an effective treatment for tape hiss as a serious problem in cassette recording. Now a new development from Dolby Laboratories significantly reduces high-frequency tape saturation as well.

Dolby HX

Dolby headroom extension, or Dolby HX for short, is new circuitry which works in conjunction with Dolby noise reduction in a recorder to improve significantly the usable dynamic range of any tape, particularly at high frequencies. As you can see from Figure 1, Dolby HX permits recording information at 10 kHz and above at a level on the order of 10 dB higher than is currently possible. In addition, as shown in Figure 2, there is a substantial reduction of the severe IM distortion that results when tape saturates. And finally, Dolby HX also optimizes performance at low and middle frequencies for minimum distortion, modulation noise, and drop-out effects.

How Dolby HX works

Dolby HX works by automatically varying a recorder's bias level in response to the changing high-frequency content of the music being recorded. At the same time, the recording equalization is automatically modified to prevent any change in frequency response. Therefore at each moment, Dolby HX provides just the right bias and equalization to optimize tape performance for the music, unlike the fixed bias and equalization of conventional decks which must compromise tape performance at least part of the time.

Much of the time on most music, the bias with Dolby HX is relatively high for best performance at low and mid frequencies. But when unusually high-level high frequencies of the type which would cause tape saturation come along, both bias and equalization pre-emphasis are momentarily lowered to increase the tape's high-frequency headroom far beyond the normal limit. Information about the high-frequency content of the music is derived from the recorder's Dolby noise reduction circuits, which are already programmed to scan the music in precisely the way required by Dolby HX.*

What Dolby HX means to cassette recording

Dolby HX makes it possible to make more accurate recordings of difficult program material, and to make accurate recordings more easily. The improvement is realized on any tape type for which the recorder is set up, so that less-costly iron oxide tapes perform like the more exotic formulations, and the more exotic formulations are further improved.

Just as important, the improvement Dolby HX provides is inherent in the recording process, so no special playback processor beyond normal Dolby noise reduction is required.

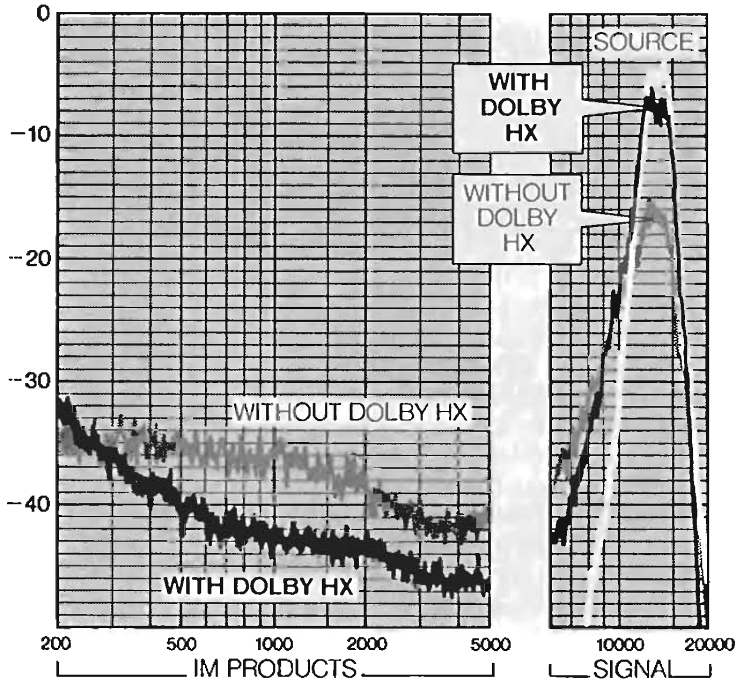


FIGURE 2. As well as increasing high frequency output, Dolby HX reduces IM distortion. *(The curves were obtained by first recording pink noise through a ⅓-octave filter centered at 12.5 kHz to simulate a musical signal such as a cymbal crash. The results were then played back and charted by a sweeping spectrum analyzer.)*

The difference will be heard when playing the tape on any deck. All decks equipped with Dolby noise reduction, and all Dolby encoded cassette recordings, will continue to be fully compatible with each other.

New cassette deck models incorporating both Dolby HX and Dolby noise reduction are on the way; watch for them over the next few months at your hi-fi dealer's. In the meantime, if you would like a complete technical description of how this new development works, please write us at the address below.

DOLBY LABORATORIES LICENSING CORP., 731 Sansome Street, San Francisco CA 94111, Telephone (415) 392-0300. Telex 34409.

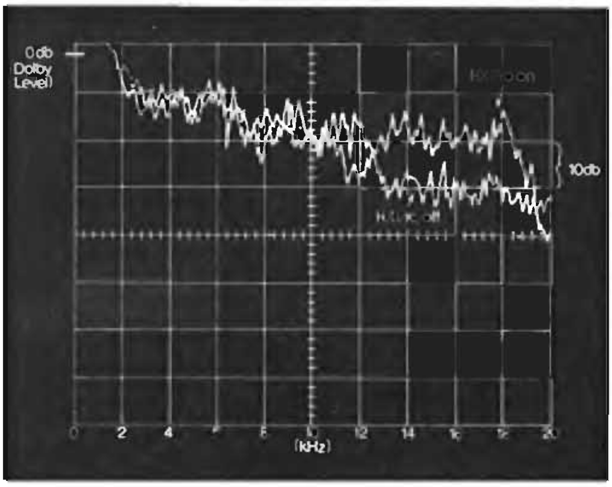
*Dolby HX is sufficiently complex to require engineering a deck from the outset for it; it is not practical to add it to an existing model.

Dolby and the double-D symbol are trademarks of Dolby Laboratories
S79/2237

Dolby®

Dolby® HX Pro

Dolby HX Pro™ headroom extension is a program-adaptive bias technique which can significantly improve the quality of cassette recordings. High-level frequencies can be recorded more accurately, without sacrificing signal-to-noise ratio, while such side effects of tape saturation as distortion are reduced. For both the home recordist and the duplicator of pre-recorded cassettes, Dolby HX Pro improves the performance of good conventional tapes to match that of costlier, more exotic formulations, and even the more expensive tapes benefit from Dolby HX Pro.



Spectral analysis of two high-speed (32 times) cassette recordings of the same selection of rock music show the highest levels accumulated over time at each frequency. Both recordings were made on conventional iron oxide tape of the type favored for commercial cassette duplicating; in this example, the high-frequency headroom improvement provided by Dolby HX Pro is as much as 10 dB.

The problem of self-bias

Even when a cassette deck is adjusted for the nominally optimum bias for a given tape, performance is nevertheless compromised under some signal conditions. In particular, music which is rich in high frequencies has what's called a self-biasing effect. The musical high frequencies act in and of themselves as recording bias on the tape, effectively adding to the external bias supplied by the recorder's bias oscillator. The net result under such signal conditions is momentarily too much effective bias, which leads to the familiar symptoms of tape saturation. The highest frequencies don't get recorded at all, and considerable IM distortion is generated at lower frequencies.

How Dolby HX Pro deals with the problem

Dolby HX Pro uses a special circuit which constantly monitors the total effective bias — a combination of bias from the recorder's oscillator and self-bias contributed by the musical signal — while the recording is being made. If it senses the total bias increasing beyond the optimum level as a result of high frequencies in the music, it instantly compensates for the increase by lowering the bias from the recorder's oscillator, thus keeping the total effective bias constant. Even on music with a great deal of high-frequency energy, the tape remains optimally biased, and so tape saturation and its side effects are significantly reduced. The improvement in high-frequency headroom can be 6 dB or more, depending on the particular tape formulation. No decoding is necessary to realize the benefits of Dolby HX Pro.

Improve both the cassettes you make and those you buy

Dolby HX Pro technology, developed by Bang & Olufsen with the assistance of Dolby Laboratories, is provided along with Dolby noise reduction in home cassette deck models from Aiwa, B&O and Harmon-Kardon.

Just as important, Dolby HX Pro can be applied to high-speed cassette duplication, where its ability to improve good conventional tape formulations is economically, as well as sonically, significant. Commercial duplicating facilities are now equipped, and the first pre-recorded cassettes made with Dolby HX Pro (as well as Dolby noise reduction) are available from the following labels: Capitol, Liberty, EMI/America, Angel, Warner Brothers, Electra/Asylum, and Atlantic.

For further information, including a complete technical explanation of Dolby HX Pro, contact Dolby Laboratories at the address below.

DOLBY HX PRO

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234 SYNCASET

4-Channel Multitrack Recorder

The TASCAM 234 "SYNCASET" 4-Channel Multitrack Recorder is essentially a compact cassette version of our 4-track open-reel recorders/reproducers with a built in 4×2 mixer.

It allows you to build rich, complex recordings—you can add one or more tracks at a time, playing all the parts yourself, until you have created exactly the sound you want. Tracks can be "ping-ponged" back and forth, combining them to create more tracks.

OTHER FEATURES

•dbx NR with On/Off switch •Record function and output select switching matrix •Locator functions •Mic and line inputs •Built-in headphone amplifier with level control •Optional remote control unit (RC-71) •Optional footswitch for punch-in/out control (RC-30P) •Standard 19" rack mounting dimensions.

234 SPECIFICATIONS

•MIC/INST. Input: −60—−10 dBV •LINE INPUT: −10 dBV •Track Format: 4-track, 4-channel •Tape Speed: 9.5 cm/sec. •Wow & Flutter: ±0.06% peak, weighted •Frequency Response: 40—14,000 Hz ±3 dB (0 VU) •S/N Ratio: 95 dB (dbx IN, weighted) •Dimensions (W×H×D): 482×147×357 mm •Weight: 9.8 kg

**The 234L is a low-speed version (normal 4.8 cm/s) of the 234. All other functions and controls are indentical to those of the 234.*

225 SYNCASET

2-Channel Simul-Sync Recorder

The 225 "SYNCASET" is a 4-track/2-channel cassette machine designed for creative production. It offers all the features and functions you need to mix inputs, overdub and "bounce" tracks on standard speed compact cassettes.

In addition to its synchronized overdubbing capability, the 225's INPUT MIX and OUTPUT PAN controls offer unprecedented flexibility in a 2-channel stereo recorder.

Engage the INPUT MIX switch and you can mix both inputs to record on a single track or both tracks. Or, you can mix one or two inputs with an existing track to record on the other track. Using "OUTPUT PAN", you can get a mono mix of two recorded tracks from the output terminals, as well as normal stereo out.

OTHER FEATURES

•Dolby B NR •SYNC switch for insert recording •RC-30P optional remote punch in/out foot switch •RM-225 optional 19" rack mount kit •Mic and line inputs for each channel.

225 SPECIFICATIONS

•MIC Input: −60 dBV •LINE Input: −10 dBV •Track Format: 4-track, 2-channel •Tape Speed: 4.8 cm/s •Wow and Flutter: 0.07%, NAB weighted •Frequency Response: 40 Hz—14 kHz, ±3 dB (CrO_2 tape) •S/N Ratio: 61 dB (weighted, with Dolby B NR) •Dimensions (W×H×D): 432×111×284 mm •Weight: 5.0 kg

133/133B

Stereo Plus Cue

The TASCAM 133/133B Stereo Plus Cue is a complete 3-channel production tool for the multi-image industry. With it you can produce a master control tape, stereo sound and sync, and then run the show with its many microprocessor controlled features. The 133 and 133B offer dual speed operation, for a choice of high audio quality or greater tape economy and compatibility. The 133 has unbalanced inputs and outputs, while the 133B offers balanced lines.

OTHER FEATURES

•Dolby B NR •Auto-Present computer system
•RC-133 optional remote control

133/133B SPECIFICATIONS

•Track Format: 4-track, 3-channel. •Tape Speeds: 4.8 cm/sec., 9.5 cm/sec. •Wow and Flutter (peak, weighted): ±0.085% at 4.8 cm/sec., ±0.055% at 9.5 cm/sec. •Audio Frequency Response, 0 VU: 30 Hz—8 kHz at 4.8 cm/sec., 30 Hz—15 kHz at 9.5 cm/sec. •S/N Ratio: 55 dB at 4.8 cm/sec. and 9.5 cm/sec. •Dimensions (W×H×D): 482×147×345 mm. •Weight: 9 kg.

122/122B
Master Cassette Recorder

The 122/122B is a standard cassette format recorder/reproducer with dual speed operation. The standard 4.76 cm/sec. speed offers full compatibility with standard consumer units, and the higher 9.5 cm/sec. speed allows the 122/122B to be used for several important studio applications. High quality reference dubs are possible for clients with two-speed playback capability, and the 122 can also be used as a spot machine for broadcasts. While the 122 has unbalanced inputs and outputs, the 122B offers balanced inputs and outputs.

OTHER FEATURES
•3-heads •Front-panel bias and level adjustment •Dolby B NR •RC-90 optional remote control unit

122/122B SPECIFICATIONS
•Track Format: 4-track, 2-channel stereo. •Tape Speeds: 4.8 cm/sec., 9.5 cm/sec. •Wow and Flutter (peak, weighted): ±0.085% at 4.8 cm/sec., ±0.055% at 9.5 cm/sec. •Frequency Response (Metal Tape, 0 VU): 35 Hz—14 kHz at 4.8 cm/sec., 35 Hz—20 kHz at 9.5 cm/sec. •S/N Ratio: 58 dB at 4.8 cm/sec., 60 dB at 9.5 cm/sec. •Dimensions (W×H×D): 482×147×345 mm. •Weight: 9 kg.

112
Master Cassette Recorder

The TASCAM 112 is a 4-track 2-channel cassette machine designed and created specifically for professional applications. Special care has been taken in the transport system to ensure unfaltering non-stop operation under the most demanding working conditions.
The 112 incorporates the Dolby HX Pro system for enhanced high frequency reproduction through increased high frequency MOL. It also has cueing in both the fast forward and rewind modes for easy location.

OTHER FEATURES
•Standard 19" rack mountable •Dolby B and C NR •Selectable front (1/4") and rear (RCA) input terminals •±12% pitch control •Headphone jack with level control •Optional RC-71 remote control unit •High-slew-rate ICs.

112 SPECIFICATIONS
•Track Format: 4-track 2-channel stereo •Tape Speed: 4.8 cm/s (1-7/8 ips) •Wow and Flutter (peak, weighted): ±0.08% •Frequency Response (Metal): 25 Hz—19 kHz •S/N Ratio: 59 dB (NR Out) •Dimensions (W×H×D): 482×133×297 mm •Weight (net): 6.1 kg.

112R
Bi-Directional Cassette Recorder

The TASCAM 112R is the deck of choice for professional applications requiring extended playback and record capability. Auto reverse operation offers obvious long play/record benefits, but an optional interface allows two or more 112Rs to be connected serially for even broader applications potential.
The symmetrical bi-directional transport with Super Acculign Rotating Head offers an identical transport environment in both the forward and reverse tape directions.

OTHER FEATURES
•19" rack mountable •Dolby B and C NR •3-head system •Optional RC-205K remote control unit kit additionally provides CPS, Intro Check, Memory Start/Stop and Block Repeat functions •Digital tape counter with TRT/Index modes •±15% pitch control •Automatic tape type selection •L & R mic inputs.

112R SPECIFICATIONS
•Track Format: 4-track 2-channel stereo •Tape Speed: 4.8 cm/s (1-7/8 ips) •Wow and Flutter (peak, weighted): ±0.06% •Frequency Response (Metal): 25 Hz—20 kHz •S/N Ratio: 60 dB (NR Out) •Dimensions (W×H×D): 479×118×290 mm •Weight (net): 4.8 kg.

SERIES 30

Recorders/Reproducers

38

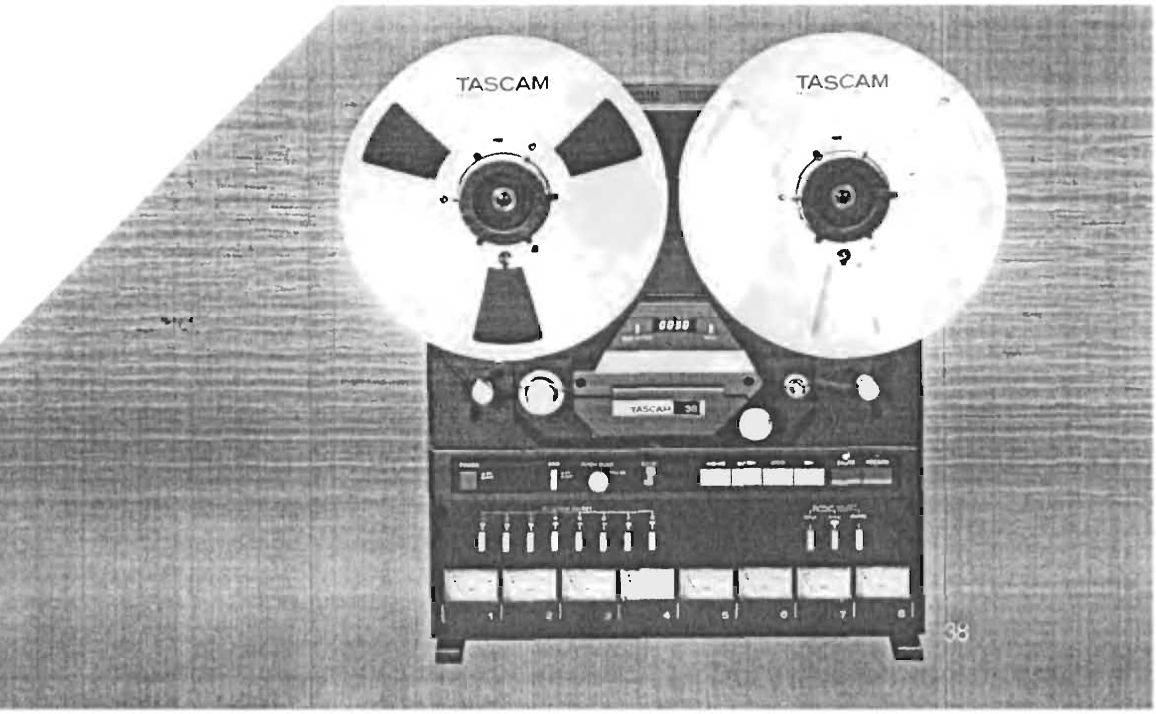
8-Track, 8-Channel

34B

4-Track, 4-Channel

32

2-Track, 2-Channel



You provide the talent, and we will provide the technology—the Series 30 recorder/reproducers. These three decks are engineered from the unique TASCAM point of view that insists on professional quality and performance made affordable to the musician, artist, engineer, etc. But that does not mean we compromise on parts, technology or versatility.

Since professional quality reproduction is the most important goal in any piece of audio equipment, we took no chance with the Series 30 decks. We designed and built every single part that goes into them. We made significant improvements starting with the heads. The Series 30 sync and repro heads offer identical response so you don't lose sound quality during track bouncing operations. Further, contour effect has been minimized and head life has been extended by 20%. Even the erase heads are improved with a material that allows more complete erasure.

A rock solid head mount assembly, closer head spacing or more precise punch-in performance, and a new chassis are further Series 30 improvements.

The Series 30 transport design and construction is superior in every respect. Special DC reel motors achieve extremely high torque to minimize wow and flutter and provide higher fast-wind speed. A belt-driven, FG servo controlled DC capstan motor ensures exceptionally precise tape speed. A microprocessor transport control guarantees smooth, positive switching with no appreciable stop between fast wind and play/record modes. The electronics in each Series 30 deck are also special. Lower noise levels are realized with the use of selected integrated circuits and amplifiers. Only the highest quality circuit components are used throughout, for the purest reproduction quality possible.

You get the most comprehensive, easy-to-use sync channel assignment system available. On each deck, each track has a FUNCTION SELECT button that places it in the SAFE or RECORD READY mode. There are three OUTPUT SELECT buttons that determine the source of the line output: INPUT, SYNC or REPRO.

Remote punch-in recording is also possible using the remote transport control unit (RC-71) or a remote footswitch (RC-30P). Each deck offers a CUE lever, ZERO RETURN function, PITCH control, expanded VU meters, and full dual process dbx noise reduction compatibility (DX-2D/4D optional). The 32 and 34B also offer 19 cm/sec tape speed along with the studio standard 38 cm/sec. Other features that are extra on the 32 and 34B are microphone inputs, MIC/LINE switches and 0/20 dB attenuators, input and output level controls and a headphone monitor output.

38 SPECIFICATIONS

•Track Format: 8-track, 8-channel, 1/2" tape. •Reel Size: 10-1/2". •Tape Speed: 38 cm/sec. •Wow and Flutter (peak, weighted): ±0.06%. •Frequency Response (0 VU): 40 Hz—22 kHz, ±3 dB. •S/N Ratio: 68 dB. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 410×461×317mm. •Weight: 27 kg.

34B SPECIFICATIONS

•Track Format: 4-track, 4-channel, 1/4" tape. •Reel Size: 10-1/2". •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak, weighted): ±0.06% at 38 cm/sec., ±0.09% at 19 cm/sec. •Frequency Response (0 VU): 40 Hz—22 kHz, ±3 dB at 38 cm/sec., 40 Hz—16 kHz, ±3 dB at 19 cm/sec. •S/N Ratio: 68 dB at 38 cm/sec., 66 dB at 19 cm/sec. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 410×461×256 mm. •Weight: 20 kg.

32 SPECIFICATIONS

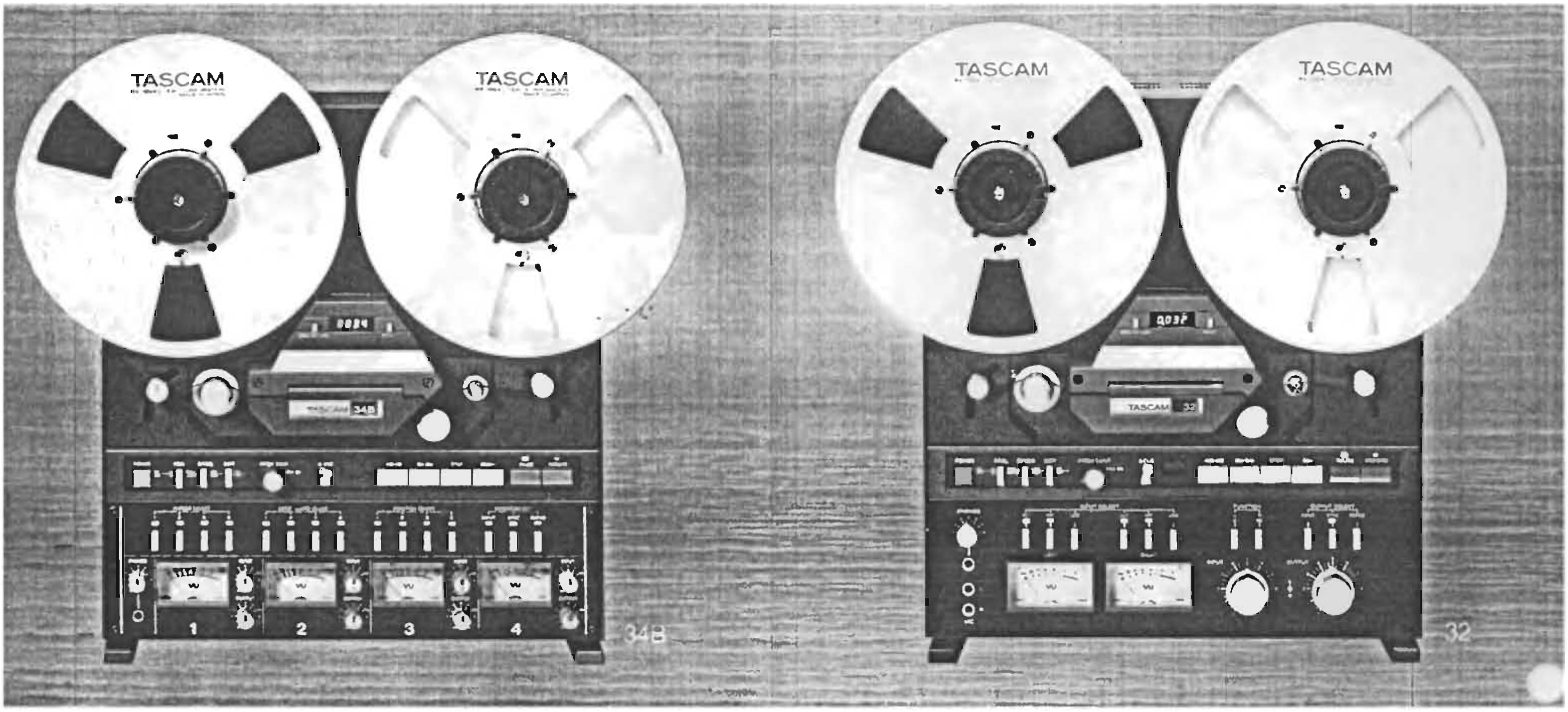
•Track Format: 2-track, 2-channel, 1/4" tape. •Reel Size: 10-1/2". •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak, weighted): ±0.06% at 38 cm/sec., ±0.09% at 19 cm/sec. •Frequency Response (0 VU): 40 Hz—22 kHz, ±3 dB at 38 cm/sec., 40 Hz—16 kHz, ±3 dB at 19 cm/sec. •S/N Ratio: 68 dB at 38 cm/sec., 66 dB at 19 cm/sec. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 410×461×256 mm. •Weight: 20 kg.

22-4 SPECIFICATIONS

•Track Format: 4-track, 4-channel 1/4" tape. •Reel Size: 7" max. •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak weighted): ±0.07% at 38 cm/sec., ±0.09% at 19 cm/sec. •Frequency Response (0 VU): 40 Hz—22 kHz, ±3 dB at 38 cm/sec., 40 Hz—16 kHz, ±3 dB at 19 cm/sec. •S/N Ratio: 61 dB at 38 cm/sec., 60 dB at 19 cm/sec. •THD: 1%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 416×410×260 mm. •Weight: 18 kg.

22-2 SPECIFICATIONS

•Track Format: 2-track, 2-channel 1/4" tape. •Reel Size: 7" max. •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak weighted): ±0.07% at 38 cm/sec., ±0.09% at 19 cm/sec. •Frequency Response (0 VU): 40 Hz—22 kHz, ±3 dB at 38 cm/sec., 40 Hz—16 kHz, ±3 dB at 19 cm/sec. •S/N Ratio: 68 dB at 38 cm/sec., 64 dB at 19 cm/sec. •THD: 1%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 410×326×231 mm. •Weight: 14 kg.

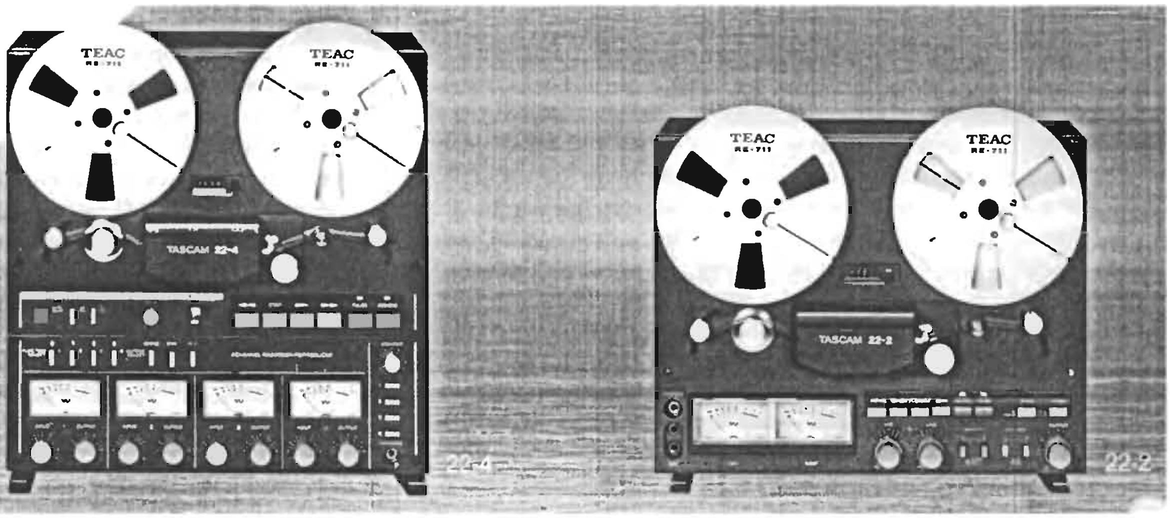


SERIES 20

Recorders/ Reproducers

22-4
4-Track, 4-Channel

22-2
2-Track, 2-Channel



Both the 22-2 half-track recorder/reproducer and the 22-4 4-track recorder/reproducer offer the quality of 38 cm/sec plus the long-play convenience of 19 cm/sec tape speeds. They also offer the sure operating precision of full logic transport control.

Both decks employ three independent motors—a DC servo controlled capstan motor specifically designed for speed accuracy keeps wow and flutter to an absolute minimum, and induction reel motors maintain smooth, even tape handling and precise tape-to-head contact.

The TASCAM 22-2 2 track is equipped with all the functions you need for studio master recording/reproduction: independent Monitor and Record mode controls for each channel, Mic/Line mixing, expanded scale VU meters with a −20 to +5 dB display range, and RP-22 remote pause capability in either record or play.

The TASCAM 22-4 is an ideal partner for the 22-2, with a full complement of functions that give you professional multitrack recording/reproduction performance. The use of Function and Output Select switches makes setting up record and sync channel assignments a simple pleasure.

Punch-in recording is another important multitrack function available with the 22-4. It lets you re-record selected sections of a track, instead of having to re-record the whole track. Other useful 22-4 features include output and input level controls for each channel, headphone monitor capability, a Memory Stop function, ±6% pitch control, manual cueing, expanded-scale VU meters, RC-30P punch in/out foot switch, RC-71 remote control capability and the ability to add an optional dbx noise reduction system.

SERIES 40

Recorders/Reproducers

48

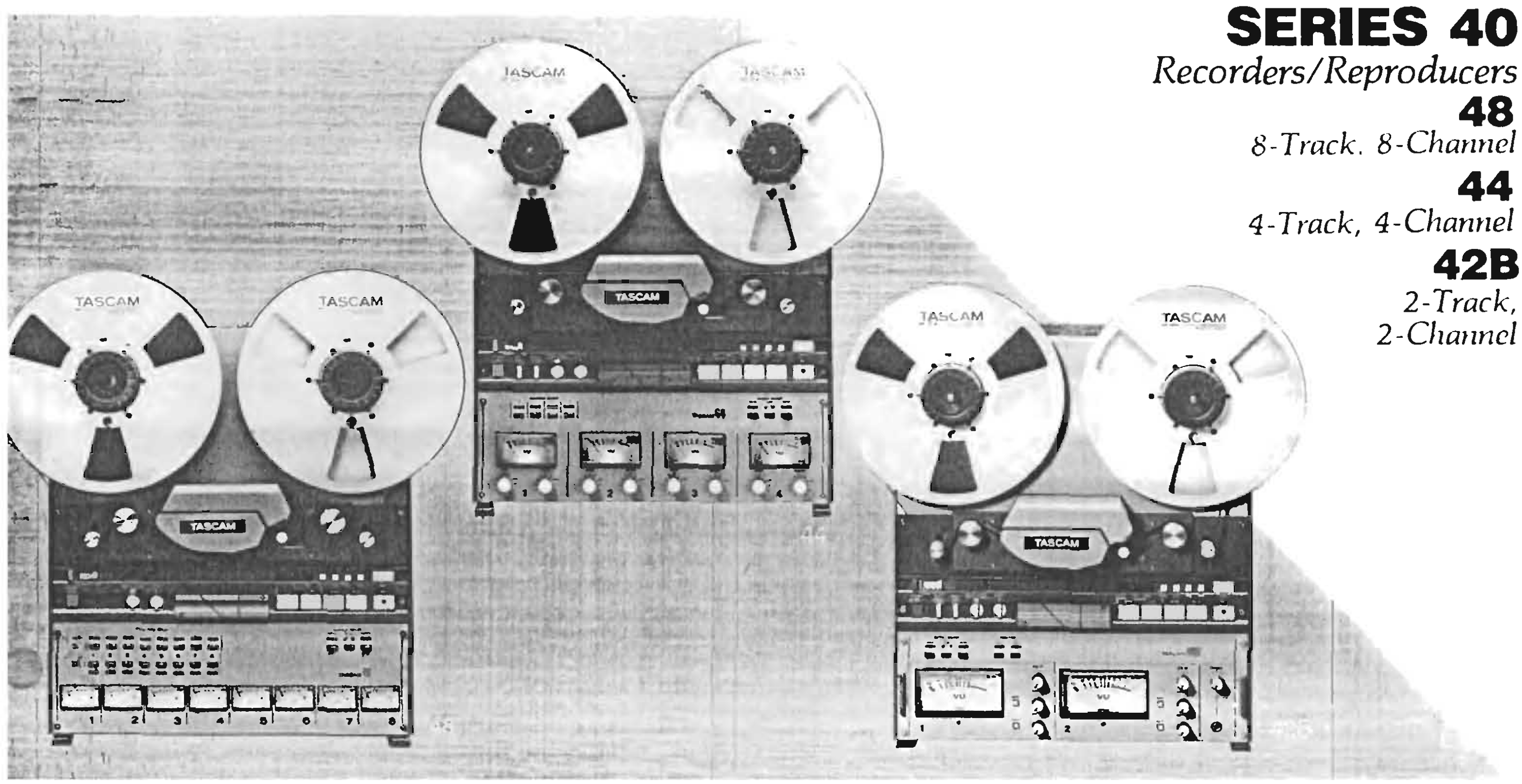
8-Track, 8-Channel

44

4-Track, 4-Channel

42B

2-Track, 2-Channel



The TASCAM "40-Series" recorders/reproducers are high-quality machines designed for production work where flexibility, reliability and synchronized operation are prime requirements. The 42B and 44 use 1/4" tape and have selectable 38 or 19 cm/sec tape speeds, while the 48 uses 1/2" tape at a fixed 38 cm/sec speed. Nominal +4 dBm balanced inputs and outputs are standard, and −10 dB RCA jacks are also provided for the broadest compatibility with all types of equipment.

All three motors—capstan, takeup and supply reel—are servo controlled so the tape comes up to speed in minimum time. The capstan motor is a high-performance low inertia, brushless DD type controlled by a precision PLL servo system. Further, a TTL I/O port is provided for direct compatibility with a broad range of SMPTE/EBU interlock systems. This is a great advantage in just about any recording situation, but particularly when synchronizing multiple recorders under SMPTE/EBU control with minimum lockup time.

Two convenient autolocator functions help to take your mind off the machinery during production: Return to Zero (RTZ) which parks the tape at 00:00, and Search to Cue (STC) which locates any preset cue point in either direction Pushing the FF or REW buttons a second time while in FF or REW modes activates the Spooling mode.

The near-straight tape path is easy to thread, and a clean control layout is easy to use. Each track has its own Function selector (Ready/Safe), and Output selectors (Input/Sync/Repro) are provided for fast, easy sync/rec mode setup. The 48 even has pre-load selectors that make it a breeze to rehearse overdubs or make inserts with exactly the right signal in the monitors. Sync response is exactly the same as repro response, so your mixing and performance values remain the same throughout all phases of production. And if you're working alone, you can punch in and out of the record mode with an optional footswitch (RC-30P).

Naturally, the electronics are superb. A high-stability bipolar power supply drives the high-performance direct-coupled amplifier circuitry, ensuring exceptionally low noise, low distortion and broad dynamic range. Direct coupling of the heads to the repro amplifiers and differental FET input circuitry ensures low-noise, precise transient response and ideal phase characteristics. Modular amp construction on plug-in glass-epoxy printed circuit boards makes maintenance and repair simple, and special metal glaze trimmer pots ensure precise, easy adjustment. DX-2D/4D dbx NR systems are optionally available for the 40-series decks, offering significantly expanded dynamic range for exceptionally clean recording quality.

OTHER FEATURES

- Ceramic capstan shaft
- ±12% pitch control
- VU meters with peak LEDs
- Dump edit/manual edit/Stop edit capability
- Optional AQ-65 Auto-Locator
- Optional RC-71 Remote Control Unit

48 SPECIFICATIONS

•Track Format: 8-track, 8-channel, 1/2" tape. •Reel Size: 10-1/2". •Tape Speed: 38 cm/sec. •Wow and Flutter (peak, weighted): ±0.08%. •Frequency Response (0 VU): 40 Hz—20 kHz ±3 dB. •S/N Ratio: 69 dB. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 432×505×315.5 mm •Weight: 37 kg

44 SPECIFICATIONS

•Track Format: 4-track, 4-channel, 1/4" tape. •Reel Size: 10-1/2". •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak, weighted): ±0.08% at 38 cm/sec., ±0.12% at 19 cm/sec. •Frequency Response (0 VU): 40 Hz—20 kHz, ±3 dB at 38 cm/sec., 40 Hz—16 kHz at, ±2 dB 19 cm/sec. •S/N Ratio: 69 dB at 38 cm/sec., 67 dB at 19 cm/sec. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 432×505×250 mm. •Weight: 34 kg.

42B SPECIFICATIONS

•Track Format: 2-track, 2-channel, 1/4" tape. •Reel Size: 10-1/2". •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak, weighted): ±0.08% at 38 cm/sec., ±0.12% at 19 cm/sec. •Frequency Response (0 VU): 30 Hz—22 kHz, ±2 dB at 38 cm/sec., 30 Hz—16 kHz, ±2 dB at 19 cm/sec. •S/N Ratio: 70 dB at 38 cm/sec., 68 dB at 19 cm/sec. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 432×505×250 mm. •Weight: 32 kg.

SERIES 50

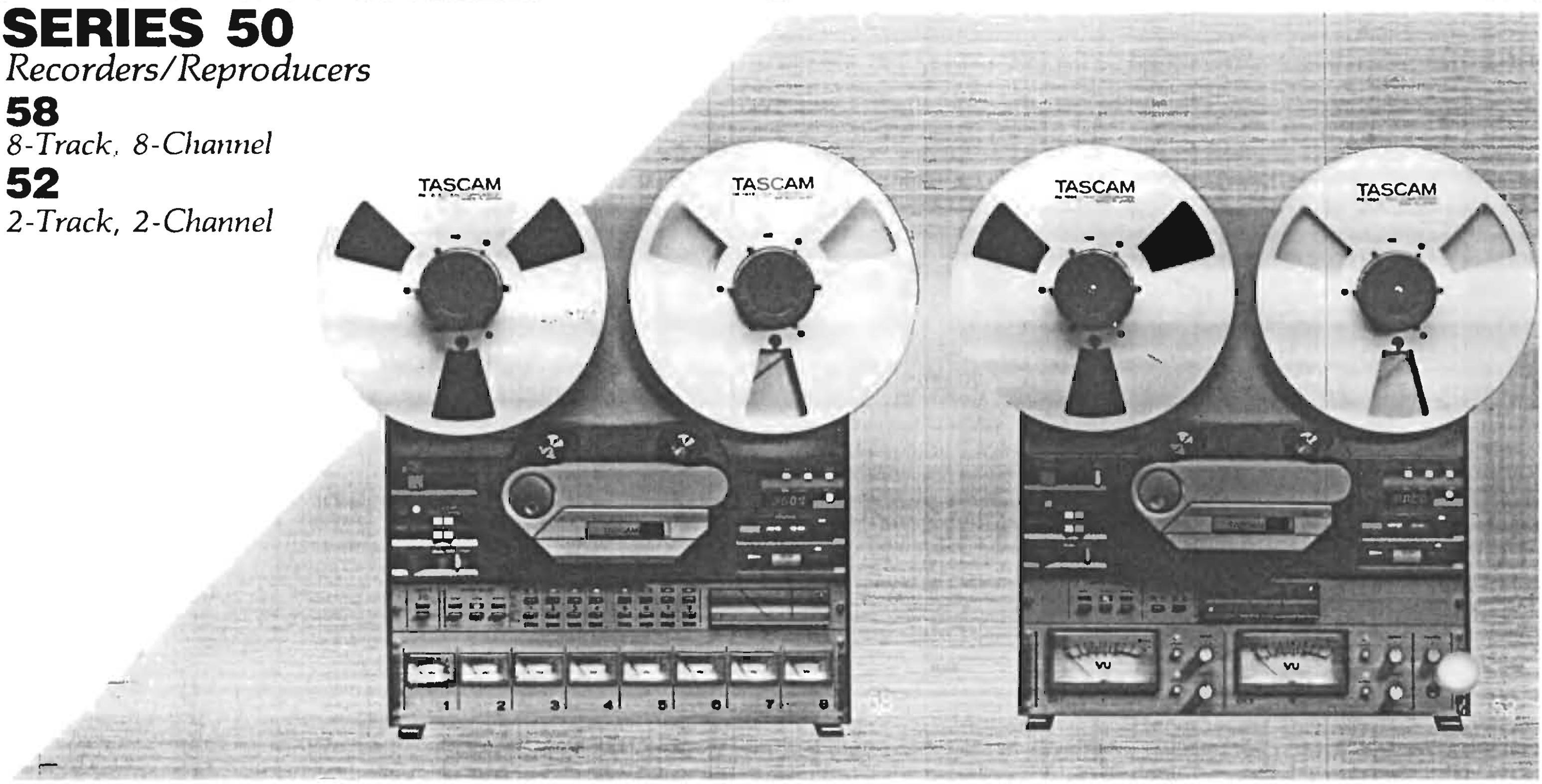
Recorders/Reproducers

58

8-Track, 8-Channel

52

2-Track, 2-Channel



The Series 50 recorder/reproducers are a pair of truly superior audio production machines. The transports are exceptionally rugged, assembled on thick aluminum alloy base plates. The extra heavy-duty motors are built to shuttle tape at high speeds, and have full servo system transports controlled by microprocessors with multiple tach and photosensor inputs to ensure reliable tape handling. The transports in both the 52 2-track recorder/reproducer and the 58 8-track recorder/reproducer are more than capable of keeping up with the stringent demands of slaved editing to video and film systems, and are compatible with SMPTE/EBU equipment. All three motors are included in the microprocessor servo system.

Hand lapped heads, made by our own craftsmen, are at the heart of the Series 50's superb audio performance. They mean lower modulation noise, lower head bumps, sync response equal to repro response. Direct coupled electronics, with ultra low noise FET differential playback head amplifiers, provide low distortion and maximum common mode rejection without transformers. There's one circuit board per channel and all trimmers are adjustable from the front, simplifying routine alignment and servicing. Balanced inputs and outputs are provided in addition to unbalanced lines for compatibility with all types of equipment.

The Series 50 decks are equipped for faster and more versatile production:

- Monitor tracks or inputs as needed using individual Pre-load and Record Mode switches, plus Output Select switches. (58)
- Sync recording capability.
- Tach driven digital display with positive or negative real-time readout.
- Return-to-Zero and Search-to-Cue.
- Dual pitch controls: Coarse ±15% and Fine ±0.7%.
- One-handed cueing.
- 52 also has input/output level controls with cal/uncal selectors and a headphone jack with level control.
- Rear Panel Accessory connector for an SMPTE/EBU interlock system or the optional AQ-65 Auto-Locator.
- Choose either the RC-50 remote control unit with all transport control function for the 52 and 58, or the RC-51 remote control unit which has transport functions as well as rec/sync channel assignment for the 58.
- Optional dbx noise reduction (DX-2D/4D).

The Series 50 decks incorporate unsurpassed audio electronics:

- Balanced and unbalanced inputs and outputs.
- All amplifiers are direct coupled for lowest distortion and optimum low-frequency response.
- First stage sync and play head amplifiers use differential paired ultra-low noise FETs, for better transient response and phase characteristics.
- Use of 8 different power supplies, including bipolar 15V for all audio amps, for optimum performance and reliability.
- VU meters plus peak LEDs.

58 SPECIFICATIONS

•Track Format: 8-track, 8-channel, 1/2" tape. •Reel Size: 10-1/2". •Tape Speed: 38 cm/sec. •Wow and Flutter (peak, weighted): ±0.08%. •Frequency Response (0 VU): 30 Hz—24 kHz, ±2 dB. •S/N Ratio: 69 dB. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 432×505×316 mm. •Weight: 35 kg.

52 SPECIFICATIONS

•Track Format: 2-track, 2-channel, 1/4" tape. •Reel Size: 10-1/2". •Tape Speeds: 38 and 19 cm/sec. •Wow and Flutter (peak, weighted): ±0.08% at 38 cm/sec., ±0.12% at 19 cm/sec. •Frequency Response (0 VU): 30 Hz—24 kHz, ±2 dB at 38 cm/sec., 30 Hz—20 kHz, ±2 dB at 19 cm/sec., 0 VU. •S/N Ratio: 70 dB at 38 cm/sec., 70 dB at 19 cm/sec. •THD: 0.8%, (0 VU, 1 kHz.) •Dimensions (W×H×D): 432×505×316 mm. •Weight: 32 kg.

We didn't have to make a better 2 track than our RS-1500. So we made a 4 track. Introducing the RS-1506.

Ingenuity is truly rare. Repeated ingenuity is true genius. Like the Technics 4-track RS-1506. It offers twice the program time of our 2-track RS-1500.

It also offers the award-winning RS-1500's "Isolated Loop" tape transport with a quartz-locked, phase-controlled, direct-drive capstan.

By isolating the tape from external influences we minimized tape tension to a constant 80 mgs. Providing extremely stable tape transport and low head wear. While reducing modulation noise and wow and flutter to a point where they are barely measurable on conventional laboratory equipment.

Electronically, too, Technics RS-1506 provides the same level of professional control as its predecessor. A separate microphone amplifier. Mixing amplifier. And separate three-position bias equalization switches. While IC full-logic function permits absolute freedom in switching modes. Also available is an optional full-feature infrared wireless remote control (RP-070). It lets you operate all transport functions and record from up to 20 feet.

For the same performance as the RS-1506 with the convenience of auto reverse, there's the RS-1700.

Compare specifications. Even with the best 2-track decks. TRACK SYSTEM: 4-track, 2-channel recording, playback and erase. 2-track, 2-channel playback 4-head system. FREQ. RESP.: 30-30,000Hz, ±3dB (−10dB rec. level) at 15ips. WOW & FLUTTER: 0.018% WRMS at 15ips. S/N RATIO: 57dB (NAB weighted) at 15ips. SEPARATION: Greater than 50dB. RISE TIME: 0.7 secs. SPEED DEVIATION: ± 0.1% with 1.0 or 1.5mil tape at 15ips. SPEED FLUCT.: 0.05% with 1.0 or 1.5mil tape at 15ips. PITCH CONTROL: ±6%.

Technics 4-track RS-1506 and auto-reverse RS-1700. A rare combination of audio technology. A new standard of audio excellence.

Technics
Professional Series

Review of the Present Status of Magnetic Recording Theory

W. W. WETZEL*

PART I

In this series of three articles, Dr. Wetzel presents the first complete discussion of magnetic tape recording theory for engineers.

IN VIEW OF THE growing postwar interest shown by engineers in magnetic recording and because of the scattered and sometimes inaccessible literature on the subject, a review of the basic theory and the present status of the art is of value. This article is designed to familiarize engineers who will ultimately make use of this recording medium with current views on some of the factors which govern the quality of sound reproduced from tapes and wires.

It seems advisable to divide the article into several short sections. Part I will explain the measurement of some basic properties of magnetic media and the effect of these properties on recordings. Part II will cover phases of recording, playback and erase. In Part III consideration will be given to noise background, distortion, equalization and the effects of media velocity.

This presentation is made on the basis of tape as the recording medium, and the text is illustrated with experimental results obtained on magnetic recording tape. This is a consequence of our laboratory having collected the predominant amount of its information on tape rather than wire.

Minnesota Mining & Mfg. Co., 900 Fauquier Ave., St. Paul 6, Minn.

All of the discussion on fundamental magnetic properties applies to wire as well as tape. This is also true of the record, playback and wipe theory. Some forms of distortion exist in wire which are not inherent in tape, and these will be evaluated in the concluding paragraphs.

Although magnetic recording has a relatively venerable history, there remain a number of phenomena upon which scientists working in the field do not agree, or for which basic explanations have not been found. An attempt will be made to present several points on view on controversial subjects.

Hysteresis Loop Tracer

Preceding a discussion of the pertinent magnetic properties of recording media it may be of general interest to describe briefly the instrument with which these properties are determined. We are interested in measuring constants such as coercive force, remanence and saturating field which are derivable from the saturated hysteresis loop. Useful information is also obtained from the unsaturated and minor loops as we shall show. Some instrument on which hysteresis loops may be studied is an essential to a laboratory studying these magnetic materials.

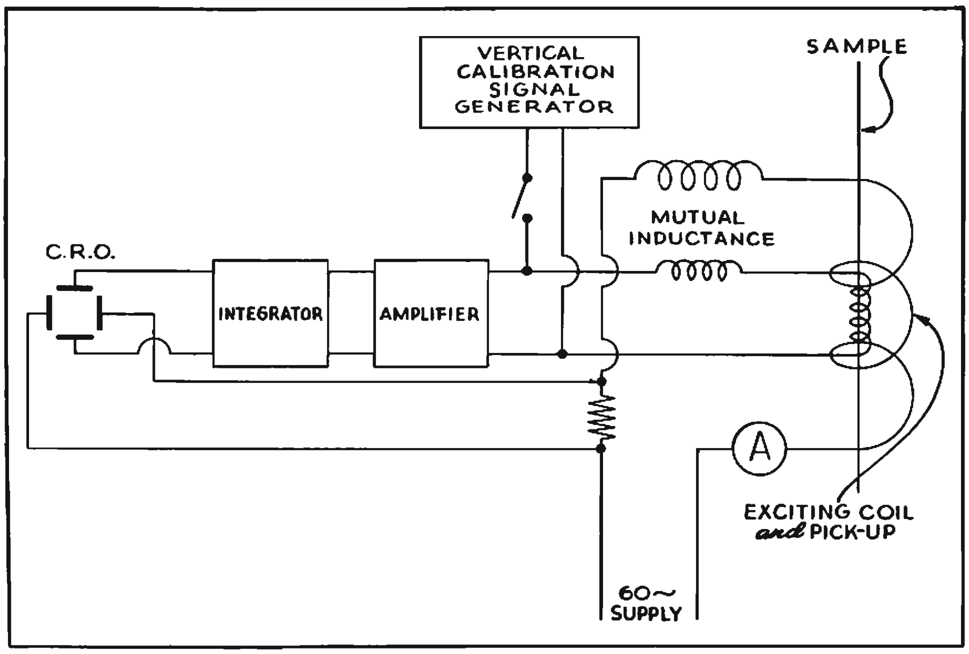


Fig. 1. Schematic diagram of a hysteresis loop tracer.

The requirement of a commercial laboratory where many samples are evaluated in a day is primarily that of speed. A high degree of precision is not demanded by most exploratory work, reproducibility of five per cent satisfying many of the requirements. High sensitivity is essential since the entire magnetic layer on a tape has a cross section of about 2.5 x 10^{-4} square inches and permeabilities are extremely low. The cross section of active material may be increased by stacking tapes in bundles for the purposes of measurement. This has the advantage of providing average values over many lengths of the tape but a marked disadvantage in the time required to prepare the sample for testing. It is considered desirable, because of the ease of sample preparation, to operate on a single tape.

These requirements for speed, reasonable accuracy and high sensitivity in a determination have been met in a number of laboratories by one form or another of a sensitive hysteresis loop tracer. This instrument produces the hysteresis loop on the screen of a cathode-ray oscilloscope. Here the values of interest may be read directly, or, if they are to be studied in detail, the loop may be photographed for leisurely analysis.

The components of a hysteresis loop tester are shown schematically in *Fig. 1.* The essential parts are the exciting and pickup coils, the mutual inductance for balancing the output to zero when no sample is present, amplifier, integrator and calibration equipment. The exciting coil, which supplies the a.c. field in which the sample is examined, is connected in series with the primary of the mutual inductance. The pickup coil is connected in series opposing with the secondary of the mutual inductor which is so designed that its output cancels that of the pickup coil in the absence of a sample. When a sample tape is inserted in the pickup coil a resultant signal is supplied to the amplifier which is proportional to the time derivative of B-H. B, the magnetic induction, is the number of lines of force per square centimeter existing in the magnetic medium. H, the applied field, is a measure of the flux lines per square centimeter of the exciting field in the absence of magnetizable material. The change in flux in the pickup caused by the insertion of a sample is the addition of a B contribution and the subtraction of the H previously existing in the space occupied by the magnetic sample. Hence this instrument responds to the B-H characteristic of the sample.

The output signal for the small samples examined requires amplification before application to the oscilloscope plates. Integration is needed to give deflections proportional to B-H rather than the derivative. H may be calculated from the current and the turns on the exciting solenoid. The horizontal deflecting voltage may be obtained from a series resistor and the gain of the horizontal amplifier adjusted to give a convenient deflection from a given value of H. Similarly the vertical scale may be calibrated through a knowledge of the number of turns on the pickup coil and the application of a standard calibration signal.

It is seen that with calibrated horizontal and vertical deflections proportional to H and B-H respectively a hysteresis loop will be traced on the C.R.O. screen and absolute values in c.g.s. units may be read from the trace.

Instead of reading B-H directly a value $\emptyset$ equal to the increment in flux in maxwells due to the insertion of a sample is usually read. If the cross section A of the sample is known, B-H and finally B may be obtained from the known value H of the applied field and the relation

$$B - H = \emptyset/A.$$

A discussion of the design details of one form of the loop tester is scheduled for publication shortly.[1]

A similar piece of equipment has been described by Long and McMullen.[2]

It should be recalled that in any properly designed permeammeter provision must be made to remove magnetic poles from the region of measurement. This is necessary to eliminate the demagnetizing forces of the poles at the point where B is determined. In loop tracers the effect of poles is made negligible by magnetizing a relatively long piece of tape in the essentially uniform field of a solenoid and confining the pickup coil to a short region in the center of the solenoid. The B-H curve traces are therefore characteristic of the basic magnetic properties of the sample. Having determined the basic magnetic properties the effects of demagnetizing forces, which depend upon the geometry of pole distribution, can be calculated in certain cases. The effects of demagnetizing forces are important in magnetic recording and will be discussed later.

Fig. 2 shows a view of a variation of the Wiegand-Hanson loop tracer. *Fig. 3* shows a photograph of a hysteresis loop for one type of tape. The possibly unfamiliar shape of this curve is due to the fact that B-H vs H is being traced rather than B vs. H. This results in the saturated condition approaching a horizontal rather than a 45° line as the asymptote.

Magnetic Constants

The magnetic constants of interest to us may be read directly on or derived from readings taken from the loop tracer.

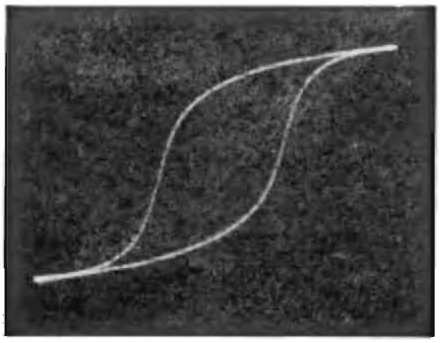


Fig. 3. Photograph of a hysteresis loop characteristic of one type of high coercive force recording tape.

Fig. 4 illustrates a possible hysteresis loop where the applied field H is plotted against $\emptyset$. Three values which concern us, the saturating field, remanent flux and coercive force, may be read directly from the chart. If the value of the cross section A of the active magnetic layer on the tape is known, we obtain B_r, the remanence, from

$$B_r = \emptyset_r/A.$$

If A is expressed in square centimeters B_r is in gauss. The remanence to a great extent governs the signal output of a tape at long wavelengths.

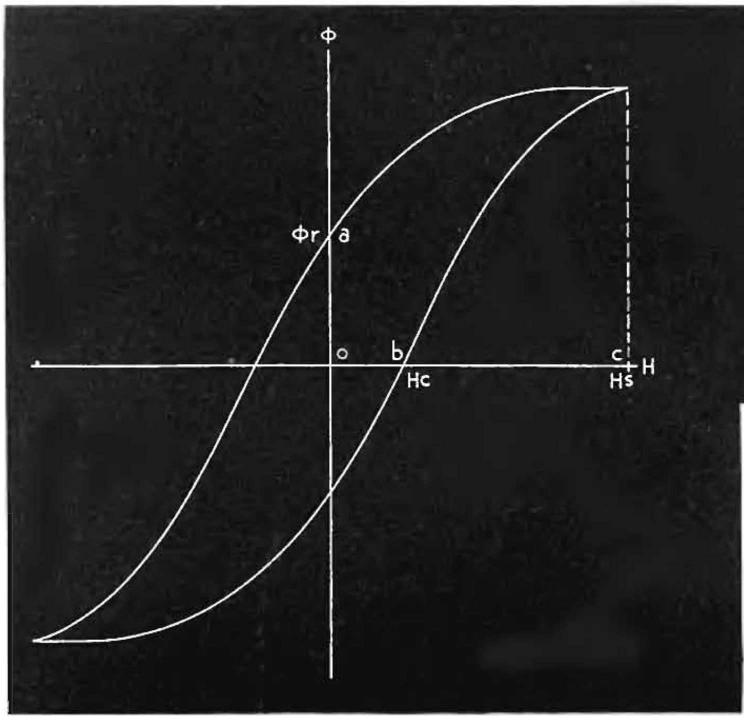
As will be shown later, the effective output of a tape is controlled at low frequencies by remanent magnetization. The dominant response is due to the material in the active layer immediately adjacent to the playing heads. The contribution of deeper layers becomes increasingly unimportant as depth is increased. For reasonable thickness of active material on a tape it is not true that a 6 db increase in output may be obtained by doubling the thickness of the coating. Doubling the coating thickness will of course double $\emptyset_r$ without affecting B_r. However, doubling B_r will be shown to double the low frequency output. It seems probable that B_r is more nearly characteristic of the behavior of a tape than $\emptyset_r$.

Because of the difficulty of measuring A within an error of $\pm$ 20% for either wire or tape, it has been proposed that $\emptyset_r$ rather than B_r be used to specify the recording medium. As B_r appears at present to more nearly characterize tape response, this proposal has not as yet been generally accepted and the materials continue to be specified in terms of remanence.

Saturation field H_s, the distance *oc* *Fig. 4*, may be obtained from the curve tracer by direct reading. For practical purposes the lowest value of H which brings $\emptyset_r$ to its maximum value may be defined as H_s. For many materials which saturate slowly this definition does not lead to easily reproducible values. As will be discussed later, H_s is of use in determining the value of the erase field required to obliterate a signal on the tape. Erase fields are now established experi-

Fig. 2. A modified Wiegand-Hanson loop tracer in operation. The long solenoid in the center background is the exciting coil in which a sample tape has been inserted. The smaller symetrically spaced coils are mutual inductors used for calibration and null point balance. The amplifier, integrator and power supply are housed in the relay rack. Traces of the hysteresis loops are observed on the C.R.O. screen. One useful modification in the instrument is an automatic cutoff of the primary current when the temperature of the solenoid reaches 100° C. The temperature sensitive element is a thermistor embedded in the windings of the exciting coil.

1. D. J. Wiegand and M. W. Hanson, "A 60-Cycle Hysteresis Loop Tracer for Small Samples of Low-Permeability Materials," *AIEE Transactions.*
2. T. E. Long and G. D. McMullen, "A B-H Curve Tracer for Magnetic Recording Wire," *AIEE Transactions, 65*, 146, March 1946.



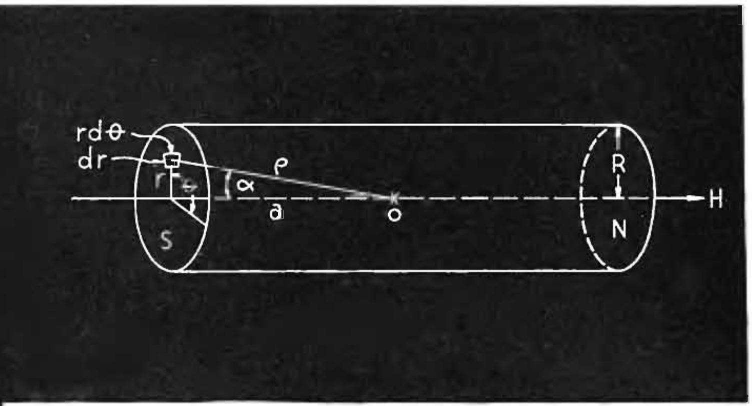


Fig. 4 (left). The form of hysteresis curve obtained on the loop tracer where the flux increment ∅ is plotted against the exciting field H.

Fig. 5 (above). Axially magnetized cylinder used to illustrate demagnetization principles in recording tape.

mentally for any given tape and H_s is only used qualitatively to predict the relative ease of wipe of two tapes. Some indefiniteness in determinations of H_s may therefore be tolerated.

The coercive force H_c is measured as the intercept *ob* on the H axis. H_c is the field in oersteds necessary to reduce the induction ∅ to zero after the material has been saturated. It is a measure of the ability of a medium to retain its state of magnetization under the influence of demagnetizing fields and, as will be explained later, is a measure of the short wavelength output of a tape.

It should be noted that, through usage in magnetic recording laboratories, it has become customary to express the coercive force of recording materials as the value of the field at which $\emptyset = O$ rather than the value at which $B = O$. From the equation relating the quantities

$$B - H = \emptyset / A$$

we see that the definitions approach equivalence only when B is large compared with H, i.e., for high permeability materials. In the low permeability materials, such as we are considering, the two definitions lead to values which differ by large percentages.

Demagnetization Effect

We have so far considered plots of ∅ where measurements were obtained under idealized conditions in a hysteresis loop tester. The conditions are idealized to the extent that terminating poles in the tape were kept remote from the region of measurement. This was done to evaluate the basic properties of the material independent of geometry of pole distribution. In recording, this ideal situation is not generally realized. While in a loop tracer the dominant poles are several inches from the region where induction is determined, the separation of the poles from the point of maximum induction in a recording is measured in tenths to thousandths of an inch. In recording, effects of demagnetization forces are usually negligible at low frequencies but become considerable at high frequencies. An explanation of the phenomena will help in understanding the behavior of magnetic records at high frequencies.

Instead of considering directly the demagnetizing forces occurring in a recording tape where geometry complicates the mathematics it will be helpful if we consider the simplified problem of demagnetizing forces in an axially magnetized cylinder. The principles are the same and general conclusions for tape may be drawn from the analogy.

Suppose a cylinder of magnetizable material is placed in a magnetic field H so that the axis coincides with the direction of the field as shown in *Fig. 5*. The cylinder will become magnetized with poles appearing over the entire surface. Let us make two simplifications, which will not affect our general qualitative conclusions, by assuming that poles are formed only on the ends of the cylinder and that the density distribution over each end is uniform and equal to m unit poles cm.[2] We are interested in evaluating the induction B at the center O of the cylinder since the higher the value of the induction the g r e a t e r the pole strength. (In a recording tape the output signal is proportional to the pole strength and therefore to B.)

In the complete absence of poles the induction is given by $B = \mu H$ where μ is the permeability of the medium. When poles are present they act to decrease the effective value of H. The presence of an elemental south pole of strength $m\, r\, dr\, d\Theta$ on a point at the origin is to decrease the effective value of H by an amount

$$mr\, dr\, d\Theta \,/\, \rho^2 \cos a$$

or

$$amr\, dr\, d\Theta \,/\, \rho^3$$

For the decrease in effective field due to the poles covering one face of the cylinder we get:

$$am \int_0^{2}\int_0^{R} \frac{r\, dr\, d\Theta}{(a^2 + r^2)^{3/2}} = 2\pi m \left(1 - \frac{a}{(R^2 + a^2)^{1/2}}\right)$$

or

$$2\pi m\,(1 - \cos A)$$

Where A is the angle subtended at the center between the axis and the circumference of the cylinder end. Since there are two faces which have identical effects at the center, we have for the demagnetizing force $4\pi m\,(1 - \cos A)$, and since it acts to oppose H, the equation for B is:

$$B = (H - 4\pi m\,[1 - \cos A])$$

It can be seen if the cylinder length is increased, keeping the radius constant, $\cos A \rightarrow 1$ and the term due to the demagnitization force becomes small. This is analogous to the situation in the loop tracer where the angle subtended by the ends of the tape at the center of the pickup coil is small enough to permit disregarding the demagnetizing force in our measurement of ∅.

As the cylinder is made shorter, A increases toward 90° and the demagnetization force increases although never actually reaching the value of H which gives rise to it.

With different geometry the same condition exists in magnetic records. With very long wavelengths where the poles are far removed from the point of symmetry, the value of the induction B is not affected. As the wavelength on the record is decreased, the demagnetizing forces increase, lowering the induction without, however, being able to reduce the value completely to zero.

Through considerations of the effects of demagnetizing forces it is possible to draw conclusions concerning the effects of remanence and coercive force on the playback characteristic of tapes. Theoretically at least, it is possible to obtain complete quantitative data on tape records from B vs. H curves and demagnetization constants. The measurements are somewhat tedious and the results as yet not as satisfactory as actual record and playback tests. The approach through calculation remains attractive since it offers a means of evaluating tapes without involving the recording and playback head characteristics.

It is evident in the example of the cylinder that the greater the induction B in a magnetic material the greater the value of m as each line must terminate at a pole. One can show that the demagnetizing field H_d for a given distribution of poles is directly proportional to B, or

$$H_d = -CB$$

This equation, where C the demagnetizing constant contains all the geometry, can be combined with a B vs. H curve, which defines the magnetic behavior of the medium freed from geometry, to obtain the effect on B of demagnetizing forces. This may be done graphically as shown in *Fig. 6*.

Lines 1 and 2 represent plots of H_d against B for two values of geometry or wavelengths on a tape. Line 1 where the value of C is small represents the effect at intermediate wavelengths. Line 2 where C is large gives the demagnetizing field as a function of B for a relatively short wavelength. The points of intersection of each line with the hysteresis loop defines the point of equilibrium for the material characterized by the loop under conditions of geometry and demagnetizing force defined by the line.

For very long wavelengths where $C = O$ an *ac* saturation field recorded on the moving medium will develop a maximum induction of B_r gauss. Intermediate and short wavelengths will result in decreasing remanent inductions of B_{r1} and B_{r2} respectively. If a material were to double its coercive force, B_r remaining constant, the intersection of the hysteresis loop and Line 2 would occur at about twice the value of B_{r2} illustrated. The low wavelength or high frequency output would in this case double without much effect on the low frequency response. Conversely, doubling the remanence while holding the H_c value constant would double the low frequency output with little effect on the high frequencies.

In a later section these predictions on the effect of coercivity and remanence will be shown to hold in terms of frequency response and output level on record-playback tests. It will be shown that, within limits, something may be done to tailor-make a tape to response specifications through the selection of remanence and coercive force of the active material.

Increase of Induction During Replay

We have examined some of the effects of the demagnetization forces in a tape after it has passed the recording head. Even though the same magnetizing forces are employed (we have been considering saturation fields), the remanent flux in any magnetic recording medium drops as the wavelength is decreased. There is an additional effect on playback where the magnetized material contacts the playback head which tends to restore the output to some degree. The per cent recovery in induction will be shown to become greater the shorter the wavelength recorded.

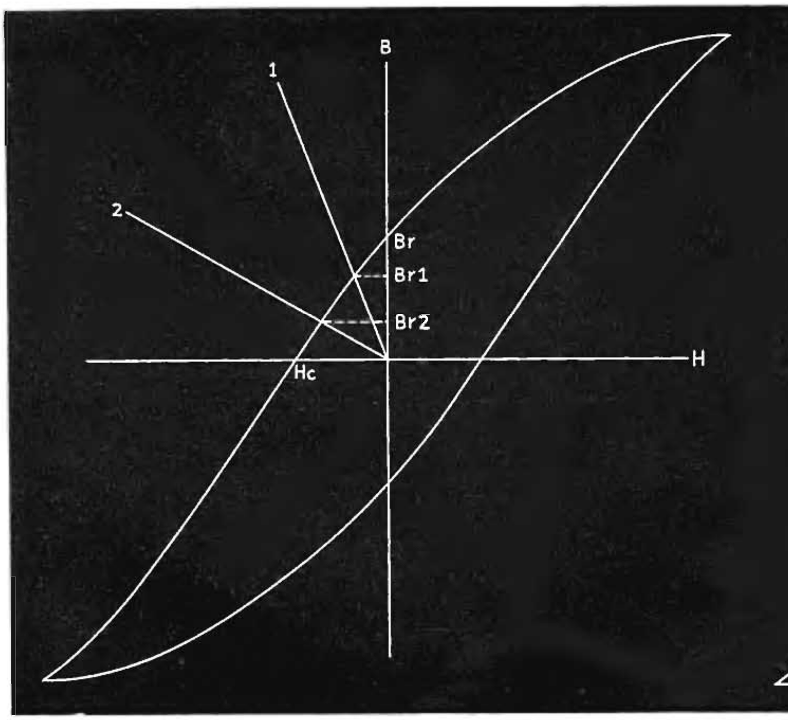
The recovery effect during replay can be understood by reference to the minor hysteresis loops *ac* and *a′c* shown in *Fig. 7*. If a hysteresis loop has been established and through some means we reverse the field at point *a* after the material has passed from *c* to *a*, instead of retracing the path *aa′b″c*, the state will traverse a minor loop *abc*. If a material, such as the rod of *Fig. 5*, has been brought to saturation (point *c*, *Fig. 7*), and the field H removed, the magnetic state traces the curve *cb″a*, coming to equilibrium with the demagnetization force at point *a*. If we then remove the demagnetizing force, neutralizing the poles by placing a "U" shaped keeper in contact with the faces of the cylinder, the condition of magnetization changes. The path described is *ab* on the minor loop and the new equilibrium point is *b*. A portion but not all of the induction lost through the demagnetizing force is recovered through the application of the keeper.

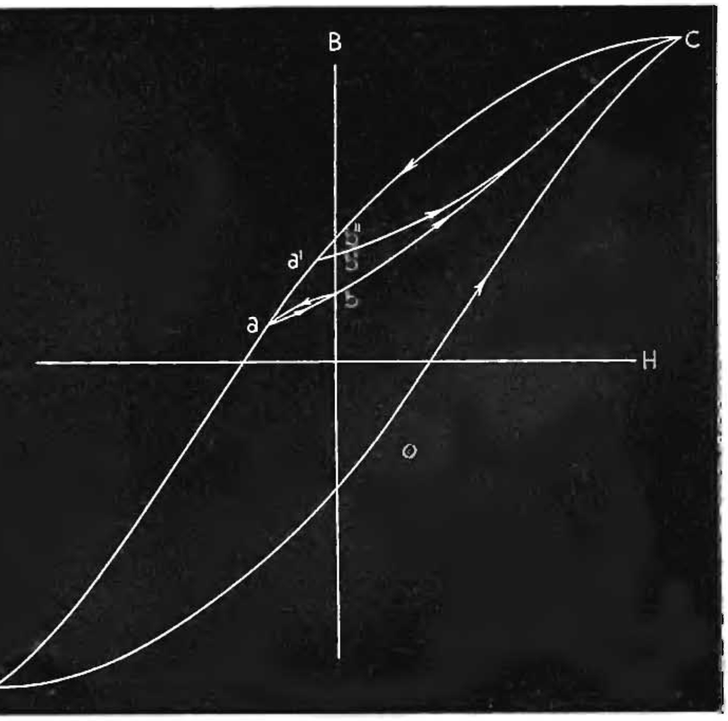
During playback of a wire or tape, the poles on the medium are to a large extent neutralized by the core of the playback head and the effective induction increased, although never to the point which would have been achieved had demagnetization forces not been present originally. This increase of induction represents useful flux increase as the output at the playback head depends on the value of B in the tape at the time of contact with the head, not the value of B either before or after contact.

Neutralization is effective only as long as the poles on the tape are in contact with the playing head. After passing the head the original state of magnetization is restored without loss to the magnetic record. The analogous case in the magnetized cylinder is the removal of the keeper. The magnetic state of the cylinder then passes from *b* to *a* along the upper branch of the minor loop as indicated by the arrow. [*Con. on page 39*]

Fig. 6 (left). This normal hysteresis loop is used to show the effects on B_r of two conditions of demagnetization.

Fig. 7 (right). The minor hysteresis loops illustrate the behavior of a magnetic tape during playback.





Recording Theory

[*from page 17*]

Since after playback the state of magnetization has returned to its initial condition, playback energy is not derived at expense of the magnetic energy stored in the media but from the energy supplied by the drive mechanism. This is directly analogous to a generator with a permanent magnet field where the output energy is drawn from the power source rotating the generator not from potential energy in the field magnets.

If demagnetizing forces for a certain wavelength bring the condition of magnetization to point a, *Fig. 7*, some longer wavelength will result in a state represented by a'. Upon playback it passes to b'. It is readily seen that the per cent change due to playback is greater the shorter the wavelength on the record.

The output of a magnetic tape is modified by two opposing tendencies. The output decreases with decreasing wavelength as the result of demagnetizing forces. The per cent recovery on playback increases with decreasing wavelength. The sum of the opposing tendencies is a net decrease in output with decreased wavelength.

Conclusion

We have seen from a qualitative discussion that variations in two of the magnetic constants affect the output of recording tapes. Actually the effects are not trivial and in the construction of recording materials careful consideration must be given to the proper balance of remanence and coercive force.

It is not proper at this point to speculate on the choice of optimum values of coercive force and remanence since all the evidence is not before us. Noise levels, erase conditions and velocity of tape drive must be taken into consideration as modifying and limiting our selection. The discussion of optimum magnetic constants will be deferred until these new factors have been considered in Parts II and III of this article.

Review of the Present Status of Magnetic Recording Theory

W. W. WETZEL*

PART II

In this series of three articles, Dr. Wetzel presents the first complete discussion of magnetic tape recording theory for engineers.

IN PART I of this article we discussed the hysteresis loop tester and how certain basic properties of magnetic materials could be obtained from readings on that instrument. We saw, as the result of demagnetizing forces, the residual induction in a recorded tape may be expected to decrease with decreased wavelength. It was further shown in a qualitative fashion that the coercivity of the magnetic material governs the remanent induction at short and the remanence governs the remanent induction at long wavelengths.

Part II is designed to familiarize the readers with the current status of erase, record and reproduction theory. Workers in the field are in general agreement on how a recording is erased or reproduced but there is disagreement on the mechanism operating during the process of recording with the aid of a-c bias. It is believed this disagreement is based on lack of appreciation of two factors: 1) the exceedingly complex magnetic history of a particle as it traverses the gap, and 2) the profound alteration of the induction picture which occurs when the demagnetization forces are brought into play upon removal of the tape from the recording head. The explanation offered here is based on two assumptions which are subject to criticism but which do have the merit of leading to results compatible with experiment.

Because it is a controversial matter more attention will be given to recording than to wipe and reproduction theory.

Minnesota Mining & Mfg. Co., 900 Fauquier Ave., St. Paul 6, Minn.

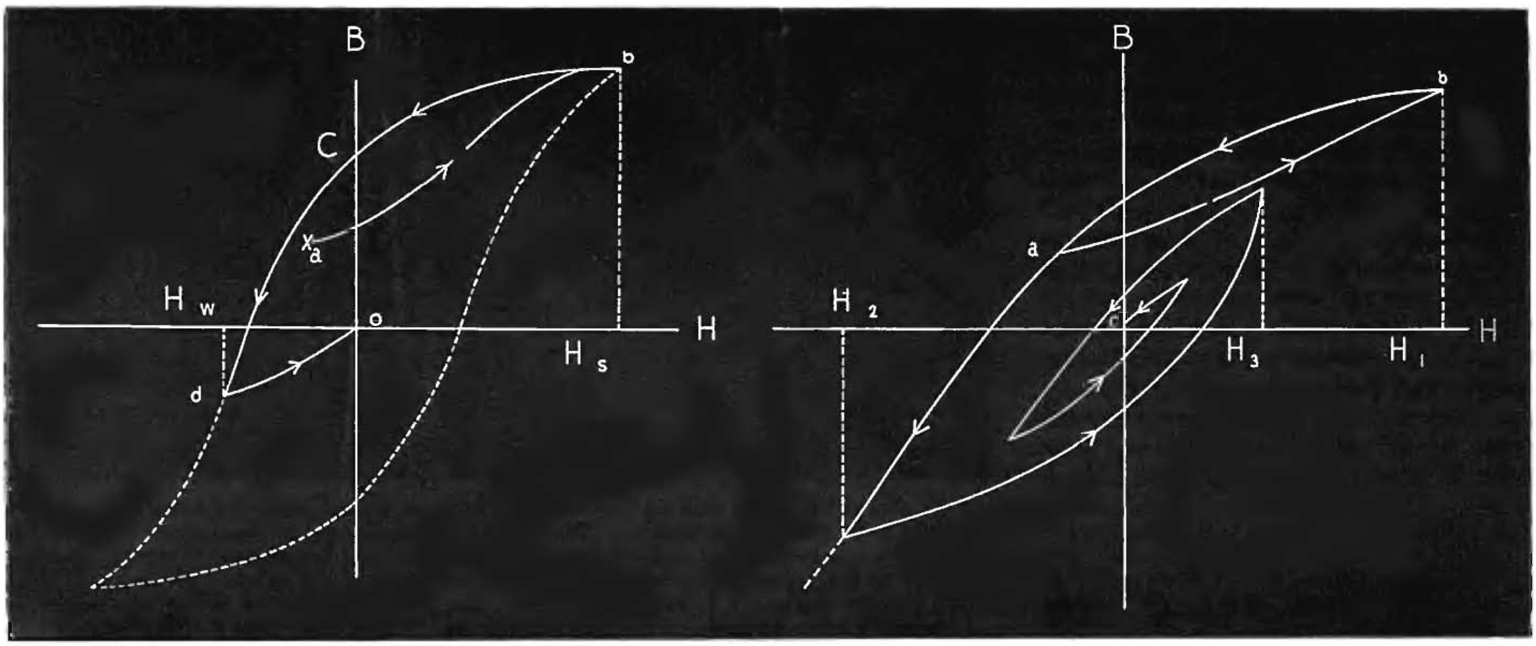
Erase

One of the outstanding advantages found in this form of recording is the fact that the records may be obliterated and re-used without deterioration of the medium. This process is known as magnetic wipe or erase. Obliteration of the record may be made with a d-c wipe in which the tape is brought to a magnetically saturated condition. The d-c wipe is required for systems employing d-c bias fields to obtain linear recording. In some cases, where the saturation field for the material is too high to permit the use of a-c erase, a d-c wipe with a permanent magnet is employed.

When a-c bias is used the recording medium should enter the recording head in as nearly as possible a neutral state of magnetization. This condition may be obtained by two methods, a-c wipe or a two step d-c wipe. *Fig. 1* shows the reversed d-c wipe accomplished by the application of two d-c fields. Starting from any state of magnetization "*a*" for the material, a constant or d-c field is applied by, say, a permanent magnet which is sufficiently strong to more than saturate the material. The application of this field brings the magnetic state to point "*b*." On removing the field the state moves along the saturated hysteresis loop to point "*c*." If following this a very precise negative field H_w is applied the state moves to point "*d*." Upon removal of this negative field the state changes along a minor loop to "*o*" or the non-magnetized condition. While it is theoretically possible to attain perfect wiping in this fashion, the precise value of H_w required is difficult to produce in practice. A further objection to this type of wipe is that the neutral condition is not stable. The noise level tends to increase with mechanical working of the tape after a wipe of this kind.

A better form of obliteration which

Fig. 1 (left). Showing the change in the state of a magnetic material during a d.c. erase. Fig. 2 (right). Diagramming the change in state accompanying an a.c. erase.



can result in noise levels below those of any usual amplifier may be attained by an a-c wipe. This form of erasure is stable in its final result and practical to apply. It is the form now generally employed.

It has been known for a long time that ferro-magnetic materials may be reduced to a neutral state if a series of decreasing fields alternating in sign are applied to the material. The process is illustrated in *Fig. 2*. If the magnetic state is originally that at "*a*," a positive field H_1 is applied which is just sufficient to bring the state to the apex "*b*" of the major loop on which "*a*" lies. This field is removed and a smaller negative field H_2 applied. Successively applied fields, decreasing in magnitude and alternating in sign will, upon removal of the last field, leave the material in state "*c*."

"*C*" will closely approximate the magnetically neutral state and will be stable if a sufficient number of reversals have been employed and the decrease in values is gradual. It has been shown that excellent demagnetization may be produced if a hundred or more reversals with the same incremental decrease are used. That is, if fields proportional to +100, -99, +98, ... -3, +2, -1, O are applied. The frequency of the application of the field is immaterial and square-wave pulses are effective. If a tape is passed over a hundred permanent magnets of the correct strength and polarity to produce the above fields, a good wipe will result.

It should be pointed out that the process illustrated in *Fig. 2* required H_1 to be a precise value, i. e., that field required to bring the state to the tip of the major loop passing through "*a*." This is not essential since somewhat smaller values than H_1 may be used initially as well as any value which is greater than H_1. In erasing a recorded tape it is necessary to approximate in wiping only the value of H corresponding to the highest value of recorded induction. However, it is generally good practice to have several cycles of the wipe field saturate the tape. This guarantees that the wipe will remove overloaded recorded passages.

It is sometimes said that a high coercive force material is difficult to wipe. This is not in agreement with the best theory of erase. It should be stated as "a material with a high saturation field (H_s not H_c) is difficult to wipe."

Figure 3a shows the flux distribution in the neighborhood of an erase head gap. A well designed erase head should saturate at the pole tips, increasing the leakage flux from points remote from the gap. The values of H_h and H_v in the neighborhood of the gap are shown in *Fig. 3b*. This differs from the desirable form of the distribution in the recording head. In recording a sharp drop of the H_h is necessary while in erase a wide gentle decrease is desired. This is to provide many alternating fields of decreasing intensity for each active particle on a tape as it passes over the gap.

It is interesting to note that an overloaded recording head will partially erase the signal it is attempting to record. This is explained by the fact that if the poles of the overloaded head become saturated the field spreads away from the gap and the magnetic material passes through a series of decreasing cycles of the bias field which tends to obliterate the signal.

Recording

As we shall see, the transfer characteristic available in magnetic recording is not linear and some means of "biasing" must be employed to produce outputs of the same wave form as the input audio signal. By analogy with radio tube usage where voltage bias is required to place the operation point on a linear portion of the input-output characteristic, the magnetic fields used to produce this effect in recording have come to be known as bias fields. The word bias is an unfortunate choice in this application since its use tends to influence our thinking in terms of vacuum tube theory rather than in magnetic terms. This has resulted in erroneous applications of the transfer characteristic curve.

Three types of bias have been proposed and used. D-C bias is employed on a previously saturated recording medium to bring the operating point to a position on the saturated hysteresis loop from which it will return to $B_r = O$ upon removal of the field. This is precisely the procedure used in d-c erase, *Fig. 1*. An audio field superimposed on the d-c bias field will leave on the tape values of B_r above or below zero which are proportional to the instantaneous value of the audio field. Many discussions of this method of recording have been published. [1,2,3] That of Lubeck is the most detailed explanation the author has reviewed.

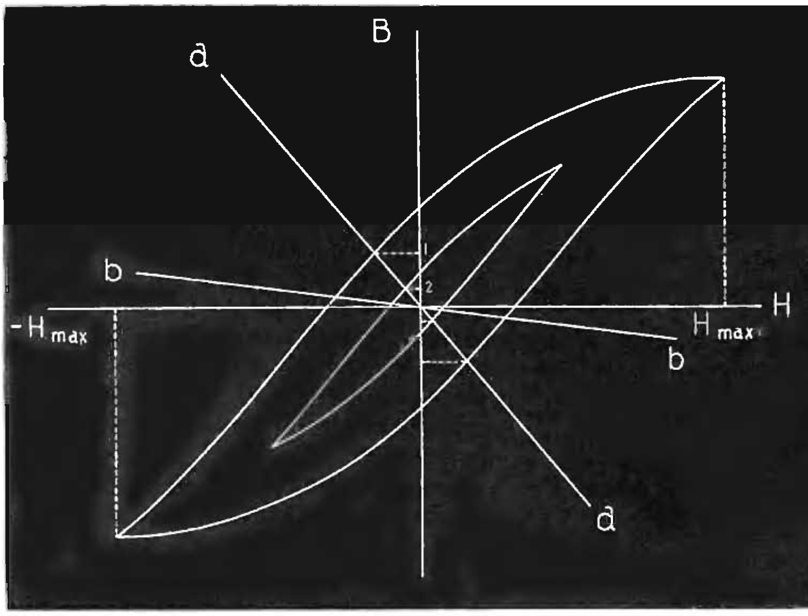
A second method of recording employs a previously saturated tape to which an ultrasonic a-c bias mixed with the audio signal is applied. This resulted in alternate points of saturated and less than saturated tape, and, when the tape is removed from the recording head, demagnetizing forces tend to average the induction to something less than the saturated state. In this type of recording, the zero of the audio induction corresponds to a state of longitudinal magnetization other than zero. As will be shown in Part III, this gives rise to back-

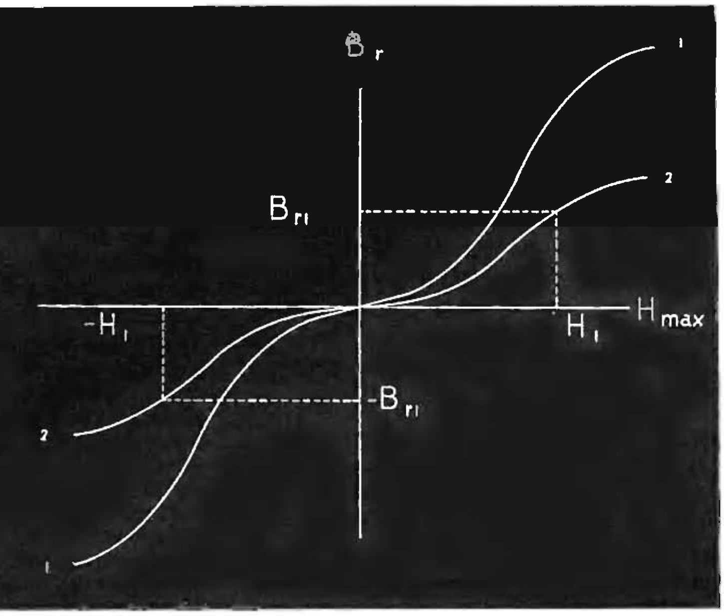
[1] Heinz Lubeck, "Magnetic Sound Recording with Tape and Ringheads," *Akustische Zeitschrift 6* 20, 1937.

[2] C. N. Hickman, "Sound Recording on Magnetic Tape," *Bell System Technical Journal, 16* 165, 1937.

[3] S. J. Begun, "Magnetic Recording and Some of Its Applications in the Broadcast Field," *Proc. I. R. E.*, p. 423, August 1941.

Fig. 5 (left). Illustrating the derivation of transfer characteristics from the hysteresis loops and the demagnetization curves for different wavelengths. Fig. 6 (right). Transfer characteristic curves. Curve 1 shows the case where no demagnetization forces are effective. Curve 2, calculated from Fig 5., illustrates the transfer characteristic for a short wavelength where demagnetization exists.





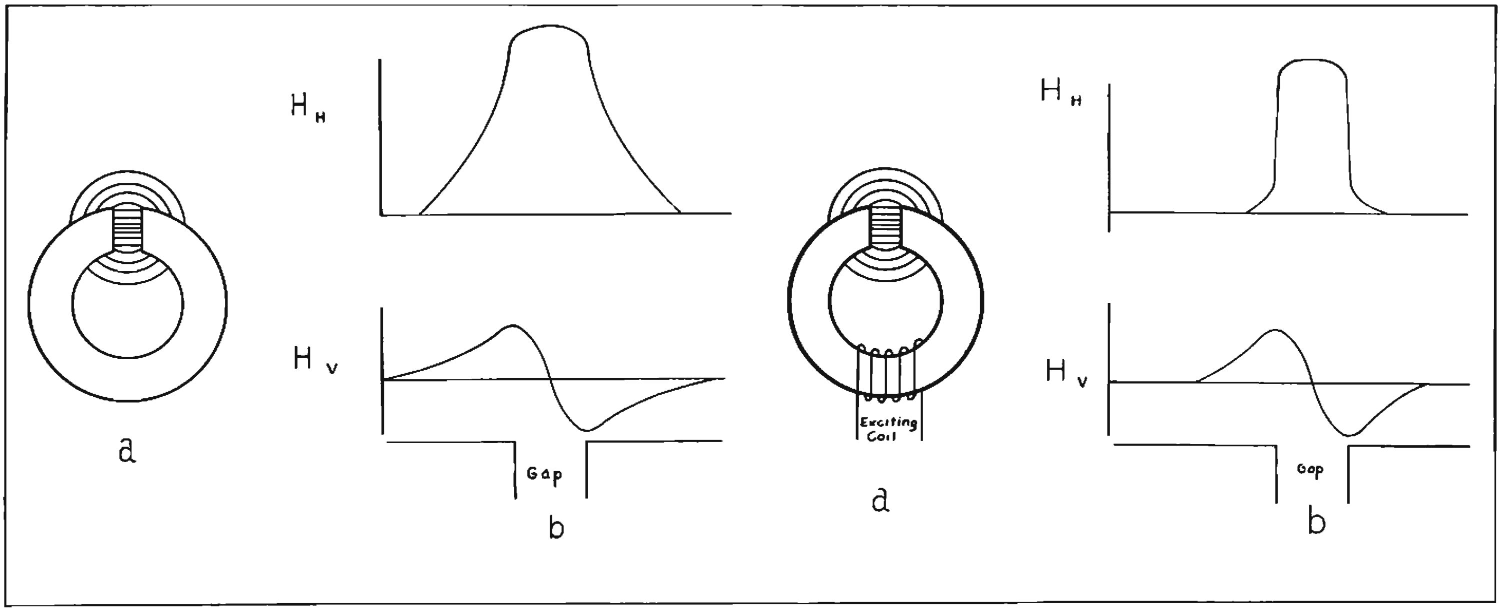


Fig. 3 (left). Sketch showing the field distribution near the gap of an erase head. Fig. 4 (right). Sketch showing the field distribution near the gap of a recording head.

ground noise. No paper on the theory of a-c bias applied to a saturated tape has come to the author's attention.

The third method which employs a magnetically neutral tape with a mixed audio signal field and ultrasonic bias field has been almost universally adopted since it results in a greater undistorted signal to noise ratio than either of the above techniques. Unfortunately no detailed account of the theory has appeared which is completely satisfying. Toomin and Wildfeuer[4] have made studies of the behavior of minor hysteresis loops superimposed on major loops and proposes an explanation based on their findings for linear transfer characteristics in recording. Holmes and Clark[5] point out that certainly the audio fields and in many cases the bias fields have not been established for the number of cycles required to produce stabilized major and minor loops. They propose an explanation in which the state of magnetization of the medium is traced as the material passes across the gap and find it to have an induction dependent to a large extent of the maximum field encountered during the transit of the gap.

This explanation is the most satisfying published. One apparent objection is that the same remanent flux is found in many successive particles under certain conditions and remanence is therefore not a smoothly varying function of position along the tape.* It will be shown that because of the profound alteration of the flux distribution during demagnetization many such irregularities in the initial flux distribution will be smoothed out.

[4]H. Toomin and D. Wildfeuer, "The Mechanism of Supersonic Frequencies as Applied to Magnetic Recording," *Proc. I. R. E.*, p. 664, November 1944.

[5]L. C. Holmes and D. L. Clark, "Supersonic Bias for Magnetic Recording," *Electronics*, July 1945.

To appreciate the complexity of the history of a magnetic particle as it passes over the recording head suppose we refer to *Fig. 4*. Sketch "*a*" represents a recording head with its gap and the associated field fringing out from the gap. It is this fringing field which is effective in producing the magnetization of the tape as the tape passes, say, from left to right across the top of and in contact with the head. The exciting coil carries a current which is the sum of an audio and an ultrasonic component. At any instant the field on some one plane over the gap is shown in *Fig. 4b* broken into horizontal and vertical components. For short distances above the head the value of the field varies approximately as the inverse first power of the distance of this reference plane from the head. Superimposing the time variations of the field, the space variations of the field, the two components of the field to be considered and the complex behavior of a ferromagnetic material in two dimensions, we begin to appreciate the difficulty of tracing the particle's magnetic history.

In order to make progress in the explanation let us assume that as a particle traverses the gap, some process leaves a horizontal component or remanence which is proportional to the instantaneous value of the horizontal component of the field at some point in the gap. If this is true, we are able to explain the action of recording with a-c bias. It will be shown, in fact, that this hypothesis is too detailed and that it is not necessary to have a remanent flux exactly proportional to the instantaneous value of the field in order to obtain linear output.

*For those interested in the details of this argument see *Fig. 4*, Reference *5*. All particles entering the gap between *t'* and *P* should leave the gap with very nearly the same horizontal component of remanent flux. This is true since the quarter cycle of the field following *S* will only extend the portion of the minor loop *R*, *S*. For all values of the quarter cycle above zero the remanence should be the same. For the remainder of the quarter cycle the remanence will be only slightly less.

Before starting the explanation let us show how we obtain a curve which Holmes and Clark call the transfer characteristic. If the loop tracer is called into use, we may examine the major hysteresis loops for a given tape for certain maximum values of the 60-cycle field. If a family of curves is observed for values of H_{max} between zero and saturation, we have information which allows us to plot the B, H curves shown in *Fig. 5*. From these curves, of which only two are shown, we read the remanent induction in the tape to be expected when the field goes to zero from each value of H_{max}. This gives data from which to plot the transfer characteristic curve shown as Curve 1 in *Fig. 6*.

Line aa', *Fig. 5*, represents a calculated demagnetization function for a certain audio wave length on a tape. Its intersection with the hysteresis loops defines the maximum value of induction which a tape will support at a given wavelength for given impressed fields H_{max}. The value of B_r at the intersections may be plotted against H_{max} to obtain Curve 2. The audio signals applied to a tape may be expected to transfer according to Curve 1 for very long wavelengths and according to curves of Type 2 for shorter wavelengths. While this is correct for d-c bias, under the influence of a-c bias this is not true since the demagnetizing forces involved are a combination of those for the a-c bias wavelength and the audio wavelength. The actual final transfer will be shown to deviate markedly from such curves.

Let us make the second assumption required for the explanation and suppose that for long wavelengths the transfer characteristic curve before demagnetization is Curve 1 of *Fig. 6*. Since Curve 1 was established on the basis of stabilized hysteresis loops, this is not strictly correct. The true transfer curve must be one similar in shape but with somewhat lower B_r values.

Suppose we borrow the type of illustration used in obtaining input-output curves for radio tubes. Let us plot the input ultrasonic field against time on the vertical axis and in the usual fashion trace the output induction as a function of time on the horizontal axis. This approach is illustrated in *Fig. 7*. So long as the tape is in contact with the recording head, the dashed output curve represents the horizontal component of induction. Upon removal from the head, poles will form on the tape and the associated demagnetizing forces will reduce the induction to a value indicated by the solid output curve. The remanent induction on the tape will be that which would have resulted had we used a transfer characteristic derived from the intersection of some line *b-b′* with the hysteresis loops of *Fig. 5*. The slope of *b-b′* is presumed to be calculated from the demagnetization constant for the bias wavelength. With no audio signal on the recording head and the bias field adjusted to reach the neighborhood of the limit of the linear portion of the transfer characteristic, it is seen that only a very low amplitude induction of ultrasonic frequency is impressed on the tape.

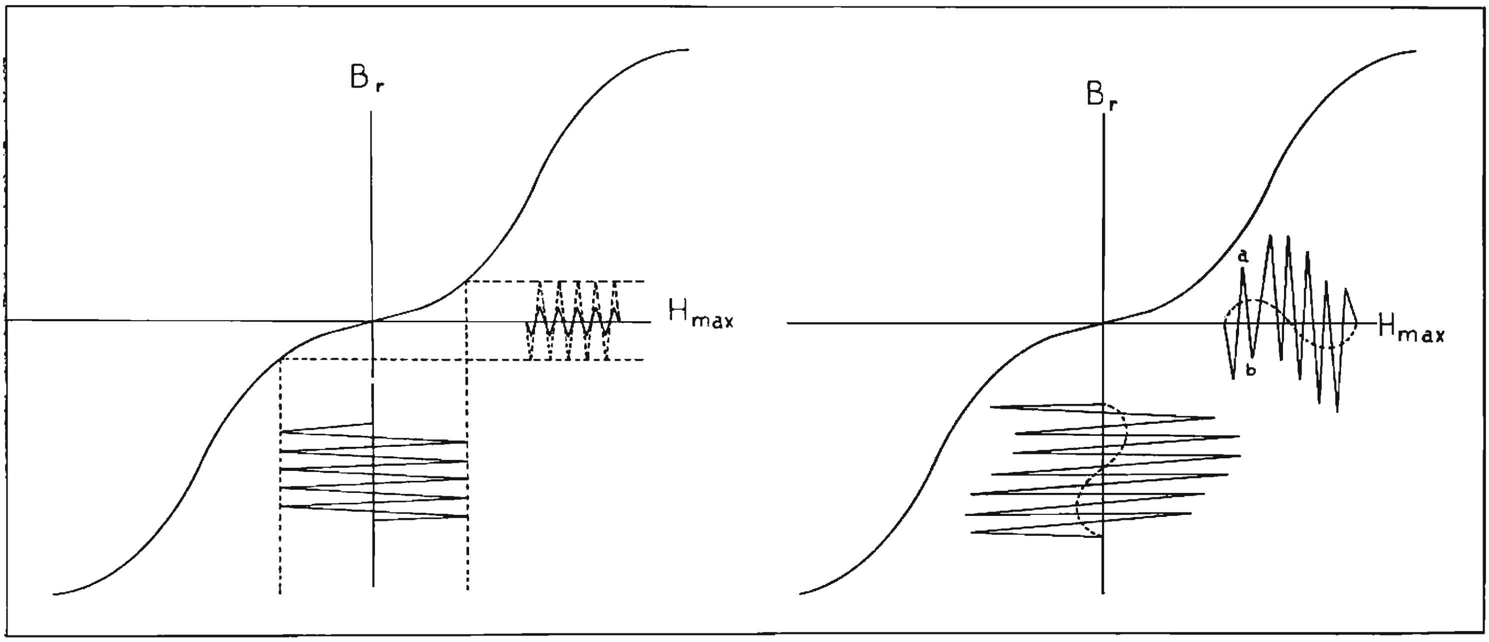
In recording with the use of a-c bias, the bias current and audio current are added in the coil of the recording head. This results in the applications of fields to the tape which are proportional to the sum of the instantaneous values of the bias and audio currents. The wave form of such an "input" field is illustrated in *Fig. 8*. (It is noted that the bias field is not modulated by the audio-frequency field as a carrier wave is modulated in radio transmission.) We assume that, before removal from the recording head, the horizontal component of induction in the tape is proportional to the instantaneous value of the field at some point in the gas and that the transfer characteristic is a curve of Type 1, *Fig. 6*. We proceed in the conventional manner to trace the output induction shown as the solid curve on the H axis of *Fig. 8*. This solid curve shows the condition of induction to be expected in a tape which has passed the gas but has not been subjected to demagnatizing forces by removal from the head.

The question of what happens upon moving the magnetized tape past the head may be answered qualitatively by considering the two sets of demagnetizing forces introduced. One set of forces is attributable to the poles associated with the bias frequency distribution of flux and the second set is connected with the average of the bias wavelength poles and may be attributed to the audio wavelength.

As a starting point in the explanation suppose we consider one bias frequency oscillation of the recorded induction *a*, *b* (*Fig. 8*) for a very long wavelength of audio signal. The oscillation *a*, *b*, which occurs at the point of maximum induction for the audio output and therefore at the point of zero demagnetizing force for the long wavelength, may be considered as the sum of a d-c component and a sinusoidal component of induction. The only demagnetizing force to be considered is that of the poles associated with the sinusoidal component. The equilibrium which will be established between induction and poles for the sinusoidal component will be that expected from the transfer characteristic curve constructed for the bias frequency. This requires that the over-all induction change to the picture of a very small oscillation about the zero value of the induction for each of the original sinusoidal components. The final picture of the output induction is shown as a dashed line drawn through the zeros of each bias frequency oscillation in the output curve of *Fig. 8* which, for simplicity, does not show the small superimposed oscillations at bias frequency. It is seen that the effect of the a-c bias superimposed on the audio signal is to impress on the tape a reproduction of the audio signal in terms of induction. The effect of the bias is to remove the first non-linear portion of the transfer characteristic curve.

Although the small bias frequency oscillations superimposed on the audio are unimportant practically, since they are subject to further smoothing action in the reproduction system and at any rate would never be heard, their existence can be used to substantiate in part the assumptions on which we have proceeded. If the output of a tape on which a low audio frequency has been recorded is examined in an oscilloscope, the bias frequency oscillations may be seen. Since the output of a pickup is the derivative of the induction in the tape, the bias oscillations of greatest amplitude are found near the zeros of the output wave and disappear near the peaks. This is exactly the position in which modulation noise is expected to reach a maximum value, so the picture in the scope is sometimes attributed to that form of noise. The sweep frequency is adjusted to the audio value and the irregular position of the bias frequency pips does resemble the picture of random noise. A second experimental check may be had by recording a low-frequency tone on a tape running at high speed. If this tape is played back at a considerably lower speed and the output examined on an audio wave analyzer, the bias frequency appears in the output at the expected

Fig. 7 (left). Showing the use of the transfer characteristic in obtaining the output induction for the bias field. Fig. 8 (right). Showing the combined audio and bias input fields and how they are transferred to induction in the tape.



new frequency with an amplitude which is larger the greater the value of the recorded audio amplitude. This increase in amplitude is what would be expected if the demagnetization of the bias component of induction takes place according to the transfer characteristic curve for the bias frequency.

We have seen that at low frequencies where demagnetizing forces of the audio pole distribution can be neglected the transfer characteristic curve does not predict the correct output directly. A transfer characteristic having something less than half the B_r values of Curve 1, *Fig. 6*, must be used if the conventional transfer procedure is employed where the peaks of the input field are used to trace the output induction.

For short audio wavelengths we must take into account the demagnetizing force due to the pole distribution associated with the audio wavelength. This results in a further reduction in the amplitude of the remanent flux. Two general procedures may be followed in obtaining the final answer. We may either calculate the transfer on the basis of Curve 1, *Fig. 6* and apply successively the demagnetization calculations for the bias, then the audio wavelength, or we may start with a transfer characteristic Curve 2, *Fig. 6*, which contains the demagnetization effect for the audio wavelength and calculate the induction from it in the same manner as was used for long wavelengths. These two procedures lead to essentially the same results for the audio frequency induction. The first method results in a bias ripple amplitude which is independent of the audio wavelength. The second method would lead to a ripply amplitude which decreases in audio wavelength. An experimental test of the amplitude of the bias frequency output as a function of the audio frequency recorded tends to support the first method of correcting for the two demagnetizing forces. Theoretically the first method of correction is the more satisfactory.

We conclude that, while the simple picture of demagnetization outlined in Part I applies directly to recording with d-c bias, the picture in the case of a-c bias, used either with d-c or a-c wipe, is complicated by the introduction of the bias demagnetization forces. The fact that strong demagnetizing forces are associated with the bias frequency, and that the remanent induction undergoes profound alteration under their influence, eases the requirements of our first assumption. It is seen to be unnecessary that the mechanism assumed in the gap produces a smoothly varying remanent induction before demagnetization. What is required is that the mean value of the induction over one cycle of the bias frequency be proportional to the value of the audio field.

Reproduction

As we have seen, a tape after leaving the recording head carries with it a value of remanent induction which from point to point is proportional to the audio signal impressed on it. After leaving the head, flux lines threading the tape establish return paths through the air which are maintained until the tape is brought into the neighborhood of other permeable material or the tape flux is altered by erasure.

In *Fig. 9a*, a tape is shown on which a sine wave has been recorded. Lines of induction in the tape and return flux through the air are shown schematically. No attempt has been made to illustrate the small flux circuits which represent the recorded bias frequency since these are not important in reproduction theory. *Fig. 9b* shows the redistribution of flux which occurs for one position of the playback head in contact with the tape. Flux which normally found a return path through the air is redistributed through the low permeability core of the reproducing head and threads the pickup coil. A voltage will be developed in the pickup coil which is proportional to the time rate of change of the flux threading the coil. For wavelengths long compared with the gap width, the amount of flux by-passed through the coil is essentially equal to the induction crossing a plane, such as *a* or *b*, *Fig. 9a*, cutting the tape at the center of the gap. As the tape moves across the head, it is easily seen that the output voltage developed is proportional to the rate of change of induction in the tape under the gap.

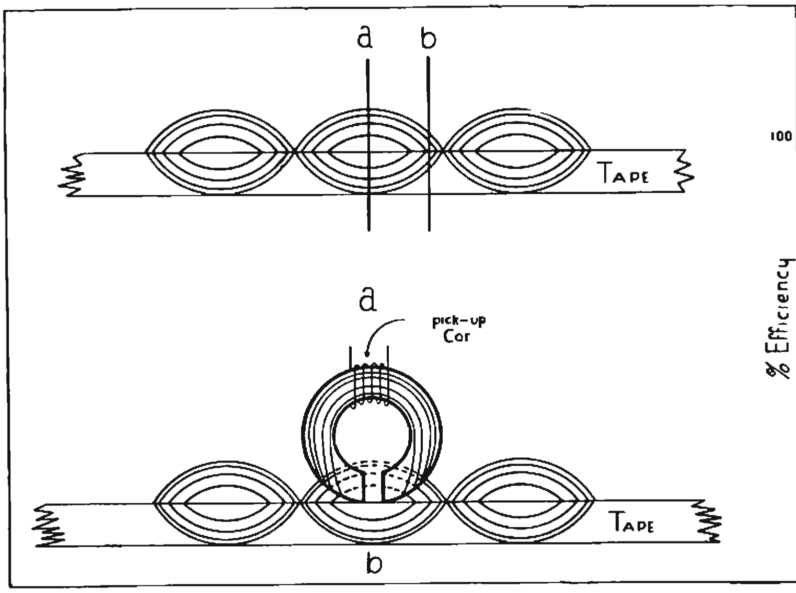
An interesting effect, which results in periodic reduction of the reproducing head output, is seen if one examines the flux distribution as the wavelength becomes increasingly shorter with respect to the gap width.

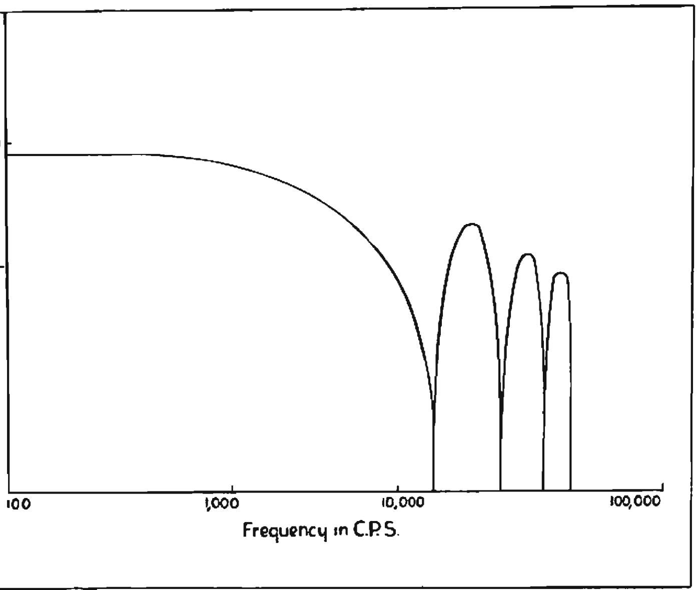
It may be shown that with decreasing wavelengths some flux lines find the air path to be that of least reluctance. These lines do not thread the coil and represent a voltage loss in the pickup. This loss increases up to the point where the wavelength equals the gap width where the output of the head falls to zero. The fall to zero output occurs at each frequency for which the gap width is an integral number of wavelengths, with partial recovery of pickup efficiency for intermediate wavelengths. A qualitative plot of the efficiency of the reproduce head as a function of frequency is shown in *Fig. 10*. The decrease in efficiency for successive peaks of the curve is due to the increase in the ratio of the core reluctance to that of the decreasing air paths' lengths, as the wavelength shortens.

This short discussion of the effects of gap width suggests that a decrease in gap opening tends to increase the high-frequency output. This is true up to the

[Continued on page 37]

Fig. 9 (left). Illustrating flux lines in the tape and air associated with a recording. The effect of placing a high permeability head on the tape is to rearrange the air flux so that it threads the pick-up coil. Fig. 10 (right). Showing the efficiency of a reproducing head as a function of frequency.





Magnetic Recording

[*from page 16*]

point where the gap becomes so narrow that efficiency again suffers from flux leakage across the gap. Obviously zero gap opening would make a very inefficient reproducing head.

Conclusion

Over wide limits, the gap width of the recording head has little effect on the recording. It should be sufficiently wide so that strong fields have the opportunity to penetrate the tape yet not so wide that excessive currents are required for excitation. The frequency response of the playback system is definitely affected by the gap width of the reproducing gead. The demagnetizing forces developed in a recording attenuate the induction at high frequencies.

In Part III of this article, we shall combine these effects and consider their influence on the over-all response. Attention will then be given equalization, noise, distortion, signal levels and the effects of velocity of the tape drive.

Review of the Present Status of Magnetic Recording Theory

W. W. WETZEL*

PART III

In this series of three articles, Dr. Wetzel presents the first complete discussion of magnetic tape recording theory for engineers.

IN THE two foregoing parts we have examined some of the properties of magnetic materials and the forces to which the materials are subjected during recording. In Part III we shall summarize the effects and illustrate the results with data taken from actual measurements. We shall then examine noise and distortion phenomena from the experimental point of view since theories of the cause of noise and distortion are far from complete.

Summary

In order to compare the over-all response of a tape as a function of frequency it appears fair to examine the outputs at different frequencies on the basis of some form of constant input. The basis usually selected is that the maximum exciting field in the recording gap be made the same for each frequency, i. e., the total maximum flux be made equal before demagnetization. This is done by keeping the exciting current in the playing head the same for each recorded frequency. The output curves so obtained are known as constant current frequency-response curves and have been generally adopted as demonstrating one characteristic of a magnetic medium.

Forgetting for the moment demagnetization and the gap effect of the reproducing head, let us see what might be expected of constant current recording. If the field of the gap induced the same maximum remanent flux Ø regardless of frequency, we should have as an expression of the flux:

$$\text{Ø} = ai \sin \omega t$$

On playback where the derivative of the induction is proportional to the response, we have for the output voltage "V" of the reproducing head

$$V = b\frac{d\text{Ø}}{dt} = abi\,\omega \cos \omega t$$

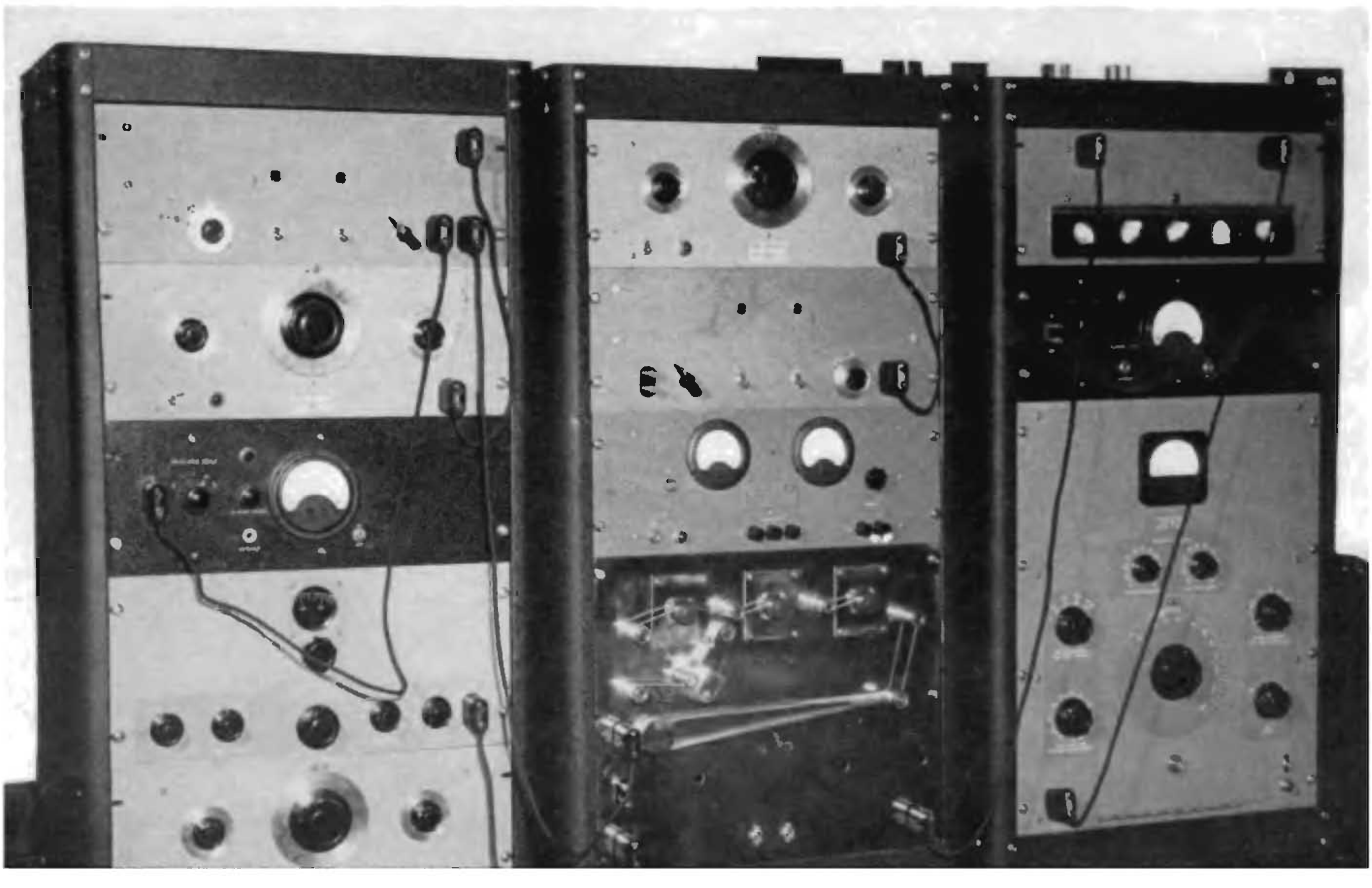
Since the maximum output voltage "V" is directly proportional to $\omega = 2\pi f$, we see that the output may be expected to vary linearly with the frequency. This represents an output which increases 6 db per octave and is illustrated by the straight *Curve 1, Fig. 1.*

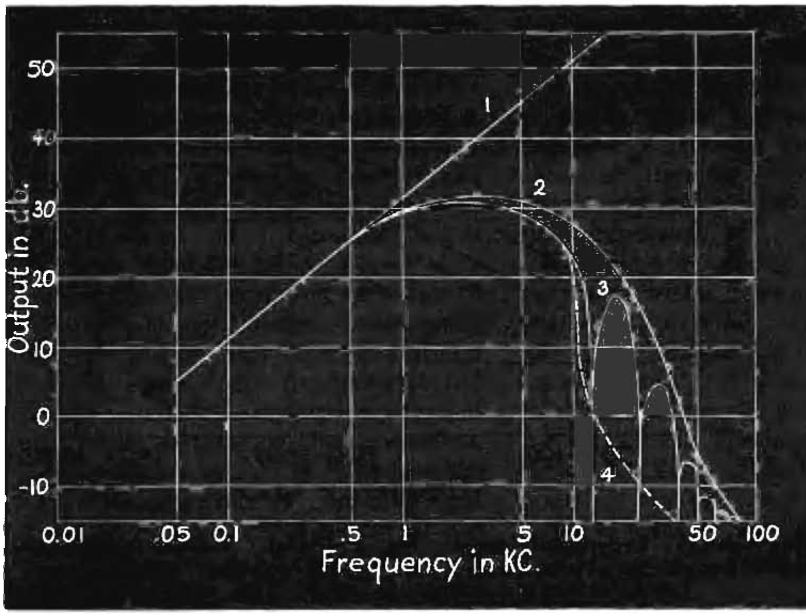
If to the above considerations we add the effects of demagnetization on the induction remaining in the tape, we obtain *Curve 2*, which indicates that a large drop in remanent induction accompanies increased frequency.

Superimposing on the above effect the gap effect of the reproducing head, we

**Minnesota Mining and Manufacturing Co., 900 Fauquier Ave., St. Paul, Minn.*

Fig. 2. Equipment assembly for testing loops of magnetic recording tape.





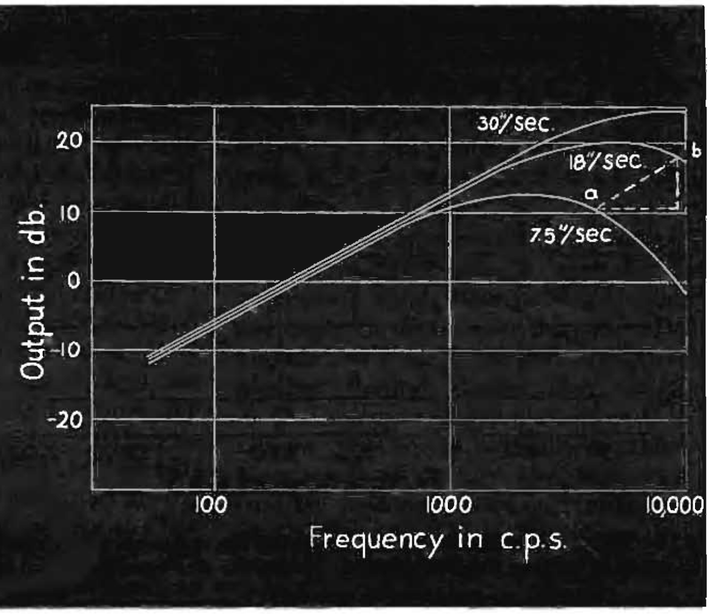


Fig. 1 (left). Showing Curve 1, the 6 db/octave increase inherent in constant current recording; Curve 2, the effect on Curve 1 of the demagnetizing forces; and Curve 3, the addition of the characteristics of the reproducing head. Curve 4 is added to illustrate the desirability of an amplifier cut-off at high frequencies. Fig. 3 (right). Showing the effect on the frequency response characteristic obtained by changing the speed of the tape drive. American tape.

obtain *Curve 3* which shows the over-all frequency response of the magnetic recording system dissociated from amplifier characteristics.[1]

The chief contributors to *Curve 3* are the remanent flux pattern on the tape and the geometry and permeability of the reproducing head. For purposes of illustration it has been tacitly assumed that the tape and its flux pattern were driven at some constant velocity across the reproducing head. This permits the plotting of the output against frequency "*f*" although the wavelength "*l*" is the basic constant upon which the flux pattern depends.[2]

We are now in a position to predict the effect on the frequency response curve of using the same recording and reproducing system at a different velocity of tape drive. For constant current recording the flux on the tape will be identical, wavelength for wavelength, independent of the velocity.

Because the output of the reproducing head is proportional to the rate of change flux, if the velocity is halved the output is reduced 6 db. If the velocity is halved, any given wavelength corresponds to half the frequency. The effect of a velocity change would be that of moving *Curves 2* and *3* along *Curve 1* until any reference point on the curves corresponding to a given wavelength for the initial velocity reaches the frequency corresponding to the new velocity.

The effect of changing the gap width of the reproducing head is more complex and will not be discussed here beyond the mention that the effective gap is not equal to the physical gap width.

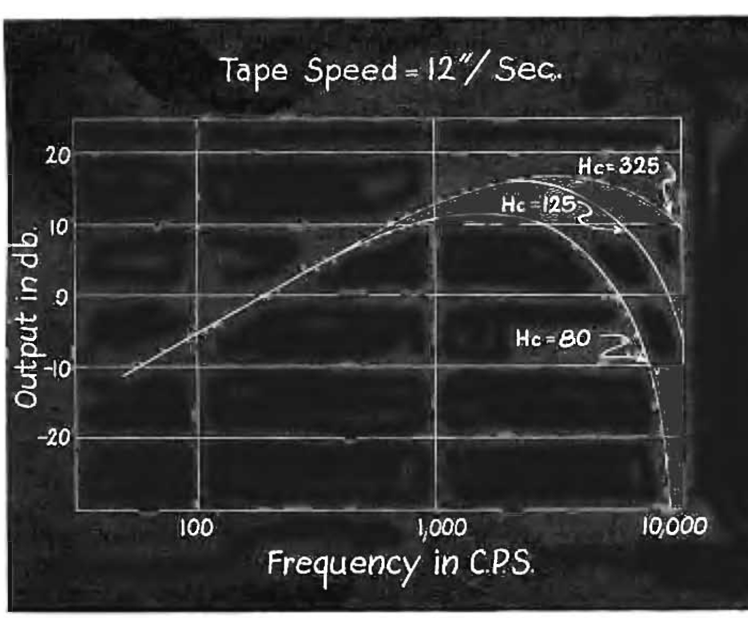
The Loop Tester

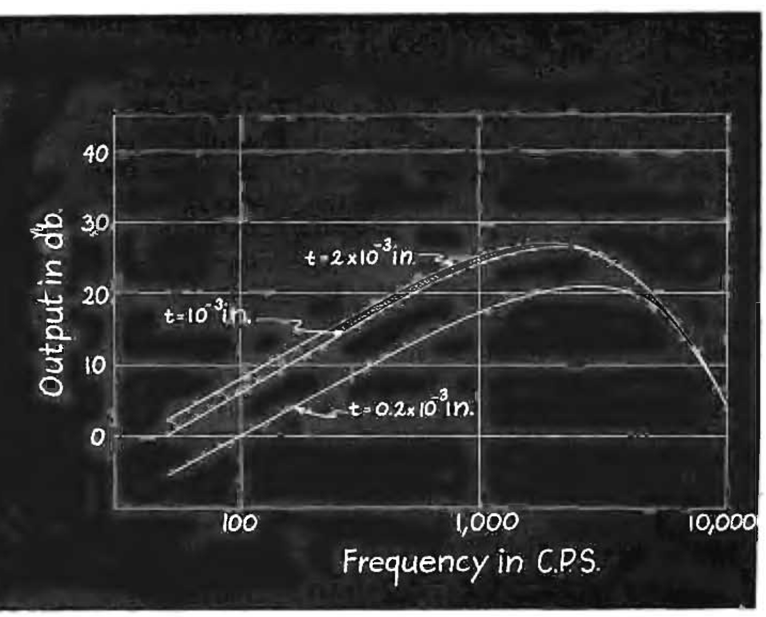
Because it offers the convenience of being able to study a small sample of wire or tape for any given length of time, most laboratories engaged in the study of recording materials employ loop testers. On this device a loop of tape or wire is driven continuously and repeatedly over erase, record and playback heads. An example of a loop tester with the associated instruments is shown in *Fig. 2*. Separate variable oscillators and amplifiers are provided for the bias

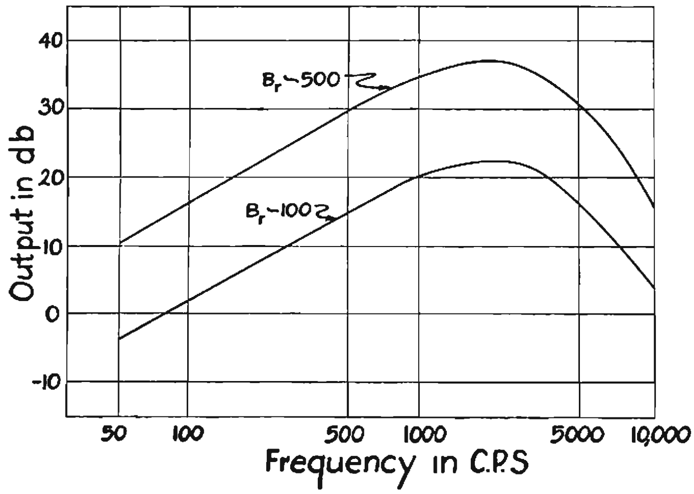
[1] It is advantageous from two considerations that the amplifiers cut off as abruptly as possible at frequencies above those which we intend to record. This sharp upper cut-off is illustrated by the dashed *Curve 4, Fig. 1*. This cut-off has these beneficial effects: *a*) it reduces the background noise to the extent of reducing the contributions in the upper frequency range, and *b*) it represses harmonic distortion to the extent that harmonics above cut-off frequency will be suppressed.

[2] The equation for the velocity "*v*" is obviously $v = fl$.

Fig. 4 (left). The frequency response is shown to vary with the coercive force of the coating material in a manner similar to that obtained by a velocity change. Experimental tapes. Fig. 5 (right). Illustrating the law of diminishing returns applied to coating thickness on a tape. Experimental tapes.







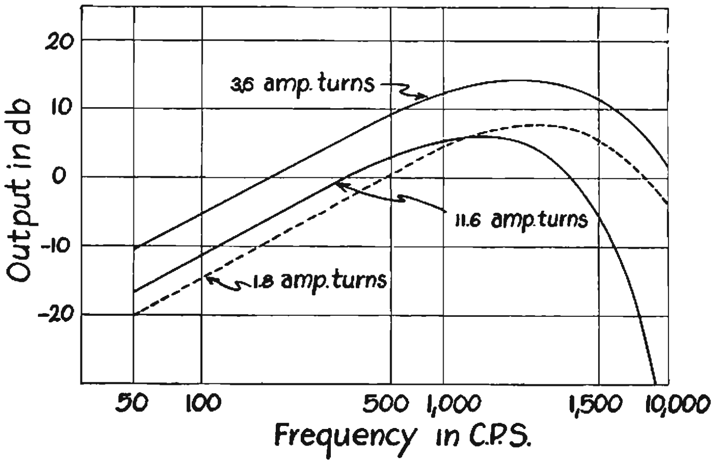


Fig. 6 (left). Showing the effect of varying the remanence of a coating material. Experimental tapes. Fig. 7 (right). Showing the influence on output of varying the bias current. Note the decrease in output at high frequencies obtained from an excessively high bias. American tape.

and erase supplies. The audio signal or signals are obtained from audio frequency oscillators which have sufficient output to supply the recording current without additional amplification.

The output of the playback head is amplified and examined in an oscilloscope, on a Ballantine voltmeter or the wave analyzer. A low pass filter cutting off at 16 kc is incorporated in the output amplifier to eliminate the pick-up of bias and wipe frequencies. Similarly, high pass filters suppress the power line frequencies in the bias and wipe amplifiers.

This combination of instruments permits one to make many of the determinations required to evaluate tapes. The graphs illustrating the remaining portion of this article were assembled from measurements taken on this equipment. The plots are generalized and occasionally represent smoothed readings. The omission of reference voltages and specifications of the heads is intended to indicate the generalization.

Miscellaneous Observations

In the discussion of the general theory we have made certain predictions concerning the effects of variables on the output characteristics. We shall illustrate a number of these effects by considering the results obtained on the loop tester.

The effect of increasing the velocity of the tape drive was shown to be the shifting of each point on the frequency response curves along a 6 db octave line. The curves shown in *Fig. 3* give the results obtained on an American tape by varying the speed holding all other factors constant. The triangle illustrates the construction proposed by Holmes[3] for calculating the response for any velocity if it is known for one velocity. Point *a* on the 7.5"/sec. curve transferred to point *b* on the 18"/sec. curve along the hypotenuse of a triangle sloping at 6 db/octave whose base is the $\log \frac{f_2}{f_1}$ or $\log \frac{v_2}{v_1}$.

Fig. 4 shows the frequency response as a function of H_c alone. Since it is difficult to obtain materials of the same remanence but widely different values of the coercive force, the curves were "normalized" by shifting them vertically until the response at the lower frequencies coincided. It will be observed that the increase in velocity has essentially the same effect as an increase in H_c. Other factors being equal, the high coercive force tapes may be used at lower, more economical speeds.

Fig. 5 shows the effect of varying the thickness of a tape and consequently the total flux Ø which would be observed in the hysteresis loop tester. Increasing the flux Ø by a factor of five increases the output of the lower frequencies by only 6 db or a factor of two. There is no change at the high frequency end of the spectrum. This tendency of the curves to approach one another at high frequencies was first mentioned by Kornei[4]. It may be attributed to the lack of penetration of remanent induction into the tape for short wavelengths after demagnetization forces have come into play. At a tape speed of 7.5"/sec. a 7.5 kc tone will record with a wavelength of .001". The spacing along the tape from center to center of the pole pattern is therefore .0005". In a thick tape after demagnetization very little contribution to the remanent flux from depths greater than .0005" would be expected. It is therefore immaterial for short wavelengths after the demagnetization equilibrium has been established whether the original flux distribution penetrated the tape to a depth of one mil or one inch. Similarly, it should be immaterial from the standpoint of output at high frequencies whether a tape be one or two mils thick. There is no reason to assume that bias frequency flux in the recording gap does not penetrate as deeply into the coating as the audio frequencies. The penetration effect observed is the result of the geometry of the flux after demagnetization.

Fig. 6 illustrates results obtained from conditions which were identical in all respects except that the value of the remanence differed by a factor of five. The predicted increase of 14 db at low frequencies is found. In this case, the high frequency determinations are questionable. Although the curves approach one another at 10 kc, because of the inaccuracy of the measurement it cannot be concluded that they prove the prediction that a change in remanence has a smaller effect on the output at high frequencies. What may certainly be concluded is that *Figs. 5* and *6* show B_r rather than Ø to be more nearly the determining factor in the output of tape.

Fig. 7 shows the effect on frequency response of bias current variations. As will be seen later the output curve as a function of bias current develops a sharper peak at high than at low frequencies. As a result a high value of bias current causes a drooping characteristic at the higher frequencies. The bias of 11.6 ampere turns was chosen deliberately to be very high to illustrate the point.

Noise

One of the more interesting studies made on a loop tester is that of the spectral distribution of noise. Since some aspects of the distribution serve to illustrate the behavior of modulation or under-signal noise, it may be well to consider such curves. *Fig. 8* shows data taken with a Brush KB919 reproducing head on a good sample of German Type C tape driven at a speed of 21"/sec. Curves *a* and *c* have been smoothed off peaks attributed to power frequencies in the output amplifier and to pick-up from the

[3] Lynn C. Holmes, "Some Factors Influencing the Choice of a Medium for Magnetic Recording." *J. Acoustical Soc.*, *19*, 395, 1947.

[4] Otto Kornei, "Frequency Response of Magnetic Recording." *Electronics 20*, 124-28, August 1947.

motor drive. All measurements were made on the 30-cycle half band width setting of a Hewlett Packard Harmonic Wave Analyzer. The noise distribution after an a-c erase is shown by Curve *a*. Curve *b* shows the distribution after a saturating d-c wipe. The effect of the d-c wipe has been that of increasing the noise level at all frequencies. Curve "*c*" shows the distribution of noise after applying an a-c wipe and recording a moderately strong 400-cycle note on the tape. It will be seen that, in addition to the presence of the fundamental and the third and fifth harmonics, a broad distribution of noise has been recorded. For frequencies well above 400 cycles the effect of recording has been the equivalent of wiping with a d-c field somewhat less than that required for saturation. At frequencies below 400 cycles the noise drops off to approach the a-c erased level.

Two rather striking features of the modulation noise are: the small peak located at 800 cycles or the second harmonic point, and the very substantial noise contributions in the neighborhood of the 400 cycle peak. The curve has not been corrected for the pass band of the analyzer, but it may be stated that the contributions near the peak greatly exceed the filter correction. As Holmes[1] has shown, the amplitude of the under-signal noise as a function of a recorded d-c field increases with the field up to a saturation value. Any portion of the 400 cycle wave may be considered to be the application of an instantaneous d-c field accompanied by a corresponding value of the under-signal noise. The greatest value of the noise will occur at the peaks of the recorded induction, falling to zero as the induction goes to zero. The modulation envelope for this noise distribution has therefore twice the frequency of the recorded note, and the small peak at 800 cycles is believed to represent this modulation frequency.

Chapin[5] has offered an explanation of the noise in the neighborhood of the parent frequency. Unfortunately, the abstract of the paper does not present his theory completely, however, the author understands it to be based on sum and difference frequencies between noise at lower frequencies and the parent 400 cycle frequency.

The noise developed during recording has come to be known as under-signal or modulation noise. The scale of *Fig. 8* was chosen to illustrate the noise, and the reader should note that the peak of the fundamental lies about 50 db above the d-c wipe level. The maximum 700 cycle signal at 2.5% distortion to over-all noise for the tape was 68 db while the signal to total d-c wipe noise was 46 db.

When a new tape is examined, the noise level is found to be quite low. It is frequently stated that wiping such a tape increases the noise background. While this may actually be observed, it usually may be traced to one or more of three difficulties. The erase fields are not sufficiently strong to saturate the tape or the erase head design does not allow for the many decreasing alternations in the field required for a good wipe. The wipe may have a d-c component which will result in recording the d.c. as "idealized" magnetization. This will result in under-signal or modulation noise. The d-c component of the wipe may be caused by a non-symmetrical wave form in the oscillator, the unbalanced plate current of a push-pull amplifier coupled directly to the head or by permanent magnetization or the erase head core. If these wipe difficulties are avoided, it is possible to reduce the noise background to a point lower than that of new or virgin tape.

[5] D. M. Chapin, "Measurement and Calculation of Under-Signal Noise in Magnetic Recording." Program 33rd Meeting Acoustical Soc. Am.

Similarly, noise contributions may arise as the result of passing the tape over the recording head. Permanent magnetization of the recording head, a d-c component in the bias current or asymmetry of the wave form give rise to modulation noise through recording of the d-c component.

Permanent magnetization of the playback head gives rise to noise by two processes. First, after passing the head, the tape will have been magnetized and modulation noise developed for the next playing. Second, during play the flux in the magnetized head varies with the reluctance of the gap. Thus, in addition to magnetization, any variations in the coating material cause a variation in gap reluctance which will show up as noise.

There are conflicting opinions regarding the possibility of generating noise on a neutral tape by passing it over a recording head which carries bias but not audio excitation. Some observers find the noise level to have increased under these conditions. Equally careful observers find that by removing any trace of a d-c field component from the recording head the noise level remains unchanged over that of the erased tape.

The noise, if present and not due to a d-c component of the recording field, must be attributed to modulation noise caused by the bias frequency. While the data of *Fig. 8* may be considered to be reliable for region of frequencies above 200 cycles, the readings below are not sufficiently accurate to draw positive conclusions. The modulation noise curve is shown approaching the a-c wipe curve at low frequencies. The fact is that it is certainly well below the d-c wipe curve, but we cannot be certain it falls off as rapidly as shown. Unfortunately, this prevents our drawing conclusions from

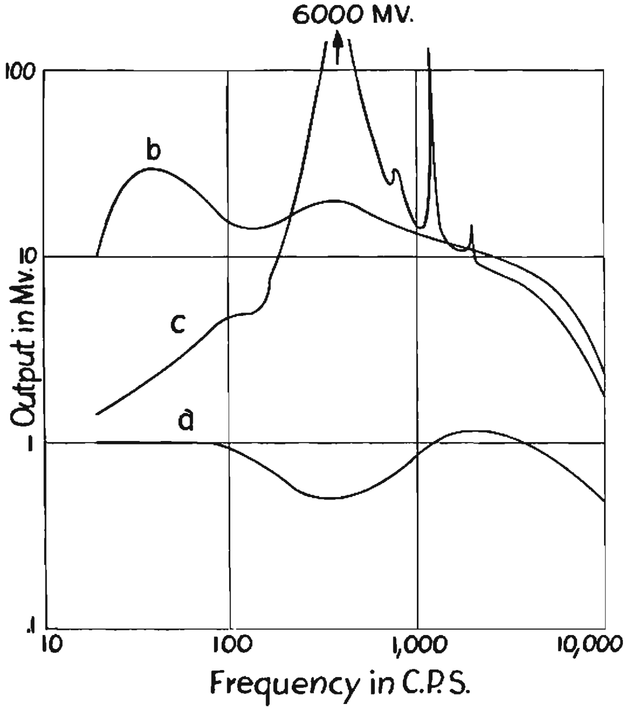
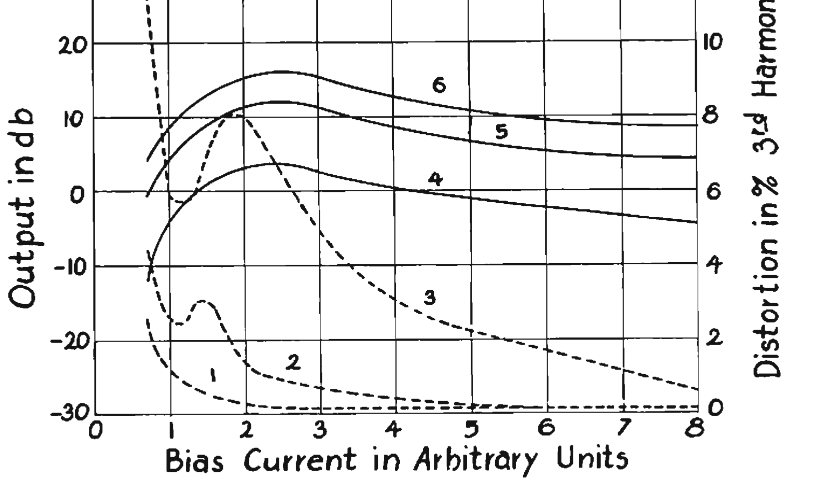


Fig. 8 (left). Noise analysis on a German tape showing the distribution for a) an a-c erase, b) d-c wipe, and c) the noise developed upon recording a 400 cycle signal. Note that the values below 200 cycles on Curve "c" are not dependable in detail but indicate the trend.

Fig. 9 (below). The third harmonic distortion and the output at 1000 cycles, 9.2" sec is shown to vary with the bias current. Distortion at 1) 5 db, 2) 10 db and 3) 20 db input levels. Output at 4) 5 db, 5) 10 db and 6) 20 db input levels. American tape.



this curve as to the possibility of a supersonic bias generating lower frequency audible noise.

Distortion

Intermodulation distortion measurements have been made on recording media by at least three laboratories, but for one reason or another this method of evaluation has not been generally adopted. When distortion is considered in tapes or wires, it is usually harmonic or amplitude distortion which is meant.

If a magnetic medium is saturated by a d-c wipe and the recording is made with either a-c or d-c bias, the operation is performed on an asymmetric transfer characteristic. This permits both even and odd harmonics to develop. In addition to the lower noise levels generated, the use of a-c bias on a tape erased to a neutral condition provides for operation on a symmetrical transfer characteristic curve which eliminates the even harmonics. Thus we see in *Fig. 8* with the exception of a small contribution to the second harmonic attributed to the modulation frequency of the under-signal noise, the harmonics observed are the third and fifth.

It is interesting to note at this point that one form of distortion which may be recorded is the beat frequency of this fifth harmonic with the bias frequency. If a 30 kc bias is used and we record say a 6,050-cycle audible tone, a 250-cycle beat note appears as distortion. This gives rise to the very practical rule that the bias frequency be at least five times that of the highest frequency one expects to record.

Returning to the subject of harmonic distortion we find two methods in general use for its evaluation. The first employs the conventional 400-cycle distortion meter which has a flat rejection filter on the band from 350 to 400 cycles. A 400-cycle note is recorded and the output through the filter measured as total distortion. Because of the peculiarities of modulation noise, this practice may be questioned, since from *Fig. 8* we may deduce that appreciable contributions to the modulation occurring near 400 cycles but outside the rejection band of the noise meter, will be counted as harmonic distortion. To this will be added the modulation noise contribution generated at all higher frequencies. The second method, which has tentatively been adopted in this laboratory, is the measurement of the third harmonic component on a wave analyzer. The fifth harmonic is usually negligibly small.

Fig. 9 shows the per cent third harmonic distortion as a function of a-c bias for recording on a demagnetized American tape. Output curves are also plotted in accordance with the practice introduced by Holmes[1]. It will be seen that maximum output is found in a region of bias for which distortion is high and lower distortion must be obtained by sacrifice of output. At first glance the regions of low distortion at low bias values may appear attractive. For two reasons they should not be used; first, they represent sharp minima which occur at different bias values for different recording levels, and second, they correspond to very low output levels. These minima correspond to recording on the toe developed in a transfer characteristic for the underbiased condition. It is the situation which causes disappointing results when high coercive force American tapes are used on machines whose bias is set for good results on low coercive force German tape. The proper condition for bias is a compromise between distortion and output on the portion of the curve beyond maximum output. Here the bias value is not critical, i. e., small shifts in bias will not cause large changes in distortion.

Recording on a machine designed with a sharp high frequency cut-off develops its distortion only at low frequencies. Suppose we expect to record up to 10 kc and provide a sharp cut-off in the output amplifier at this point. Fifth harmonics of 2000 cycles and third harmonics at 3333 cycles and above are eliminated. At high frequencies distortion may be neglected and output alone considered in choosing the bias.

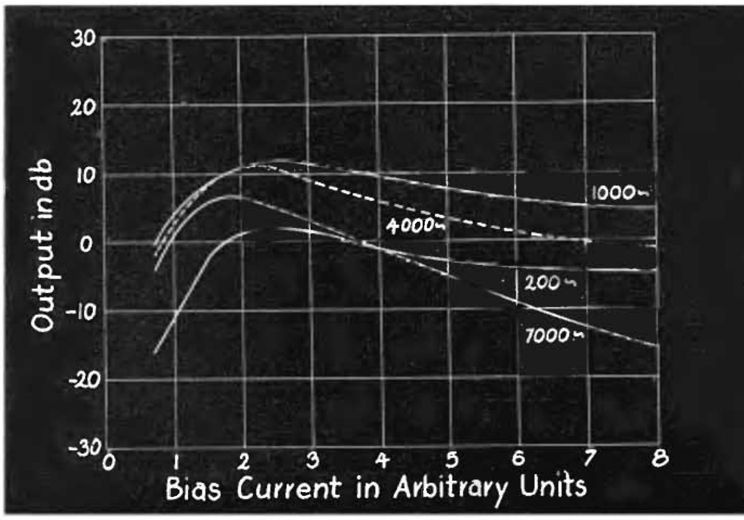
Fig. 10 shows the output of a tape as a function of bias current for a number of frequencies. It will be seen that at the higher frequencies the fall from maximum output is steeper than at low frequencies. These observations will be found to correlate with the curves of *Fig. 7*.

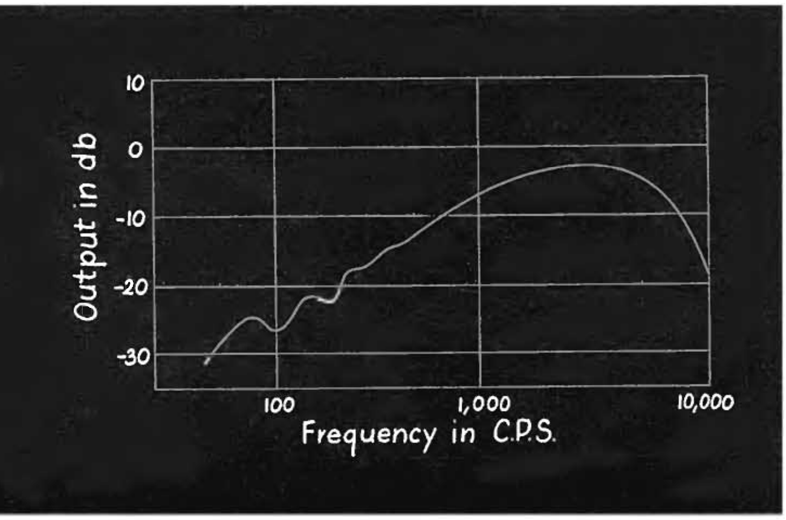
One form of distortion inherent in wire recording does not occur in tape. This is the effect of rotation of a wire on high frequency response. In the "U" shaped recording gap through which wire travels, only the portion of the wire in contact with the head is magnetized at high frequencies. This corresponds to the penetration effect in tape (*Fig. 5*) as a function of wavelength. If on playback the wire is rotated 180°, the weakly magnetized portion contacts the reproducing head resulting in serious deterioration of high frequencies. If wire is used carefully, unless a break occurs and splicing is required, there are only accidental forces which tend to rotate the wire. There is some probability that upon repeated playback the orientation remains essentially the same.

Fig. 11 shows a second effect in wire recording which causes non-uniform increase in response with increasing frequency at the lower end of the spectrum. The non-linearity of the frequency response at low frequencies is attributed to poles arriving at and leaving the reproduce head simultaneously. Some flux from these poles which is by-passed through the pick-up coil may aid or oppose the flux pick-up in the gap. That the effect should occur in wire and not tape is attributed to the surface area of wire being small compared with tape. This allows for precision contact of wire entering and leaving the head. The

[*Continued on page 46*]

Fig. 10 (left). An American tape used to illustrate the effect on output caused by bias variations at different frequencies at the same level (10db) of recording. Fig. 11 (right). Showing the irregularities in the response curve of a wire recording. These variations from linearity are not observed for tape.





Magnetic Recording

[*from page 30*]

greater width of the tape allows an uneven approach and departure contact which averages out the flux return. If this explanation is correct, the same irregularities should be observed in tape if a very narrow recording track is used. This experiment has not been reported.

Printing or signal transfer from a recorded turn to an adjacent turn of the medium on the storage roll exists in both wire and tape. Because of the separation of the active layers of a coated tape by the magnetically inert backing some reduction, about 6 db, is obtained in signal transfer in tape as it is used over the situation in which active layers of tape come in contact. The much greater reduction which might be expected due to the physical separation of active layers is partially lost because of pole geometry. The relatively long line of poles formed across the tape during recording, which results in greater output than in a 4 mil wire of comparable magnetic properties, also results in somewhat stronger fields available for printing. In either medium the effect falls off exponentially with the level of the original recording. It may be shown that the resulting transfer is not appreciable in wire if the recording level is kept below the overload point. Tape may be expected to exhibit 6 db less in transfer than a wire of comparable magnetic properties.

Conclusion

The author announced his intentions of including in this review a discussion of equalization. The complexity of the subject together with a rapidly approaching dead-line rules out its consideration at this time. For information on equalization, the reader may see two excellent papers partially devoted to the subject.[6,7]

Finally, the author wishes to express his indebtedness for many of the ideas presented in these articles to the friends with whom he has had the opportunity to discuss magnetic recording. These are R. Herr of Minnesota Mining and Manufacturing Company, Lynn C. Holmes of Stromberg-Carlson Company, S. M. Rubens of Engineering Research Associates and R. B. Vaile, Jr. and R. E. Zenner of Armour Research Foundation.

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[6] L. C. Holmes and D. L. Clark, "Supersonic Bias for Magnetic Recording," *Electronics*, p. 126, July 1945.

[7] A. E. Barrett and C. J. F. Tweed, "Some Aspects of Magnetic Recording and Its Application to Broadcasting," *Jour. I.E.E.*, p. 265, March 1938.

Factors Affecting Frequency Response and Distortion in Magnetic Recording

J. S. BOYERS*

Methods of improving fidelity in magnetic recording are discussed.

As ONE physics professor often remarks to his classes, before one can make rabbit soup one must first obtain the rabbit. Such is the case in magnetic recording—before one can intelligently design and use a system one must first obtain certain fundamental information concerning its operation. It is the purpose of this discourse to shed a little light on some factors affecting the frequency response and distortion occurring in magnetic recordings.

It is well known that the speed at which a recording medium is moved past the recording and reproducing heads affects, to a very great extent, the frequency response to be expected. This is true regardless of the method of recording, but it is very easy to demonstrate in the case of magnetic recording.

High Frequency Response

All things being equal, the high-frequency response of a magnetic recording system varies nearly directly as the speed of the medium is varied. This can be readily understood when one considers that a certain minimum wavelength can be reproduced by the playback head. A certain wavelength will be recorded for a given frequency and recording medium speed. Thus it follows that by increasing the speed a shorter wavelength will be recorded which the reproducing head will be able to resolve.

The finite scanning width of the gap used in the recording and reproducing heads has a considerable effect on the high frequency response. Generally, the shorter the scanning gap the greater will be the resolution and consequent high frequency response. Heads for recording on and reproducing from a 0.004 inch wire are usually manufactured with a small piece of 0.001 inch nonmagnetic material inserted in the magnetic structure to provide the gap. However, heads for use with tape may be constructed with practically no physical gap, the magnetic gap being caused by a butt joint in the pole piece structure. The resultant discontinuity causes the effective gap.

It is interesting to note that the gap in the *recording* head is not too important because the high-frequency response is a function of the sharpness of the field at the leaving edge of the head.[1] Heads have been constructed which give a very good high frequency response when the wire is run over them in one direction while running the wire in the other direction resulted in very poor high-frequency response. This phenomenon was due to the field at one side of the gap being very sharp while the other side was very broad. In the case of *reproducing* heads such is not the case because the resolution depends upon the sharpness of the entire gap field. Of course, it is usually the case that the recording head serves also as the reproducing head so it is necessary that this head be very carefully constructed to give a symmetrical and sharp field distribution.

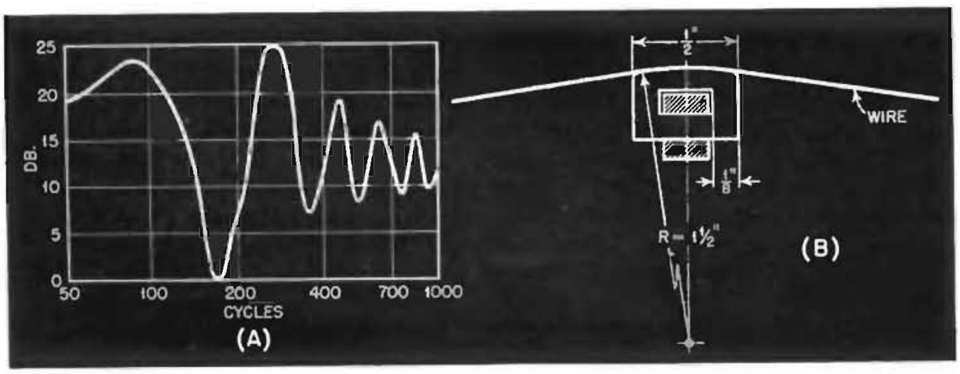

Fig. 1A. Irregularities in frequency response due to residual poles in reproducing head. Fig. 1B. Dimensions of head producing curve shown.

Heads can be constructed which have a very peculiar frequency response. Reference to *Fig. 1* will illustrate this fact. The frequency response curve is for a wire running at four feet per second, and it will be noticed that the first peak occurs at approximately 90 cycles. Consideration of the dimensions of the head will indicate that the first peak should occur at 96 cycles, which is the frequency at which the head structure is one-half wavelength long. These bumps are a result of residual poles which occur at or near the edges of the pole piece used in the head. Various dodges have been used to overcome these irregularities. They usually involve making the head long with respect to the longest wavelength to be reproduced. One head known as the closed type, when properly constructed, gives very smooth response at low frequencies resulting in a curve having a slope of approximately 6 db per octave, increasing with frequency. This head, however, has a very great disadvantage in that the wire must be threaded through the coil which completely surrounds the wire. Similar effects are noticed in tape heads but to a lesser extent than in wire heads.

Magnetic Characteristics

The magnetic characteristics of the medium and their effect on the frequency response have been well discussed in the literature.[2] It has been shown that, from a consideration of the principles of magnetism, the high-frequency response is a function of the coercive force while the output at low and middle frequencies is a function of the residual magnetism. This generally is true, but some data have been obtained which tend to show

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[1] Field Measurements on Magnetic Recording heads, Clark & Merrill, Page 1580. Proc. I. R. E. & Waves & Electrons, Vol. 35, No. 12, Dec. 1947.

[2] Theoretical Response from a Magnetic-Wire Record. Marvin Camras, Proc. I. R. E. & Waves & Electrons, Vol. 34, No. 8, Aug. 1946.

TABLE I.

Hc	Ratio of output at 12 kc to output at 1 kc
320	-9.5 db
285	-2 db
285	-6 db
285	-5 db
260	-13 db
260	-9 db

Speed – 4'/sec.
Bias frequency – 65 kc.
Amplitude adjusted for maximum output at 1 kc.

otherwise. *Table 1* illustrates this fact by showing that the wire with coercive force of 320 gauss has a lower high-frequency output than one with a coercive force of 285 gauss, and is about the same order as one with a coercive force of only 260 gauss. It will also be noted that the three wires with a coercive force of 285 gauss have rather wide variations in their 12 kilocycle outputs. These data were taken, and very carefully checked, using a wire drive of four feet per second, a bias frequency of approximately 65 kilocycles, and with the bias amplitude adjusted for the maximum reproduced signal at 1000 cycles. The exact cause of this effect is not known but there seems to be reason to believe that it lies in the composition of the material. Wires of other alloys than the widely used 18-8 stainless steel greatly affect both the output and frequency response with the result that some wires give as much as 10 or 12 db higher output for the same recording level.

The supersonic bias, used to enhance the recording characteristic of a medium, affects the high frequency output of a magnetic record in the following manner. As the bias is increased from a very low value, the output from the reproduce head increases with little change in frequency response. However, after the value of bias which gives maximum output at a medium frequency is reached, any further increase will cause a decrease

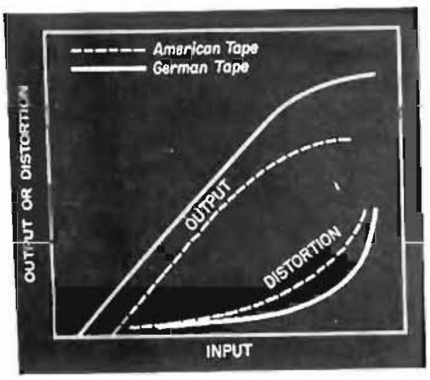

Fig. 2. Relation of input-output curves to degree of distortion.

in the high frequency output. This effect, which may be very serious at high bias currents, is apparently caused by self-erasing of the recording due to the strong bias field.

It is obvious that the amplifiers used with a magnetic recording system must be capable of reproducing the desired frequency response. It is well known that the output from a magnetic reproducing head in the low frequency region is a differential function. In view of this fact, it is necessary to add integration to the "reproduce" amplifier to compensate for the decreased output at the low frequencies. This imposes very stringent requirements on high quality systems in that the hum originating in the reproduce amplifier must be held to a very low value. Likewise the stray pickup of the reproducing head must be very low. This leads to various shielding and hum bucking schemes, none of which works to perfection! In wide band *wire* recorders particularly, the reproduce amplifier must have very low noise for best results because the output from the playback head at, say 50 cycles, is in the order of 200 microvolts at the first grid with a signal having low distortion. Tape systems can be designed to give considerably greater outputs than this, making

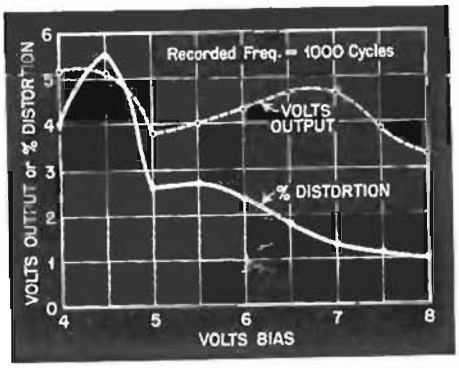

Fig. 3. Distortion as a function of bias applied to the recording head.

amplifier noise requirements less stringent.

The distortion in magnetic recordings is affected principally by the bias and medium used, and to a lesser extent by the amplifiers associated with the equipment.

As will be seen in *Fig. 2*, the input versus output curve of a magnetic recording medium is a fairly good indication of the distortion characteristics. The dashed curve is characteristic of most available magnetic recording media in which the straight portion of the input-output curve is relatively short, resulting in appreciable distortion at relatively low outputs. The solid curve is characteristic of some newly developed material in which the input-output curve has a relatively longer straight portion resulting in higher output for a given amount of distortion. However, it should not be overlooked that the two approach the same value of distortion at high recording levels. Furthermore, the medium with the longer straight portion will give very serious distortion on overload if it is not operated properly. The fact that some magnetic recordings do not "blow up" on serious overload, as is the case in other recording systems,

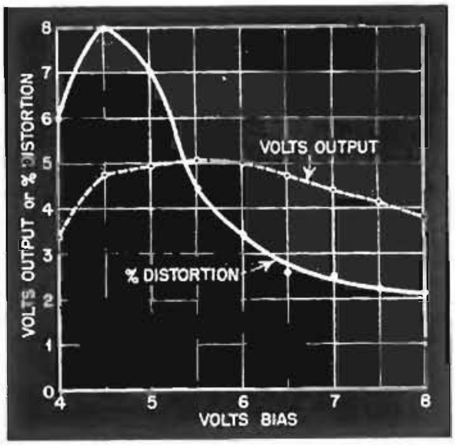

Fig. 4. Distortion vs. bias curves at 100 cycles;

may be attributed to the shape of the distortion characteristic as compared to the recording level.

The decision as to the shape of the distortion curve at maximum recording levels could very easily be difficult in some materials. Naturally it would not be desirable to use a material which had a very steep and abrupt curve because the distortion would increase very rapidly with even slight overload. Conversely, a medium would not be particularly desirable which has a very shallow curve because appreciable distortion may be generated at very low recording levels, thus limiting the signal-to-noise ratio available. It follows, then, that a compromise must be made in which a suitable signal-to-noise ratio is obtained with a satisfactory distortion curve.

When a system using direct current for bias or erasing purposes is designed the engineer must take into account the fact that serious even order distortion will result therefrom. This is not the case in a system using alternating current for bias and erasing purposes. It should be noted that some materials have more susceptibility to even order distortion arising from d-c bias or erase than others. Furthermore, it should not be assumed that just because a system is using a.c. for bias and erase that no even order components are present. Some peculiar effects arise, from time to time, due to accidental magnetization of improperly treated heads through switching transients, head construction, and other causes. Usually it is possible to detect magnetization in heads through the increase in background noise.

Bias Level

In general, the higher the value of the supersonic bias used on a recording head the lower will be the distortion in the reproduced signal. Here a compromise must be made between the maximum level which can be recorded and the desired frequency response because, as mentioned above, higher bias reduces the output at high frequencies. *Figure 3*

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shows the distortion in a reproduced signal of 1000 cycles from a 0.004 inch 18-8 stainless steel wire running at four feet per second as a function of the bias applied to the recording head. Also shown is the output resulting for the bias used. The input to the head was maintained at a constant value for these data. It can be seen that for the higher values of bias considerably less distortion is generated. However, this curve does not show the reduction in high frequency response caused by the higher bias. *Figure 4* is similar to *Fig. 3* except that a frequency of 100 cycles was used instead of 1000 cycles. It will be seen that the shape of the distortion curve has changed, and also the output curve is much broader than for the 1000-cycle case. The distortion for 100 cycles is higher than that for 1000 cycles even though the same recording level was used. This is not always the case. Wires have been tested which have just the reverse of this operation in that the low frequency distortion is much lower than that at medium and high frequencies. No data is available on tape concerning this phenomenon but it would seem that similar operation should be found.

Again, as was the case regarding frequency response, the amplifiers associated with a magnetic recorder must be capable of handling the output and input circuits without generating appreciable distortion if the full capabilities of the system are to be realized. Because magnetic recording, when properly handled, does not generate even order distortion it would seem advisable to use an amplifier which has the same general characteristics. This tends to lead to the use of push-pull output stages at least.

Audible Beats

One factor in the design and testing of a magnetic recorder, which has not received much attention, is the production of audible beats between the recorded material and the bias frequency. This effect generally occurs at the higher audio frequencies. The spurious signals appear to be the sum or difference, or both, of the harmonics of the audio signal beating with the supersonic bias. This effect is not appreciable except at recording levels which somewhat exceed the maximum permissible level for low distortion. However, most magnetic recorders use some sort of pre-equalization in which the current in the recording head is boosted at the higher audio frequencies. The level is usually adjusted for maximum signal at some low or medium frequency and, if the input which gives this level is maintained at the higher frequencies, beats, due to overload, are nearly certain to occur. However, if the input is adjusted in accordance with the pre-equalization the beats will not be present because little or no overload will occur. If the pre-equalization is properly designed it will take into account the maximum probable energy at a given frequency and adjust the head current so that overload will not occur at any point under normal conditions.

The designer of a magnetic recorder must consider all of the above factors in working out a unit. However, many of them at this time are nearly entirely empirical and thus must be determined by trial and error methods. Thus a great deal of experimental and developmental work is necessary to arrive at a satisfactory solution.

Finally, it may be said that, in magnetic recording, as in most other things, nearly anything can be done if one just wants to do it badly enough, but sometimes the desire must be extremely intense. Excellent frequency response, with low distortion and good signal-to-noise ratio may be achieved if one is willing to expend the time and effort necessary. However, the art of magnetic recording is essentially just beginning, and the means to achieve these ends are sometimes extremely difficult.

Magnetic Field Distribution of a Ring Recording Head

S. J. BEGUN*

The magnitude of the magnetic field components acting on the recording medium is determined.

IN most of the prior discussions of the theory of magnetic recording, the assumption has been made that the recording medium is subjected in the recording process either to a longitudinal or to a perpendicular magnetizing field only. An inspection of the field distribution around the gap of a ring recording head, however, indicates that a perpendicular field component is always present and that in many cases the magnitude of the perpendicular component cannot be neglected. W. W. Wetzel[1] shows qualitatively the longitudinal and perpendicular field pattern around a typical magnetic ring head but does not evaluate the effect of the perpendicular component.

Furthermore, no proper allowances have been made for the fact that in many recording mediums the strength of the biasing and signal fields are not constant over the thickness of the sound track in the zone of the recording gap. The longitudinal field component in air has been experimentally determined by Clark and Merrill[2], but these measurements relate to the average field strength over a considerable cross section, since a relatively substantial air column was enclosed by the wire loop which was used to scan the field along the path of the medium.

Magnetic Field Analysis

In an effort to get a better understanding of the various factors which might affect the recording process, a graphical analysis was made of the magnetic field to which successive layers of the active cross section of the recording medium parallel to the surface of the pole pieces are subjected. In view of the small physical dimensions, it is difficult to obtain this field distribution by direct measurements even in air. In addition, it seems next to impossible to determine experimentally the field pattern in the presence of a typical recording medium. Fortunately, graphical methods supply a sufficiently accurate picture of the field distribution. *Figure* 1 shows the field pattern with a recording medium bridging the gap region of a ring-type head. Since the field pattern is symmetrical on both sides of the center of the gap, only one-half of the field pattern has been plotted[3]. The field lines were drawn for a gap length which is equal to the coating thickness and for a recording medium with a permeability 3. These assumptions approximate commercial conditions for powder-coated tapes.

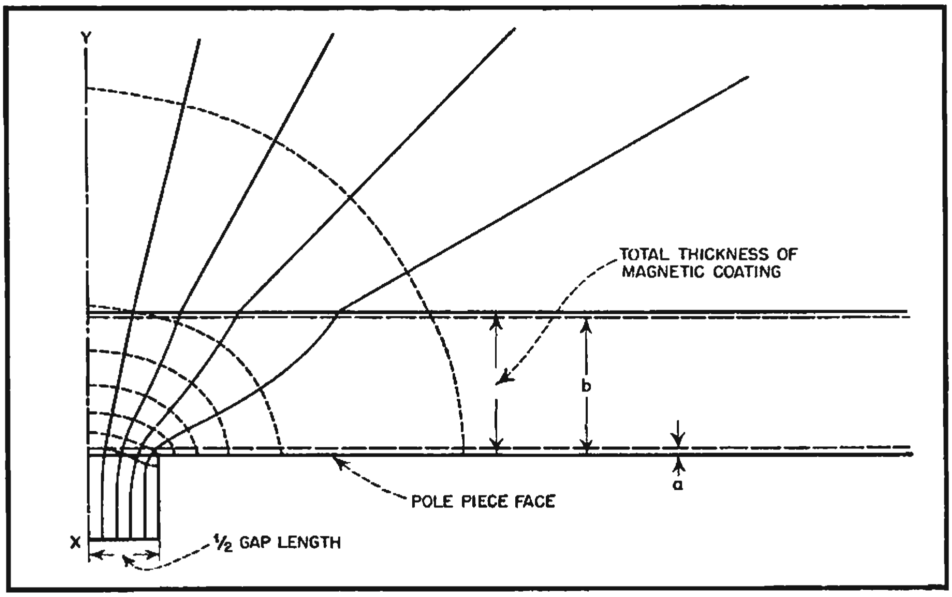
In *Fig.* 2 (plotted from the analysis of *Fig.* 1), Curve *A* shows the magnitude of the longitudinal field component and Curve *A'* the magnitude of the corresponding perpendicular field component as function of the distance from the center of the gap for a tape layer at a distance *a* (see *Fig.* 1) from the plane

*Brush Development Co., Cleveland, Ohio.

[1] Audio Engineering, *Dec.*, 1947.

[2] *Proc. of the I.R.E., Dec.*, 1947.

[3] *These plots were made by Prof. W. R. Smythe of the California Institute of Technology for The Brush Development Company.*



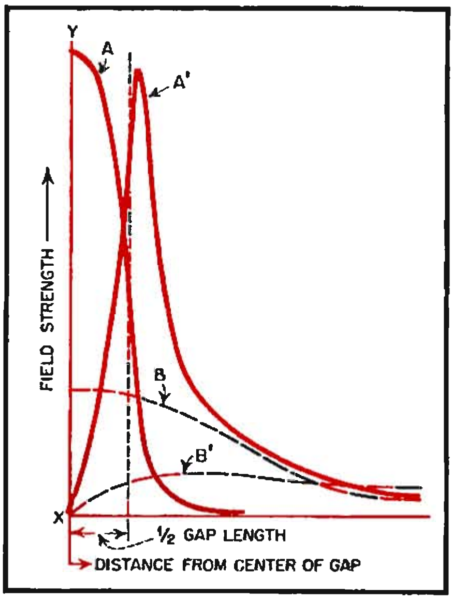


Fig. 1 (left). Magnetic field distribution from recording pole piece into and through coating of a powder-coated magnetic recording tape. Flux direction and density are represented by the direction and density of dotted lines. Field gradient is mapped by solid lines. Fig. 2 (right). Plot of the horizontal and vertical field components of the magnetic recording tape. See text.

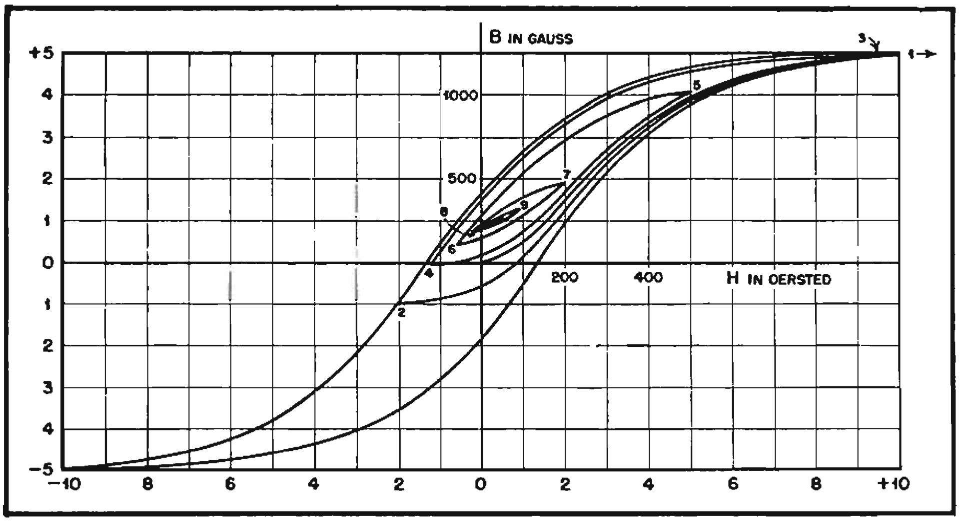
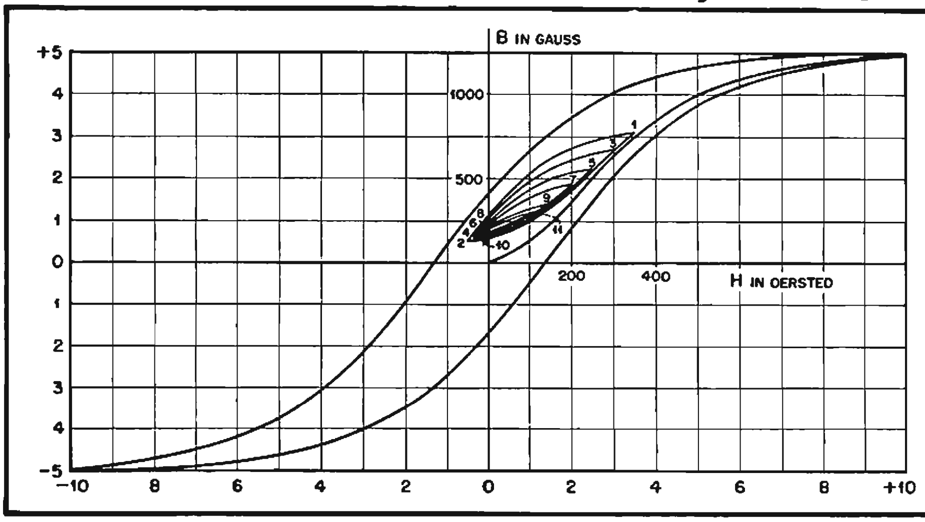


Fig. 3. The major hysteresis loop represents the magnetic properties of the coating. The minor loops, identified by 1, 2, 3, 4, etc., show the various steps through which an incremental part of a nearby layer passes when it moves away from the center of the recording gap and is subjected to decreasing field strengths of the combined d-c signal and high frequency a-c bias. The final induction reaches a value to the d-c signal.

Fig. 4. Similar to Fig. 3, but showing the various steps through which an incremental part of a far layer passes when it moves away from the center of the recording gap and is subjected to decreasing field strengths.



of the pole-piece faces. Curve *B* and *B'* are plots of the longitudinal and perpendicular field components at a distance *b* (see *Fig.* 1) from the plane of the pole-piece faces. Distance *a* corresponds to a layer very close to the surface of the coating, and distance *b* to a layer which is separated for an average coating thickness from the plane of the pole faces. These curves show that at the center of the gap the longitudinal components of the induction for layers close to the pole faces are almost four times greater than for the deepest layers. They also show that the effect of the perpendicular field component is much more pronounced in layers adjacent to the pole faces.

Considering the tape moving to the right from the center of the gap, the longitudinal field is at a maximum at the center of the gap and decreases very rapidly for close-by layers and relatively slowly for remote layers. On the other hand, the perpendicular field is zero at the center of the gap, rises to a maximum beyond the gap, and then decreases again steeply for adjacent layers and more gradually for deep layers.

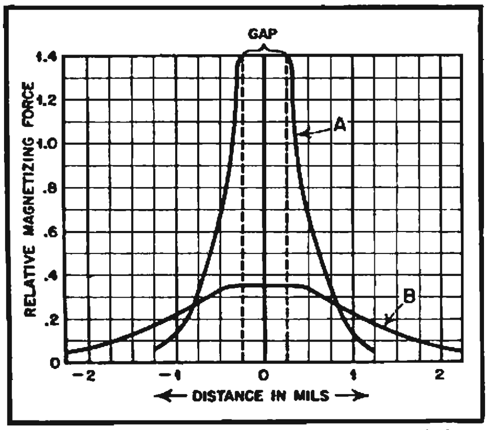


Fig. 5. Idealized total magnetizing forces as function of distance from center line of recording pole-piece gap. Curve A is for a layer close to the surface of the coating, and Curve B is for a layer well inside the coating.

Experience has shown that in a-c biasing the bias field strength must be of a certain magnitude to assure a linear transfer characteristic. In accordance with most theories explaining the mechanism of a-c bias, the peak value of the minimum bias field strength must be sufficient to produce induction values in the medium which lie beyond the instep of the normal magnetization curve. When applying such minimum bias field strength to the farthest layer at the center of the gap, the nearby layers are subjected to field strengths about four times greater, as shown by *Fig.* 2. Such large fields will cause saturation of the nearby layers in most recording materials. While the nearby layers are driven into saturation, no recording is effected. Only after the field strength has decreased to values below the saturation region will recording take place. Assuming that the bias field strength in the center of the gap was properly chosen to provide a linear transfer characteristic for the remote layers and assuming that approximately the same field strength will be most appropriate for recording in any layer of the material, it then becomes apparent from the field distribution curves of *Fig.* 2 that the final induction laid down in the farthest layers will be predominately longitudinal, whereas the final induction in the near layers will be mainly vertical. It is fortunate for this hypothesis that the ring head is an efficient pickup device for either component.

Graphical Analysis

It is of some interest to study the recording mechanism by means of a graphical analysis. It is assumed that the magnetic recording medium has the magnetic properties given by the hysteresis loop in *Figs.* 3 and 4. This is typical for a powder-coated tape. To simplify the problem, it is furthermore assumed that the signal is of such low frequency that it can be represented by a d-c value. In *Fig.* 5, Curve *A* shows in an idealized form the magnitude of the total magnetizing force produced by the d-c signal along the path of the motion of the tape across the gap of the head for a layer close to the pole faces. Curve *B* shows the magnitude of the total magnetizing force acting on far layers. No direction is associated in these plots with the magnetizing forces. Curves *A* and *B* were determined by vectorially adding the vertical and horizontal field components of *Fig.* 2. No attempt has been made to plot Curves *A* and *B* accurately, since the idealized curves as shown provide all information needed. If the d-c signal level is changed, the height of Curve *A* and Curve *B* will change correspondingly.

In a similar manner, the a-c bias magnetizing forces are plotted superimposed upon the d-c signal, giving the Wave Curves *A'* in *Fig.* 6 and *B'* in *Fig.* 7. The peak values 1, 2, 3, 4, etc., of Curves *A'* and *B'* were used to plot in the *B-H* diagrams of *Fig.* 3 and *Fig* 4 a series of hysteresis curves corresponding to the individual cycles of the magnetizing forces. The transfer of the wave pattern *A'* and *B'* to the B-H diagram takes only the decreasing magnetizing forces into consideration, starting from maximum values as they occur in the center of the gap. This simplifies the graphical representation without introducing any undue errors.

Induction Diagram

Points 1, 2, 3, of *Fig.* 3 and *Fig.* 4 are presented in an induction diagram in *Figs.* 6 and 7. Correlating the instantaneous magnetizing forces to the induction in the recording medium while it passes over the head, it is interesting to note that the remanent induction in this case is approximately the same for the nearby and far tape layers[4]. This, however, will not always be so. The remanent induction values left in different layers will depend upon the bias condition, the magnetic properties of the recording medium, and the recorded wavelength.

Making a sufficient number of plots for different conditions, namely different biasing and signal forces, the remanent induction can be obtained either as function of the bias magnetizing force for a constant signal or as function of the signal level for a constant biasing force. Such curves have been plotted and found to be in good agreement with experimental data.

This particular graphical representation, however, must be viewed with some caution since no attention has been given to the change of direction of the magnetizing force while the magnetic recording medium moves away from the center of the gap. For nearby layers, the remanent induction must be essentially perpendicular, since they are driven to saturation until the peak value of the magnetizing force is smaller than that identified by Point 3 in *Figs.* 3 and 6, which coincides with the position of the tape in the head where the perpendicular component is predominant. This conclusion can be properly drawn on the basis that a saturated ferromagnetic material shows no hysteresis losses when the field direction is changed[5]. For the remote layers, on the other hand, conditions are more complex and only additional studies can tell whether a graphical method describes the conditions properly.

After the recording medium leaves the pole pieces, demagnetization will take place mainly because of the perpendicular field in the surface layers. Would there be true longitudinal recording made with a d-c signal field, no demagnetization should be expected, since a magnet is formed under these conditions which is extremely long in relation to its cross section. No allowances have been made for effect of demagnetization.

The graphical analysis can be made to show also that a slowly tapering field, as shown by Curve B in *Fig.* 5, is unsuitable for recording high frequencies. The relatively fast varying signal intensity which is superimposed upon the bias field has a self-erasing effect. It is for this reason that short wavelengths will be retained much better by the nearby layers which are subjected to a sharply decaying field pattern. Experimental evidence confirms this explanation. When frequency curves are taken with tape coatings of different thicknesses, the playback level from these tapes in the lower frequency region is approximately proportional to the thickness, thus indicating that the complete sound track cross section contributes to the threading of flux lines through the reproducing head. In the high frequency region, on the other hand, the output level from these tapes does

[Continued on page 39]

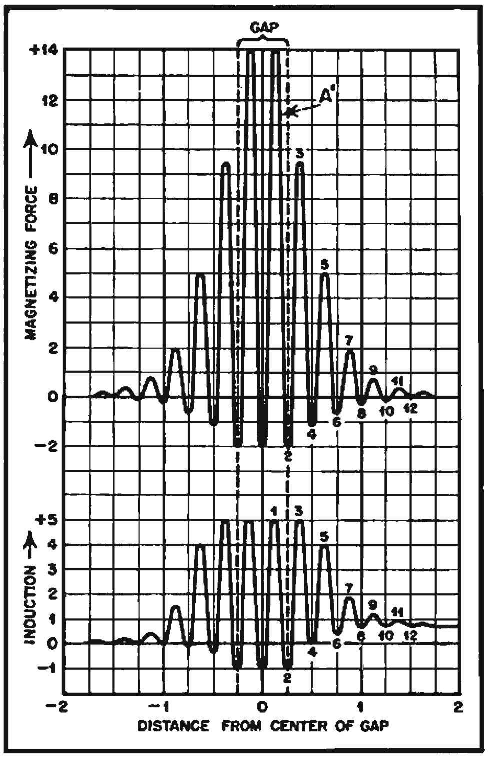


Fig. 6. Magnetizing forces on and resulting induction in an incremental unit of a magnetically coated tape layer as it passes across the recording gap of a magnetic recording head being excited with a d-c signal superposed on a-c bias. The layer is assumed to be one close to the surface of the magnetic coating.

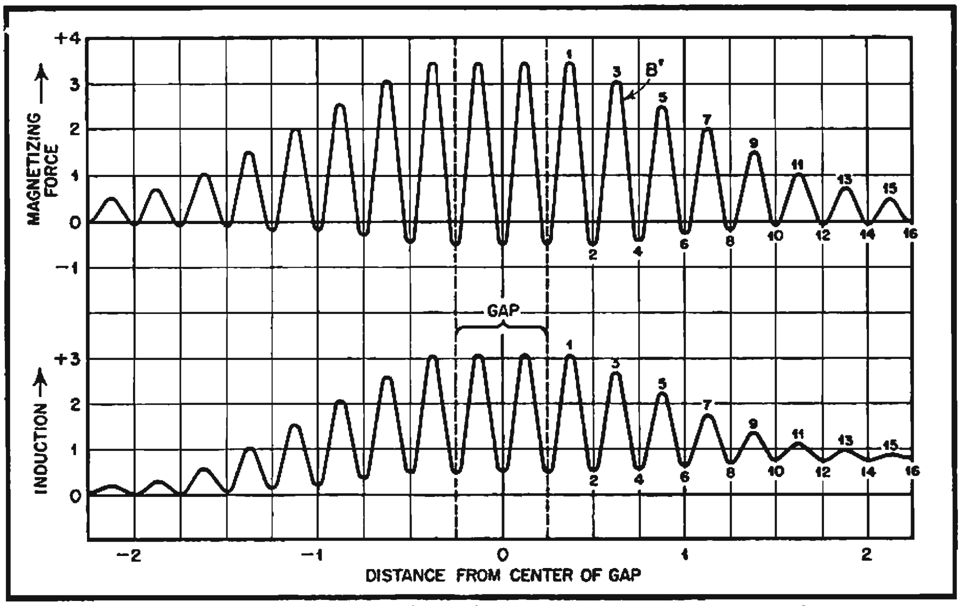


Fig. 7. Same as Fig. 6, except that the unit of magnetic powder coating is assumed to be located far inside the magnetic powder coating, away from the recording pole pieces.

[4]*These plots were prepared by J. E. Shomer of The Brush Development Company.*

[5]*Bozorth, R.M., "Magnetism"*, Rev. of Mod. Phys., *Jan.*, 1947 (*Vol.* 19), *p.* 47.

[6]*Kornei, O., "Frequency Response of Magnetic Recording"*, Electronics, *Aug.*, 1947.

RING RECORDING HEAD

[*from page 13*]

not correspond any more to the thickness of the magnetic coating[6].

From the graphical analysis, it can also be concluded that overbiasing is not as harmful as underbiasing. One must, however, see to it that the biasing forces which produce the proper recording conditions for the nearby layers do not fall into "tail region" of the field distribution pattern where its decay becomes more gradual, since this will result in loss of output level for high frequency signals.

Much additional work has still to be done in order to learn more about all factors which control the recording process. Because of the complexity of the problem, certain assumptions have been made which may not prove to be completely justified. For example, in the above discussion no allowance was made for the fact that the permeability of the recording medium drops when saturated. A change of permeability will lead to a modification of the field distribution (*Fig.* 1).

The conclusions advanced here are not intended to provide a complete picture of the field strength phenomenon in the recording process, but thinking and experimental work along are intended to encourage additional similar lines.

HEAD ALIGNMENT

With Visible Magnetic Tracks

B. F. MURPHEY* and H. K. SMITH*

It is reasonably obvious that, in order to obtain optimum results in magnetic recording on tape, it is necessary to align the recording and reproducing head gaps to a certain degree of parallelism. In order that tapes recorded on one machine may be reproduced on another without loss of high frequencies, it is necessary that the gaps be made perpendicular to the edge of the tape. An easy technique has been developed to obtain these alignments which seems to be worth calling to the attention of audio engineers.

The first order approximation for amplitude loss due to skewed gaps is reviewed, and a technique for microscopic examination of the recorded track is described.

Theory of Alignment

Schott[1] first discussed the effect of nonparallel gaps in magnetic recording, although certainly the equivalent situation in photograph recording received prior attention. Imagine a line gap which is within an angle α of being exactly perpendicular to the motion of the tape. It is possible to compute the ratio of the voltage induced in a misaligned head to the voltage induced in a perfectly aligned head. The calculation, given in the appendix, leads to the following result.

$$E_{\alpha_{max}} \Big/ E_{o_{max}} = \frac{\sin\beta}{\beta}$$

where $\beta \cong \pi\alpha\, d/\lambda$

The width of the tape is denoted by d and the wavelength on the tape by λ.

From these equations we may predict a decrease in output for a misaligned head which is greater, the greater the gap length d, the greater the angle α and the smaller the wavelength λ. *Fig.* 1 illustrates the effect to be expected for certain practical values of these constants.

Table 1 shows the values from which *Fig.* 1 was plotted.

From a plot such as this, values for other wavelengths and gap lengths may be obtained readily. For instance, the effect of doubling d is the same as doubling α, while doubling λ is equivalent to halving the angle α. *Table II* lists the value of α required to give certain attenuations for various values of d and λ in use today.

If we assume that a given method of slit alignment will assure accuracy of perpendicularity to within an angle of plus or minus δ, the alignment which assures interchangeability of tapes between machines is obviously $\delta \leq \alpha'/2$ where α' is the angle for the the permissible attenuation. From *Table II* it is seen that δ for 1 db attenuation must be less than 4′ 30″ in cases 1 and 2, less than 3′ 36″ in case 3, and less than 2′ 15″ in case 4. For constancy of reproduction to within 0.5 db, these values of δ are not halved but multiplied by 2/3, since the curve of *Fig.* 1 is not linear.

A method is described in the following section which allows for adjustment of aligument to within a half db in cases 1, 2, and probably 3; to within 1 db in cases 1, 2, and 3, and probably case 4.

TABLE I

α	$d = 0.1''$ Sin β/β	$\lambda = .001''$ db Loss
2′	.993	.07
4′	.978	.18
10′	.865	1.3
20′	.531	5.5
30′	.141	17
34.5′	0	∞

Microscopic Technique

Visual examination of the track recorded on magnetic tape may be carried out in a manner analogous to the mapping of magnetic fields by means of iron filings. Using suspensions in ethyl acrylate of Fe_2O_3 particles about 1 micron in diameter, Bitter[2] obtained the first pictures of the domain pattern on a ferromagnetic surface. A similar method has been used before to make the pattern associated with

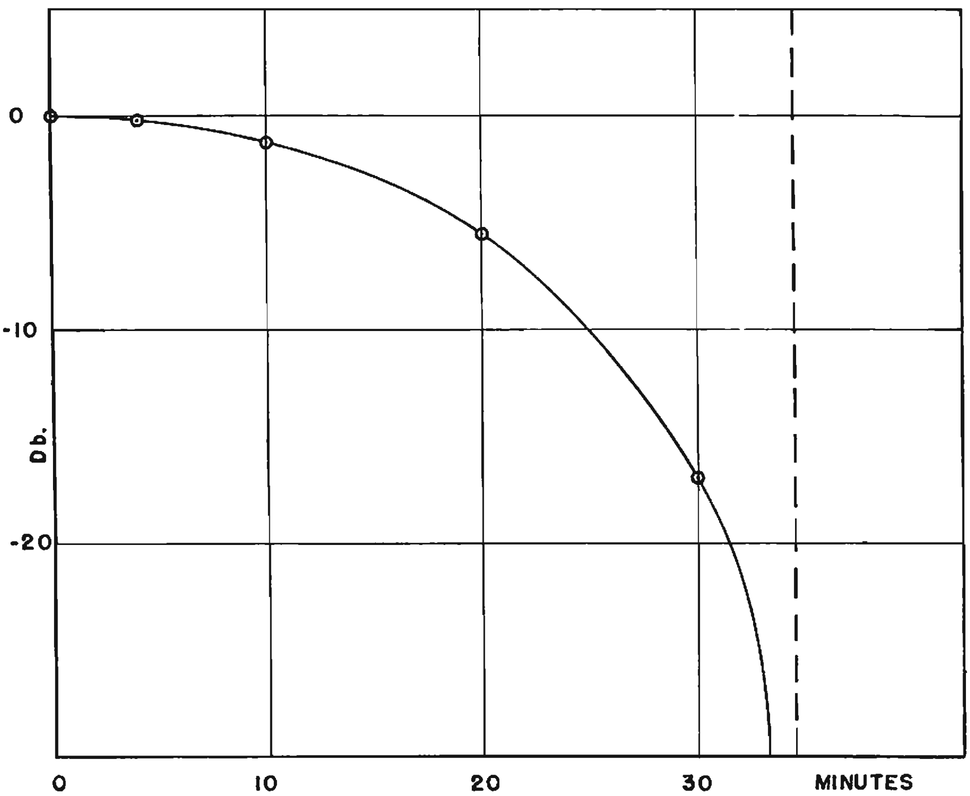


Fig. 1. Db loss versus α in minutes of Arc. d = 0.1 inch, λ = 0.001 inch.

*Minnesota Mining and Manufacturing Co., St. Paul, Minn.

[1] Z. Techn. Physik 17, 275 (1936).

[2] F. Bitter, Physical Review 41, 507 (1932)

the track on magnetic tape visible. The process consists simply of covering the surface of a magnetic tape recorded to saturation with an oil suspension of Carbonyl Iron particles. There appears, almost at once, a series of straight lines at right angles to the length of the tape which is visible to the naked eye. Photomicrographs of typical patterns are shown in *Fig.* 2. Wavelengths as short as 0.001 inches have been examined where the alternating poles are one-half mil apart.

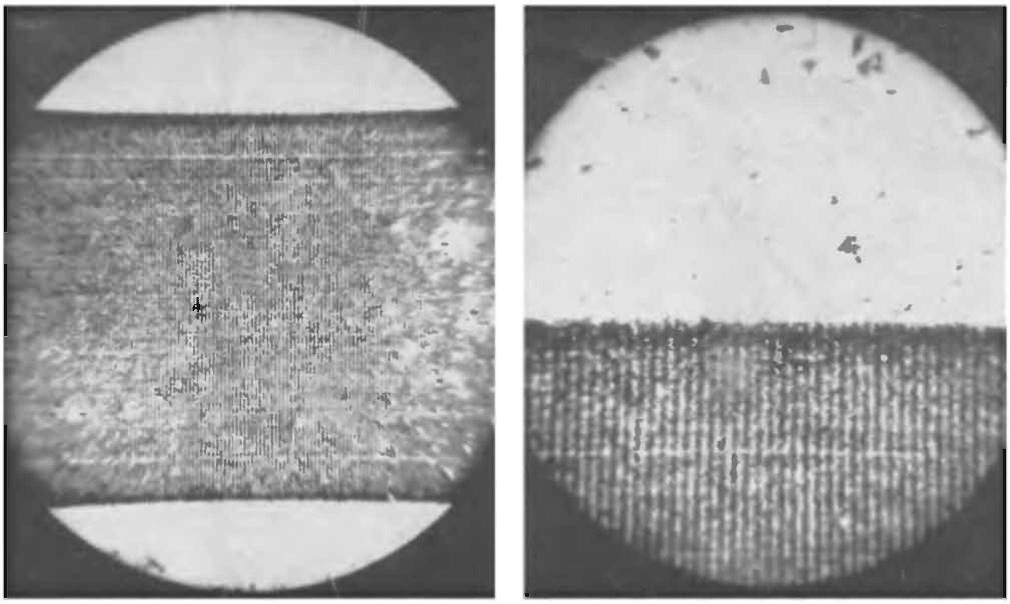


Fig. 2. 0.0023 inch wavelength signal at magnifications of about 20X and 60X.

The materials used for producing a visible pattern of the recorded signal on magnetic tape vary. Black magnetic oxide of iron, gamma ferric oxide, or iron powder may be used. These should be very fine particles, 3 microns or less, and dispersed in a liquid which does not evaporate readily and which has a sufficiently high viscosity to prevent flocculation. Flocculation, to any great extent, interferes with the sharpness of the lines and consequently with the accuracy of the measurements. The powder selected should give as much color contrast as possible with the tape being used. Carbonyl Iron of about 3 microns diameter made by General Aniline Company, Grasselli, New Jersey, having a grey color which reflects light well, has given very satisfactory results when mixed with raw linseed oil or a mineral oil such as "Nujol".

Using this method of making the track visible, the process of alignment of magnetic recording heads is as follows: A saturated signal of one mil wavelength is recorded on the tape. A portion of the tape is covered with the oil suspension of Carbonyl Iron and examined by means of a microscope provided with cross hairs and a rotating stage in order to determine whether or not the lines corresponding to the recorded track are perpendicular to the edge of the tape. Adjustment of the head, recording and microscopic examination of the track are repeated until the track is aligned within the desired angle.

Once the recorded track is correctly oriented, the recording head is obviously aligned. To adjust the reproducing head, a long recording is made at the highest frequency the system is designed to produce. A meter is connected to the output of the system and, using the long recording, the reproducing head is adjusted to give maximum output. From the theory of misalignment, it is apparent that the greatest sensitivity for adjustment is found at the shortest wavelength.

We were required to align a set of heads which fitted the conditions of Case 1, *Table II.* The microscope used was provided with a carefully centered rotating stage and a vernier attachment graduated to 03′. A 28 mm. objective and a 12.5X eyepiece gave ample magnification. The stage was shifted until the horizontal cross-hair fell on the track edge, then the vertical cross-hair was used to measure the angle of the recorded pole[3].

If pains are taken to make the length of the tape parallel to one of the motions of the stage, then the stage motion at right angles may be used to follow the track across the tape. The alignment can then be checked by observing the fraction of a track width that the tape is misaligned. Knowing this, the wavelength and width of the track, the angle of misalignment may be computed.

Centering the track edge in the field avoids the "pincushion" effect so apparent in *Figure 2a.* With this equipment it was relatively easy to obtain track perpendicularity to less than 0°03′.

For measuring the angularity of the recorded signal, a magnification of about 30 to 40 times is all that is necessary. A greater magnification than this leaves such a short part of the recorded signal in the field of view that accuracy of measurement is reduced.

The visual examination of the track not only provides a means of aligning tape, but also reveals peculiarities in head construction. In *Fig.* 2 for example, there appear streaks along the length of the recording. Gaps between laminations will show up in such a manner, but will be much more pronounced. In one instance the gap of the record head was made as small as possible by removing the foil. Upon examining the track recorder from this head, it was found that each lam-

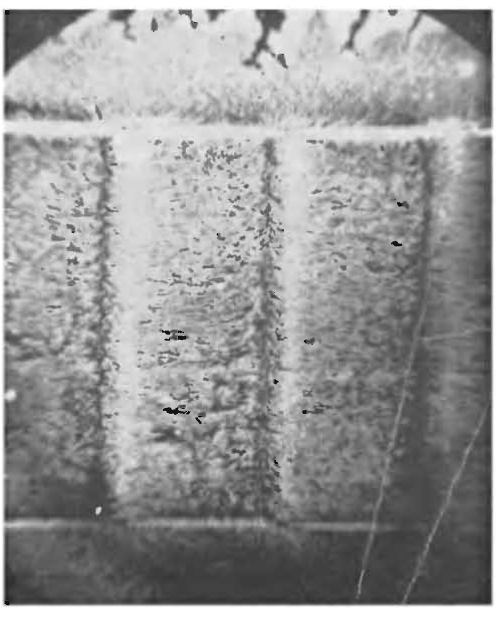
[*Continued on page 38*]

TABLE II

Case	d	λ	f in cps	V in./sec.	α′ for 1 db att.	α′ for .5 db att.
1	0.1″	.001″	7500	7½″	09′	06′
2	0.2	.002	9000	18″	09′	06′
3	0.25	.002	15000	30″	07′ 12″	04′ 48″
4	0.25	.00125	12000	15″	4′ 30″	3′ 06″

[3]Instruments such as the Gaentner Coordinate Comparator, which may be adjusted to a smaller angle, 01′, might also be used. Some limitation on the method is imposed by slight imperfections along the edge of the tape and by lack of definition of the track.

Fig.3. 92 mil wavelength signal, magnified approximately 20 times.



Magnetic Recording Head

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ination was acting individually so that there were as many tracks as there were laminations. Offset laminations also show up in a striking manner. Some notion of the fringing effect at the edge of the gap may be obtained by looking at long wavelengths. There is shown in Fig. 3 a 92 mil wavelength track where fringing is in evidence. It should be noted that in order to obtain clear tracks, a fairly strong signal must be recorded on the tape.

In conclusion, the authors wish to express their thanks to the management of Minnesota Mining & Manufacturing Company for permission to publish this material.

Appendix—Theory of Alignment

Schott's attack on the effect of non-parallel gaps in magnetic recording is essentially as follows:

For simplicity, assume a line gap and further suppose the flux intercepted by the reproducing head to be the sum of the induction in the tape under each increment of the gap length.

Let a tape whose length lies in the y direction be recorded with an induction B varying sinusoidally such that

$$dB = C \sin \frac{2\pi y}{\lambda}\, dX; \quad \text{for } \frac{-d}{2} \leq X \leq +\frac{d}{2}$$

which expresses the fact that the induction is uniform from —d/2 to + d/2 in the X direction. Let the reproducing gap make an angle α with the Y-axis. Any element dX of the gap will, therefore contribute to the flux pickup in the head an amount

$$dB_\alpha = C \sin \frac{2\pi}{\lambda} (y - X \tan \alpha)\, dX$$

where

$$\frac{2\pi X \tan \alpha}{\lambda}$$

represents the phase angle of the flux at X resulting from misalignment.

The total flux pickup is then given by

$$B_\alpha = C \int_{-d/2}^{+d/2} \sin \frac{2\pi}{\lambda} (y - X \tan \alpha)\, dX$$

or

$$B_\alpha = Cd \frac{\sin \beta}{\beta} \frac{\sin 2\pi Y}{\lambda}$$

where

$$\beta = \pi d \tan \alpha / \lambda \qquad (1)$$

Substituting $y = Vt$, where V is the constant tape velocity and t is time, we may calculate the voltage developed in n turns of the pickup winding.

$$E_\alpha = -n \frac{dB_\alpha}{dt} = c'd \cdot \frac{2\pi V}{\lambda} \frac{\sin \beta}{\beta} \cos \frac{2\pi Vt}{\lambda}$$

The peak value of E_α occurs when

$$\cos \frac{2\pi Vt}{\lambda} = 1$$

or

$$E_{\alpha_{max}} = c'd \frac{2\pi V}{\lambda} \frac{\sin \beta}{\beta}$$

For a perfectly aligned gap

$$\alpha = 0, \beta = 0 \quad \text{and} \quad E_{o_{max}} = c'd \frac{2\pi V}{\lambda}$$

or the fractional decrease in voltage may be written as:

$$E_{\alpha_{max}} / E_{o_{max}} = \frac{\sin \beta}{\beta} \qquad (2)$$

For small values of α

$$\alpha: \quad \beta \cong \pi \alpha\, d/\lambda \qquad (3)$$

Magnetic Tape and Head Alignment Nomenclature

N. M. HAYNES*

Suggested terminology for expressing causes of malfunctioning of experimental and commercial tape recorders.

SOMEBODY ONCE SAID that an art drops its swaddling clothes when it loses its ambiguous expressions, and becomes a science when its terminology acquires both conciseness and accuracy.

The development of some phases of the art of magnetic tape recording has been handicapped by inadequate terminology. This handicap has somewhat hindered the free exchange of ideas between experimenters and technicians.

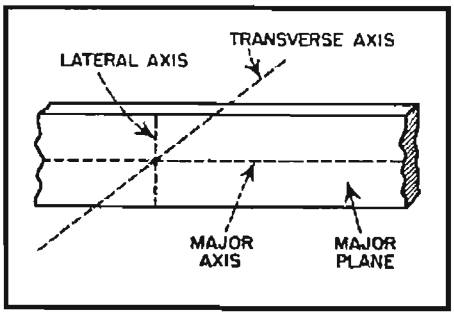

Fig. 1. Tape axis nomenclature.

Although some work has been done to compile a glossary for magnetic tape recording all efforts have been focused on definitions instead of derivations of much needed terms.

The inadequacy of our present terminology was humorously exemplified when a designer of recording heads found it extremely difficult to transmit precautionary instructions to a subcontractor without using his hands. Subsequently, two project engineers (both college graduates) were found discussing head alignment problems in the sign language. (The left hand, with fingers extended represented the tape, the fingers pointing in direction of tape travel and the palm representing the coated side. The right hand, similarly held but at right angles to the left hand, represented the head gap. Tilting, skewing and rotating the right hand effectively portrayed common types of gap misalignments.)

Early efforts attempted to tie misalignment to astronomical and gun control terms. For a short while azimuth, elevation, and meridianal deviations became meaningful. When the tape transport mechanisms were redesigned for vertical (rack mount) operation, junior engineers were literally standing on their ears to reorientate their terminology. Subsequently geographical terms were applied. Longitudinal, lateral, polar, and transverse deflections took on some meaning. Difficulties, however, became evident when attempts were made to correlate the working head gap with the tape.

For example, the actual gap *length* determined the magnetic track *width*. The gap *width* was used to determine the effective resolution of recorded wave *lengths*. It was finally decided that inasmuch as the tape was the most determinative characteristic element of the process, all terminology was to be in terms of the dimensions of the tape, the orientation of the tracks on the tape and the direction of tape travel.

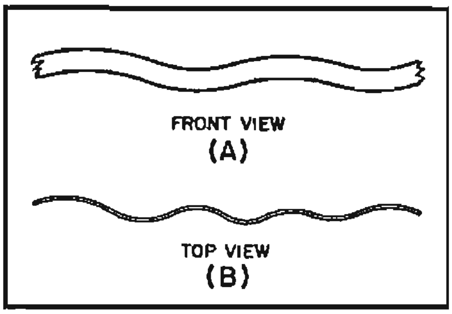

Fig. 2. Tape motion nomenclature: (A) Lateral weave; (B) transverse weave.

Tape Nomenclature

Major Plane: The major plane of the tape is its largest surface. Its boundaries are determined by its length and width.

Major Axis: The longest axis on its major plane.

Directional Nomenclature

Longitudinal: Along the longest dimension of the tape (length).

Lateral: Across the width of the tape (second largest dimension).

Transverse: Through the thickness of the tape (third tape dimension).

Longitudinal Axis: An imaginary line coinciding with the major axis.

Lateral Axis: An imaginary line on the major plane perpendicular to the major axis.

Tranverse Axis: An imaginary line perpendicular to the major plane and major axis. (See *Fig.* 1)

Tape Motion Nomenclature

Lateral Weave: Movement of the tape in the direction of its lateral axis. This kind of movement is usually caused by inadequate tape guides and, if excessive, will result in improper tracking between the record and playback heads and the production of amplitude variations. When improper tracking between the erase head and pickup head occurs, incomplete erasure is sometimes evident because the erase track does not consistently "blanket" the pickup track.

Transverse Weave: Movement of the tape in the direction of its transverse axis. This type of tape travel is usually caused by improper pressure pads, wrinkled tape, or obstructions in the normal tape path and results in amplitude variations particularly pronounced in the high frequencies. (See *Fig.* 2)

Longitudinal Weave: An irregular movement in the direction of the tape travel caused by eccentric rotary elements, uneven rotary torque in the drive system, or variations in tape drag or tape takeup. Produces frequency modulation (flutter, "wow" and "drift").

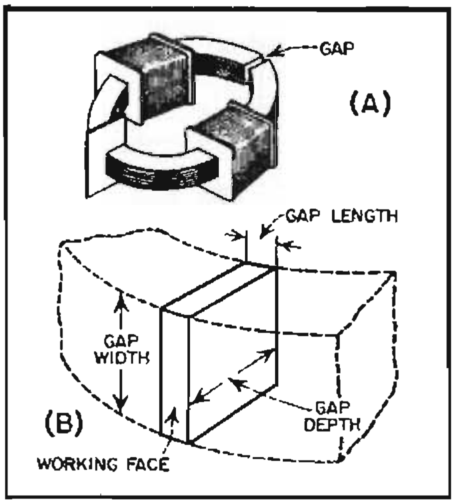

Fig. 3. Recording-playback head terminology: (A) Gap placement in recording head; (B) gap dimensional terminology.

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Recording-Playback Head Gap Terminology

For sake of simplicity the dimensions of the gap in the record or playback head are correlated to their respective effect on and in relation to the tape.

Gap Length: Dimension of the gap along the longitudinal axis of the tape. This dimension determines the scanning resolution of the head. (The shorter the gap, the lower the pickup level and the higher the frequency it resolves.)

Gap Width: The dimension of the gap along the lateral axis of the tape. This dimension determines the magnetic track width. The wider the gap the higher the signal level and the greater the dynamic range. (Doubling the track width increases dynamic range by 3 db.)

Gap Depth: The dimension of the gap along the transverse axis of the tape. Short depths provide less leakage and enable full magnetic tape modulation with lower recording levels. (See *Fig.* 3).

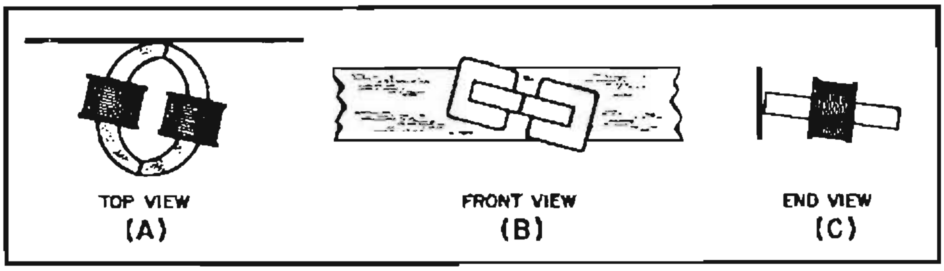

Fig. 4. Gap alignment terminology: (A) Longitudinal deviation; (B) lateral deviation; (C) transverse deviation.

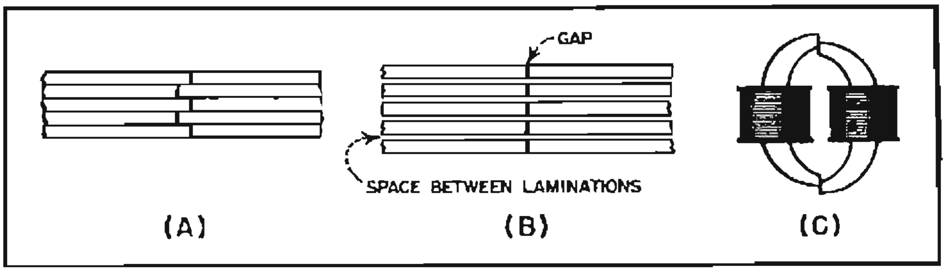

Fig. 5. Gap displacement terminology: (A) Longitudinal displacement; (B) lateral displacement; (C) transverse displacement.

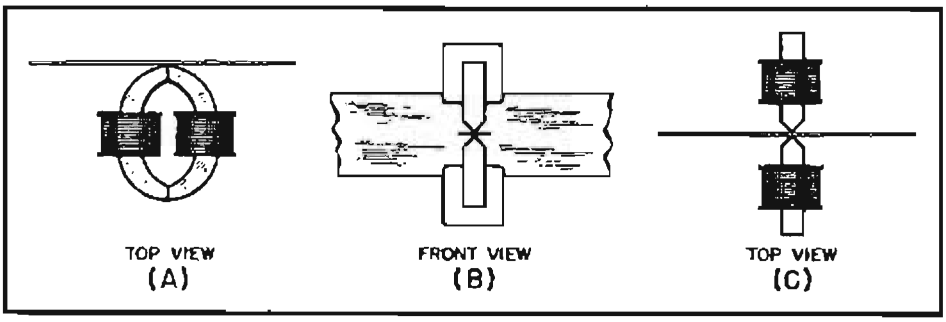

Fig. 6. Direction of magnetization terminology: (A) Longitudinal; (B) lateral; (C) transverse.

Gap Alignment Terminology

The working face of the gap can be misaligned in three different planes in relation to the tape. These misalignments are known as angular deviations expressed in degrees and are of the most serious type for they prevent tape interchangeability between machines.

Longitudinal Deviation (also known as tangential, azimuth, and polar deviations): Angular displacement of the working face of the gap in an arc tangential to the longitudinal axis and major plane of the tape. Perfect tangential contact of both gap edges cannot be maintained easily when longitudinal deviation exists. This type of deviation results in loss of both amplitude and high frequencies. In a two-way drive system frequency response characteristics vary according to the direction of tape travel and the head tends to clog up much sooner.

Lateral Deviation: Angular displacement of the working face of the gap in the major plane and about the transverse axis of the tape. The type of misalignment is the most serious for it contributes largely to loss of high frequencies and is most deleterious to tape interchangeability between different machines.

Transverse Deviation: An angular displacement of the working face of the gap in the transverse plane of the tape about the longitudinal axis. This misalignment prevents the full width of the magnetic track from coming in contact with the working face of the gap and results in lowered recording and pickup levels. It has the same effect as reducing track width. (See *Fig.* 4)

In terms of the magnetic track on the tape, the gap in the recording or playback head may be set improperly. The maladjustments are known as displacements.

Longitudinal Displacement: (Longitudinally stepped) An irregular and discontinuous gap caused by misalignment of each gap formed by the individual laminations in the core structure of the head. Misalignment takes place along the longitudinal axis of the tape and also prevents interchangeability between machines. (See *Fig. 5A*)

Lateral Displacement: A discontinuous gap caused by actual separation of the laminations along the lateral axis of the tape. (See *Fig. 5B*)

Transverse Displacement: (Transversely stepped) An uneven alignment of the edges of the gap along the transverse axis of the tape. Results in poor high frequency response. (See *Fig. 5C*)

Gap maladjustments may exhibit any combination of two or more deficiencies which are usually easily detected by examination through a low power microscope.

Direction of Magnetization Terminology

It is unfortunate that, in the early stages of the art, all possible variations and directions of magnetizations on tape had not been considered carefully before usage popularized the terms "longitudinal" and "perpendicular." Both these terms are well suited and self-explanatory for two-dimensional wire recording but confusing for tape. Although subsequently carried over to tape, an additional term was needed for the third-dimensional type of recording (across the width). Transverse was inappropriately chosen to form a rather conglomerate trio of longitudinal, perpendicular and transverse expressions. The term "perpendicular" is both ambiguous and unrelated to longitudinal or transverse.

If "longitudinal" be reserved for lengthwise magnetization and "lateral" for widthwise magnetization, then transverse is a natural term for magnetization through the medium. (The length of the magnet would be exposed in a transverse cross section of the medium). With these expressions a correlated family of terms results which nicely ties in with the suggested magnetic tape and head alignment nomenclature.

Longitudinal Magnetization: Magnetization of a recording media in its major plane along its longitudinal axis. (See *Fig.* 6A)

Lateral Magnetization: Magnetization of a recording media in its major plane along its lateral axis. (See *Fig.* 6B)

Transverse Magnetization: Magnetization of a recording media perpendicular to its major plane along its transverse axis. (See *Fig.* 6C)

Any combination of two or more types of magnetizations can be described easily with this nomenclature. For example, oblique magnetization indicating a lateral deviation in degrees concisely defines a specific mode of magnetization of a recording media. Oblique transverse with lateral deviation expressed in degrees similarly describes a combination of all three types.

It is the writer's hope that these or similar terms will soon become standard so that technicians working in the field of magnetic tape recording can more easily effect an exchange of ideas and more readily express results of their investigations.

Magnetic Tape Erasure by Permanent Magnets

ROBERT HERR*

Applications of d-c pulses may be used to obtain erasure almost comparable to the results possible with a-c bias, providing the conditions are carefully controlled.

For good erasure of magnetic tape recordings, two requirements should be met. The first is complete obliteration of previous signals. This condition is met if, at some time in the erasing process, the magnetic material is saturated in at least one direction. The second requirement is demagnetization of the tape. This is important in order to achieve minimum background noise and minimum distortion in the subsequent recording. This condition is best met by subjecting the magnetic material to a large number of cycles of alternating fields of symmetrical waveform which at some point in the process reach substantial saturation and, thereafter, decrease gradually through many cycles to zero.

A third consideration of some practical importance is that the process be insensitive to the magnetic characteristics of the tape to be erased, so that a given erasing means may be used without change for any of a wide variety of tapes. The alternating field process, with a high enough maximum field, will work on any tape. Some permanent magnet processes will obliterate the signal from any tape, but leave various states of magnetization on various tapes.

Thus from a magnetic point of view, demagnetization by alternating fields is satisfactory, but from a manufacturing point of view it is attractive to erase tape by one or more permanent magnets because such erase offers economy, reliability, simplicity, light weight, and freedom from servicing. This paper discusses some of the points to consider in the use of permanent magnets for erase.

If the erasure is to be followed by recording with d-c bias there is no problem. Here the erase should not be designed to demagnetize the tape but to saturate it, and a single saturating magnet is all that is required. However, with this type of recording, a high noise level is unavoidable and most present day recorders use a-c bias. The following discussion assumes that the erase will be followed by a-c bias recording.

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D-C "Pulse" Erasing

With a single permanent magnet of sufficient strength to saturate the tape, the signal may be easily obliterated but the noise level of even a perfect tape would be fairly high and that of actual tapes is very high. For example, noise levels of 20 to 30 db higher than those normally obtained with a-c erase are common. In addition, the polarized condition of the tape leads to serious even order harmonic distortion in the subsequent recording, a condition which is absent with a good a-c erase. The expedient of a "single pulse" permanent magnet (or d-c) erase can therefore be justified only when cost or simplicity is the prime consideration.

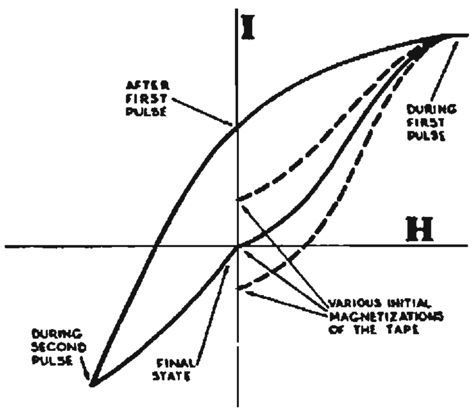

Fig. 1. Curve showing principle of demagnetization by two d-c pulses.

The next step is to consider two d-c "pulses." In principle such a cycle can demagnetize, as shown schematically in *Fig.* 1. The first pulse should be saturating (in either direction); the second pulse is of opposite polarity and of just such strength as to leave the material with zero magnetization after its removal.

At this point several considerations enter:

1. The hysteresis loop of the material which we obtain from any gross measurements represents an average over many billions of oxide particles. These particles are crystals or oriented aggregates of crystals, each of which has preferred axes of magnetization. For a given direction of magnetization, each crystal orientation will result in a different contribution to the observed hysteresis loop. Under these conditions, a cycle of two pulses cannot be adjusted to demagnetize all the particles. However, a gross average of zero magnetization can be obtained, and when it is, the noise will be much reduced from that obtained with one pulse because,

   a.) The magnetization of individual particles, while not zero, will likely be less than that resulting from saturation, and

   b.) Noise resulting from tape irregularities of relatively long wave length will be effectively reduced.

2. Any given erase will not be optimum for a variety of tapes, although it may be better than a single-pulse erase for all of them.

3. In actual use, the tape when heard has passed not only the erase head but also the record head. In this process the bias field acts on the tape, even when no audio is recorded. Thus the demagnetization result discussed above is of purely academic interest. What is important is the condition of the tape after erasure and biasing. In biasing, the tape is subjected to an a-c field of increasing and then diminishing strength. For a demagnetized tape the bias (if of good design) changes conditions very little, but for a tape erased by one or a few d-c pulses the change may be considerable. To some extent the bias field acts as an incomplete (i.e., non-saturating) a-c erasure and tends to reduce noise. It may, however, tend to alter an average tape magnetization from the zero value in which a permanent magnet erase cycle left it, and thus tend to increase noise. This will be made clear below.

A characteristic of a tape in which an average zero magnetization has been achieved by a few d-c pulses is that it is composed of a collection of particles all of which may be magnetized. If a small a-c field is applied and gradually reduced to zero, it may demagnetize some of these elements or alter some domain boundaries that are relatively unstable, while not affecting others. When this happens the average magnetization is no longer zero.

Method of Obtaining Data

To take quantitative data, a large solenoid was used to apply known fields to a ¼-inch square rod, 12 inches long, of oxide dispersion. The magnetization was measured by passing the rod through a pickup coil connected to a ballistic galvanometer. A d-c pulse was administered by setting the desired current from storage batteries in the magnetizing solenoid and passing the rod through the solenoid. Diminishing a.c. fields were obtained in the same way, using a 60 cycle supply with a Variac. The diminishing was done by the movement of the rod rather than by the Variac. After each pulse of d.c., the magnetization of the rod was measured, and after a suitable erase schedule, the rod was subjected to a-c fields and measured after each "shake" beginning with weak and increasing to saturating a-c fields. The units for magnetization were left arbitrary, although they could be converted to absolute values of flux lines or gauss; the field values were measured in oersteds.

The data are summarized in *Fig.* 2. The schedules of d-c fields used to erase are shown at the left for erasures of from one to five d-c pulses. Beneath each pulse applied is tabulated the residual flux, measured following that pulse. To the right of each schedule is plotted the magnetization as a function of the peak value of the a-c field to which the sample was subjected after erase.

Leaving aside the bottom curve, the curves form a family which illustrate the basic phenomenon. The top curve is a single-pulse d-c erase, which leaves a large magnetization which is hardly reduced by ordinary bias fields (of the order of 200-300 oersteds). The next two curves show two-pulse erasures with obvious improvement over the single-pulse erase. The first of these achieves zero average magnetization after erase, but the a-c field corresponding to the bias increases the magnetization very markedly. The second of these two shows an erase so designed as to leave zero magnetization after biasing with 280 oersteds. Various three, four, and five pulse erasures follow and it may be seen that:

1. Gross, or average, magnetization may be made no more complete but enormously more stable by increasing the number of pulses in the erase schedule.
2. One can alter the shape of the curves deliberately and predictably by choosing the sequence of pulses. As the a-c field is increased following a multi-pulse d-c erase, the material retraces its past history, "forgetting" first the last or weakest d-c pulse, then the next to last, and so on. This shows the extent to which the erase schedule failed to demagnetize the material and achieved instead merely a zero average flux, as described above.

Experimental D-C Pulse Head

The schedules shown are, of course, not all the data taken. For a given result the magnitude of pulses had to be chosen with great care. In practice, one of the fields from commercial magnets could not be controlled within the fraction of one per cent necessary to get the results shown for a five pulse erasure. A magnetization of 10 to 25 units on the vertical scale of the curves would be a good practical result.

Another factor of great importance in actual tape erasure is that fields applied to the tape are not uniform through the tape. Thus for a given array of magnets the layers of tape next to the magnets might experience fields proportional to but stronger than the fields acting on a more remote layer of the tape. The last curve of *Fig* 2 illustrates the inadequacy of a five-pulse schedule which is similar to the one directly above it but with all values increased by the factor 1.86. It does a poor job, and if one must contend with such variations, either more pulses must be used or poorer results expected.

The schedules shown in *Fig.* 2 are also non-universal with respect to the magnetic properties of the tape. A schedule best for one tape will not be best for another, although it will be better than a single pulse. Universality is necessarily tied in with a very large number of pulses such as approximate true demagnetization.

In actual tape recording, not only are there gradients in the erasing fields but also in the biasing field. For conventional ring-type record heads, the bias-field gradient across the thickness of the tape may introduce a factor from two to five between the fields near the front and back of the tape. Thus

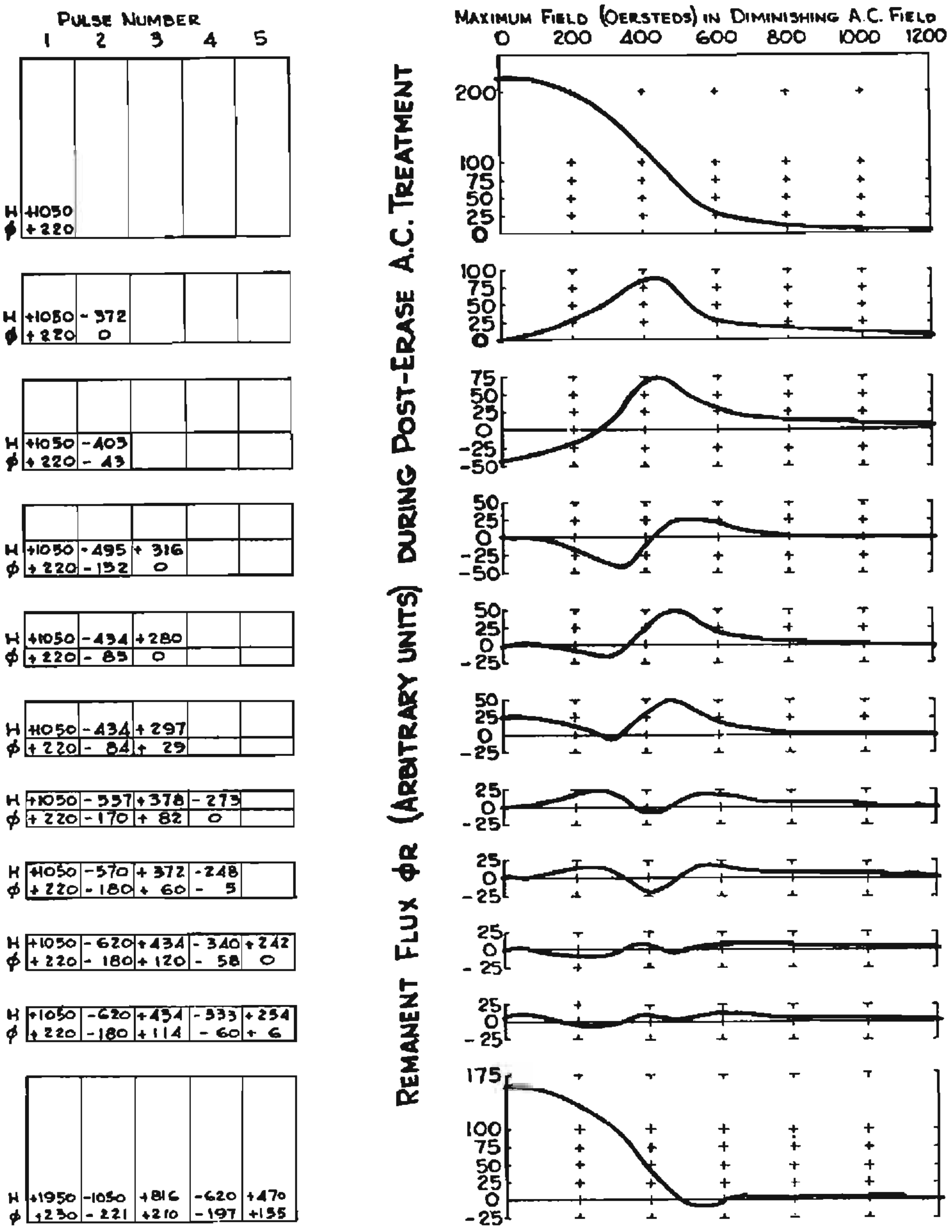

Fig. 2. Chart showing characteristics of erasure by various schedules of d-c pulses, using rods as the magnetic medium.

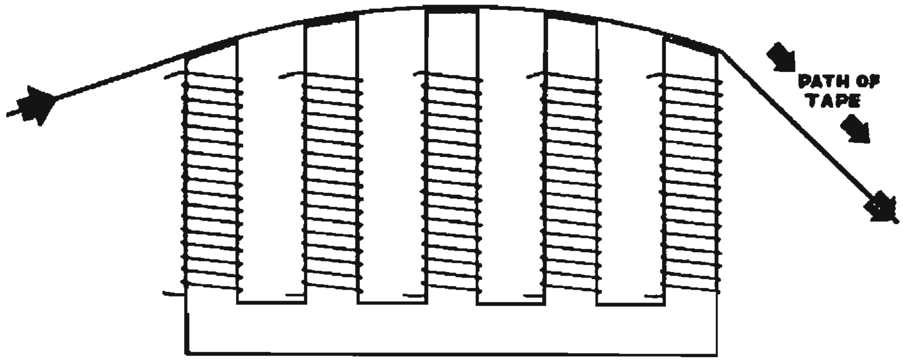

Fig. 3. Experimental d-c erase head providing five separate poles which may be magnetized to any desired degree and polarity.

each particle has its own post-erase history and the curves of *Fig.* 2 are valuable chiefly in understanding the basic phenomena. However, it is reasonably safe to predict from them that a careful arrangement of three to five successively opposite permanent magnet fields will give substantially quieter erase (10 to 15 db) than a single magnet. The data also show the importance of adjusting the magnets by observing noise after the tape is subjected to the bias field.

Comparison of Results

To see to what extent the above phenomena are reproduced in actual d-c erase of recording tape, the erase head shown in *Fig.* 3 was constructed. The core is soft iron and the legs were wound with separate coils which could be energized with independently controlled d.c. This made possible the erasure of tape by a succession of various numbers of d-c pulses in a manner analogous to the experiments with rod and galvanometer. In this case the erase was followed by the usual record head employing a-c bias of variable amount in place of the a-c solenoid shaking field, and the residuals were measured in the form of noise at the playback head in place of the galvanometer measurements of magnetization.

When first tried, this head gave disappointing results. The use of two poles lowered the noise about 5 db below that for single-magnet erase, but no adjustment of any number of additional poles gave any substantial further improvement. This was traced to the large field gradients existing near the pole tips which had been machined with fairly sharp covers. To reduce the local field gradients, all pole faces were covered by a layer of "Scotch" cellophane tape (.003 inches), and the good performance described below was obtained. Perhaps still greater field uniformity would have yielded still better results. In any case, it indicated that a good practical design would not employ very sharp magnet edges next to the tape.

The results for various pulse erasures by this head are shown in *Fig.* 4. Tabulated on the left are pulse ampere-turns in successive legs of the special erase head, and on the right are graphed the noise levels for various bias currents in the record head following erase. For the tape and head used, a typical bias current would be .07 or .08 amps. The noise levels are plotted in db relative to the so-called d-c noise which results from a single saturating d-c pulse followed by no bias.

Practical Results

In comparing the graphs of *Fig.* 2 with those of *Fig.* 4 one must remember that the vertical scale of the former is linear whereas that of the latter is logarithmic. It may be seen that, as predicted, the best erasure conditions must be chosen with the bias operating, since the pulse schedules for best erase and best erase-plus-bias are not the same. It is also evident that the bias may act to increase the noise as well as to reduce it, in accordance with the basic data of *Fig.* 2. It is not possible to relate the number and strength of pulses to the shape of the curves so easily as in *Fig.* 2, presumably because of the gradients in erasing and biasing fields. The improvement in noise for multiple-pulse erase over that for single-pulse erase is about as would be predicted by the data of *Fig.* 2.

The improvement of the two-pulse erase over the one-pulse method is evident, and the addition of a third pulse offers significant further improvement. Beyond this point improvement is very slight, although more pulses could be used advantageously if a more gradient-free field were supplied or if it were desired to make the erase efficient for a wide variety of tapes.

In general, it would seem that a well designed multi-pulse d-c erase system

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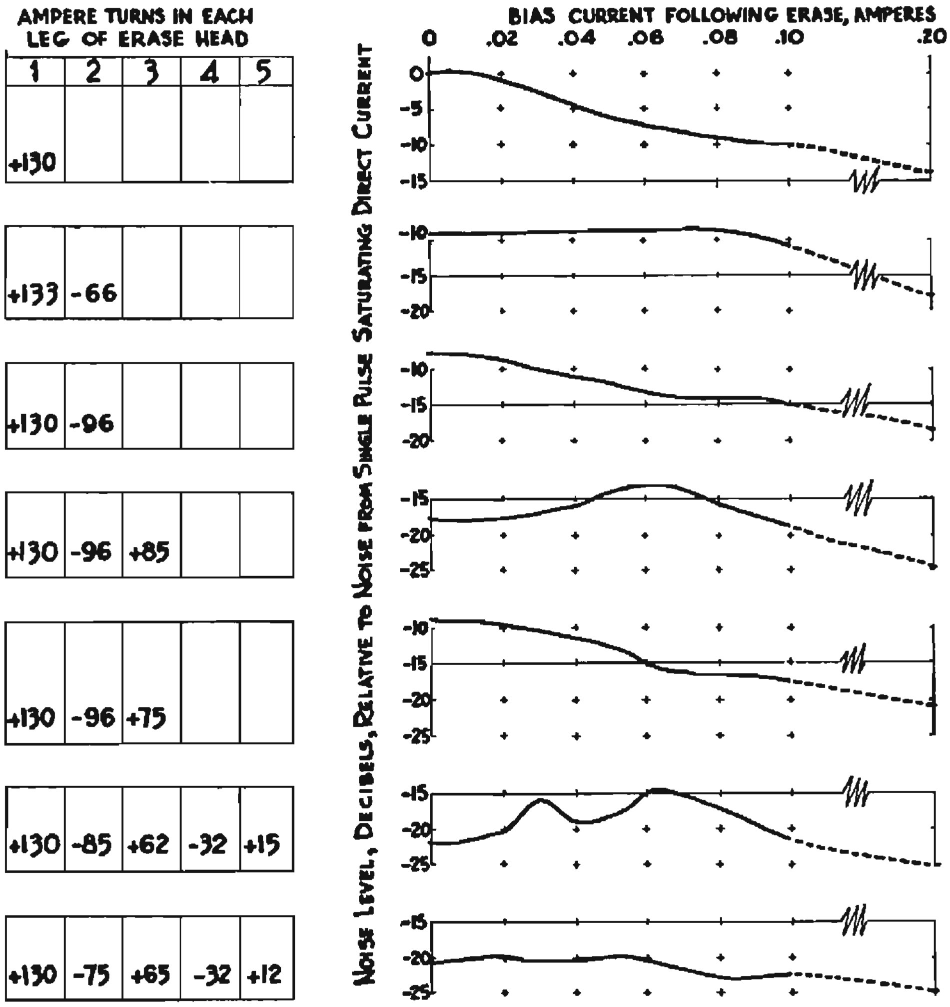

Fig. 4. Erasure characteristics for various schedules of d-c pulses applied to magnetic tapes, using noise measurements to indicate efficacy.

TAPE ERASURE

[*from page* 16]

will yield noise levels (after bias) from 10 to 15 db lower than is obtainable with a single saturating pulse. This represents a considerable sacrifice compared with what can be done with a really excellent a-c erase, although it may be only slightly poorer than a typical cheap a-c erase and bias system in which there may be bad wave form and/or d-c components. D-c erasure is, therefore, not to be considered for machines in which maximum quality is necessary; for the cases where it is to be used, the information given above may be of value in design and testing.

The author wishes to thank the management of the Minnesota Mining and Manufacturing Company for permission to publish the data contained in this paper, which was obtained in connection with research on "Scotch" Sound Recording Tape.

Magnetic Recording in Motion Pictures

M. RETTINGER*

PART I. The fundamental aspects of magnetic tape recording, particularly for motion pictures, including a description of magnetic recording, reproducing and erasing head construction, and a discussion of a.c. biasing, together with experimental results.

THE PURPOSE of this paper is to outline the present state of magnetic tape recording. While much of the following is applicable to wire recording, too, emphasis is given to magnetic tape—particularly plastic tape with a ferro-magnetic coating—since it has brought about a revival of this sound recording method. Wire recording has been known for over 50 years and is still employed extensively, but it was tape which, by its non-twisting character and easy means of splicing, has engendered an increased interest in magnetic transcriptions.

In the past, high-quality recording of sound was effected chiefly by optical means, as in the case of motion picture sound film recording, and to a lesser degree, by mechanical devices, as in wax disc recording (although lacquer disc recording has produced very notable quality, too). *Figure* 1 shows advantages and disadvantages for optical and magnetic recording.

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TABLE I

	Perpendicular	Transverse	Longitudinal
Advantages	Constant-aspect ratio[2]	Constant-aspect ratio	May be used with wide, coated tape
Disadvantages	Not suitable with coated tape	Requires narrow tape to avoid spreading of flux lines with resulting poor high frequency response	Aspect ratio decreases with frequency

In magnetic recording, the signal along the sound track takes the form of variations in the remanent magnetization of the magnetic medium. Three types of magnetic recording are recognized—longitudinal, perpendicular, and transverse. When the recording medium is tape, longitudinal recording exists when the magnetic force is parallel to the direction of motion of the medium; perpendicular recording occurs when the magnetic force is perpendicular to both the direction of motion of the tape and the face of the tape; transverse recording is produced when the magnetic force is perpendicular to the direction of motion of the medium and parallel to the face of the tape.[1] The three types of recording are illustrated in *Fig.* 2. When the recording medium is a round wire, transverse and perpendicular recording are equivalent. Table I shows the advantages and disadvantages of the various types.

	Optical Recording	Magnetic Recording
Advantages:	1. No direct contact between modulator and film. 2. Ease of duplication—direct contact printing. 3. Permits visual inspection of developed sound track.	1. Permits immediate monitoring of recorded sound-track. 2. No processing required. 3. Sound-tracks may be erased. 4. No noise reduction amplifier required. 5. Track has no definite overload point. 6. No optical system required. 7. No exposure lamp or power supply for it required. 8. Recorder unit need not be light-tight. 9. No light-tight film magazines required. 10. No light-tight film loading facilities required.
Disadvantages:	1. Requires developing before playback. 2. Track itself cannot be monitored immediately. 3. Medium must be handled in the dark. 4. Unexposed film has expiration date. 5. Recording apparatus more expensive.	1. Head contacts film—producing wear, clogging problems. 2. Sound track cannot be easily and accurately inspected.

Fig. 1. Compilation of advantages and disadvantages of both optical and magnetic recording methods.

Terminology

While a technical discussion of magnetic recording might well be prefaced by a definition of magnetic terms, *Fig.* 3 illustrates the more common designations. Currently, three systems of electromagnetic units are in use: the c.g.s., the nonrational m.k.s., and the rational m.k.s. Engineering texts, usually, employ the c.g.s.; physics books, including Prof. G. P. Harnwell's well-known "Principles of Electricity and Magnetism," use the "rational," while others employ the "non-rational." Table II gives conversion factors for the units of the three systems.

It should be noted that remanent and residual induction are the same only when, in a closed magnetic circuit (that is, in one without free poles), the mag-

[1] Suggestions have been advanced to reverse the meanings of transverse and perpendicular recordings as defined in the past and above. See N. M. Haynes, "Magnetic Tape and Head Alignment Nomenclature," AUDIO ENGINEERING, June, 1940.

[2] By aspect ratio is meant the ratio of length of magnet ($\lambda/2$) to width of magnet (width of sound track). The smaller this ratio is, the greater is the tendency for the individual magnets to demagnetize themselves. Constant aspect ratio is not necessarily an advantage unless the ratio is such that demagnetization is not serious. If the ratio is such that demagnetization exists at all frequencies, the output voltage at the low frequencies will also be down, and noise will become a greater problem.

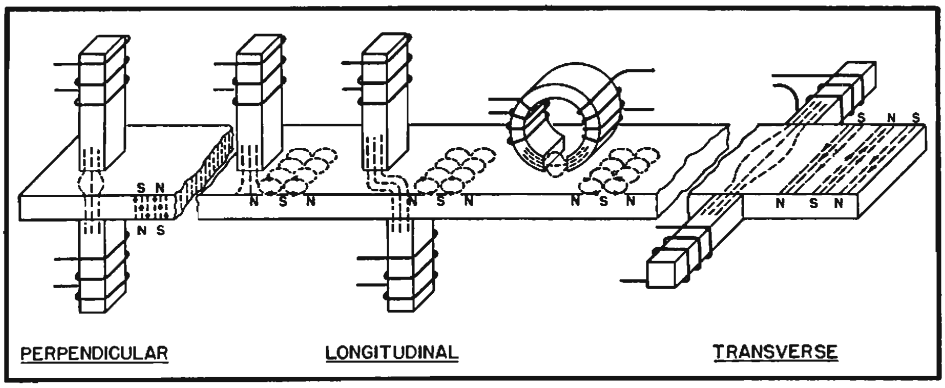


Fig. 2. Diagram to illustrate three types of magnetic recording.

netizing force is reduced to zero. When this is not the condition, as when the ferromagnetic core has an air-space, the flux per unit area remaining in the core when the external magnetizing force has been removed, is spoken of as the remanent induction, since it exists in the presence of a demagnetizing force. Thus, the sound track of a magnetic recording is characterized not by a variation of residual induction of the medium, but by a variation of remanent induction.

The flux in the iron placed within a coil may be considered to be made up of two fluxes—one set up by the circulating atomic currents in the iron and the other set up by the magnetizing coil current, acting on a vacuum. This can be expressed by the equation

$$B = \beta + \mu_0 H$$
$$\text{or } \beta = B - \mu_0 H$$

where B = flux density, gauss
μ_0 = permeability of vacuum, gauss/oersted
H = magnetizing force of magnetizing coil, oersteds
β = intrinsic flux density, or flux density caused by the currents in the iron, gauss

Figure 4, applicable for powdered iron media, shows a plot of β, and it is seen that the intrinsic flux density for high magnetizing forces becomes constant, and that the "intrinsic coercivity" is greater than the normal maximum demagnetizing force. Powder-coated tapes usually have a coercivity of from 100 to 500 oersteds and a remanence varying from 300 to 1000 gauss.

It appears natural in a description of magnetic recording to mention first the construction of the elements employed in the process and then to explain their functions. The following, therefore, contains first a description of conventional magnetic recording and reproducing heads, which is followed by a discussion of the recording and reproducing process, together with experimental results. It may also be said that magnetic recording is still in its infancy and that improvements in both its theory and practice are likely to occur.

Ring-Type Recording and Reproducing Heads

In magnetic recording and reproducing heads of the ring type, the magnetic material forms a quasi-toroidal enclosure with one or more air-gaps. The

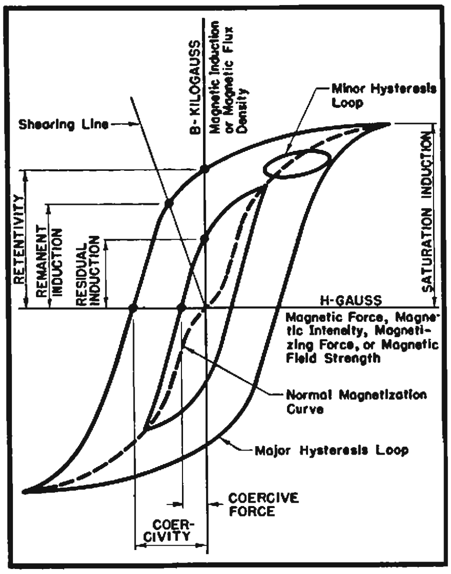


Fig. 3. Typical hysteresis curve showing designations commonly employed in magnetic recording literature.

magnetic medium bridges one of these gaps, commonly spoken of as the front gap, and contacts the structure on one side only. When a second gap is inserted in the core so as to divide it into two symmetrical halves, the second interstice may be termed the back gap. Each hiatus usually contains a nonmagnetic spacer—the "front" and "back" spacer—although some ring heads have butt joints. One or more coils may be wound around the core, and the entire head assembly is usually placed in a mumetal can to act as magnetic shield. It is the purpose of the following to discuss various features of such magnetic recording and reproducing heads.

The inductance of the head varies with the thickness of both the front and back gap spacers. *Figure* 5 shows the variation in inductance of a typical head. It is obvious that increasing the thickness of either spacer will lower the inductance. Curves of *Fig.* 5 were obtained by supplying constant current through the windings of the coil. It will be found that, in general, the inductance of the head varies somewhat with both frequency and current, as will the a.c. resistance and the "Q" of the coil. This is illustrated in *Fig.* 6. It is, of course, desirable to make these quantities as independent of these parameters as possible to avoid the necessity of equalizing networks in the circuit. The wire for the "recording" coils should also be thick enough to carry sufficient audio and bias current for saturating the recording medium without unduly heating the head.

Knowing the inductance of the toroidal head, it is possible to calculate the effective permeability of the construction. For the simple case of butt joints at the front and back gaps, the permeability at the appropriate maximum alternating flux density is approximately given by

$$\mu = \frac{Ll10^8}{.4\pi N^2 A}$$

where L = inductance of head (henries)
N = number of turns
A = cross-section of iron (not including the area of insulation between the laminations), cm²
l = length of magnetic circuit in iron, cm

Similarly, if the permeability of the ring is known, the inductance may be determined by

$$L = \frac{.4\pi N^2 \mu A}{l}\,10^{-8}$$

where the letters refer to the same quantities as before. *Figure* 7 gives both accurate and approximate flux and inductance formulae for toroids of circular and of rectangular limb cross-sections.

When the inductances of the two coils have been measured separately, or when the number of turns of each coil is

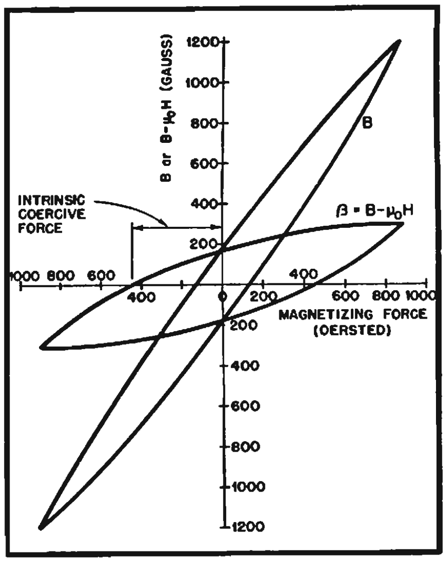


Fig. 4. Plot of curves for B and β.

TABLE II

C.G.S. VALUES	NON-RATIONAL M.K.S. VALUES	RATIONAL M.K.S. VALUES
m.m.f. = 1 gilbert	= 10 pragilberts	$= \frac{10}{4\pi}$ ampere-turns
= .1 "	= 1 "	$= \frac{1}{4\pi}$ "
$= \frac{4\pi}{10}$ "	$= 4\pi$ "	= 1 "
R = 1 magnetic ohm	= 10^9 (n–r) m.k.s. units	$= \frac{10^9}{4\pi}$ (r) m.k.s. units
= 10^{-9} "	= 1 "	$= \frac{1}{4\pi}$ "
= $4\pi 10^{-9}$ "	$= 4\pi$ "	= 1 "
ϕ = 1 maxwell	= 10^{-8} webers	= 10^{-8} webers
= 10^8 "	= 1 "	= 1 "
H = 1 oersted	= 10^3 praoersteds	$= \frac{10^3}{4\pi}$ ampere-turns/meter
= 10^{-3} "	= 1 "	$= \frac{1}{4\pi}$ "
= $4\pi 10^{-3}$ "	$= 4\pi$ "	= 1 "
B = 1 gauss	= 10^{-4} webers/square meter	= 10^{-4} webers/square meter
= 10^4 "	= 1 "	= 1 "
μ = 1 gauss/oersted	= 10^{-7} webers/praoersted sq. meter	= $4\pi\, 10^{-7}$ webers/meter amp.-turn
= 10^7 "	= 1 "	$= 4\pi$ "
$= \frac{10^7}{4\pi}$ "	$= \frac{1}{4\pi}$	= 1 "

m.m.f. = magneto-motive force
R = reluctance
ϕ = flux
H = magnetizing force
B = induction
μ = permeability

known, and the coils are then placed in series, the total inductance is not proportional to the square of the combined number of turns, but must be determined separately. The total inductance is given by

$$L = L_1 + L_2 + 2M$$

where L_1 = inductance of one coil
L_2 = inductance of the other coil
M = mutual inductance

Magnetic recording heads of the ring-shaped core type are usually constructed with a low-impedance winding, and reproducing heads with a high-impedance winding. The reason for the low-impedance winding of the recording head is that the line to the head has a certain amount of capacitance, which represents a leakage path for the high-frequency bias current if such is used. If the recording head had a high-impedance winding, the line leakage would constitute a considerable loss. In the case of the reproducing head, where no bias frequency is employed, a high-impedance winding on the head is satisfactory.

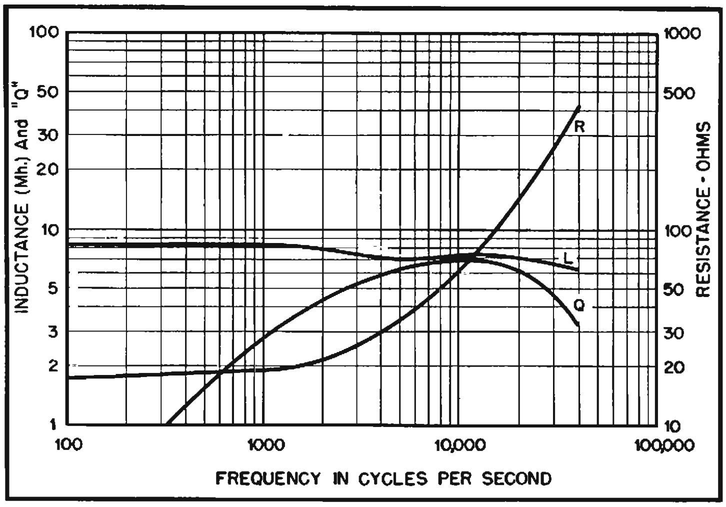


Fig. 6. Impedance characteristic of a typical recording head.

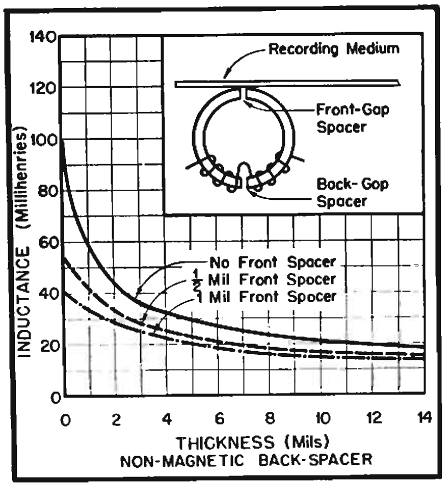


Fig. 5. Curves showing effect of gap dimensions on inductance of typical head.

Front Gap

Figure 8 shows the variation in flux distribution about the front gap of a ring-type magnetic recording head in the absence of a recording medium when

spacers of different thickness are inserted in the gap. The spacer serves the double purpose of maintaining the air gap parallel and to avoid the accumulation of ferrous dirt, which would change the performance characteristic of the head. The figures, obtained by maintaining constant voltage across the head, indicate that considerable leakage flux exists about the gap even when a butt joint is used. While a 1-mil spacer shows greater leakage flux than a half-mil spacer, a loss of high frequencies would result if the 1-mil head were used for reproducing. It has been determined[3] that the output of a reproducing head varies as

$$20 \log \frac{\sin(\pi d/\lambda)}{(\pi d/\lambda)}$$

where d = effective gap length (note that the gap length is actually the shortest dimension of the parallelepiped air-space)
λ = recorded wavelength

A minimum exists where $d/\lambda = 1$ (see *Fig.* 9). In the case where the medium travels with a velocity of 18 in. per second, a frequency of 15,000 cps would have a wavelength of 18/15000 = .0012 inches, so that it might be assumed that a gap length less than this magnitude would be sufficient to "resolve" such a wavelength. However, since the effective length of such slits may be from 10 to even 100 per cent greater than the actual gap length, a spacer of the order of .0005 inches or smaller is commonly employed when reproduction of 15,000 cps is desired.

Measurements have shown that the shapes of the curves of *Fig.* 8 remain substantially independent of frequency as long as core saturation is prevented.

The curves for *Fig.* 8 were obtained by moving the head past a single loop search coil of #46 Formex wire. The

[3] Wien Schott, "Einfluss der Schragstelling des Spaltes bei Intensitatschrift," *Zeitschrift der Tech. Physik,* V. 17 (1936) p. 275.

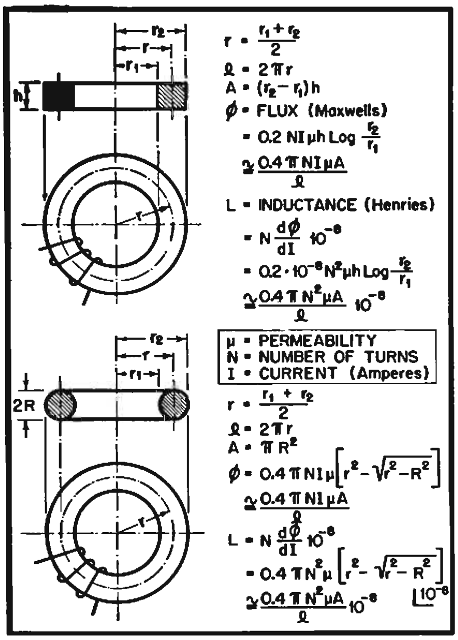


Fig. 7. Formulas for inductance and flux of toroids of circular and rectangular sections.

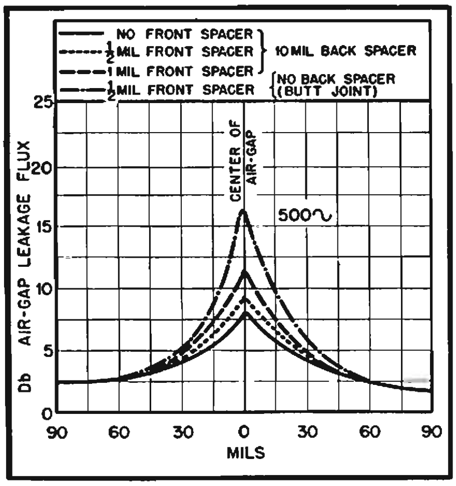


Fig. 8. Effect of gap width on flux distribution about the front gap of a ring head.

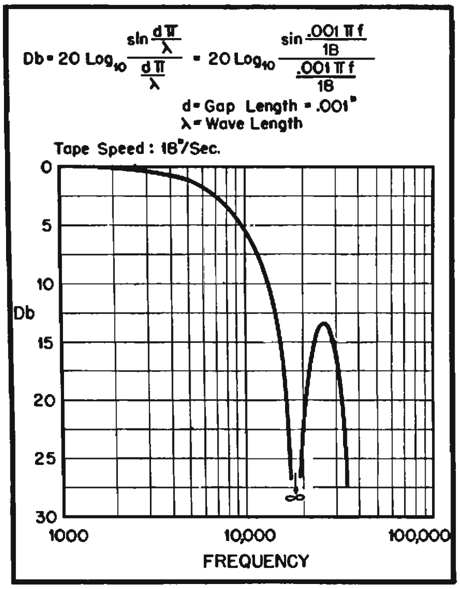


Fig. 9. Typical output curve for 18-in./sec. tape speed and gap length of .001 in., with formulas for calculating loss due to gap.

head was fastened to a brass rack, and the pinion for the rack was driven by the curve recorder; in this manner, a measure of the distance traversed by the head was secured on the curve paper.

It should be noted again that the curves were obtained without a magnetic medium over the gap. Hence, they are useful chiefly for comparing different heads, and do not necessarily show the actual leakage flux distribution when the head is in operation—that is, when magnetic tape is lying on the head or when it is passing over it. No experimental method has come to the writer's attention which will measure the flux distribution in the presence of a magnetic medium over the gap. It is believed that the flux distribution will be somewhat altered in the presence of a tape over the gap.

It may be observed that the curves of *Fig.* 8 are practically symmetrical. Clark and Merrill[4] have noted that when this is not the case—as when one side of the "branch" is much steeper than the other—it makes a considerable amount of difference in the resulting frequency response in which direction the tape is passed over the head. In particular, when the "entering" half of the branch is steeper than the "leaving" half, the high-frequency response is less (due to an erasing action of the trailing edge) than when the tape is passed in the other direction, where the "leaving" edge of the branch is steep.

In a ring-shaped recording head such as shown in *Fig.* 5, the inductance is approximately given by:

$$L_1 = \frac{KN_1^2}{r_1 + R_1} \text{ henries}$$

[4] D. L. Clark and L. L. Merrill, "Field Measurements on Magnetic Recording Heads," *Proc. I. R. E.*, December, 1947.

where r_1 = reluctance of front gap

$= \frac{l}{wd}$

l = gap length, cm
d = gap width, cm
w = gap depth, cm
R = reluctance of back-gap

$= \frac{l_o}{w_o d_o}$

l_o = gap length, cm
w_o = gap width, cm
d_o = gap depth, cm
N = number of turns
K = constant

The above equation is true as long as the reluctance of the core material is small compared to the reluctances of the gaps.

Likewise, the inductance of another similar head is given by:

$$L_2 = \frac{KN_2^2}{r_2 + R_2}$$

or

$$\frac{L_1}{L_2} = \left(\frac{N_1}{N_2}\right)^2 \frac{r_2 + R_2}{r_1 + R_1} \ldots\ldots (1)$$

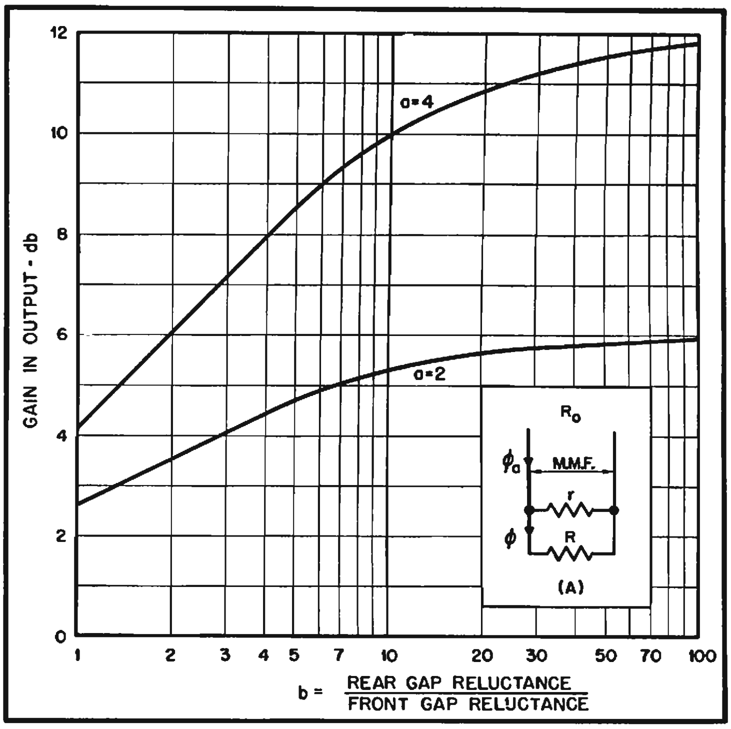
The action of a reproduce head may be explained by means of *Fig.* 10A. Assuming the tape to be a constant flux generator providing a flux ϕ_o, we have across the tape head a magneto-motive force equal to:

$$M.M.F. = \phi_o R_o$$
$$= \phi' \frac{rR}{r + R}$$
$$= \phi R$$
$$\phi = \phi_o \frac{r}{r + R}$$

where ϕ = flux through the core
r = reluctance of front gap
R = reluctance of rear gap plus core reluctance

[*Continued on page 32*]

Fig. 10. (A, insert) Equivalent circuit for magnetic tape as a generator acting upon head gaps. (B, graph) Effect of varying front and rear gap reluctances.



MAGNETIC RECORDING

[*from page 12*]

The voltage generated in the coil of head No. 1 will be:

$$E_1 = \frac{d\phi}{dt} = \frac{K'r_1}{r_1 + R_1}$$

The voltage generated in the coil of another, similar head No. 2 will be:

$$E_2 = \frac{K'r_2}{r_2 + R_2}$$

or

$$\frac{E_1}{E_2} = \frac{r_1\,(r_2 + R_2)}{r_2\,(r_1 + R_1)}$$

or

$$db = 20 \log \frac{E_1}{E_2} = 20 \log \frac{r_1\,(r_2 + R_2)}{r_2\,(r_1 + R_1)} \ldots (2)$$

Assuming two reproduce heads, of the same inductance, but with unequal number of turns, so that equation (*1*) can be written as:

$$\left(\frac{N_2}{N_1}\right)^2 = \frac{r_2 + R_2}{r_1 + R_1}$$

then equation (*2*) reduces to:

$$db = 20 \log \frac{r_1}{r_2}\left(\frac{N_2}{N_1}\right)^2$$

Also, if we now make the following substitution:

$$r_1 = ar_2$$
$$R_1 = R_2 = R$$
$$R = br_2$$

we can write equation (*2*)

$$db = 20 \log \frac{a\,(1 + b)}{a + b}$$

It is seen from (B) of *Fig.* 10 that an increased reluctance at the front gap increases the output of a reproduce head; this increase is to some extent dependent on the reluctance of the back gap. In this curve the abscissa represents the factor by which the reluctance of the rear gap is larger than that of the front gap, and in which the ordinates show the db gain in output resulting from increasing the front gap reluctance by the factor "*a*". For example, a head having a rear gap reluctance equal to the front gap reluctance ($b = 1$) will provide a 2.6 db gain in output when its front gap reluctance is doubled ($a = 2$); a head having a rear gap reluctance which is ten times as great as the front gap reluctance ($b = 10$) will provide a 5.3 db gain in output when its front gap reluctance is doubled ($a = 2$).

In addition to the gain achieved when the reluctance of the front gap of a reproduce head is increased, another gain is secured by the additional number of turns of wire required to provide a reproduce head with a certain inductance. Increased reluctance for a reproduce head can ordinarily only be achieved by

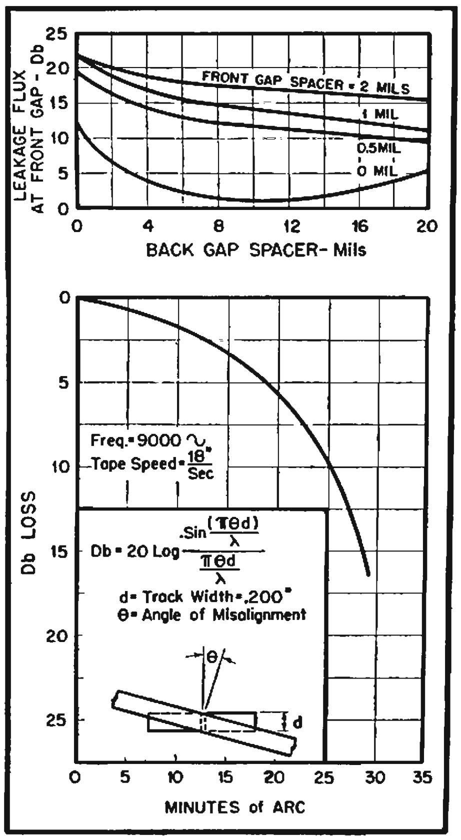


Fig. 11. (A, above) Leakage flux at front gap as a function of back-gap spacer. (B, below) Reduction in output as a function of the angle of misalignment.

reducing the depth of pole face, since the length of the air gap is fixed by frequency-response considerations. Still, there are limits also for the pole-face depth: a very small depth, say 5 mils, means that the life of the head is rather short, since the wear on the head by the tape will decrease the inductance quickly, with a corresponding change in the performance of the unit. In general, when the inductance of the head has been reduced by 20 per cent, due to wear, the head should be replaced.

Figure 11(A) shows the leakage flux at the front gap of a magnetic recording head as a function of the thickness of the back-gap spacer; the curves were obtained for different front-gap spacers, maintaining constant current through the head for all measurements. It is seen that the leakage flux decreases with increasing thickness of back-gap spacer, and that it increases with increasing thickness of frontal spacer. The head with the frontal butt joint represents, within the limits considered, an anomaly in that the leakage flux increases for very thick back spacers.

Another condition frequently observed, particularly with rectangular reproducing heads, concerns the increase of low-frequency reproduction when the head width equals a half wavelength of the recorded frequency. While the author has never measured as much variation in the low frequency response as was noted by J. S. Boyers,[5] an increase of 2 db at 100 cps can frequently be realized by permitting the tape to run over the entire width of the (rectangular) core of the head.

Spacers are usually made of beryllium copper or some other hard non-magnetic material to prevent burring of the pole tips. Following is a table giving the Brinell hardness for various materials:

TABLE III

Material	Brinell Hardness
Aluminum, annealed	23
Aluminum, work-hardened	44
Copper, annealed	30
Copper, work-hardened	105
Brass, annealed	60
Brass, work-hardened	150
Iron, annealed	67
Iron, work-hardened	220
18-8 Stainless Steel, annealed	135–185
18-8 Stainless Steel, work-hardened	180–330
Phosphor Bronze, annealed	73
Phosphor Bronze, work-hardened	234
Beryllium Copper, annealed	125
Beryllium Copper, heat-treated	300–350

[5] J. S. Boyers, "Factors Affecting Frequency Response and Distortion in Magnetic Recording," Audio Engineering, May, 1948.

From the foregoing, it is seen that beryllium copper represents an ideal material for a non-magnetic spacer: it is even harder than material used for the core.

To avoid high-frequency losses, it is important that the gap of the reproducing head, in respect to the sound-track, be in the same position as the gap of the recording head was when the recording was made. Of course, if the same head is used for both recording and reproducing, no misalignment losses are suffered. *Figure* 11 shows the db reduction in output as a function of θ, the angle of misalignment, when the tape speed is 18 in. per second and the sound-track width is .200 in. The figure also gives the general formula for calculating the db loss for any wavelength, λ, and angle of misalignment, θ.

Back Gap

A back gap is frequently introduced in ring-type magnetic recording and reproducing heads to reduce d.c. magnetization with a consequent lowering of the noise produced by such magneti-

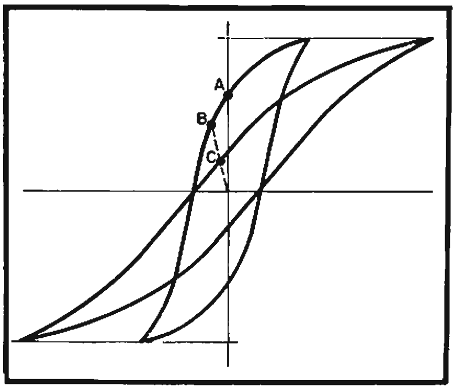


Fig. 12. Sheared hysteresis curve.

zation. This is effected by "shearing" the hysteresis curve, as shown in *Fig.* 12. When the magnetizing force in a closed ferrous ring is reduced to zero, the ring will have a residual induction, as indicated by the point *A* in the figure. When free poles exist in the ring, as in the case of a very fine air gap in the core, and the magnetomotive force is removed, the ring will have a remanent induction *B*. When a large back gap is inserted in the toroid, however, a "shearing" of the hysteresis curve is effected as shown in the figure, and the remanent induction *C* becomes rather small.

Cores are usually built up of laminations to reduce eddy-current losses. A coil having eddy-current, hysteresis, and copper losses may be represented as shown in *Fig.* 13. While the eddy-current loss increases with the square of the lamination thickness, it should be noted that very thin laminations make for a poor "stacking factor." In the case where a 200-mil thick core is built up of fifty 3-mil laminations, the insulation between laminations comes to 50

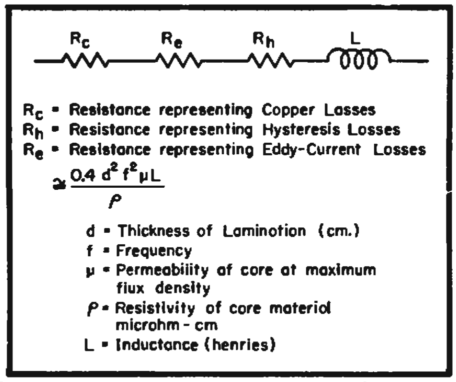


Fig. 13. Equivalent circuit of typical head to represent losses.

mils (allowing a one-mil insulating layer between laminations) or 25 per cent of the core thickness. When 6-mil laminations are used, it can easily be calculated that, for the same thickness of insulation between layers, the insulation comes to only 14 per cent of the core thickness.

Erasing Heads

Erasing heads employing supersonic current for "wiping" are similar in construction to recording heads, but have a much longer and somewhat wider front gap. The gap length may be as much as 0.020 in., and the width is usually 10 per cent greater than the width of the sound-track. As the tape moves over the front gap, it passes through an alternating magnetic field whose peak value is great enough to saturate the medium and which decays so gradually as to reduce the magnetism on the tape to zero. The number of decreasing field reversals required to "erase" tape magnetism is chiefly a function of head construction. The peak value of the erase field is usually equal to several times the coercivity of the medium.

The laminations for an erasing head usually consist of silicon steel, and not of mumetal, since the former material saturates at approximately 20,000 gauss, compared to 8000 gauss for mumetal. To avoid undue heating of the non-magnetic front spacer, this wedge is frequently made of a non-metallic material, such as plastic or mica. It has also been suggested to employ head shape such as shown in *Fig.* 14 to effect a more gradual reduction in the leakage flux distribution on the side where the magnetic medium leaves the head.

(To be Concluded)

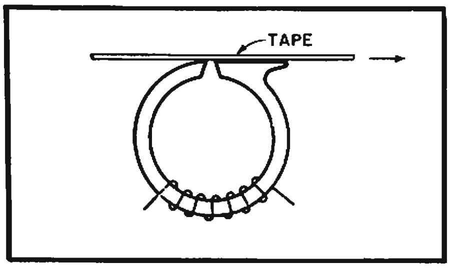


Fig. 14. Suggested head shape to improve leakage flux distribution.

Magnetic Recording in Motion Pictures

M. RETTINGER*

PART II. The fundamental aspects of magnetic tape recording, particularly for motion pictures, including a description of magnetic recording, reproducing and erasing head construction, and a discussion of a.c. biasing, together with experimental results.

IN ORDER TO discuss the frequency response of a magnetic recording, it appears desirable to assume constant-current input to the recording head. In practice, constant-current input is approximately accomplished by either connecting the head to a high-impedance source, such as a pentode, or else by placing a high resistance in series with the head. While a pentode is the most economical generator for the purpose, the harmonic distortion from such a source is considerably higher than from a triode. A series resistance, of course, incurs a power loss, which may be as much as 20 db or more at 1000 cps. To illustrate the condition, consider a recording head of 8-mh inductance. On the assumption that its inductance is independent of frequency and its resistance small compared to its reactance within the frequency range considered, the impedance at 100 cps will be 5 ohms and at 10,000 cps it will be 500 ohms. A 5000-ohm resistance in series with the head will provide substantially constant current to the head when the combination is connected to a 500-ohm constant-voltage generator.

Neglecting for the moment demagnetization and gap effects, constant sinusoidal current through the recording head should induce, in the recording medium, a constant remanent flux given by:

$$\phi = \phi_{max} \sin \omega t.$$

During reproduction, the open-circuited induced voltage from the reproducing head having a coil of N turns—again neglecting some factors to be discussed later—will be proportional to the rate of change of flux, as given by:

$$E = -N\frac{d\phi}{dt}10^{-8} = -N\,\phi_{max}\,\omega 10^{-8} \cos \omega t$$

$$= 4.44\, f\, \phi_{max}\, 10^{-8}\, v, \ (Eff.\ Value)$$

When the recording medium is moving past the reproducing head at a speed different from that at which it was recorded, a change in frequency occurs. The new frequency is given by

$$f_1 = f_0 \frac{V_1}{V_0}$$

where V_0 = speed of medium employed in recording
V_1 = speed of medium employed in reproduction
f_0 = recorded frequency

The change in output level is given by

$$db = 20 \log \frac{V_1}{V_0}$$

Thus, as the tape speed is doubled in reproduction, the output level will increase by 6 db. That is, the output level of what was formerly f_0 will be 6 db greater at the new reproduced frequency $2f_0$. It may be noted that the output voltage from the reproducing head at low medium-high frequencies is independent of the velocity as long as the signal is reproduced at the same velocity with which it was recorded.

Demagnetization (the reduction in remanent flux on the recording medium as the wave length is decreased) is a function of the geometry of the recorded flux pattern and of the coercive force of the recording medium. The greater the coercive force, remanence remaining constant, the less will the little magnets on the tape be able to demagnetize themselves. As far as the high frequencies are concerned, the effect of increasing the coercive force, thus, is similar to increasing the speed of the recording medium; that is, the output at the high

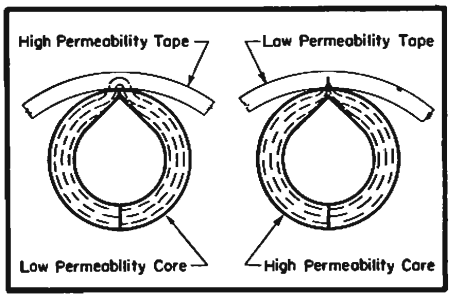


Fig. 15. Distribution of flux for heads and tapes of differing permeabilities.

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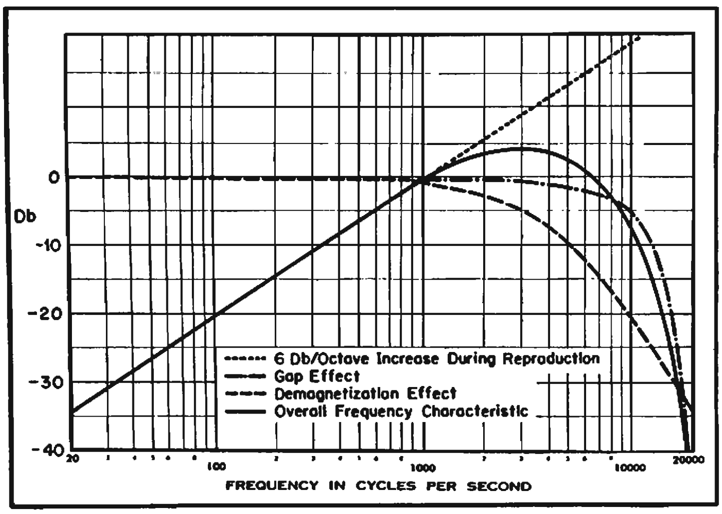

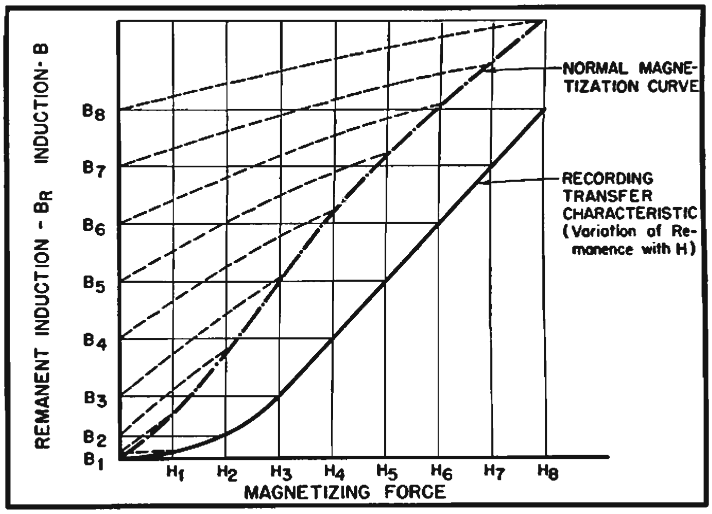

Fig. 16. (left) Factors entering into the development of output curve for reproducing head. Fig. 17. (right) Comparison of remanent induction for recordings made with and without high-frequency bias.

frequencies is increased while that of the low frequencies remains the same. Unfortunately, the larger bias current required for the high-coercive force media in turn effects a reduction in high-frequency output from the tape so that the net result, in practice, may be inconsequential. Increasing the remanence, coercive force remaining constant, effects a greater output level for all but the very high frequencies. A decrease in output level at the high frequencies results when a high-permeability medium is used, as illustrated in *Fig.* 15. The leakage flux lines from the air gap of the recording head, seeking the path of least reluctance, lose their peaky distribution character and become bulgy, the more so the smaller the permeability of the core relative to that of the medium.

During playback, there is a slight recovery in induction at the shorter wavelengths as the recording medium passes over the reproducing head, since the poles on the medium become neutralized by the magnetic core of the head. This neutralization exists only as long as the poles on the tape are in contact with the head, after which they return to their

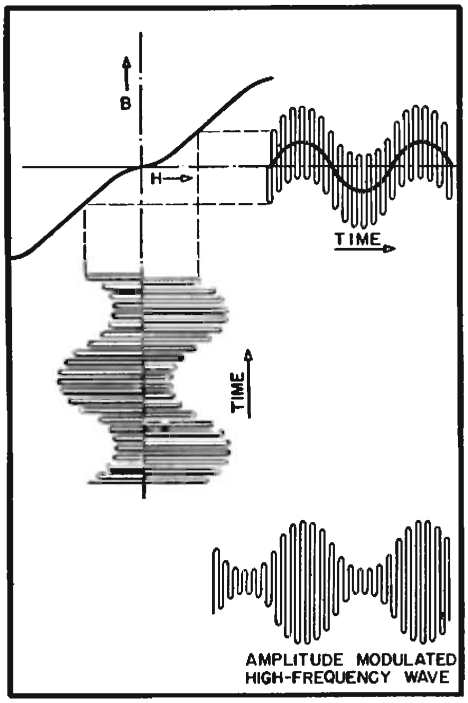

Fig. 18. Input/output curve for tape recording when high-frequency bias is used.

original state of magnetization after leaving the head. The head thus acts somewhat as a "keeper."

The dotted line of *Fig.* 16 shows the 6 db per octave increase in the output of a reproducing head when the recording head is energized with constant current, demagnetization and gap effects being neglected, The dot-dash line of *Fig.* 16 represents the high-frequency attenuation due to the gap effect (discussed in Part 1). The dashed line shows for a particular type of recording medium the effect due to the demagnetizing forces. The solid line of *Fig.* 16 gives the theoretical frequency response of a magnetic recording when gap and demagnetization effects are considered.

A flat frequency-response characteristic is provided for the over-all system when the frequency response of the reproducing amplifier is made equal to the inverse of the solid curve of *Fig.* 16. Sometimes this equalization is "split," with part of the high-frequency compensation introduced in the recording amplifier.

What happens when a magnetic re-

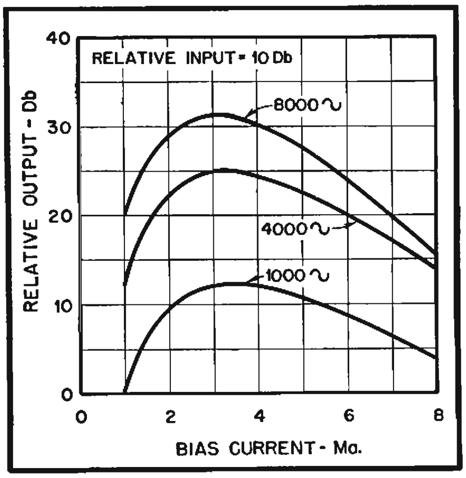

Fig. 19. Effect of bias current on output level for different frequencies.

cording is made on a magnetically neutral tape in the absence of a bias may be illustrated in *Fig.* 17. The dot-dash line shows the normal magnetization curve of the recording medium, with B_1, B_2, B_3, etc., indicating the remanent induction values on the medium after it has passed the recording gap. The solid line represents this remanent induction as a function of the magnetic field. This curve is known as the recording transfer characteristic for low and medium-high frequencies; the non-linearity of this curve about the origin gives rise to a marked distortion when no bias is used.

The addition of a supersonic bias causes the recording characteristic to be linear about the origin, as well as symmetrical in the first and third quadrant, thus preventing the production of even harmonic distortion. According to Toomin and Wildfeuer[1], when the bias current is such that the magnetomotive force which it produces approaches the coercivity of the medium, the added signal current produces a shifting of the minor (bias) hysteresis loops vertically inside the major loop in such a manner that the remanent (signal) induction on the medium is proportional to the distances on the straight portions of the

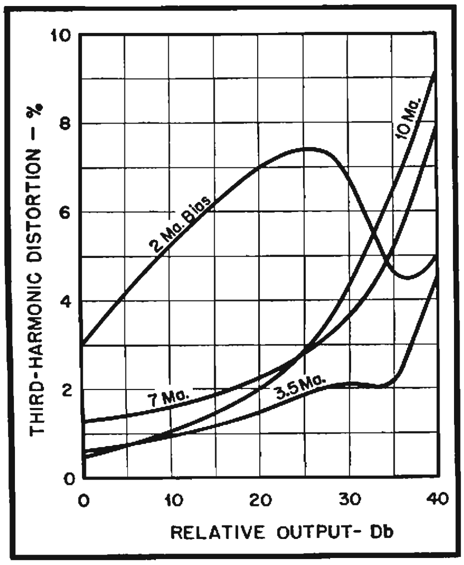

Fig. 20. Distortion/output curves for varying amounts of bias current.

major loop along which the minor loops are shifted by the signal magnetomotive force. Thus, a quarter cycle of a 100-cps signal current superimposed on a 50,000-cps bias current "envelops" 125 complete alterations of the bias current; every minor hysteresis loop produced by these 125 alterations of the magnetomotive force is progressively displaced within the major loop, the tips or peaks of the minor loops sliding with uniform velocity along a branch of the major loop. It should be noted, however, that the symmetrically alternating bias current alone produces in the recording medium a symmetrically cyclically magnetized condition in which the mean values of both induction and magnetizing force are zero. The bias current thus tends to keep the recording medium in a magnetically neutral state when no signal is recorded, with a consequent reduction of background noise.

According to Holmes and Clark[2], the action of the supersonic bias can be explained in terms of input-output curves similar to those used for radio tubes, when the input of the bias and audio field is plotted against time on the vertical axis and the output remanent tape induction is plotted against time on the horizontal axis. Such a curve is shown in *Fig.* 18. The solid signal curve on the horizontal axis is the resulting remanent induction on the tape after the considered portion of the tape has left the air-gap and demagnetization forces have taken their effect. It should be noted again that the signal current is merely superimposed on the bias current; it is not modulating the bias frequency in the manner an audio wave modulates a radio frequency carrier, as shown in the lower right-hand corner of the figure.

[1] H. Toomin and D. Wildfeuer, "The Mechanism of Supersonic Frequencies as Applied to Magnetic Recording," *Proc. I.R.E.*, November, 1944.

[2] L. C. Holmes and D. L. Clark, "Supersonic Bias for Magnetic Recording," *Electronics*, July, 1935.

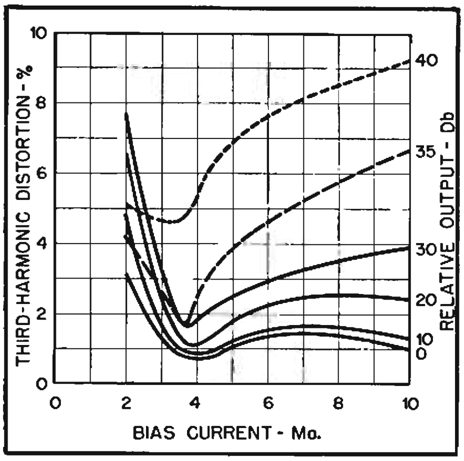

Fig. 21. Distortion/bias current curves for differing output levels.

Experimental Results

Figure 19 shows the effect of bias current on recording medium output level for 1000, 4000, and 8000 cps. It is seen that this level rises at first rapidly and then decreases more slowly with increasing bias current. This decrease is somewhat more pronounced for the higher frequencies and may be ascribed to an erasing action on part of the bias flux in the recording head air-gap. The curves were obtained by supplying constant current to a particular recording head and employing a reproducing amplifier with a flat frequency response; also, the same head was used both for recording and reproducing. It may be noted that another kind of head may not produce peak output at the same bias current for the three frequencies shown in *Fig.* 19.

When harmonic distortion is plotted against bias current, we find that the bias current I_m which gives maximum output from the tape does not, with some exceptions, produce minimum distortion, but that a bias current either slightly larger or smaller than I_m is more suitable in this respect. This is shown in *Figs.* 20 and 21. Whether to use a somewhat larger or smaller bias current—assuming each will provide the same low amount of distortion—will depend on the resulting frequency response: if the larger bias current will reduce the high-frequency response excessively, the lower bias current should be used—unless this lower bias reduces the over-all output excessively. For this reason, it is generally desirable to employ a type of tape which shows a broad maximum when output is plotted as a function of bias current. The distortion measurements were made with a reproducing amplifier whose frequency response above 400 cps decreased 6 db per octave, as it would most frequently under normal operating conditions; with a reproducing amplifier having a flat frequency response, the third-harmonic distortion percentage would, of course, be greater than the amount shown. Third-harmonic distortion measurements are frequently preferred over total distortion measurements, since the latter may contain noise components which would obscure the effects produced by the non-linearity of the magnetization curve of the recording medium.

Figure 20 is of chief interest to the technician operating a magnetic recorder, since it enables him to select the bias which gives greatest output with least distortion. By drawing a horizontal line corresponding to a chosen distortion, he can determine the bias which gives maximum output without exceeding the preselected value of distortion. It should be noted that the curves pertain to a particular type of tape; another type may produce greater output at a similarly low preselected value of distortion, but, at the same time, may require a higher recording level. This latter requirement is usually not a serious consideration, however, since the necessary audio wattage, at any event, is not very large—at least as far as commonly available recording amplifier output capacities are concerned. This is true only as long as the larger recording level does not increase the distortion from the head due to higher flux densities in the recording head core.

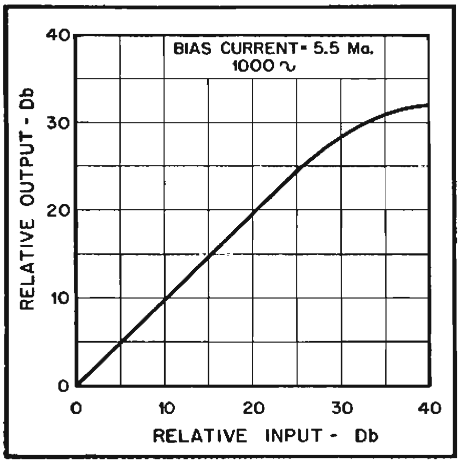

Fig. 22. Typical overload characteristic.

Figure 21 may be used to determine if minimum distortion occurs when the bias is adjusted for maximum output. It may be noted here that bias values on curves of this type are frequently expressed in "ampere turns," instead of in amperes, in an attempt to provide more general information. However, it has been found that even when the same type of recording head is used on another recorder, somewhat different values of bias are required to achieve identical results, due to capacitance effects in the recording head and associated wiring.

Figure 22 shows the overload characteristic of a magnetic film, and it is seen that there is no sharp break in the curve, as would be the case for variable-area recording when "overshooting" takes place.

Figure 23(A) shows the effect of insufficient contact between film and reproducing head, and it is seen that the resulting loss for a 1-mil air-space between film and reproducing head is far more pronounced for the high than the low frequencies. This can be explained by a consideration of the "elementary" magnets which make up the sound-track, as shown in the insert. It is seen that the flux lines for the longer or low-frequency magnets extend much farther into space than do the flux lines for the shorter or high-frequency magnets. The number of lines, indicative of the remanent induction, is the same for each type of magnet, the lines for the shorter magnets being crowded more closely to the "dipole."

Noise

Two types of signal-to-noise ratio are of interest in magnetic recording. One, conveniently expressed as signal-to-background-noise, gives the ratio of the maximum (440 or 1000 cps) signal that can be recorded and reproduced with a limited amount of distortion (say, 2 per cent) to the no-signal background noise generated in the reproducing head when the frequency response of the playback amplifier is adjusted to provide a flat response for the entire magnetic recording system. The second type of signal-to-

[*Continued on page 42*]

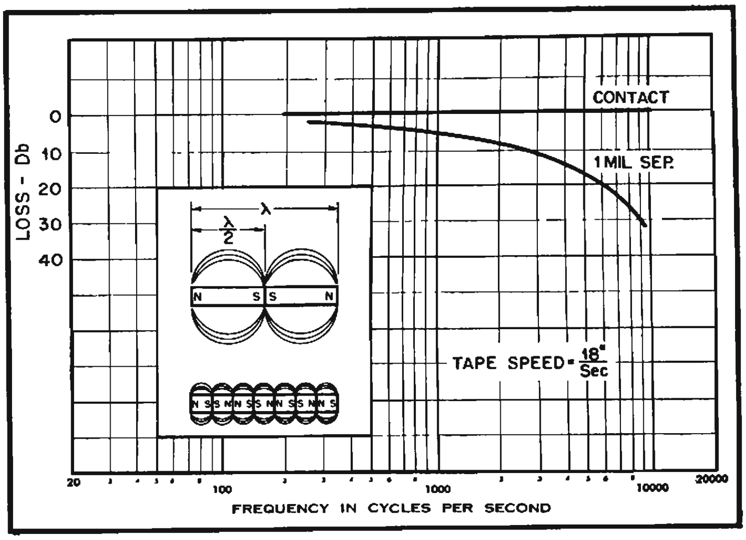

Fig. 23. Curves showing response due to poor contact between head and tape. (Insert) Enlarged section of coating to show magnetic fields.

MAGNETIC RECORDING

[*from page 20*]

noise ratio, for which no term has as yet been coined, concerns the ratio of the maximum signal to the noise-behind-the signal, the noise being a function of the signal strength. This latter type of noise (often termed modulation noise), while readily visible on an oscilloscope screen as a crest-distorted wave when a pure sine signal is recorded, is less easily measured, although sharply tuned band rejection filters have been employed for the purpose.[3,4]

Noise in magnetic recording may be due to one or a number of the following factors:

1. Non-uniformity in the ferric oxide dispersion of the magnetic layer on the film. This type of noise is usually spoken of as the "ground-noise" of the magnetic medium.
2. Incomplete erasure of a previous signal on the film.
3. Unsymmetrical wave shape of bias current.
4. D.c. magnetization of head, guide-roller, spool, etc. Anyone experienced with magnetic recording operations knows that considerable effort must sometimes be exerted to produce quiet recordings. The writer's tools—screwdriver, pliers, even the shears for cutting the magnetic film—are made of non-magnetic 18–8 stainless steel. It is also a good policy to demagnetize recorder parts frequently with a demagnetizing coil which may be energized directly from the common 110 v. 50 or 60 cps current supply.
5. "Print-through" or "echo" effect, when magnetization on one part of the film on a roll is transferred to an adjacent layer. This effect is more pronounced in the presence of a strong alternating field and at high temperatures.
6. Clicks and pops due to improper splicing.
7. Modulation noise. This type of noise is a function of the signal strength.
8. Variations in contact between medium and heads.
9. Variations in cross-section of soundtrack, scratches, etc.
10. The presence of foreign particles in the head gaps.

[3] S. J. Begun, *"Magnetic Recording,"* New York: Murray Hill Books, Inc., 1949; p. 214.

[4] S. J. Begun, "Measuring Procedures for Magnetic Recording," AUDIO ENGINEERING, April, 1949, p. 19.

APPENDIX I

It is sometimes desired to know what the diameter of a roll of film or tape will be when a certain length of film is wound on a core of a given diameter. Conversely, one may wish to know how many feet of film a roll contains when its diameter and the core diameter are known. The formulae are:

$$D = t + \sqrt{(d-t)^2 + \frac{4tl}{\pi}}$$

$$l = \frac{[(D-t)^2 - (d-t)^2]\pi}{4t}$$

where D = diameter of roll of film, inches

t = thickness of film, inches
d = diameter of core, inches
l = length of film, inches

For instance, when 400 feet of .006″ thick film are wound on a 2″ core, the diameter of the roll will be

$$D = .006 + \sqrt{(2-.006 + \frac{4 \times .006 \times 400 \times 12}{3.14}}$$
$$= 6.37 \text{ inches}$$

The Art of Tape Recording—1

JOEL TOLL*

The first of a series of articles on the practical aspects of magnetic tape recording and editing.

THREE BASIC METHODS of recording and reproducing sound are now in use—the mechanical or phonographic, the photographic or sound-on-film, and the magnetic. Of these, the last has now come into general use and promises to outstrip the other two in fidelity.

The first tidings of the advent of successful magnetic recording were recorded in the patent offices of Europe and America in 1899 when Valdemar Poulsen, a Danish inventor and engineer, applied for original patents on "methods of and apparatus effecting the storing up of speech or signals by magnetically influencing magnetizable bodies." Poulsen's unique machine attracted the attention of the scientific world at the Paris Exhibition in 1900. The Telegraphone, as his machine was called, could have many uses, but in his patent application, Poulsen noted only three: as a substitute and improvement on the phonograph; for recording and imparting communications over telephone wires with no human assistance; for telegraphic purposes, to record code messages at high speeds and play them back at much lower speeds so that the messages could easily be transcribed.

Within a few years several different types of Telegraphones had been developed, one using a solid steel disc, another steel tape, still another, wire. But no further development, except the addition of d.c. bias by Poulsen, was noted for about two whole decades. One possible reason for this stalemate in the development of magnetic recording may have rested in the lack of appropriate means for amplifying the reproduced sound with any great fidelity. Not until 1912 was DeForest's audion being manufactured in quantity and, before Dr. DeForest's invention of the three element tube, electronic amplification of sound was not possible.

Further improvements in the art of magnetic recording were developed in 1924, this time by Dr. Kurt Stille in Berlin. Influenced by Stille, the Ludwig Blattner Picture Corp. Ltd., of London, together with the Telegraphic Patent Syndikat, of Berlin, jointly announced a new system of magnetic recording on steel tape in 1929. Talking motion pictures, using steel tape for the accompanying sound, were exhibited and the "Blattnerphone," it was hoped, would speedily supplant the phonograph discs which were so difficult to synchronize with the pictures. The Blattner Corp. also noted that its machine could be used to record telephone conversations, a purpose for which Poulsen had, almost expressly, designed his Telegraphone.

In 1930, Dr. Kurt Stille brought out his "Dailygraph," a dictating machine which could record either on steel tape or steel wire. The steel tape used on the Dailygraph was a highly-developed steel alloy about one quarter of an inch in width and approximately three thousandths of an inch (.003) in thickness.

First Tape Broadcast—on BBC

That same year, 1930, the British Broadcasting Corporation began using the "Blattnerphone," and its first broadcast using steel tape occurred when King George V's New Year's Day address was re-broadcast from a Blattnerphone recording. This same year, a patent was issued to Dr. Pfleumer covering the use of paper or plastic tape coated with iron dust. This innovation was immediately taken up by the Allgemeine Electricitats-Gesellschaft (AEG) and I. G. Farben with the idea of producing a recorder for general use less expensive to operate.

Radio broadcasting companies both in Germany and in England were continuing to experiment with magnetic recording. By 1934, the BBC, in collaboration with Dr. Heising of Stille Laboratories, Ltd. had developed a recorder of good broadcast quality, incorporating frequency-correcting circuits.

The Marconi Company predicted a great future for tape recording in broadcasting and, by the pen of N. M. Rust, notes that "a continuous-band machine is being developed, which will have several interesting and useful applications for broadcasting. By its use in conjunction with special circuit arrangements, artificial echo effects can be produced simply, and with relatively little apparatus, as compared with methods ordinarily in use." By the end of 1936 the AEG in Germany had put a recorder on the market using a coated film tape and the Lorenz Company was marketing a machine using a special steel tape. Both machines were being used by the German Broadcasting Company, and several were made for portable use in automobiles and sound trucks.

Supersonic Bias

Except for refinements in quality of reproduction, in mechanical drives, and in the use of d.c. bias, the basic principles laid down by Poulsen in the 1900's still governed magnetic recording. The AEG and I. G. Farben, who had merged their magnetic recording activities in the Magnetophon Co., had, it is true, made use of film coated with iron dust or oxides as the recording medium, but Poulsen's direct current method of magnetizing was still in universal use. Because of this, there was always a great deal of noise in the reproduction of a magnetic record. In 1921 W. L. Carlson and G. W. Carpenter of the General Electric Co had designed a "Radio Telegraph System" in which recording was accomplished by means of high-frequency bias current, (the Carlson patent was granted in 1927) but it was not used in commercial equipment until 1941 when Dr. Braunmuhl and Dr. Weber rediscovered a.c. bias. They observed, while testing a Magnetophon, that the use of high-frequency magnetizing currents improved the signal to noise ratio. The magnetophon immediately began to use this "supersonic" method of recording and so did, and still do, all other magnetic recorders in general use.

After World War II, tape recording utilized the technical knowledge gained during the war. Copies of the German Magnetophon were produced in Europe and the United States. This country is now far in the lead in both quality and quantity of recorders manufactured. Refinements are being added and machines are being designed for special purposes.

In tracing the progress of magnetic recording since its invention at the beginning of the twentieth century, we have followed the development of the art from the first unique machine of Poulsen up to present day recorders capable of almost exact reproduction of sound. The increasing use of tape recorders—by the public, by industry as a whole, and by broadcasters—points to its great value in any situation where sound must be faithfully recorded and easily edited. With developments yet to come, there

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* *Columbia Broadcasting System, New York.*

TAPE RECORDING

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is reason to believe that tape will become the universal method for recording sound in every kind of application.

This new art opens up such a vista of possibilities in all directions that we should recognize it as another step forward in man's struggle to preserve his works and ideas; the struggle that began in civilization's dawn when the first of our literate ancestors made his own unique mark of ownership upon his first clay pot. The marking on a clay pot became more or less permanent when the clay dried; the magnetic marks on wire or tape can be visualized along the same general line of reasoning.

How A Tape Recorder Works

When Valdemar Poulsen, in 1899, applied for his patent on magnetic recording he used the following words:

> "To all whom it may concern:—
> Be it known that I, Valdemar Poulsen, have invented certain new and useful improvements in methods of and apparatus for effecting the storing up of speech or signals by magnetically influencing magnetizable bodies." etc.

From the above we can readily understand that in order to picture the action taking place when a magnetic record is being made, or when the same or a similar record is being played back through an amplifier and loudspeaker, it is necessary to know a few facts about "magnetism."

Everyone, at one time or another, has noticed how a magnet attracts a needle or other small bit of iron or steel. But, if you were to take two magnets and place their ends together, if the ends were both "north poles" or "south poles" they would push apart from each other. But if they were not alike in "polarity" they would attract each other. If two unlike magnetic poles were forcibly kept separated, a force would be exerted that would try to bring them together. The area in which this force is exerted is called the "magnetic field." Any magnetic material that is placed in or near this magnetic field will tend to become magnetized itself. The ease with which such a material (iron or steel, for example) becomes magnetized determines its *permeability*. (The property of permeability is one of the factors considered in the making of magnetic tape coatings.) *Figure* 1 shows a source of direct current electricity, a switch, a bar of iron, and a coil of wire surrounding the iron bar. When the switch is closed,

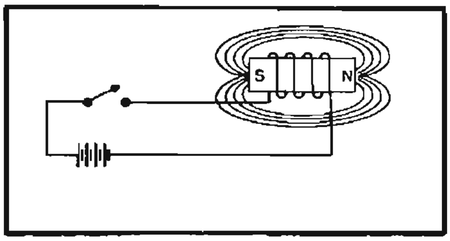

Fig. 1. Simple d.c. magnetizing circuit. The magnetic field is established when the switch is closed, and collapses when the switch is opened.

the iron becomes a magnet and a magnetic force is exerted from one pole to the other. If a strip of magnetic material passed across either pole, this force would magnetize the nearest end of the strip to an *opposite polarity*. That is, if the material were magnetized in passing a north pole, it would try to take on a south polarity and vice versa. It is possible to make a magnetic record on tape just by opening and closing the switch in *Fig* 1. Then, if the same coil were disconnected from the battery and switch and connected to the input of an amplifier, by moving the tape along the end of the iron bar you would hear a "click" every time the switch had previously been opened or closed.

A method similar to the above, in essence, was used by Poulsen in his Telegraphone. One coil of wire, through which direct current flowed, was the magnetizing coil. Another coil of wire was connected to the source of sound (telephone wire, for example) and, while the steel wire or tape was being magnetized, influenced the magnetism so that it varied in both magnitude and frequency in accordance with the original sound.

At the present time tape recording is effected in much the same manner as in the Telegraphone, with the exception that an alternating magnetizing current, commonly called bias current, is used. The frequency of the bias current, its strength, the width of the magnetic gap of the recording head—these and many other points are questions of the design of the tape-recorder and of the type of tape used.

The Physics of Recording

Since this article is designed for the guidance of the operating technician who is confronted with a new medium, nothing will be gained by repeating in detail the whole theory of magnetic recording. Adequate descriptions of the complete process will be found in AUDIO ENGI-

NEERING[1] and in Dr. S. J. Begun's book "Magnetic Recording." In essence magnetic recording is a method of "storing up speech or signals by magnetically influencing magnetizable bodies." The modern tape recorder uses iron oxide coated tape as the "magnetizable" body and uses a.c. bias instead of d.c. bias. The remaining differences are due to improvements in drives, amplifiers, heads, and so on.

The first unit that the tape encounters in its recording cycle is the "erase" head, as indicated in *Fig.* 2. All authorities

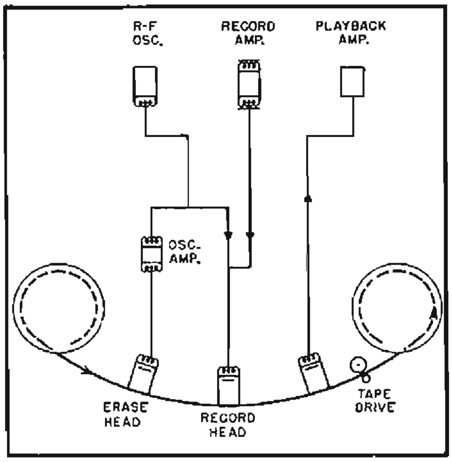

Fig. 2. Normal arrangement of erase, recording, and playback heads in a modern tape recorder.

agree that thorough demagnetization is necessary before tape can be properly recorded. The demagnetization is accomplished by subjecting the tape to a strong saturating supersonic field which is created by the erase head. Of course, erasing by means of a series of ordinary permanent magnets of alternate polarity is also effective in some degree. But at best the permanent magnets, besides wiping out any previously recorded magnetization in the tape, leave some slight noise on the tape. For this reason professional tape recorders make use of the supersonic erase method. The erase head coils, which are energized by an r.f. amplifier that follows a supersonic oscillator, must be sturdy enough to withstand possibly 5 watts of power. The erase field must be able to wipe out any previous signal completely, otherwise the cross talk and noise on the tape will be quite obnoxious. Indeed, recording engineers have found that tape that has been erased properly is often appreciably quieter than brand new tape.

The second head the tape passes is the recording head, *Fig.* 3. What occurs magnetically at this point determines how faithfully the tape will "store" sound. The tape has already been thoroughly demagnetized by the action of the erase head. The record head is also energized, in part, by the same kind of supersonic current that actuated the erase coils. This head is connected to the output of the same oscillator that feeds the erase r.f. amplifier, or to a separate oscillator at a different frequency. The strength of the magnetic field emanating from the poles of the recording head is only a fraction of that of the erase field. The audio signal that will be recorded on the tape mixes with the supersonic current at the record head.

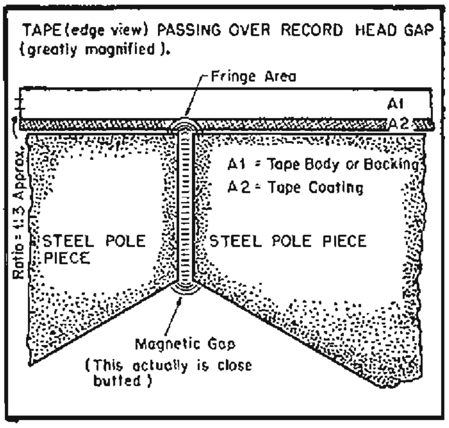

Fig. 3. Edge view of magnetic tape passing over recording-head gap, greatly enlarged.

Most recorders employ supersonic oscillators that operate frequencies from 30 kc to over 100 kc. The majority em-

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[1] W. W. Wetzel, *"Review of Present Status of Magnetic Recording Theory,"* Parts 1, 2, and 3; Nov. and Dec. 1947 and Jan. 1948.

ploy a frequency near 60 kc. The higher the frequency the less possibility of "beat frequencies" between bias and audio signal that, of course, introduce a kind of distortion. The rule followed in designing tape recorders requires the use of a supersonic frequency at least five times the top frequency of audio to be recorded. A still higher frequency would produce less beats but, unfortunately, r.f. losses increase with frequency and economy dictates a compromise.

If the wave shape of the erase field or of the bias field is not symmetrical there will be induction remaining in the tape that is proportional to the degree of non-symmetry. The record field is designed to be symmetrical and so, if it does not contain an audio component, should leave the tape "unmodulated" so to speak. But, if audio is mixed with the r.f. bias current in the head, it will cause differences in the peak values of flux in the recording field that correspond to the audio frequencies introduced. Thus there is remanent induction in the tape that varies exactly the same as the audio frequencies applied to the recording field.

The playing of a recorded tape again brings into action the magnetic forces we have already described. The actual record in the iron coating of the tape may be visualized in the form of very small magnetic structures. The maxima and minima of these infinitesimal magnets are spaced according to the frequency of the sound that was impressed on them by the audio component of the magnetizing current, and they will stay in that position until subjected to a strong magnetic field. When we play a recorded tape back, these tiny magnets, moving against the steel pole-piece of the playing-head, cause minute currents of electricity to rise and fall within the coil of wire wound around the steel pole-piece. After they have been sufficiently amplified, these currents produce the exact sound that was originally recorded on the tape.

From now on we will be concerned solely with techniques of recording, editing and mixing tape recordings, with the maintenance of the machines, and with ideas for making use of tape recordings in radio production.

Part II of this series will deal with recording "lay-outs" and methods and will contain a detailed discussion of the factors making for good magnetic tape recording.

The writer wishes to acknowledge the continued assistance of Howard A. Chinn and Price E. Fish for their assistance in the preparation and editing of this material.

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The Art of Tape Recording—2

JOEL TALL*

Continuing the discussion of magnetic recording, with especial attention to the practical aspects of operation under various field conditions.

Quality of recording depends primarily upon the degree of excellence of the facilities available and and secondarily upon the recording engineer's ability. In contrast to other media (film and disc) it is possible for the rankest tyro to get fairly good tape recordings if his recorder is in good condition. But the engineer whose job it is to get consistently excellent results will be interested in finding out exactly how to get them. Top-grade recording on tape requires a fundamental comprehension of the governing factors, whether they be mechanical, electrical, or esthetic in nature.

Some Causes of Distortion

Overload distortion on tape makes itself evident by the apparent loss of low frequencies and by high-frequency "fringe" noise. A heavily distorted tape will sound "choppy" and harsh. Distortion may be measured by considering the whole recorder as an audio amplifier and proceeding accordingly. Total harmonic distortion at 1000 cps should not exceed 2.0 per cent at maximum output. Distortion arises almost wholly in the magnetic recording process and tends to be more pronounced at the lower end of the recording spectrum. The more iron oxide in the tape coating, the lower the distortion at the low frequencies, provided that the recording level is the same. Distortion above 100 cps should be somewhat less than 2.0 per cent if the bias current has been properly adjusted. For a given amount of distortion there is a definite relation between the strength of the bias current, the audio level fed into the recording head, and the kind of tape used. Thus, as has been noted, recording at excessive levels results in overload distortion; at too low a level, the signal/noise ratio will suffer. However, it is better to undermodulate than overmodulate.

* *Columbia Broadcasting System, New York.*

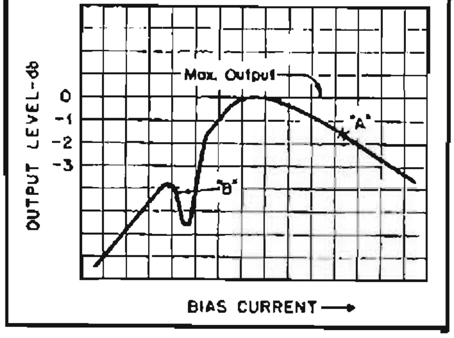

Fig. 1. Curve showing relation between output level and bias current for determination of correct operating point.

Fig. 2. Partially completed dual Magnetophon Unit. (Photograph by Wm. R. Busch, Chief Engineer for RIAS.)

Practical Bias-Current Adjustment

One method for adjusting bias current for optimum results is as follows:

1. Record at 1000 cps with an input that normally results in 2.0 per cent total harmonic distortion. This should take place at maximum output.
2. Vary the bias and observe the output level of the tape playback system. When the output level passes maximum and drops between 1 and 2 db. as at "A," the bias is correct for over-all good recording (see *Fig.* 1).
3. A temporary dip may occur, as at "B" during the bias adjustment operation. Make certain to pass this point to arrive at the correct operating bias point.
4. When minimum distortion at low frequencies is required, the bias may be increased still more than noted in *Fig.* 1, perhaps by 20 to 30 per cent. The practical effects will be: increased fidelity at low frequencies; loss in tape output voltage; and some loss at high frequencies due to erasure caused by the higher-than-normal bias current.
5. If in doubt, due to lack of measurement facilities, adjust the bias on the high side. Restriction of dynamic range is preferable to distortion.

Most recorders use constant signal current up to about 3 kc (at a speed of 15 in. per second). Above this point the recording signal current is increased in order to overcome, as much as possible, the effects of high-frequency losses in the head and self-demagnetization of the tape. Thus there is pre-emphasis on the high frequencies and post-emphasis (in playback) on both low and high frequencies, resulting in practically flat reproduction from the recorded tape. It is the best practice to play the tape back on the same machine on which it was recorded or, failing that, on the same make and type of recorder. Follow the manufacturer's advice as to which tape to use and how heavily to modulate it. When you begin to use a new type of tape, noise and distortion tests should be made at several frequencies. It will then be apparent, in line with the foregoing section on bias current adjustment, whether the bias current and audio level are correct. In the output of any tape recorder there will be a certain amount of "tape noise" which should be close to the lower limits of audibility and inaudible at normal listening levels.

Tape Qualities

The problems attending the manufacture of high quality magnetic coated media are receiving a great deal of attention in the industry. Paper-back tape continues to be used where cost is a factor and a slight increase in noise and lower tensile strength are not deterrents to its use. Homogeneous tape, in which the iron oxide is mixed with the plastics, is not used to any extent in the U. S. Enough iron for good overall response cannot easily be mixed with the plastic; it weakens the tensile strength too much. Homogeneous tape also prints excessively. The tape used in the U. S. is almost wholly the coated type, several kinds of which are easily available. The important characteristics of tape are:

1. Frequency response
2. Signal/noise ratio
3. Sensitivity to recording (Output from the recorded tape)
4. Ability to erase completely *easily.*

The necessary physical characteristics of tape are:

1. Limpness (for good head contact)
2. Good tensile strength (with no "stretch")
3. Even coating (and non-flaking)
4. Smoothness of backing material

Tape is cut from large sheets or rolls into ¼-in. strips and then wound on reels of various types. On large tape reels a number of separate strips may be patched together. The conscientious engineer will check the tape splices before recording. Sometimes these splices are carelessly made and will provide a disconcerting "hole" in the recording. A rapid check may be made by recording tone at low level and monitoring. Another method is to run the tape off at high speed and "feel" for the splices.

The tape speeds encountered in radio broadcasting are 7½, 15, or 30-in. per second. Combinations of 7½ and 15 or 15 and 30-in. speeds may be obtained in most recorders. All three speeds can be had on special order from some manufacturers. It is best to operate at the highest available speed that the budget permits. Fidelity increases with speed up to the point where excessive demagnetization takes place at the high frequencies. The higher the tape speed the easier editing becomes. However, response curves at 15-in. speed are good enough for FM use and where a network standard is needed the 15-in. speed is practical and economical. It should be noted that the 15-in. speed is the primary standard, and that the 7½ and 30 in. speeds are secondary standards.

Fig. 3. CBS-NY Studio 30A. Tape engineer Hendrickson at the controls.

Overheating of the recorders should be avoided. When operating in hot locations the temperature of the magnetic heads may rise to the point where the tape recording will become distorted. The cure is to devise some method of keeping the head temperatures down. Under normal conditions and with well-designed machines and heads, no cooling is necessary. But, when heads are operated above their normal current carrying capacity or are so mounted as to absorb excessive heat from the motors, a small blower may be used to cool them. Besides the distortion effect, excess heat may soften the tape coating binder and cause it to stick and peel, resulting in eccentric operation. Excess cold, on the other hand, may cause the tape to shatter easily, and the bearing grease may possibly congeal.

Proper Location for Recorder

Tape recorders should be so located that the mechanical noises they create while operating cannot be picked up by a microphone. The best possible location, obviously, would be in a soundproof room. Recorders for broadcast use should be installed in pairs, with convenient switching, metering and monitoring facilities. Note the partially-completed installation of a dual Magnetophone unit in the studios of RIAS (Radio in American Sector) in Berlin, (*Fig. 2*). One set of amplifiers is used for two mechanical units, with provisions for switching from one to the other on both record and playback. (Incidentally, the blower mounted at the bottom of the drive motor assembly functions both as a motor-cooling device and as an air impedance for drive stability.) Recorders used in the U. S. are complete units and may be installed as shown in *Fig. 3* where the machines can be used separately or simultaneously for either recording or playback.

The placement of the recording studio with respect to r.f. interference merits consideration. Strong r.f. signals may enter the record or playback amplifier with attendant cross-talk and noise. As a rule of thumb, the closer to earth the recorders are located the less r.f. interference. Test for r.f. pick-up in the feed lines, microphone lines, and in the playback amplifier. Interference may be filtered out with a low-pass filter or tuned out with a band-rejection filter, depending upon circumstances. Recorders used for field work should be well-shielded and should have r.f. by-pass capacitors installed, if necessary, from grid to cathode (or grid to ground) of the first-stage tubes in both record and playback amplifiers. In both studio and field installations avoid ground loops with the concurrent hum and noise increase.

Fig. 4. The man-behind-the-mike is Lee Bland, former Director of Special Events for CBS News.

When recording in an automobile or sound-truck, dressing of audio leads and careful grounding should be sufficient to keep out ignition noise. Audio cables should not be placed next to any of the electrical wiring of the car or truck. High- and low-level cables should be grouped separately and the recorder should be well-grounded and placed as far away as possible from the ignition system. Avoid using the car or truck battery as a primary power source for the recorder a.c. supply. No difficulty should be experienced while recording in airplanes since their ignition systems are normally well shielded. Recording in motion (except in trains and conveyances offering 115 volts, a.c.) requires the use of either a battery-operated recorder or an a.c. supply system of some sort. Where it is necessary to use a low-wattage a.c. operated recorder, a heavy duty 12-volt storage battery and an inverter supply of the proper size will perform very well. Whether an inverter or a converter is used, make certain its output is well-filtered and its acoustical noise deadened before attempting to record. As an illustration of a hookup using a vibrator supply and a low-wattage (175 watts) recorder in an airplane, the following is typical. The project was to record the conversation between the pilot of a helicopter and one of his passengers during a demonstration flight (see *Fig.* 4). The recorder was was placed in the center of the padded rear seat and securely strapped in place. An inverter, placed on a 2-in. rubber mat, was bolted to the floor on one side of the helicopter cabin with the 12-volt battery on the other side. The mechanical noise of the inverter was then muffled by felt sheets and short ground wires were connected from both the recorder chassis and the vibrator supply to "ground" which, in both cases, was the metal floor of the helicopter cabin. This hookup worked well throughout a thirty minute flight and several landings, including one "dead-motor" landing.

Since this helicopter flight was made, a completely self-contained battery-operated recorder has appeared on the scene. *Figure* 5 shows the size and ease of operating the Minitape. It records at either 7½ or 15-in. per second but has no provision for re-winding or playing back. It permits a recording time of 15 minutes at the lower speed and 7½ minutes at the higher. With proper care, very good performance can be expected of this recorder. Another concern has developed and is now marketing a miniature tape-playback amplifier. This is a pocket-sized unit which can be connected quickly and easily to any recorder to permit monitoring the recorded tape *while recording*. It also can be used, in conjunction with a manual winding mechanism, to check tape for quality while in the field.

Where the machines are used continually in one location for both recording and playback, there are no serious problems of a.c. supply. But when recording in the field with equipment using synchronous motor drives, some method must be found of controlling the frequency of the a.c. supply to the drive motor. One way of doing this is to use a frequency-regulated motor-generator set. Another method, used by John Mullin, recording and editing engineer of the Bing Crosby show, arrives at the proper frequency by triggering a pair of thyratrons by a vibrator whose frequency can be varied and measured on a vibrating-reed indicator.

Do's and Don'ts of Recording

The normal amount of care should be expended on getting proper mike placement for the response wanted in the recording. The value of making sample "takes" cannot be overstressed, especially where it is impossible to monitor the tape while recording. With machines that have separate playback systems, it is possible to listen to either the incoming program or the recorded tape and thus make valid quality checks. Incidentally, the human system of hearing is not a "high-fidelity" system but distorts excessively when the "loudness" of sound attacking it is too great. Normal hearing also may "hear" sound where such sound may not exist. The ear will tend the reconstruct some sounds out of harmonies, especially in the low-frequency range. The recording engineer, therefore, should monitor the recorded tape at normal sound level for comparison with the original sound.

The "liveness" of the studio should be adjusted for best results depending on the program to be recorded. Normally a slight amount of reverberation makes for cleaner speech and more sparkling sound. Remember that pre-emphasis in the recording amplifier keeps the highs up; therefore avoid excessive sibilance and paper "rustle". Place mikes to avoid heavy lows, which may overload the tape and also mask the intelligence frequencies. Music should be recorded full range and may be given special treatment in re-recording before playback to air.

If the recording is made with editing in view, the following things should be avoided. It is not possible to remove reverberation from recorded tape; therefore avoid it in recording for editing. It can be put back later in the re-recording process. Keep levels constant, within reason, to permit easy blends in editing. Record several minutes of clear background sound whenever there is a change in the story location or idea. The suggestion of impending change by means of changing background sound patterns is a valuable tool in tape work. Record over again, on the same location, any sequence in conversation that may have been marred by a sudden change in background, a change in position of speakers' voices (unintentional) or by unwanted interjections. Interviewers should, of course, train themselves not to interrupt or trail off into the background with "hm-m." Speakers or "interviewees" must be allowed time in which to settle down to normal voice and pace before recording. Sometimes, obviously, it is better to record a "practice" interview which may prove to be better than the "air" interview.

Always fade into backgound slowly and fade out the same way; the fades may be needed in editing the show for air. Background sounds, after editing,

[*Continued on page 43*]

Fig. 5. The author operating a Minitape recorder. There is no level control, only a stop-start switch. The headphones monitor the incoming audio only, not the recorded tape.

TAPE RECORDING

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assume an importance all out of proportion to their original significance while recording. Consider the plight of a brilliant radio writer who recorded interviews in Europe with the DP's of World War II. The records he obtained were fine examples of recording but, when the tape was edited and the parts he wanted joined together, the effect was anything but homogenous. One sentence contained, in the background of the first part, the roar of an airplane overhead. The airplane sequence then cut abruptly and the second part of the sentence was backstopped by the bawling of an infant. The writer had made the common mistake of trying to compose his script without considering background sound.

Thus it can be readily perceived why, if recording with editing in view, it is so important that background sounds be taken into account. Of course, when recording voices in the open, one does not have full control of any sound that may arise. But it is possible, if in a noisy location, to reduce background sound to a minimum by using uni-directional microphones close to the speakers. Then, after the interview is completed, background noises may be recorded by themselves and utilized in the editing and re-recording process to provide a solid sound background. It is the skillful recording and use of background sound that makes it possible to create outstanding radio shows.

Acoustic Variation

Just as background and special sounds have a character all their own, room acoustics also vary with weather and the number of occupants. It is preferable to make certain that all recordings made in one location for a given program are made under the same acoustical conditions. There are many shades of acoustical "color" and vibration in the acoustics from one sentence to the next in an edited tape does not make for a good performance. Listeners, even though unconsciously, are affected by reverberation, by deadness, by whispering stillness, or by blank quiet. The coherence of the acoustical background of a tape show is one of the factors that will gain for it complete credibility, thus creating the wanted illusion of reality.

The human voice with all its variations and moods is a very complex sound. In order to capture this sound completely and reproduce it with no factor of its composition lacking, emphasis should be placed upon the central theme of the subject matter that the "voice" is discussing. To illustrate this, one may use a recording made by Bill Downs of CBS News, who, in addition to being a fine reporter is evidently an artist in the use of atmosphere and background. In the case in point, Bill was recording a sequence in the underground retreat in Berlin where Hitler was presumably killed and later carried out and cremated. Downs and his engineer manoeuvered around in this bunker until the reverberations of his voice carried just enough sepulchral quality to create the correct impression. Even his footsteps going downstairs and coming back up were authentic. Histrionically speaking, his voice *carried* through the recording and placed his audience right in the bunker with him. If he had made the mistake of speaking close to the microphone this atmosphere would have been spoiled or lost entirely.

The tape-recording engineer must keep in mind, during all his recording, the purpose of the recordings. It is not enough just to record at random a multitude of voices and sounds with the hope that out of all that mass there might be enough material to construct a documentary show. If this hit-or-miss method is indulged in, it will generally be found that many necessary ingredients for a good show will have been missed. If he is to turn in a professional job, the engineer must know the purpose for which his recordings are to be used. It would help immeasurably if the engineer were given a resume of the prospective show so that he could become familiar with the dramatic idea. Then, with the help of the director, the engineer could record sequences that would enhance the dramatic appeal of the show by utilizing techniques available and known to him.

Occasionally a sequence is recorded where not enough time is available to make tests and to place microphones. In cases like that of Bill Downs in Berlin, an appropriate background could be dubbed in later to create the proper illusion of "place where" the recording was made. Where enough equipment is available, the voice recording can be played back on one machine, background from another, and both recorded on a third machine utilizing any degree of filtering and reverberation necessary to complete the wanted illusion.

It is advisable when recording voices to choose microphones for good high-frequency response, since a great deal of intelligence is conveyed in the higher voice frequencies (above 1500 cps). It seems also there is more dramatic impact in the high-pitched voice. This may be explained by the fact that hearing is excited more by high sounds, and that low frequencies tend to mask, or obscure, high frequencies in the mechanism of hearing. Another obvious reason is that voices under tension become shrill and listeners' memories react with excitement to an excited, high-pitched voice.

For the last several years point-to-point overseas transmissions have provided a great deal of work for tape recording engineers. There are several reasons why this method of recording news from foreign points is valuable. For one thing, a tape recording may be edited, after it is received, to fit a time-slug in a news program. In common with other means of recording it permits a correspondent to live a more or less normal life (for correspondents!) in that he does not have to get up at unreasonable hours of the morning to appear on a late evening broadcast. Also it provides a fairly cheap method of recording transmissions at the best possible transmission time. Transmissions, or broadcasts, via short wave from foreign countries are generally not crisp enough in quality to bear reproduction without some filtering. Generally speaking, cutting out all frequencies below 150 cps will improve the intelligibility without destroying much voice quality. Very often the communication company can filter this out for you. However, a valuable piece of equipment for any tape recording engineer is a variable-step sound filter, which permits cutting down, by small degrees, any band of sound frequencies. With it, disturbing heterodyne "whistles" can be diminished, modulating hums eliminated etc. It depends upon circumstances whether filtering is to be done before recording or after recording. Ordinarily there is more time available after recording to check the tape, on playing it back through a filter like the one mentioned above. It happens that a recording will be unintelligible as broadcast but completely "readable" when played through the proper filter combination.

It is obviously impossible, in an article of this nature, to include many more specific examples of good engineering. The reader is probably aware of other examples which will serve as guides through the ramifications of tape-recording.

The next part of this series will undertake an explanation of the re-recording or "dubbing" process in tape recording.

The Art of Tape Recording—III

JOEL TALL*

A discussion of one of the more important operations in the assembly of a completed tape-recording of a radio show.

THE PROCESS OF RE-RECORDING, or dubbing, magnetic tape is intrinsically a simple operation. It is necessary, of course, to have at least two machines and to match, or bridge, the input of the recording machine to the output of the playback machine. Re-recording may be done before or after editing, depending on the reason a copy of the original tape is needed. In some cases re-recording may be wholesale, and a battery of recorders may be set up to make copies of a program for distribution to regional broadcast stations, or some system such as Minnesota Mining and Mfg. Co.'s method of "printing" copies for commercial purposes may be employed. Basically there is no loss of fidelity in re-recording and practically no noise is added to the copies. In some respects, copies can be considerably better than the original as we shall point out later on in this article.

The Re-Recording Studio

The re-recording studio should be so laid out and equipped that the engineer can control the various processes with comparative ease and with a minimum of lost time. The following set-up for a re-recording studio should be sufficient to permit almost any re-recording operation that will be found necessary.

The studio should be equipped with at least three similar tape recorders, one of which should have a tape drive capable of being varied in speed so that tape recorded in the field at above or below normal speed may be re-recorded at the correct speed. (The turntables of this machine should be adaptable to any type of tape reel.) The method of arriving at the correct speed does not matter as long as the tape can be played back for re-recording at the speed at which it was originally recorded. The other two machines should be standard in all respects and should be identical. The frequency response and noise levels of all three machines should be exactly the same, both on record and playback positions.

A four-position mixer should be available, the output of which can be patched to the recording input of any one machine. The outputs of any two machines may then be patched into the mixer for re-recording on the third. Each of the positions on the mixer has available in its circuit a variable equalizer filter, which can be either switched in or faded in on the mixer panel. Also, each mixer position has echo available, controllable in any degree. There should be a VU meter for each position, since a very close control of individual levels is indispensable in re-recording. The two spare positions are then used, when necessary, for live voices and for any other effect or sound—whether it comes from a third and fourth tape machine, from discs, or from mikes. The mixer should also have available the stopping, starting, record, and play controls of the recorders. In this setup the re-recording engineer has at his fingertips all of the ingredients for an excellent re-recording job (see *Fig.* 1).

At this point perhaps we had better stop to answer the obvious question, "Why all the emphasis on re-recording? Why should a description of this process consume so much space?" The answer has much to do with this engineer's conception of the future of radio broadcasting. One trend in broadcasting is to utilize top-grade artists on TV "live," if possible, and to record the audio for AM. Also, as explained before, some artists prefer to record their programs wholesale or at times and places that cannot be scheduled for direct air broadcast. The processes of re-recording and editing permit making TV audio palatable for AM use—plus any of the operations that follow.

Correction of Off-Speed Recording

A tape recorded off-speed in the field or in countries where 60-cps a.c. supply is not available can be re-recorded in the studio from a variable-speed machine to a standard-speed machine. However, there is a limit to what can be done in correcting tape recorded at abnormal speeds. If the incorrect speed has been constant all through the recording, all will be fine. But if there has been tape slippage, no ordinary technique will do the job. A variable-frequency or variable-voltage power supply may be used to operate the drive motor at different speeds, depending upon whether it is a synchronous motor or a simple series-field motor. A variable-taper mechanical drive might also be used for varying the capstan speed. But there is no real cure for a tape that has slipped during recording.

Reduction of Noise and Hum

Occasionally tape will be recorded with a noise or hum component that was not heard during the recording session. Short-wave transmissions may contain a characteristic generator whine, or may have been filtered too much or too little by the communications companies concerned. In cases like these, re-recording through a variable equalizer can provide a usable recording. In general, overseas recordings may be advantageously re-recorded after filtering out everything below 150 cps and everything above 4000 cps. The engineer must use his own judgment and re-record for good intelligibility. The writer has improved recordings from overseas by using reverberation after filtering which added to the "apparent loudness" of the voice without accentuating the noise content. In some cases it is necessary to "roll off" low frequencies beginning at about 250 cps to get rid of the more flagrant hum harmonics.

Although the writer assumes that his confreres always will record at the proper level, as outlined in Part II, any lapse from normal may be corrected by re-recording. Material recorded at low level cannot be re-recorded at an increased level without also increasing the level of the noise. Therefore, if the re-recording is to be put on the air without going through a mixing studio, it would be better to "roll off" the more noticeable noise frequencies (about 6000 cps and up) during the dubbing operation. Remember to keep levels up. Wide dynamic range is a fine thing to theorize about, but a great majority of listeners want to hear plainly without straining and without twiddling at their receiver gain controls.

The uses of a properly designed equalizer in reducing noise and hum during re-recording has been discussed. The same equalizer may be used to advantage to bring dialogue in sharp focus or to change presence. It can be used to cut down or increase sibilance, as needed. A director may wish to change a musical recording so as to accentuate one band of frequencies or another or to simulate varied effects. He may want music "re-

* *Columbia Broadcasting System, New York.*

verberated" and re-recorded or any combination of filter and reverberation may be used to gain an effect.

Present practice indicates that one of the chores of tape engineers will be that of recording TV audio for AM broadcast at another time. Due to many factors, some controllable and some not, TV audio quality does not always compare with that of AM or FM radio. With microphones out of camera range or hidden, filtered P.A. systems, actors' changing positions, etc., TV audio may have very little at the low end, a hump in mid-frequency, or over-accentuated high frequencies. Re-recording through an equalizer can prove effective in improving the quality of such a recording. It would be still better, and result in better signal-to-noise ratio, if the equalizing were done *before* recording in such cases.

Blending of Program Material

With the approaching standardization of tape recording machines, the assembly of taped programs from material recorded at different places and at different times will become commonplace. The dialogue may be recorded in Hollywood, the music in New York, and the commercials in Chicago. There need be no difficulty in blending all these ingredients into a good show provided that there is no audience. If there is an audience, its reactions and applause should be recorded only at one point, say New York, and enough "audience only" sound recorded to serve as background throughout the whole program. Announcements and commercials may thus be inserted in the tape before a "live" audience, no matter where recorded. This whole process requires a close control of timing, levels, and sound (filter and reverberation). The re-recording engineer will have to make use of a cue sheet and control every process accurately. Needless to say, the "blending" should be rehearsed adequately before re-recording the final tape.

One of radio's most used devices for indicating a change of place, time, or mood is the "bridge." A bridge may consist of music, sound effects, background sound, or absolute silence, or any combination of these four factors. Bridging is a technique that finds many interesting applications in tape re-recording and deserves full treatment here.

In Part II it was suggested that background sounds be recorded every time they change in character. These same sounds may be used for bridge material as indicated by the director. The obvious purpose of a bridge is to provide a vehicle for "carrying" the listener from one sequence to another. A good bridge establishes the idea it is meant to picture without being too obtrusive or shocking. As ordinarily used, background sound bridges are smooth blends and may be made in the following manner: Suppose that you have recorded a program, one part of which takes place in a farmhouse and a second part in a railroad car. The transition requires at least three background sounds, possibly more. The minimum blend requires sound inside the house, outside the house, and the railroad car sound. An elaborate bridge could use these, plus auto starting, running, auto stopping, station sounds, train outside, train starting (perhaps with whistle), car inside, and then inside background with voices. The technique of making bridges effective requires that each sound registers in the listener's mind and establishes its intended meaning. The degree of level and the length of time necessary to establish a sound will vary according to the recognizability of the sound.

Where a bridge from one sequence to another is absolute silence, be careful not to mar the effect with any mechanical sound. Fade out completely and fade the new sequence in *after* starting the tape playback.

Procedure in "Matching" Sound

Suppose that five programs have been recorded from which one additional "warp-up" program is to be created. The program director and the engineer have listened to all the material and have noted variations in level, background, and reverberation. From the director's point of view the dramatic value of the program requires that a "dead" portion of recorded tape be placed between two "live" portions. The engineer then must re-record the "dead" portion. It is possible to make a recording "live" by sending it through an echo chamber or a mechanical or tape reverberator. But it is not possible to make a "live" recording "dead." If the operation has been performed carefully, the tape portions will then all match and can be edited as needed. Suppose that one of the above programs had been recorded in a studio which had in it a smaller audience than normal. There might then be an apparent increase in "liveness," and the applause patterns would be entirely different. This situation could be partially cured by re-recording at a lower level than normal and by cutting off the extreme highs. If necessary, the applause could be doubled or quadrupled in "number of people applauding," by re-recording the applause two or more times. The first re-recording would be a simple copy of the applause. Then both "applause" dubs should be played back on two separate machines and their outputs recorded on a third machine. If the two "applauses" are started at the proper time interval, the resulting recording will sound like the applause of an audience twice as numerous as it actually was. An artistic job of re-recording will be entirely acceptable here; do not, however, mutilate the applause pattern. Applause bursts are rare. Ordinarily people begin to applaud in a scattered manner, build up to a crescendo and then taper off. One cannot easily duplicate the scattered applause effect, and it should not be attempted unless every other method is inapplicable. The hallmark of poor tape

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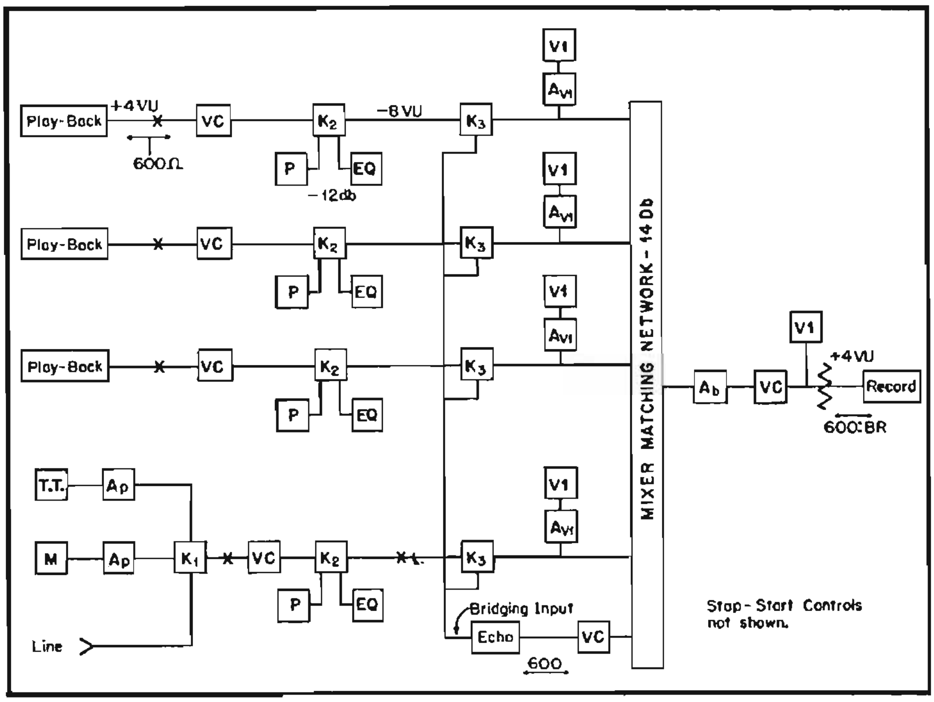

Fig. 1. Suggested block diagram suitable for the re-recording operation using several tape machines in conjunction with other sound sources.

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recording is repetitive applause which has palpably been dubbed from one single applause pattern.

You may occasionally become involved in the following operation: A program to be edited calls for a change in the ending, which was originally recorded over applause. It is obvious that the changed portion should be recorded in the same way. In order to do this an applause sequence of the proper length, taper, and timbre must be found. If necessary, it can be manufactured from short dubs of other applause sequences. The applause dub can then be recorded, mixed with the "live voice" or recorded voice, containing the new ending. Have the announcer use the same voice level as in the original. You may even have to accentuate the high frequencies in his voice, since most announcers' voices increase in pitch during applause that is heavy and sustained. It might be a good idea to play the applause through a loudspeaker in the same room with the announcer in order to duplicate the actual conditions under which he made the original announcement. Accoustical response must always match the original recording, however, no matter what method you use.

In recording on location, it is not possible to have full control of the surrounding sounds. A regular background sound that repeats itself at intervals may then have been recorded in back of voices. If the admonitions in Part II have been heeded, enough background sound will have been recorded in the clear to make a corrective operation easy to perform. Where a repetitive sound pattern, like the tick of a clock, will not be regular after the proposed editing, edit first and then dub in the repetitive sound at exactly the proper times and level.

For special effects that may be required in the production of a program, the tape engineer may rely on his own ingenuity. Voices may be modulated in air by propellers to simulate conversation in back of a plane. The duplication of a needed effect should be sought in the natural way first. If that is not possible, re-recording with sound effects in some combination will prove effective.

Sound Library

The tape engineer should be provided with a library of tape recordings of background sounds, applause, laughter, and general "bustle" and audience sounds that can be used for simulating the needed background for re-recording. Thus, by using the proper equalization and reverberation, he can match exactly the mood of the master recording. It is sometimes possible to dub, from the original tape, enough of applause and background to make two continuous "bands" of recorded tape. If these "bands" are then placed on two separate playback machines, they can be faded in from the mixer at the appropriate time and re-recorded, together with the needed "live" sequence, upon the master. There are two cautions about the foregoing operations:—

1. Pick out a background section that does not contain any noticeable variation so that it will not be repetitious.
2. Be extremely careful not to make the background level too high or too low. It must play back at exactly the same level as the "live" background in order to complete the illusion of "same place."

The operation of re-recording, in all of its complexities, calls for a degree of tonal perception on the part of the tape recording technician. It is quite useless, in most cases, to make a wide-range tape recording of transcriptions or commercial pressings when the final result will contain mostly distortion and noise. It is wise to restrict what is re-recorded to the useful sounds only and to filter out the rest. In this way a recording will at least have the virtue of some degree of uniformity. The writer's experience, during the composition of the Columbia Record Album, the first "I Can Hear It Now," was that it was best to restrict the range of the better transcriptions to the approximate quality of that of the poorer, so that there will be no startling discrepancies in the tonal quality of the finished product.

Part IV of this series, to follow in next month's issue, will deal with the methods of editing tape and the exact procedure in editing various kinds of programs.

The Art of Tape Recording—IV

JOEL TALL*

The mechanics of tape editing—covering the actual methods of cutting and splicing magnetic tape recordings into a finished program.

Disregarding every other quality which makes tape recording valuable to professional workers in audio, tape would still continue to be valued because of its facility of editing. How well the tape can be edited depends upon the judgment of the production personnel, the expertness of the tape editor, the recording itself, and last and possibly least, the facilities that are available. In any case, the edited version of the recording can be no better than the original as far as tonal quality is concerned, although some improvement in tonal balance can be achieved through the various processes of re-recording.

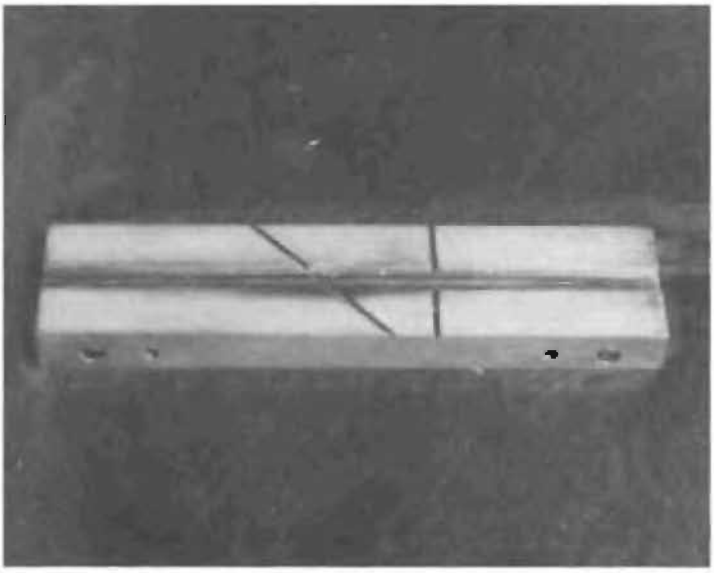
There are certain logical steps which must be taken in the production of a completely edited tape before the tape is ready for playback. Consider one simple form of tape editing, the process by which a talk or speech is condensed without disturbing its meaning or impact. Assume that a half-hour recording of a political speech must be cut to nine minutes and thirty seconds air time. The permission of the speaker must be obtained to condense the speech, and its sense must not be disturbed. Obtain a typed draft of the speech, if it is available, or have a competent stenographer make a typewritten transcript from the tape itself. After this transcript of the speech has been studied, a capable director can condense the speech into the required time *on paper*. (A word of warning may be in order at this point. Occasionally a speaker will wander from his script and "extemporize." In such cases the tape editor will be at a loss unless he makes a practice of not discarding any part of the recording until the editing job is finished.) After the speech has been edited on paper, the production personnel and the tape editor should monitor the tape two or three times: first, to make certain that the excerpts preserve the ideas the original tape contained; and second, that the parts that are to be joined together are "joinable."

* *Columbia Broadcasting System, New York.*

That strange word "joinable" means this: A speaker will try to convince his audience by the use of oratorical devices. He will change mood, level, pace, and inflection. He may be sad, joyful, sarcastic, cynical or earnest. It is impossible to edit tape without taking into consideration all of these factors. To attempt to separate mood from pace or level from inflection is useless. They must be considered together, in the many combinations of sounds that make the human voice a most expressive musical instrument. The art in tape editing lies in the editor's ability to interpret correctly the many factors contained in speech and to utilize them in producing a coherent and authoritative product. Because of this, there can be no hard and fast "rules of editing." There are some fairly adequate generalizations that will result in better productions. For example, it would create a weird effect if a phrase spoken in a high-pitched, excited voice were joined to another phrase in a calm, low-modulated voice. Or suppose that the speaker were popular and frequently had to over-ride applause during his speech. A phrase from the "applause in background" portion could not be joined to a phrase from the "no applause" portion without dubbing-in applause where it is needed.

Where to Cut and Why

As the advertisements tell us, tape is edited by cutting with scissors and splicing the two ends together with an adhesive. The actual methods now in use will be described later. Right now the question is "Where shall I cut the tape?" There are many places where you *can* cut it but there is only *one* spot that is exactly right. In order for you to understand why there is only one right place to cut the tape, we shall diverge from editing to a short discussion on the subject of "hearing."



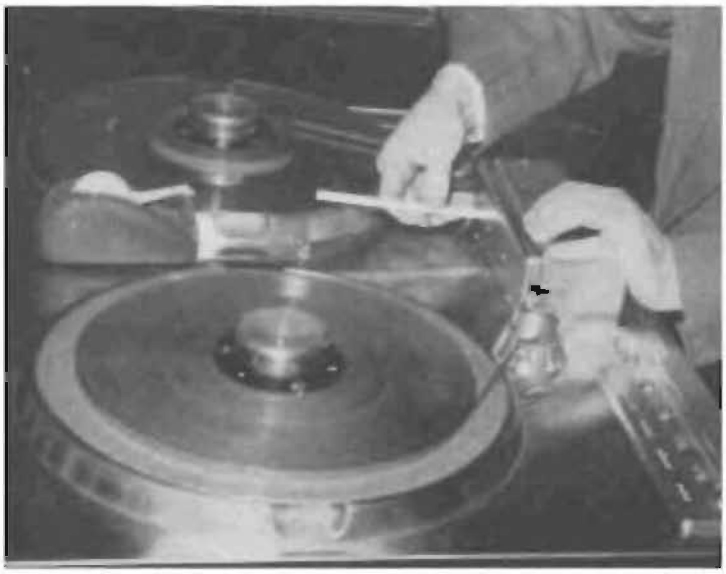


Fig. 1, left. Adjustable splicing block. The original was designed by the author in 1947 and made by Victor Piliero, CBS engineer. Fig. 2, right. Method of marking a cut with a grease pencil inserted through the marking aperture which is at the center of the playback head.

The human ear, when it is behaving normally, can understand, or perceive, an unrelated sound following another sound after a period of time approximately 0.14 seconds long. At the standard tape speed of 15 inches per second, that length of time represents about 2.27 inches of tape. (The ordinary key-click at the above speed would occupy from ¼ to ½ inch of space on the tape.) This 2.27-inch space, in normal speech, represents about twice the distance between words. Therefore, to edit accurately, you must find *the right place* to cut within a space of one to two inches. Within that space you must try to match, as closely as possible, the background sound recorded at the beginning of the next phrase to be joined. After you have noted that the backgrounds match, you must still observe the speaker's natural "pace." The speaker's "pace" or "gait" will vary according to what he is saying and his natural pace must be observed and followed in the edited version so that it is the same as in the original recording. In this connection do not forget that the man who speaks must breathe. Allow time for this function even though it is not audible. The expert tape editor will observe natural breathing habits and edit accordingly. Occasionally this will force him to clip out "breaths" of various lengths and intensities, and sighs and hesitant speech sounds, and insert them in their proper places in the edited tape.

Mood, pace, level, and inflection should be considered together. When a speaker becomes angry or excited, he generally speaks more rapidly, at an increased level and in a higher pitched voice. The tape editor should be able to judge, before cutting the tape, whether two wanted sequences of a tape recording can be joined with reasonable naturalness and credibility. If it is evident that they cannot be matched together as originally recorded, they may be matched by re-recording, utilizing some of the methods previously noted. If re-recording is inconvenient, and the sequence is absolutely necessary to the show, a pause of matching background sound may be put between the two segments to give the sequence some flavor of actuality.

It is not desirable to end a sentence with an "up" inflection unless the speaker is meant thus to interrupt himself or to be interrupted by another voice or sound *immediately*. There must be no pause whatsoever on an interruption of this nature. It should be completely evident to the listener that it *was* an interruption and no background sound should intervene.

At other times the editor will find places in his show where a pause is required for an effect, dramatic or otherwise. In such cases it is important that the background sounds in the pause match the background of the end of the preceding tape segment and that of the beginning of the following one. If there is an unavoidable change in the character of background sound from one tape sequence to another, there is only one short-cut to making the whole thing believable. That is to leave in the background of one *or* the other sequence and clip the other sequence close to the first word. Another way to achieve homogeneity is to dub both sequences accompanied by another masking sound at fairly low frequencies, but intelligibility will then be diminished.

Where a transition effect on tape is needed it may be obtained either by re-recording and cross-fading or by using a recorded fade-out of one background and a fade-in of the other background. Effects of this nature on tape are limited only by the ingenuity of the engineer and his experience in the medium.

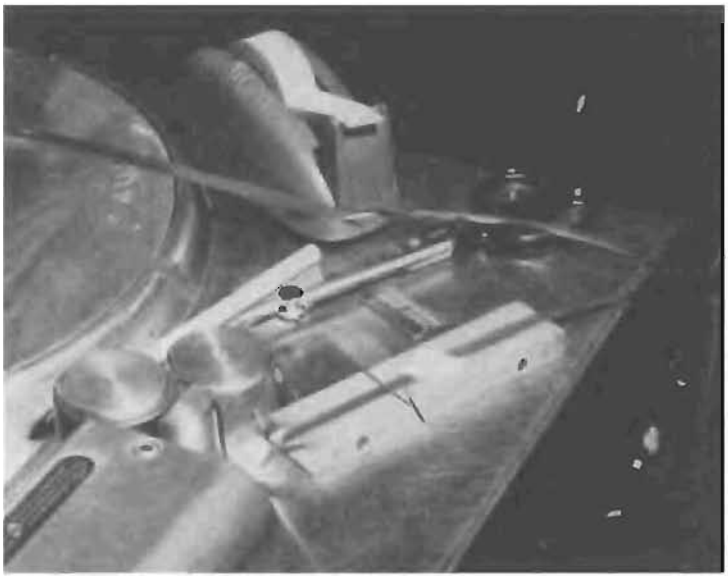


Fig. 5. Tape ends are joined together in the splicing block. The #41 Scotch tape is applied at an angle to the magnetic tape so there will be the least possible disturbance to smooth motion of the tape through the head assembly.

Editing Quiz Shows

Radio showmen who are alive to the possibilities of tape consider it is best used for editing audience-participation shows. It is only by using tape that some

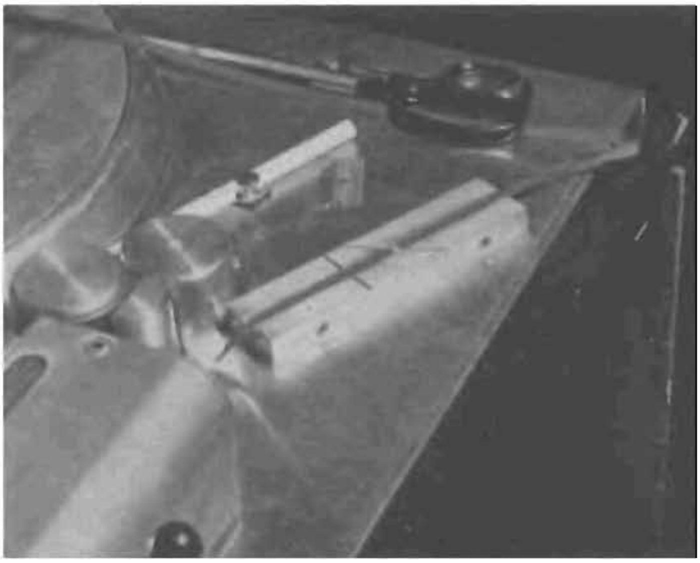


Fig. 3, left. Marked tape ready to be cut at the first mark. Use a slicing action when cutting to avoid fracturing the tape ends. Fig. 4, right. Running off tape to be discarded. This section of the tape is between points previously marked.

artists turn in consistently excellent performances. The artist (or master of ceremonies, if you prefer) is then relieved of the fear of a "dead" or "wise" participant, for he can be clipped out easily.

There is no great obstacle for the engineer who edits a first-class quiz-show. If he understands his medium and knows the routine of the show, he can turn in a very creditable job. The same conditions prevail as in the editing of a speech, except that the pace is generally faster, there are more interruptions and the backgrounds change rapidly. Perhaps it will be of use to outline the procedure in editing a typical quiz show.

The show was recorded in duplicate and lasted approximately forty-five minutes. The personnel of the show knew it was being recorded for editing. If an error in a music cue or commercial was made, the show was stopped and the error corrected. (In most cases, however, because of background change, it proved to be better to use the original than the amended version.)

When the original show was performed it was also recorded on disks for the use of the producer. He and his staff would then time the sequences needed, figure out the necessary editing and approximate the position of each part in the finished show. Thus when the time came for editing the tape, the editor knew approximately whether or not the parts would fit together. Rarely, however, would the show be put together in anything like the sequence in which it was recorded. To get listeners' interest, the first contestant in the original show might become the fifth in the edited version and vice versa. In order to fit these parts of a tape-puzzle together it was necessary to match wherever possible. Some pieces joined on applause, others on a laugh. Once, when it was necessary to cut the show by one chorus of a song the same note in two different choruses was cut in half and the two half notes joined together. Luckily the musicians stayed in pitch. However, it is possible to edit musical numbers in this way only when the music is played rapidly and there is little reverberation. It is not workmanlike to edit music unless it is possible to hit the same note exactly, played by the same instruments in the identical manner. All factors must match, otherwise a re-recording session and cross-fading is called for. After the show was edited to the proper time (29 minutes and 45 seconds), the edited tape was played back and a copy made on fresh tape. While recording the copy, which was for air use, the levels were corrected so that the overall effect was as smooth as possible.

Fig. 8. Tape editor A. J. Sisco splicing tape at NBC cutting room. (NBC photo)

It can be readily understood that the tape editor must be more than a mechanical splicer of loose ends of tape. Whether the show be musical, dramatic or quiz, the "feeling" of the show determines how it should be edited.

Many special instances could be cited, but the following is an example of what can be accomplished through the exercise of judgment and common sense.

Several years ago a recording of a native woman made on one of the South Pacific Islands required editing. She had recorded, in English, her distrust and distaste for the invaders of her island. As she said the word "Soldiers!" explosively, she followed it with a nervous laugh. In the sequence in which her words had to be used, the little nervous laugh after "those soldiers" would have been completely misunderstood. Thus, in order to retain what dramatic value there was, the laugh was changed into a sob by inverting its inflections, which left the sequence entirely in character.

No doubt it is understood that recognition of sound depends upon the speed at

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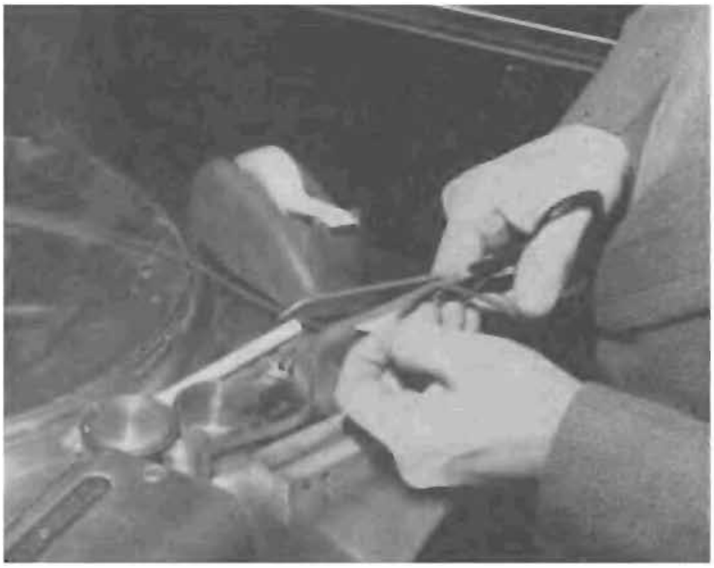

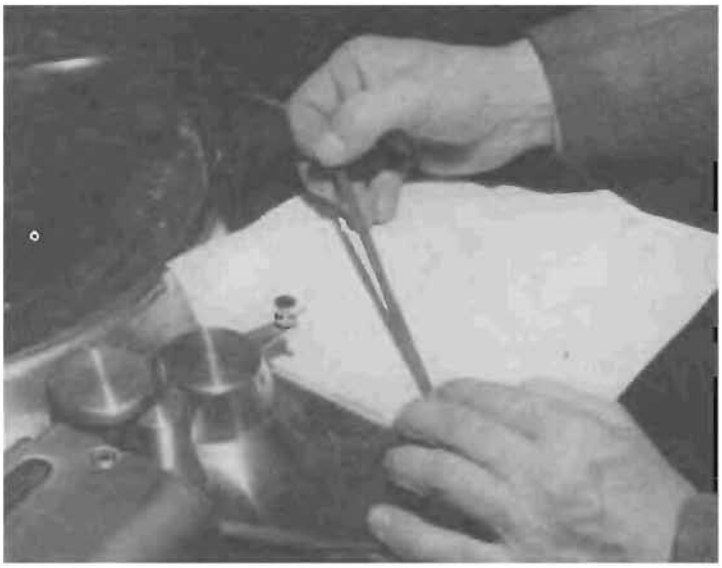

Fig. 6, left. Undercutting at the splice. The cut should taper out at both ends. Fig. 7, right. A completed splice. Note that the cut can be seen. A perfect splice cannot be noticed easily when running through the playback machine.

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which the tape moves past the head. The faculty of recognizing sounds at various speeds must be well-developed if the tape editor expects to achieve any appreciable speed and finesse. Constant practice is required, so that the ear becomes familiar with all commonly encountered sounds.

The easiest sound to recognize—and therefore to edit—is the sound of *s* and similar sibilant sounds such as *ch, sh, ts, tz,* and so on. The hard sounds of *t, p, b* and similar sounding combinations of sounds are also quite easy to recognize. The sounds of *r*, both round and gutteral, are more difficult to determine, especially when they occur in the middle of a word. Compound sounds, such as the beginning *y* of *you* and other *y* and *u* sounds, are difficult to apprehend and sometimes are recognizable only at normal tape speed. The tyro editor would do well to practice the recognition of sounds and to exercise himself in the art of editing tape by cutting out slurred *r's, u's* and other sounds as mentioned above.

Editing of musical recordings is done in the same manner as that of voice and sound, with the exception that the editor of music should be acutely conscious of rhythm, pitch, and "overhang." By "overhang" is meant those lingering tonal beats, especially of string instruments, that are somewhat similar to reverberation. Because of these lingering overtones, it will often be difficult to edit a musical piece without the use of re-recording techniques. However, in some cases, by cutting at the beginning of a bar exactly, an acceptable job can be done.

Tape Splicing

The actual mechanics of tape editing are fairly simple. Some tape editors work with scissors and Scotch tape. Some use a patented cutting and patching mechanism similar to a motion picture film splicer. Since 1947, the author has used a cutting block which he designed and which has proved satisfactory. The final result to strive for in the mechanical process of editing is a smooth splice, with the ends of the tape abutting each other with no discernible space between. A diagonal cut which eliminates the 90-deg.-cut "clicks" has been found most satisfactory. In addition to eliminating clicks, which are very disconcerting to the listener, the diagonal cut helps to make background blending easier. The splicing block is made of a piece of brass 7½ in. long and 1½ in. wide and ¾ in. thick. A slot ⅛ in deep and .248 in. wide is machined in this block. The ¼ in. tape will then fit snugly in the block so that it can be cut accurately. In the middle of the block, a 45-deg. angle cut is made, as shown in *Fig.* 1, extending to the bottom of the tape groove, wide enough to admit a single-edged razor blade (See figure #1).

When editing on any tape machine the place to mark the tape for cutting is at the magnetic gap in the play head as in *Fig.* 2. Any sound you have heard will have passed this point. It is best, if possible, to mark both sides of an excerpt *before* cutting. Then it is only necessary to cut at both marked spots and join together. (See *Figs.* 3 and 4.)

A number of methods for splicing tape have been tried. Tape can be cemented together if overlapped, but the cement may disturb the binder in the magnetic-coating, and an overlap splice is neither as accurate nor as quiet as a butt splice. Heat-vulcanizing has been tried on plastic tape with great success, but it is not recommended where a splice might have to be removed for a change in script. Scotch tape #41, which has a minimum of "tackiness" and does not "bleed", permits quick and permanent splicing and is the favorite adhesive in the industry to date.

The simplest way to make a good splice is to use a block such as has been described. There are several good tape editors who mark their "spots" and cut *both* diagonally with scissors at the same time. However, in that case, it requires practice to keep the tape ends lined up perfectly until they are patched together.

After about a one-inch length of #41 Scotch tape has been stuck to the recording tape ends, it should be firmly pressed to the tape so that the adhesive is thoroughly engaged, as shown in *Fig.* 5. When splicing tape in a jig, be careful not to press the tape with such force that the recording surface is depressed at the splice. When this surface is pressed in it will not be in good contact with the magnetic heads and the high frequencies will be attenuated at the splice. Then the superfluous Scotch tape should be trimmed with scissors, cutting smoothly *into* the recording tape for a depth of about .002 in., as in *Fig. 6*. This undercutting is advised so that no adhesive will appear on the surface of the tape and, subsequently, on the recording and play heads. Undercutting also makes the compliance of the splice more nearly the same as that of the tape itself, thus assuring better head contact and less "skip." Care in splicing, resulting in a finished splice similar to the one shown in *Fig.* 7, pays off by making the completed show sound perfectly natural, avoiding any sign whatsoever that it has been edited.

Where it is necessary, to save time, strips of adhesive tape (#41) may be cut in advance, in widths of 3/16 in. or so, and used to make splices that are not of a permanent character. An experienced editor can make a good, permanent splice in approximately thirty-five seconds, while a temporary splice may take ten seconds less. *Figure 8* shows a tape editor at work in an NBC tape cutting room.

If the tape is to be re-used time and time again (which is one of the reasons why tape is used) it is better to make perfect splices of the permanent kind. Then, after the tape has been erased and used again for recording, there will be no embarrassing "holes" in the recording. The adhesive tape, needless to say, does not record magnetically.

Auditory Fatique in Editing

We have thus far covered editing in two of its aspects—how to cut tape and why and how to make a clean splice. The psychological aspect of tape editing becomes the next problem. You will find that the concentration required of your hearing system during an editing session is extremely fatiguing. Since, the monitor system must be set at a fairly high level in order to hear extraneous noises at low levels and soft sibilant endings of words, auditory fatigue sets in rather rapidly. Hearing-fatigue lowers the ability of the ear to detect sounds by almost 50 per cent and causes more than normal distortion in the system of hearing itself, so the reason for the above opinion is self-evident.

Editing tape is in itself a profession and like any other work, the more editing you do, the better you become at it. There is nothing that will take the place of practice and the exercise of your own acumen. You may devise methods of your own for editing that will prove to be better than those outlined. The main object is to become familiar with the workings of your equipment; the rest is practice and common sense.

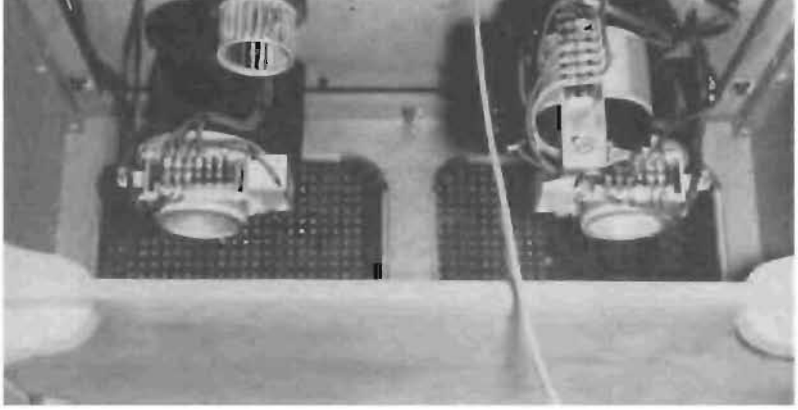

Fig. 1 (left). Top view of professional-type recorder (Ampex). The tape feeds from the loaded supply reel at the left past a tape guide which automatically actuates a motor switch. The control panel is shown below the supply reel, with the magnetic head assembly to its right. The drive capstan and pressure roller are at the right of the head assembly, with the take-up reel above. Fig. 2 (right). Motors and brake-drum assembly of the same machine.

The Art of Tape Recording—V

JOEL TALL*

The author describes methods of maintaining tape-recording equipment so as to produce consistently good recordings both in the studio and in the field.

Broadcast engineers will concede that the proper maintenance of standard broadcast equipment is a chore requiring extensive experience and systematic logical reasoning. When one adds the maintenance of mechanical components, such as are found in tape machines, to that of electronic equipment, the job of the maintenance engineer becomes highly specialized in character. It is the writer's earnest hope that this article will serve, in a necessarily general manner, as a primer in the work of tape machine maintenance.

Whether tape recorders are in continuous service or not, a periodic check should be made of the complete machine at least every 100 hours of use. The operation of the mechanical portions should be tested before and after any necessary greasing and oiling and a speed check made with stroboscopically-marked tape or with a measured length of tape. The brakes should be adjusted, if necessary, relays and switches cleaned as indicated by inspection, and tubes should be tested. After the first rough inspection, the erase and bias currents should be checked against specifications and a noise and distortion test made. If noise tests indicate magnetized heads, they should be demagnetized and the noise and distortion test should be repeated. When this distortion and noise prove satisfactory at 400 cps, a frequency check at the upper limit of the recorder's spectrum should be made in order to check head alignment. Another test, using normal input and output loads, should provide the finishing touch.

The Tape Drive System

Shown in *Fig.* 1 is a typical drive system. The function of this mechanism is to move the tape at a constant speed past the magnetic heads. Anything that causes any aberration from a constant speed will cause audible "flutter" and "wow," defects which, at the present high stage of tape recorder development, should not be tolerated. Typical "flutter" or "wow" specifications run less than 0.1 percent which cannot be heard by even the most discerning listener.

The basic parts of the modern tape drive system are

1. The capstan
2. The capstan pressure-idler
3. The drive motor

In most modern machines the drive capstan is positioned directly after the head assembly so that the tape is pulled through the assembly at a constant speed. In at least one very well designed machine the tape drive is located between the supply reel and the head assembly, and its action serves to feed the tape at constant speed *into* the head assembly. Then the tape is kept taut by the pull of the take-up motor. Whatever system of drive is used, the maintenance engineer should check for any trouble that would make for any variation from *constant longitudinal travel* on the part of the tape. Any "binding" of the tape will result in vibration-modulated sounds being recorded on the tape, plus the accompanying distortion. A slow variation in speed will cause wow, a fairly rapid variation will cause flutter. Any part of the tape-travel path that introduces vibration *in any plane* should be repaired.

The Drive Motor

Modern professional tape recorders utilize hysteresis motors for the drive system. A properly-designed motor of this type, provided that it has sufficient torque for the job, will maintain constant speed within certain limits. The frequency of the a.c. supply should not vary more than one cycle for best results, and the voltage should be above the minimum specified by the manufacturer. Drive motor speed may vary slightly according to temperature, since the grease in the bearings exerts more friction when cold than when warm. It is, therefore, recommended that speed checks be made at normal operating temperatures.

The drive capstan and pressure idler assembly should be checked to make sure it is perpendicular to the tape. If it is out of order in this respect, the tape will try to travel up or down on the capstan, a condition which will be immediately apparent to the operator. The pressure-idler should be checked for proper pressure against the capstan when in operating position. The pressure

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* *Columbia Broadcasting System, New York.*

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should be accurately maintained at the value specified by the manufacturer. Too much pressure may affect speed and cause over-heating of the drive-motor; too little pressure will cause slipping and erratic speed. The rubber-faced pressure idler should be inspected for glazing or any out-of-round condition. Cleaning with alcohol is recommended for the idler facing, as well as for other parts contacted by the tape.

The Magnetic Brake System

In the majority of machines used in broadcasting, the supply reel and the take-up reel are actuated by their own motors. The supply motor field (and in some instances, when in reverse, the take-up motor field), when the machine is recording or reproducing a tape, is hooked up to exert a pressure *opposite* to the direction of tape-travel. The e.m.f. necessary to produce just enough pressure is controllable and should be adjusted so that the tape is held flat against the heads at all times during the normal operations of recording or playing back. Once a machine has been thus adjusted, simple reading of the voltage at the motor terminals should be sufficient as a maintenance check. Note that if this adjustment is not correct the tape will be too taut or too slack. Tautness will cause the tape to curl toward the coated side, thus making for poor recording; obviously, a slack tape will not contact the heads properly, resulting also in poor recording.

Early recording machines designed in Germany coincident with the Magnetophon utilized a mechanical brake on the supply motor both for tape tautness and for rapid braking. The modern tape recorder utilizes a mechanical braking system similar to that on an automobile, except that it is actuated by solenoids. These brakes should be so adjusted that there will be no tendency for tape either to break or to spill.

On forward or rewind high speeds (five to ten times the normal speed) the tape may wind unevenly on its platter or "throw loops." This condition may be due to unevenly cut tape, to eccentricities of tape manufacture, or, most commonly, to a too-high speed. The normal cure is to adjust the rewind and fast-forward speeds downward by reducing voltage at the motor fields. Where it is desirable to keep the fast speed, a cover-plate for one-sided reels should be utilized to insure against spilling the tape.

Present day power supplies for professional and semi-professional tape recorders are well-designed and trouble free. Ordinary care should ensure long life. It would be well to check voltage-regulator tubes and to replace them periodically. They have been known to produce peculiar audio oscillations and flutters. Test all tubes regularly and make sure ventilation is sufficient for the temperature of the room.

The high-frequency oscillator (60–100 kc and over in professional type recorders) is generally a modified Hartley circuit. Its main function is to generate a perfect sine wave. The output of the oscillator may be taken directly to the erase and record heads in the case of semi-professional recorders. However, in most instances the oscillator in broadcast equipment is followed by r.f. amplifiers. It is axiomatic that the more complete the erasure the better the recording will be. There are two points to check periodically in the oscillator system. First, make sure that its output does not contain distortion. Second, see that the output measures up to specifications in wattage delivered. Most machines now incorporate a meter for reading bias and erase current, or if not it can easily be obtained. The manufacturer of the equipment will provide instructions for adding this useful feature.

Note that the bias current mixes with the audio current at the record head or in the transformer feeding the record head. It should not be allowed to get into the audio system at any other point.

Bias Current Adjustment

Bias current should be adjusted only when equipment is available to check its level and the effect on the recorded tape. Generally, in the present day highly-equalized recorder, the bias current setting is a compromise. An increase in bias will improve low-frequency response but cause the high end to drop because of excessive erasure. If, again, the bias is left too low, the low frequencies will distort excessively. (See Part II on "Recording.") When measuring equipment (noise and distortion test set or signal generator and oscilloscope) is not available, the approximate bias level may be found in the following way: Record a low-frequency signal (100–250 cps) at the correct level to the input. Increase the bias until the tape output reads maximum level. Beyond this point there is a choice. If better lows are desired, the bias may be increased still further, but the dynamic range will be compressed by so doing.

The record and playback amplifiers, generally built in separate units, are not extraordinary in design or construction and do not require any extended treatment. The chief troubles to guard against are r.f. interference and tube noise. Radio-frequency noise may be trapped out at the input to the playback amplifier. Tube noise and residual hum should be at least below the level of the tape noise in a well-designed professional machine (at least 55 db below maximum output).

The magnetic heads are the most carefully designed parts of the recorder. They require careful use and maintenance to remain in good working order. Do not run tape at high speed in contact with the heads, because the tape then acts as an abrasive and wears the pole pieces. Heads that are constructed like the one shown in *Fig.* 3 may then

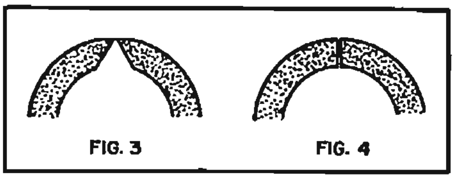

Fig. 3 (left). Type of core assembly liable to damage from tape abrasion. Fig. 4 (right). Core design not injured by abrasion.

need to be replaced. Others, like that of *Fig.* 4, will not be damaged appreciably, but may only require re-alignment.

The head gaps should always be kept in alignment, perpendicular to the tape travel. Any mis-alignment between the record and play heads results in losses, especially at the higher frequencies (see *Fig.* 5, showing exaggerated mis-alignment). All professional recorders incorporate some means of alignment in the head mounting, and the machines should be checked periodically to make certain it is correct. In order to make alignment checks, an audio osillator, a test tape with several high-frequency cuts, and an output VU meter are required. The playback head should be aligned for greatest response at the top frequency. Then a new recording on fresh tape should be made at the same frequency from the oscillator output. The playback output level of this new recording should equal that of the test tape. If it is not equal, the record head should be re-aligned. While making these tests, be certain that the same type of tape is used throughout.

The heads should be kept absolutely clean. They should be cleaned with a lint-free cloth moistened in alcohol every half-hour of use. A coating of dust or oil, even the residue that may be left after recording with a "lubricated" tape, may result in a muddy recording, lacking in highs.

A considerable increase in tape noise may occasionally be noticed which cannot be traced to improperly erased tape, uneven bias-current wave-shape, or a noisy record amplifier. The most likely cause of the noise is a magnetized head, which leaves a d.c. noise-signal on the tape. The instruction book will probably designate the correct procedure for demagnetizing the heads. One good method is to record and playback tone at any frequency (the lower the better), increasing slowly to about four times normal level, holding it there a few seconds, then decreasing slowly to zero. When demagnetizing by this method do not exceed the wattage rating of the heads. Another method is to use a 115-volt a.c. demagnetizer, now available, which requires only a few seconds to neutralize the core structures thoroughly. It is wise to disconnect the heads from their respective amplifiers if an external demagnetizer is used.

The ability of magnetic tape to capture and keep a signal depends upon its physical and magnetic properties. Almost any tape manufactured today will deliver good quality reproduction. Research in tape coatings and plastic processing results in better products every year. However, it is important to adjust the bias for the tape used, and to observe care in the handling of the tape. It should be stored in dust-free cabinets where it is not subjected to any stray magnetic fields. A normal room temperature of 70° F. and a low humidity will ensure its long life.

The problems of maintaining tape recording equipment in the field are similar to those found in studio operations in most respects. The outstanding problem of the recording engineer travelling with his equipment is that of obtaining a satisfactory a.c. supply. The most commonly used supply is a motor generator set driven by storage batteries and regulated manually to 60 cps. How to care for such equipment is known by most broadcast engineers and will not be touched upon here. Magnetization of heads by lightning is naturally much more common in the field than in studio operations, and it is good routine to check *before* playing in order not to ruin a valuable recording. (An ordinary compass may be a valuable device in this operation.) Altogether, maintaining equipment away from the studio requires more ingenuity and experience of the engineer. Unique problems arising from unusual conditions may require unorthodox methods of maintenance, but whatever the conditions, the engineer with the "know-how" of tape recording will do a creditable job.

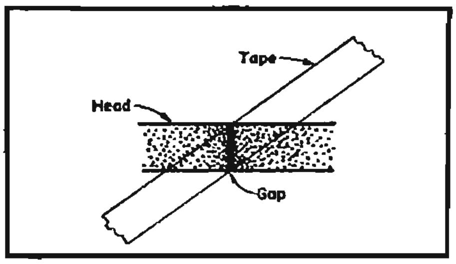

Fig. 5. Exaggerated tape-head misalignment.

Top Tape Recording Performance

HAROLD REED*

A discussion of some of the non-electronic factors affecting quality recording and reproduction of sound.

In addition to maintaining the electronic section of a tape recorder in such condition that it will give the most efficient performance of which it is capable, it is of the utmost importance—in the interest of faithful recording and reproduction of sound—that the mechanical system be kept in peak operating status. This entails checking and testing of the various component parts of the mechanical layout, and especially—if the equipment is subjected to long hours of use, such as occurs in radio broadcasting and recording studios—this inspection and investigation into the competency of the system to approach recognized standards must be frequent.

One of the most important considerations is the alignment of the recording and playback heads. When only one recorder is being used and this machine is employed both to record and to reproduce the program material, and provided the recorder utilizes one head for both recording and reproduction, then some misalignment of the head does not necessarily result in sub-standard performance. In this circumstance the tape is drawn over the same head during both the recording and playback operations and, as the misalignment factor is identical for each operation, quality reproduction response may be obtained. This harmonious state of affairs is not in evidence when several recorders are in service and reproduction may be assigned to any one of the machines, nor when the mechanism employs separate record/playback heads, unless all heads are in proper alignment. Further, inferior results may be obtained when playing tapes received from outside sources.

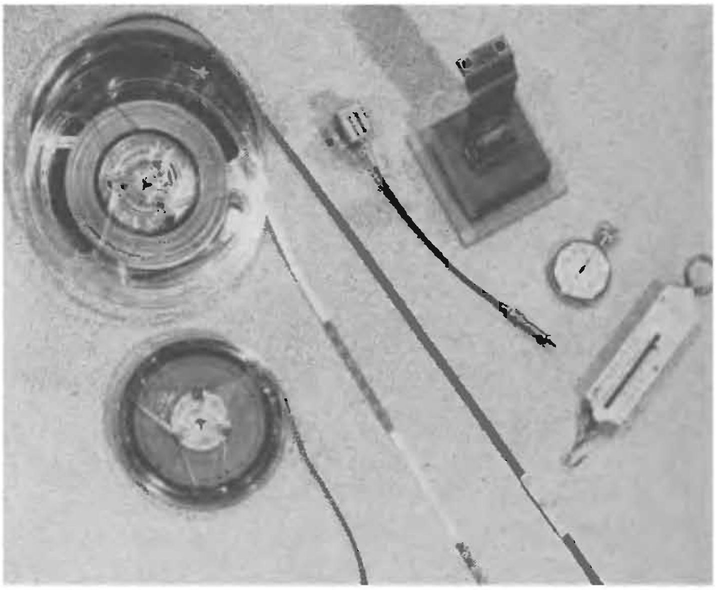

Fig. 1. A collection of several useful items needed in the maintenance of a tape recorder.

Incorrect head alignment manifests itself in poor quality in the form of muddy, distorted response, lacking in brilliance due to deficiency in reproduction of the higher audio frequencies present in the original program material.

In the moderate price range, Magnecord recording equipment has proven to be quite popular and considerable thought and attention has been given to maintaining its mechanism in a high state of efficiency. The procedures outlined in this discussion, although based on tests with this particular mechanism, may generally be applied to other recorders.

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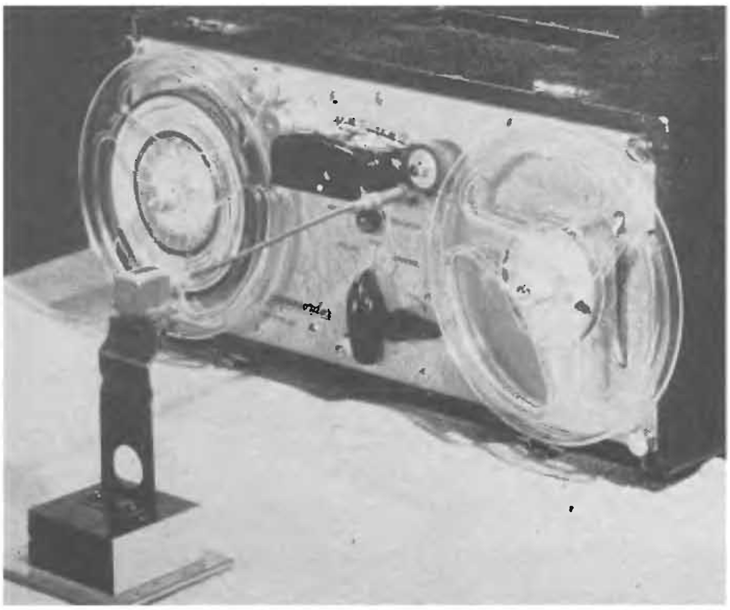

Fig. 2. Stand-mounted Veeder counter is coupled to the capstan by means of a flexible shaft and the fitting described.

"Standard" Tape

To afford interchangeability of tapes between recorders the head gaps must necessarily be in alignment with each other. One suggested procedure, for a recorder with a single record/playback head, is to feed the signal output from an audio oscillator at about 7500 cps for 7½-in. machines, or 15,000 cps for 15-in. machines, into one of the amplifiers, recording this signal for use as a "standard" for all others. This so-called standard tape is then reproduced on each of the other machines and the head adjusted to obtain maximum output from each recorder by observing the volume indicator or other signal level indicating device. The head adjustment consists simply in turning an adjustment screw in or out with a screwdriver to change the physical position of the head gap with respect to the tape, until maximum response is acquired. This procedure will ensure good high-frequency response for all the machines aligned in this manner, even if the unit used as a standard is in misalignment. However, as indicated previously, tapes received from outside sources that may have been recorded on a mechanism in perfect alignment may not faithfully reproduce the program material when played on recorders aligned as indicated above.

A more satisfactory way to accomplish head alignment is to employ a standard alignment tape such as is used in the factories of recorder manufacturers, and obtainable from most large jobbers. It is simply then a matter of adjusting each machine for maximum output as the standard tape moves over the head.

So far we have considered the recorder containing a single head for recording and reproducing, and of course, an erase head. In the case of machines containing a three-head assembly—that is separate record, reproduce, and erase heads—the same standard alignment tape is used but the procedure is not quite as simple as with the single head unit.

The playback head is aligned first. These assemblies contain locking screws at the bottom of both heads, which must first be loosened. Then with the recorder set on playback, and the alignment tape running over the assembly, back off the alignment screw—accessible through a small hole in the front of the case of the head—to establish definitely that the head is out of alignment, as indicated by the volume indicator. The locking screw under the playback head is then tightened, and the alignment screw turned in a clockwise direction to bring the head into alignment, as noted by maximum indication on the meter. If the alignment screw is turned too far—that is, beyond the peak response indication on the volume indicator—it will be necessary to start again at the beginning of the process.

The record head must now be brought into alignment. The standard tape is removed from the recorder and a blank tape installed on the machine. With the standard alignment tape on another recorder, its output is fed into the machine being adjusted, which is now set in RECORD position. Adjustment procedure as described for the playback head is followed, again obtaining maximum indication on the volume indicator connected so as to measure the ouput of the machine being adjusted. This completes adjustment of the three-head assembly. If the second recorder is not available, an audio oscillator can be used as the source of the high-frequency signal for this alignment procedure. On heads without locking screws it is advisable, after alignment, to place a small dab of speaker cement on the adjusting screw head where it rests against the head case, to prevent changes due to vibration of the machine. This cement is easily broken loose when further alignments are to be made.

Worn heads will result in poor frequency response, this condition manifesting itself in a falling off of the higher audio frequencies. The heads can be expected to give satisfactory service for about 1000 hours or more, depending upon such factors as the care with which the mechanism is handled and the operating speed employed. Tape should never be in contact with the heads during the high speed rewinding process. Dirty heads also contribute to a loss of the higher audio frequencies. Some of the minute particles of the tape coating wear off and cling to the head laminations. Thus the heads should be cleaned frequently with a lintless cloth and absolutely clean carbon tetrachloride.

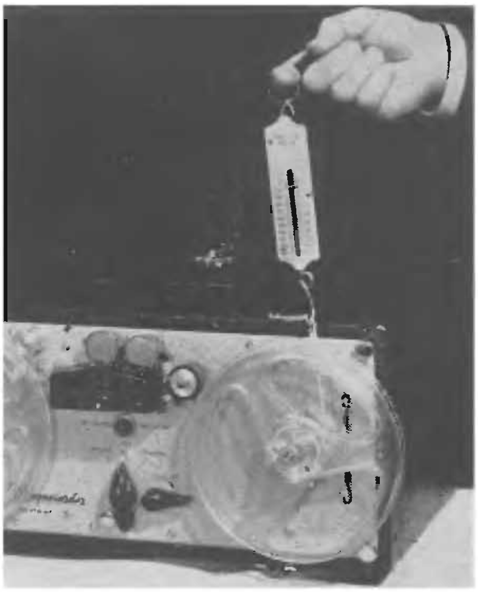

Fig. 3. Method of using string and spring balance to measure the pull of the mechanism in ounces. for later transformation to torque in inch-ounces.

Erase heads normally have considerably greater life than the record/playback heads and usually require no physical adjustment. If the bias oscillator is supplying the proper erase voltage and the tape is not being erased completely, the erase head should be replaced. Heads may be obtained on a revolving stock basis, credit being given for return of the worn head to the factory.

Speed Accuracy

To obtain top performance, the recorder mechanism must be maintained at the correct rotational speed. Without close speed control, not only will timing errors in reproduction occur, but pitch variations may become noticeable. Several methods were tried in an attempt to determine easily the speed of any recorder, to correct speed discrepancies, and to keep timing errors within a reasonable tolerance.

It was estimated that at the popular 7½ in./sec. tape speed, 562.5 feet of tape should pass over the heads in 15 minutes, 1125 feet passing over in 30 minutes. For this test a new recording tape was marked off with a steel tape measure, and white splicing-tape markers were attached at 187.5, 375, 562.5, and 1125 feet, corresponding respectively to 5, 10, 15, and 30 minutes. In normal operation of the mechanism the marker of any particular footage should be drawn over the head at the corresponding time required to pass this amount of tape.

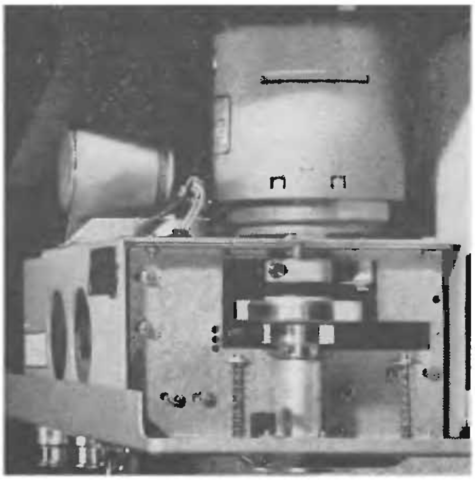

Fig. 4. Take-up assembly of Magnecorder, showing the clutch with the knurled split adjustment nut at the end of the shaft.

Another method is to employ a leader and timing tape which is available in 150-ft. reels in various forms to indicate known lengths, with marking means at fixed intervals. Several of these timing tapes can be spliced together and markers attached at any time intervals desired. *Figure* 1 shows a typical timing tape, a marker on a piece of standard tape, and several other aids to recorder maintenance.

An interesting technique tried in this speed control problem consisted of counting the revolutions per minute of the recorder capstan. In testing new Magnecorders it was learned that the rotational speed of the capstan was about 6.35 revolutions per second, 381 per minute, 5715 per 15 minutes and, 11430 per 30 minutes. A tachometer was used to clock the capstan speed and although a practical method, there was concern about the accuracy because of the possibility of reducing the speed due to the pressure required to engage the tachometer to the capstan.

What was considered a more satisfactory method was in the use of a Veeder counter. The torque necessary to turn the counter is negligible so that no appreciable error occurs when it is connected to the capstan. A short length of ¼-in. copper tubing was used to couple the flexible shaft from the counter to the capstan. With a hack saw, a slot was cut across the tubing diameter so that it fitted in the coupling at the end of the counter flexible drive. A machine screw with the correct thread to fit the capstan shaft, was soldered into the opposite end of the copper tubing. This affords direct drive between the capstan and Veeder counter. With the recorder and counter running, the rotational speed in revolutions per minute is clocked with a stop watch. The counter was fitted with two small banana plugs, which in turn fit into banana jacks mounted on a metal support. This assembly is attached to a wooden base, which rests on the bench or table and eliminates the need for holding the counter during the checking. The arrangement is shown in *Fig.* 2.

Tape Tension

A number of factors contribute to timing errors and speed variations. Clutches at the rear of both the supply and take-up reel spindles are employed to produce the proper tape tension to prevent either throwing or stalling of the tape. Clutch friction on the supply reel should be just great enough to prevent throwing the tape when the mechanism is stopped. Excessive resistance at

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TAPE RECORDING PERFORMANCE

(from page 27)

this point results in the mechanism running slow. The friction supplied by the take-up reel clutch should allow the tape to move at normal speed and prevent tape throwing when the machine is stopped after the rewinding operation. These adjustments can be made by using a small spring balance. With an empty standard reel on the take-up spindle, a length of string should be wound around the reel hub, the other end of the string being attached to the balance. With the spring balance held in a vertical position, as shown in the *Fig.* 3, start the recorder and read the pull of the mechanism in ounces. It is recommended that this tension be from 5 to 6 ounces, and this may be obtained by adjustment of the clutch at the rear of the spindle, *Fig.* 4, to obtain more or less friction, as required. The supply reel clutch should be similiarly adjusted, except that the motor is not used, but the tension is found by placing the empty reel and string on the supply spindle and pulling the spring balance upward, observing the reading in ounces at the instant the reel begins to turn.

These tension values are in inch-ounces, which can be ascertained by multiplying the value indicated by the spring balance by the distance in inches from the center of the reel spindle to the tangency of the string where it leaves the reel hub. In the case of the standard 7-in. reel, this distance is very close to one inch, so that the absolute indication on the spring balance can be taken as being correct. When using the new larger hub reels, the tension in inch-ounces can be obtained by multiplying the pull in ounces as indicated on the spring balance by a factor of 1.375, since the radius of the new hub is approximately 1⅜ in.

The spring attached to the rubber pressure roller that holds the tape in contact with the capstan must also supply the correct tension. Too little pull from this spring results in slippage and may even cause complete stoppage of tape movement. Extreme care must be exercised in oiling the mechanism. The smallest amount of oil on the rubber idler wheels produces serious slippage, with resulting wows and timing discrepancies. Both the pressure roller and idler wheels should be kept clean by wiping with a cloth just dampened with carbon tetrachloride.

Sticky tapes have contributed to speeds variations and, in extremely humid climates, have actually stopped the mechanism. Most presently available tapes are being produced with adequate lubricant in the coating which is resistant to high humidity and temperature. The newer tapes eliminate the squeal often heard as the tape moves over the heads.

The new 7-in. professional reel is another step toward eliminating speed variations and timing discrepancies. This new reel has a hub diameter of 2¾ in. which approaches the NARTB standardized 2½-to-1 ratio specified for the 10½-in. professional reel. In the normal operation of a tape recorder there is less tape tension with a full tape on the supply reel than when the reel is almost empty, and tape speed is faster at the beginning, becoming slower as the end of the tape is reached. The larger hub reduces the differences in tape tension and speed between the beginning and end of the reel, resulting in greater timing accuracy and reduction in pitch changes, noticeable particularly on classical musical selections when, due to the length of the composition, more than one recorder is employed.

Using the techniques outlined, the recorders have been adjusted to a timing accuracy within 5 seconds or less in 30 minutes. To hold this close timing in mechanisms without synchronization, constant surveillance of the mechanism performance is required. It has been observed that greater accuracy results when speed checks are made over longer periods than just a minute or so. It is interesting to compare the results obtained when several methods are employed in any particular test.

Design Of A Professional Tape Recorder

WILLIAM F. BOYLAN* and WILLIAM E. GOLDSTANDT**

Design and operational features of a new high-quality machine.

The present state of the magnetic recording art has made possible the design of recording equipment with extremely high performance standards—excellent frequency response, low distortion and flutter—and so on. While performance standards are of the greatest importance, a survey of individuals and requirements in the professional recording field made it plain that certain purely operational features were very much in demand as well, such as ease of tape handling, precise and instant control of tape motion, good editing facilities, accurate timing, portability, and other similar points.

These factors were taken into prime consideration in the design of the new tape recorder pictured in *Fig.* 1. In addition to fulfilling the high performance standards required in quality professional work, the machine provides extremely quick and easy operation. It consists of two major units, the tape transport and the record-reproduce amplifier unit which includes the high-frequency oscillator. Both major assemblies may be mounted in the console as shown, in a portable carrying case which is furnished, or in a standard 19-inch relay rack.

Tape Transport Unit

The complete tape transport, shown in *Fig.* 2, is constructed on a standard 19-inch rack-mounting panel. The panel is 12½ inches high and the mechanism extends 8 inches behind the panel.

One of the principal design objectives for this project was to produce a mechanism which would afford easy operation and tape handling. In order to accomplish this, considerable effort was directed to the problem of panel layout. Among the people interviewed, the general trend of opinion indicated that the normal tape direction should be from left to right in both relay rack and console mounted machines. This arrangement dictated a front panel 19 inches wide with the height of the panel kept to a minimum to preserve portability.

In any professional recorder certain minimum components should be included in the panel layout. These are a constant-speed capstan and pressure roller assembly, a tape tensioning or hold-back device, a take-up system to spool the tape after it passes the capstan, erase, record, and reproduce heads, compliance arms, and an inertia-stabilizing roller to filter out tape speed irregularities produced as the tape leaves the pay-off reel. The layout of these components not only governs the performance of the unit to a great extent, but also affects the simplicity of operation. In the machine being described the tape path is straightforward and free of loops and curves around guiderollers.

A 3-position function control lever is included. Its positions are marked OPERATE, LOAD, and EDIT. When the lever is placed in the LOAD position the compliance arms, tape guide, and head covers are put in such a position that "straight-line" or "slot" loading of the tape is permitted. This feature eliminates most of the inconvenience normally encountered in tape threading. The head covers are open wide in the LOAD position for ease of head inspection and marking of tape.

After the machine has been loaded with tape, the control lever is placed in the OPERATE position. This brings the compliance arms and head cover back into their normal positions. In the OPERATE position, the tape does not come into contact with the heads except during normal forward operation. This allows the tape to be run at high speed forward or rewind without excessive wear on the heads. When the tape is moving in the normal forward direction, as for playback or record, the tape guide and head cover are solenoid-actuated to cause the tape to engage the heads.

When the control knob is placed in the EDIT position the pushbutton switches are locked out of the circuit, and the tape is held against the heads. If one

Fig. 1. The new Magnecord M-80 recorder in its console cabinet. A portable case is also provided.

* *Assistant Chief Engineer and ** Mechanical Engineer, Magnecord, Inc., 225 West Ohio Street, Chicago 10, Ill.*

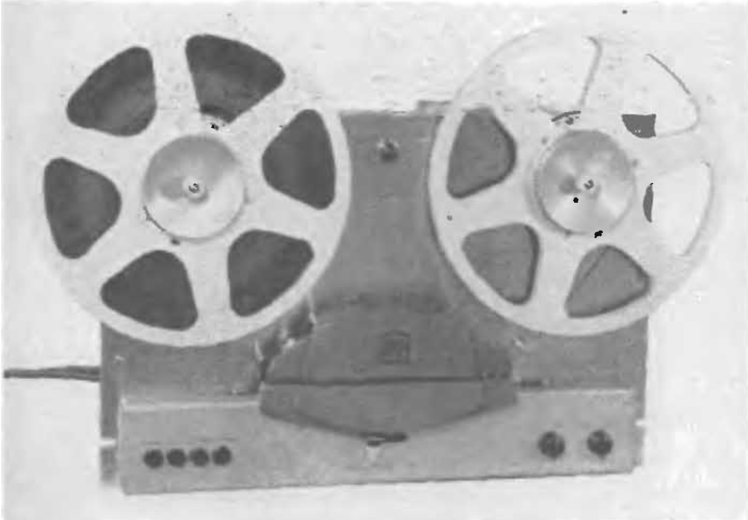

Fig. 2. Top view of the tape transport unit. "Drop-into-slot" tape threading makes for easy operation.

reel hub is then grasped in each hand the tape may be moved back and forth across the heads, permitting easy and rapid cueing.

In addition to the function control lever, other controls located on the front panel include four pushbutton switches labeled REWIND, STOP; FORWARD, and HI FWD (high speed forward). Two rotary switches are located on the lower right-hand side of the panel, one of which controls the tape speed, 7.5 or 15 inches per second. The other rotary switch turns the recorder mechanism on and off.

A record warning indicator light is provided on the transport unit panel which tells the operator when the unit is ready to record. This indicator light is lit when the record-playback switch (located on the front panel of the amplifier) is in RECORD position. When the tape is not in motion the indicator is lit by d.c. from the amplifier, and when the tape is in motion it is excited by the high-frequency bias supply, which operates only when the tape is moving forward at recording speed.

The complete transport unit consists of eight unitized assemblies. These are two reel motor and brake assemblies, capstan-drive assembly, stabilizer roller assembly, head assembly, front-panel assembly, pushbutton control box housing the control relays, and the high-frequency bias-erase oscillator which also includes the power supply for the solenoids.

Each of the two reel assemblies consist of a torque motor mounted directly on the front panel, as illustrated in *Fig.* 3, with a brake assembly mounted on the rear end bell of each motor. Solenoid actuated band brakes provide for the necessary differential braking to keep the tape at constant tension during braking so that it does not throw loops when it is stopped after high-speed running. During normal forward operation a reverse torque is applied to the pay-off motor, which maintains a relatively constant tape tension while the tape is being unwound from the pay-off reel. Slightly greater torque is applied to the take-up motor to provide sufficient torque to spool the tape.

The tape is pulled by a direct-drive assmbly consisting of a 600/1200-r.p.m. hysteresis synchronous motor with an integral ground capstan. A flywheel is mounted on the rear shaft extension of the motor. The capstan drive operates with the same characteristics in the horizontal as in the vertical position. Timing accuracy in the neighborhood of 3 seconds in 30 minutes can be expected.

The stabilizer assembly consists of a flywheel, tape roller and bearing housing, compliance arm guide, and tape-break switch. It filters out effects of tape sticking as well as high-frequency flutter components caused by the pay-off reel.

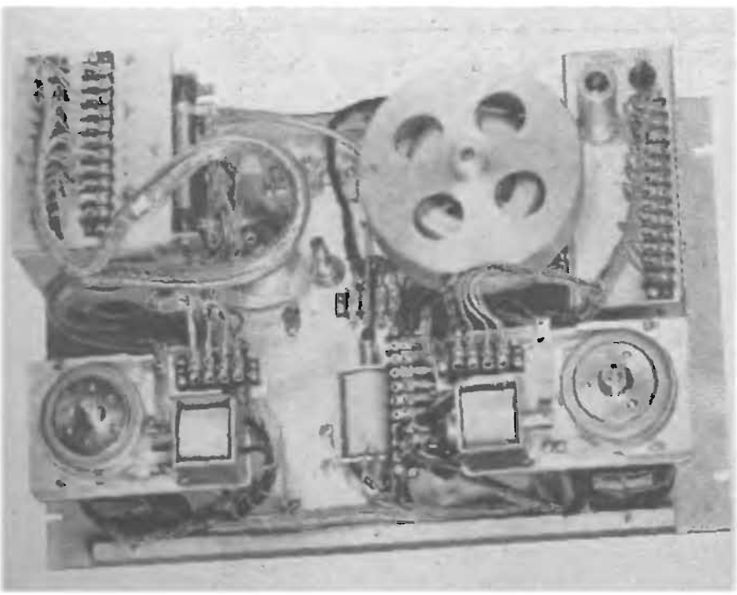

Fig. 3. Underside of the transport unit.

Head Assembly

The high performance characteristics of the M-80 recorder are dependent to a large extent on the precision record and playback heads as well as the head mounting assemblies and tape guides. There are three heads in this assembly, illustrated in *Fig.* 4.

The erase head is a low-impedance device requiring approximately 1 ampere of ultrasonic current. It is of the double-gap variety, which assures complete erasure.

The record head is a new unit developed specifically for this recorder. Its impedance is approximately 50 ohms at 1,000 cps. Precision lapping such as that used in optical work has been employed to assure a very straight and uniform gap. Since recording is done with the trailing edge of the gap, the straightness of the gap edges is more important than the gap width itself. The record head gap length is in the order of .0007 to .001 inch. The reason for this large gap is to reduce the noise recorded on the tape due to residual localized d.c. magnetizations of very small magnitude.

Fig. 4. The three heads are mounted in a complete subassembly.

The reproducer head is similar to the record head in construction except that the gap length is of the order of .00025 inch. The reproduce head is even more critical than the record head and greater care is taken in manufacture to assure uniform gap length as well as gap edges. The impedance of the playback head is approximately 6,500 ohms at 1,000 cps. There are two determining factors which dictate the impedance of the head. As the impedance is increased, greater output voltages are obtained, lowering the requirements of the playback amplifier equivalent input noise. However, as the impedance is increased, lower resonant frequencies of the head and connecting cable are encountered. The resonant frequency of the head and cable should be higher than the highest frequency to be reproduced.

The heads are mounted in triple-shielded Mu-metal cans to reduce hum pick-up. Glass tape guides are employed.

The front panel assembly contains mountings for the control box and oscillator unit, plus the pressure roller assembly, compliance arms, tape guide linkages, and the power on-off switch and tape speed selector switch. *Figure* 3 is a rear view of the unit showing all the assemblies in place.

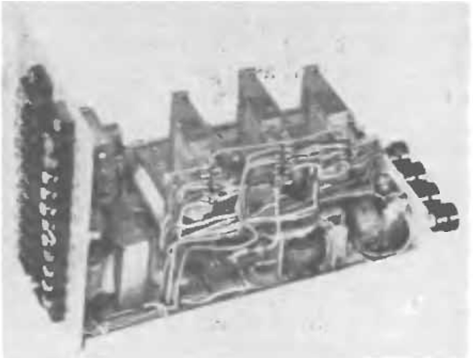

Fig. 5. The control box assembly contains the relays and relay power supply as well as the control buttons.

Control Box Assembly

The control box assembly, shown in *Fig.* 5, is contained in a special mounting shell. All relays, pushbutton switches, and the relay power supply are mounted on a single chassis and are readily accessible for servicing during operation by simply removing the chassis from the mounting assembly. Four relays are located on this chassis, three for normal forward, rewind, and high forward. These three relays have interlock circuits to prevent any two operations from occurring at the same time. To prevent tape breakage, the normal forward relay is locked out of the circuit until the machine is stopped.

An additional time-delay relay—the time delay being acomplished by an R-C network—is included to supply a voltage surge to the take-up motor when the machine is initially started in a normal forward mode. This feature prevents the occurrence of a loop of tape between the capstan and the take-up motor when the machine is started, resulting in smoother starting.

Terminals are provided on the control box assembly for remote control connections.

The oscillator assembly contains the high-frequency erase and bias oscillator circuits, as well as a 115-volt L-C-filtered d.c. power supply used to energize the solenoids. The oscillator is a push-pull Hartley as shown in the schematic of *Fig.* 6, using a 12BH7. This tube has a higher plate dissipation rating than the previously used 12AU7 and should give longer service without replacement. A special, improved, high-Q oscillator coil has been employed, to result in lower even-order harmonic content.

The oscillator coil secondary feeds the erase head in series with an 0.5-μf coupling capacitor. This capacitor has the funtion of resonating the erase head inductance at the operating frequency of the oscillator, which is approximately 70 kc. Approximately one ampere of high-frequency current is supplied to the erase head. The voltage drop caused by the reactance of the 0.5-μf capacitor is used as the supply voltage for the record-head bias. Lower harmonic content is achieved by feeding the record head from a voltage source developed across a capacitive reactance, which results in lower noise being recorded on the tape. The bias current is fed through a rheostat which is used for bias adjustment.

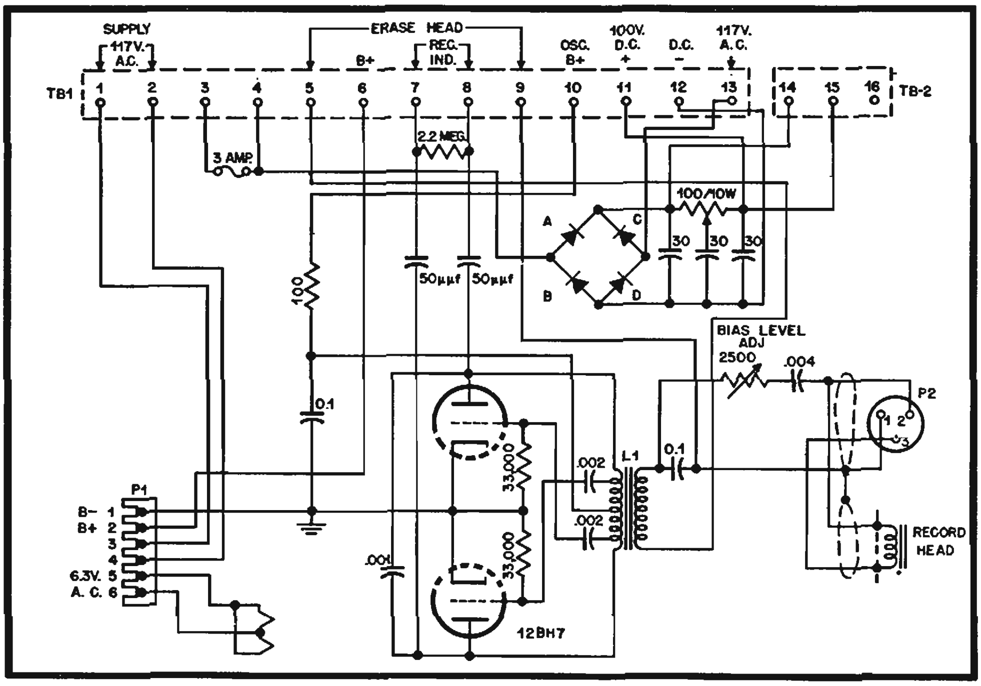

Fig. 6. Schematic of the bias oscillator and relay power supply.

Record And Reproduce Amplifiers

The amplifiers are mounted on the standard 5¼ × 19-inch rack mounting panel shown in *Fig. 7*. *Fig.* 8 shows the chassis mounting the tubes and major components in a horizontal position. The front panel contains the controls. An input selector switch provides for operation from a high-impedance balanced or unbalanced bridge input or a 50-ohm microphone. The VU meter function switch has three positions, record level, playback level, and bias measurement. The record-playback switch includes a REMOTE position allowing the record or playback function to be controlled from a remote location. A monitor jack is provided on the front panel to provide monitoring directly from the program source or from the tape. The monitoring voltage is switched simultaneously with the VU meter. An equalizer switch suits equalization to 7½- or 15-inch-per-second tape speed.

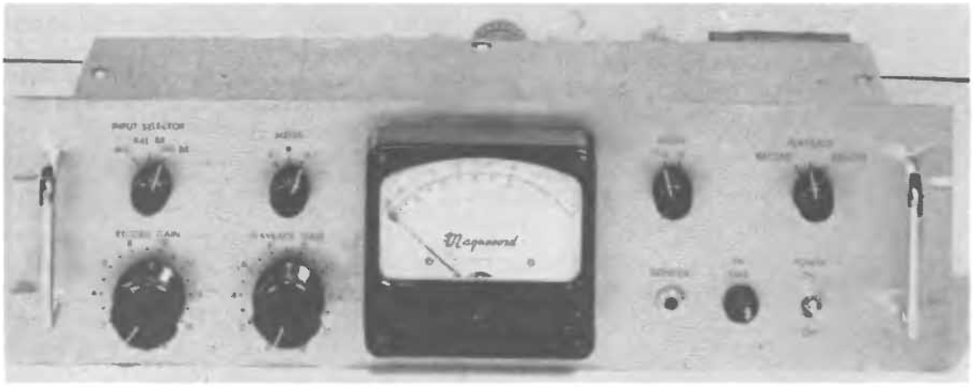

Fig. 7. The amplifier unit is mounted on a 19-inch rack panel.

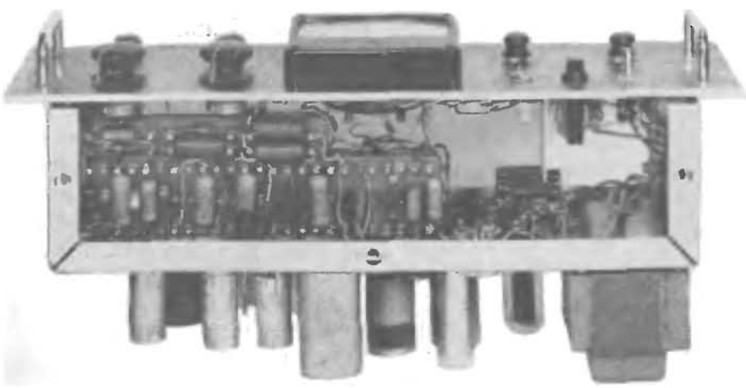

Fig. 8. Side view of the amplifier unit with side panels removed shows "dish mounting" with tubes horizontal. Components are board mounted for easy replacement.

The record amplifier consists of three stages of voltage amplification V_1, V_2, followed by an output stage V_{3a} which supplies current to the record head and contains the pre-equalization.

The input stage is a newly developed form of the cascode circuit described by

(Continued on page 60)

PROFESSIONAL TAPE RECORDER

(*from page 22*)

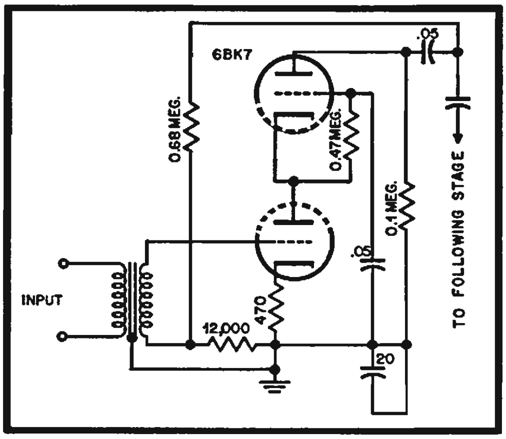

Fig. 9. Schematic of the cascode preamplifier stages in the amplifier section.

Valley and Wallman[1] and shown in *Fig.* 9. The use of this input stage results in lower equivalent input noise than most microphone preamplifiers commercially available. The production units have shown an input noise figure of the order of –127 dbm. The newly developed 6BK7 tube is used. The high amplification factor and low plate impedance of this tube make it well suited for cascode application. The input stage is capable of handling input levels at high as –20 dbm without exceeding 1 per cent total harmonic distortion. The large input signal handling capability and the low noise figure result in an ideal microphone amplifier input stage. Approximately 6 db of inverse voltage feedback is carried around the input stage to increase the level handling capabilities.

The output stage V_{8a} consists of a constant-current, inverse-feedback amplifier, with a large cathode resistor to provide the feedback. Equalization is accomplished in the cathode circuit by employing a small bypass capacitor across the cathode resistor, resulting in higher gain with higher frequency. This method is used for the 15-inch tape speed. At 7½ inches a greater slope of the equalizer curve is required and a series resonant circuit is used to shunt the cathode resistor. Equalizations are of the order of 8 db at 15 kc for 15 ips, and 16 db at 15 kc for 7½ ips.

The response of the recording amplifier is flat from approximately 30 cps to the point where the high-frequency preemphasis begins. It is the opinion of the authors that low-frequency pre-emphasis should not be used in the record amplifier, to eliminate the possibility of overloading the tape with high-level, low-frequency tones.

The plate of the output stage connects directly through an 0.5 μf capacitor to the record head. A potentiometer is provided in the ground return of the record-head circuit. The bias current passes through this potentiometer, and a voltage proportional to the bias current is developed across it. A portion of this voltage is tapped off and fed to the VU meter as the bias reading.

A second voltage amplifier also feeds a cathode-follower stage which supplies record monitoring voltage to the VU meter and to the phone jack. The output impedance of the cathode follower is approximately 600 ohms. This impedance, together with the 3,300-ohm VU meter series resistor, constitutes the proper impedance and time constant for proper damping of the VU meter. The over-all apparent gain of the record amplifier at the microphone input is 95 db. This means than an input level of –95 dbm is required for a zero level recording.

Reproduce Amplifier

The output voltage from the reproducer head is fed directly to the first grid of a 12AU7 input stage connected in a cascode arrangement. Several types of tubes were tried in the input stage; these included a low-noise pentode and various dual triodes. However, it was found that the 12AU7 connected in the cascode arrangement resulted in the lowest over-all noise figure. The output of this stage feeds the first section of a 12AX7 dual-triode cascode amplifier.

The output of the first section of the 12AX7 is fed back through an inverse feedback loop to the cathode of the input stage. This is a frequency-selective voltage feedback network which results in over-all frequency characteristics very closely in agreement with the standard NARTB playback curve. This curve consists of approximately 25 db of additional gain at 50 cps above the 1000-cps gain, while the gain at 15 kc is approximately 10 db below the 1-kc gain.

The over-all gain of the playback amplifier is approximately 75 db at 1000 cps. There is an additional gain of approximately 25 db at 50 cps. Since most of the noise consists of low-frequency components, the gain affecting the noise output is of the order of 95 db. Calculations will show that the equivalent input noise due to the input stage is phenomenally low. This is a criterion which must be satisfied to achieve over-all high record-playback signal-to-noise ratios. It may be safely stated that with any high-quality recorder the determining noise should be confined strictly to the tape output noise, and the amplifier noise should be held at least 6 db below that.

The playback gain control is located between the second and third stages of the amplifier. The gain control is followed by two stages of triode amplification and an output stage consisting of one-half of a 12AU7. The output transformer has two 600-ohm secondary windings, one of which is carried directly to the output terminal, a barrier strip on the rear of the chassis. The other secondary winding is used for developing constant-voltage inverse feedback around the output and driver stage, and also to supply voltage to the monitor jack and the VU meter. There is approximately 20 db of feedback around the output and driver stages, which results in distortions of less than 1% at the full rated output of +16 dbm.

When the equalizer switch is in the 7.5 position an R-C network is connected across the cathode of the driver stage to increase the high-frequency response by about 6 db at 15 kc so that frequencies up to 15 kc may be played back at 7½ inches per second.

The power supply for the amplifiers consists of a transformer-fed, single-phase, fullwave rectifier, followed by an R-C low-pass filter. Filament voltage for the input stages of each of the two amplifier sections is supplied from a rectified 18-volt winding of the power transformer to maintain low hum content. A.c. power for the amplifier is obtained from the tape transport unit through a power cable and connector. The plate and filament voltages are supplied to the bias oscillator on the tape transport unit through this power cable.

Performance Figures

The over-all performance of the M-80 is well within professional standards, not only in an electrical and mechanical sense, but also in terms of fulfilling professional operational requirements. Starting time for the tape is less than 0.1 second and stopping takes place within 2 inches of tape when using the operating speeds. Flutter and wow are under 0.1 per cent r.m.s. at 15 ips and less than 0.15 per cent at 7.5 ips. The signal-to-noise ratio is better than 58 db, based on a level giving a maximum of 3 per cent harmonic distortion.

The frequency response for the over-all recording-playback operation is shown by the two curves of *Fig.* 10 to be within 2 db from 30 to 15,000 cps at 15 ips, and within 4 db at 7.5 ips. The total harmonic distortion for maximum indicated level of 0 VU on both record and playback is less than 1 per cent.

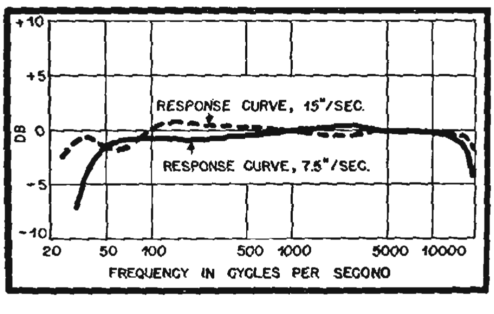

Fig. 10. Response curves of the M-80 at 7.5 and 15 inches per second.

[1] Valley and Wallman, *"Vacuum Tube Amplifiers,"* McGraw-Hill Book Co., 1948.

"PRESENCE"

(*from page 57*)

R_{15}	50,000 ohms, linear pot, (combined with Sw_3)
R_{17}	1800 ohms, ½ watt
R_{20}	18,000 ohms, ½ watt
$R_{21a,\ b}$	Centralab Senior Compentrol
R_{23}	1000 ohms, 1 watt
R_{24}	6800 ohms, 1 watt
R_{25}	13,000 ohms, ½ watt
R_{26}, R_{30}	11,000 ohms, ½ watt
R_{27}	8200 ohms, ½ watt
R_{29}	12,000 ohms, ½ watt
R_{31}	10,000 ohms, ½ watt
R_{32}	7500 ohms, ½ watt
R_{33}	5600 ohms, ½ watt
R_{35}	10,000 ohms, 1 watt
R_{36}	5600 ohms, 2 watt
R_{37}	3900 ohms, 2 watt
R_{38}	1.8 meg, 1 watt
Sw_1, Sw_2	Preamplifier switch, Centralab 30c type, special (see text)
Sw_3	Centralab 30a type, combined with R_{15}, special.
J_1, J_2, J_3	Amphenol 80-PC2F receptacle
J_4	Amphenol 80-C receptacle
Tone Control	Centralab Couplate No. 12614, with 1.0-meg front section and 0.5 meg tapped rear section dual-concentric controls
V_1, V_2	Genelex Z729
V_3	12AX7
V_4	12AU7

A New Idea for Measuring Wow and Flutter

Robert E. Berglas*

Before describing the system I have worked out for an accurate measurement of Wow and Flutter using digital counting techniques, some attention should be given to the weaknesses inherent in existing methods.

Present methods do not lend themselves to a central criterion of scientific investigation: namely, reproducibility of data. Given the art as it exists, published results are more a function of the particular test instruments, than the actual matter under review. Printed Wow and Flutter specs simply cannot be reproduced unless the manufacturer's set-up is known (and his test equipment happens also to be on the user's bench).

Recognizing this embarrassing fact of engineering life, the industry has attempted to bring some method to this madness by hitting upon a mind-boggling variety of "weightings" and *a priori* suffixes. In sum, these are equations which lop off equipment eccentricities and, unfortunately, "fudge" results.

To be more specific, consider for a moment the numerical manner researchers devised for specifying degree of speed variation—of course, the cause of Wow and Flutter. The unit of measurement became RMS percentage of speed variations. And, yet, the RMS notion is no more than a convenience—a) to transform a periodically changing quantity into a steady quantity, and, b) a convenient average value that subverts more than it explains.

Fundamentally, Wow and Flutter is *frequency modulated* (FM) program material; and the RMS concept is, at heart, amplitude modulation (AM) of an AC voltage. It simply will not do to transform peak-to-peak voltages into "peak-to-peak percentage speed error." It may be convenient, but it is also a fudge beyond the definitions we are working with.

Another inproportionate conversion takes place within existing Wow and Flutter meters—i.e., the conversion of AC to DC (to drive the meter). The DC voltage in these instruments is generated by a "detector." The problem here is the same as was for the RMS concept—namely, that this is *not* "detection" in the true sense. Detection refers to the recovery of amplitude modulation from an RF envelope. Wow and Flutter components, however, change the *frequency* of the incoming signal, and converting to DC is inappropriate and not proportional. In the process of making the meter happy, investigators have fundamentally confused AM with FM.

A New Digital Wow and Flutter Method

Analog devices such as meters are fine for measuring voltages; but they just cannot contend with rapid fluctuations in frequency.

With recent advances in digital IC propriety devices, we have at our command, and for the first time, accurate and precise systems for frequency determination. Indeed, my friends at Texas Instruments developed their remarkable TTL digital IC line largely for counting purposes, where, often, the input would be variations in frequency.

The new digital method for measuring Wow and Flutter that is described here, is made at once simple and incredibly accurate by pressing into service the latest generation of Frequency Counters, built around TTL logic. With this instrument and a suitable stable test signal, frequency modulated variations in turntable and tape recording gear can be measured with repeatability and authority. The procedure described is simple, easy to perform, and, to tell the truth, a remarkable experience for the investigator.

Wow and Flutter components will frequency modulate a test signal, and if we monitor resulting output tones with a Frequency Counter, we will not see displacements from the standard input signal.

Anticipation and an understanding of the phenomenon serves to prepare one for the actual procedure. Wow and Flutter give off specific frequency variations: Wow is known to affect frequencies under 6Hz; while Flutter exists in a higher range, up to 250Hz. Further, these frequencies modulate very often both above and below the test signal (known as positive and negative speed variations).

Let us say we have a 1000Hz test signal, either on a record (say, the CBS BTR-150, or the one put out by Stereo Review), or recorded on a test tape. Moreover, let x equal a possible wow component of 6Hz or less; and y equal flutter existing at up to 250Hz. Then, if we monitor the playback on either a turntable or tape machine, we will get the following interesting phenomena:

$$\text{Wow} \ldots\ 1000 \pm x;$$
and
$$\text{Flutter} \ldots\ 1000 \pm y;$$

The readout usually starts off, obviously, with the 1000Hz reference. But, then, with transients which repeat themselves with periodicity, the modulated compromise is flashed out as the instrument is toggled by variations in count. We can now calculate the true percentage of speed variation, manifest as percent FM:

$$\frac{(1000 \pm x) - x}{(1000 \pm x) + x} \times 100$$

Flutter determination is similar.

Choice of Frequency Counter

The Heath Company today puts out some of the most advanced and inviting Frequency Counters on the market. They are designed around TI's TTL logic IC's, and a first generation Schlumberger unit was used by the author.

(Continued on page 80)

*University of California

(Continued from page 26)

A word of explanation should be noted about other Counters. There is some difference between the general purpose Counter/Period Counter, and one designed specifically for high-speed frequency work.

For one thing, the readout should be non-blinking, and reflect both a continuation of count and change only when the input's frequency varies. The unit, then should be *latched*. We do not want the readout to reveal a count-up or count-down. With latched decade counters, the display continues to put out a given frequency while a new one is being counted. With the general purpose Counter, the count period is a divided down timing interval of the clock frequency.

The input shaping circuitry show some differences, too. Counters/Period Counters use a comparator input which is best suited for reducing noise or zero-crossing, while the other kind of unit uses well designed amplifiers and a precision Schmitt Trigger. The latter is better suited for rapidly and idiocincratically changing frequencies.

Readers who try out the above test procedure will probably notice what appears to be either a mis-cue or falsely recorded as FM of the least significant digit. The "$\pm$ 1" flicker of this digit is inherent to most Frequency Counters, and should not be noted down as a speed variation. (It occurs when the count gate opens and a plus-or-minus one-count ambiguity exists.)

Present specs are far from perfect. The DIN is not really an improvement. RB's idea is an interesting one but it has snags too.—*Ed.*

BEHIND THE SCENES

BERT WHYTE

8-Track Tape Cartridges

RCA VICTOR RECENTLY announced that it had produced its eight-millionth Stereo-8 tape cartridge since the introduction of this format three years ago. As of a few months ago, Ampex was producing in excess of 20,000 8-track cartridges *per day!* Other duplicating companies around the country are reporting production statistics which are equally impressive. To an old reel-to-reel man like myself, these figures are staggering. In fact, RCA states that its Stereo-8 sales over the past three years are *five times as great* as all the reel-to-reel tape sales for the past fourteen years!

Obviously, the 8-track tape cartridge is a howling success, in spite of the gloomy predictions in certain quarters that it would fail because of the "gimmicky format," "impossible to maintain production tolerances," "mechanical difficulties," etc., etc. Actually, many of these criticisms had a certain degree of validity and there were unforeseen problems, many of which have yet to be resolved. Unquestionably, the stimulus of the multi-million car market and its vast profit potential resulted in a money-is-no-object approach to solving the major problems, and the production facilities today are far more sophisticated than they were in 1965. Naturally, the quality of 8-track tape cartridges varies from company to company. Some are better than others in terms of hiss, crosstalk, and other distortions. However, I think it would be safe to say that, at least as far as *production* is concerned, most 8-track tape problems are "manageable." It is when we get to the nitty-gritty nuts-and-bolts aspect of the playback of 8-track tape cartridges in the car or home that we run into some problems, some of which are unique.

It is always good practice to examine a new tape cartridge carefully before you insert it into the slot of your playback machine. Check the pinch roller, to determine if it revolves freely. Ninety nine per cent of the time it will be all right, but you will occasionally find the pinch roller frozen in its bearing. When you insert a cartridge and after a reasonable period you don't hear any sound, chances are that it is this pinch roller problem. Also check the cartridge tape for proper slack. Pull the tape over the pinch roller, with your thumb simulating the capstan shaft. The tape should move freely down into the cartridge and onto the reel. If the tape doesn't move, insert a pencil under the tape and pull upwards, forming a loop of tape you can grasp. Sometimes a firm pull will free the tape sufficiently to make the cartridge usable. If reasonable pressure won't free the tape, further effort is useless and you must seek recourse with your dealer. You will also encounter cartridges which are operable, but the little felts of the pressure pads are missing and the bare metal rubbing against the tape can cause trouble.

Now picture this situation. You've checked a cartridge and everything appears to be okay. So you insert it into your machine and you are tooling down the highway blissfully enjoying your Sinatra in stereo. All of a sudden there is a horrible sound . . . Awk! Grunnnchh! . . . and finally no Sinatra, no sound at all. You pull out the cartridge. At last you try to pull it out and find to your chagrin that you have two strands of tape coming out of the cartridge and the apex of the strands appears to be caught in the machine. Zounds, what a mess!

Well, first things first. Let me assure you that there is no way to extricate the tape cartridge without cutting it free. I know. I've tried. Not once but three times this has happened to me and on three different machines. So steel yourself and *cut* the tape. *Don't* try to break the tape with your hands. There is a good reason for this I'll tell you about a little later. Now that the cartridge is out of the way, if you can look into the tape slot, you'll find that the tape has wrapped itself around the capstan shaft! What happened is this: most 8-track playback machines have a drive system wherein a heavy flywheel is part of the capstan shaft and it is used to help smooth tape motion. The capstan shaft is under the center of the flywheel, perpendicular to it, and seated in a bearing situated in the bottom part of the tape machine structure. The flywheel rides in a bearing on top of a yoke, which is connected to the side of the bottom bearing and in which the capstan shaft "rides." Apparently this yoke supports the flywheel and affords a degree of rigidity to help prevent wobble in the shaft. This yoke is not a true bearing and of course, it isn't lubricated. Tolerances are fairly tight, but there is enough clearance between the capstan shaft and the yoke that under certain circumstances the tape is caught between the shaft and the yoke and quickly wraps itself around the shaft. When the clearance is exceeded, the shaft seizes in the yoke.

Why does this happen? I've asked quite a few "experts" and have received quite a variety of answers. The concensus seems to favor the idea that something impedes the smooth flow of tape between the pinch roller and capstan shaft, momentarily "snagging" on the shaft, but enough to upset the equilibrium of the endless loop and thus get wound around the shaft. One expert blamed it on the dynamics of the endless loop itself. As you know the 8-track tape is back-lubricated with graphite. While this is pretty slick stuff, I suppose it is conceivable that some sort of bump or dirt or other irregularity on the back surface of the tape could interrupt the motion and cause a snag. One other idea was that as the tape is played back many times, the edges become "scalloped" and this could upset the dynamics. Mebbe so . . . but I have had it happen with a brand new tape.

One of the most logical-sounding reasons was that of a "bleeding" splice in itself, or this kind of splice worsened by excessive heat. No doubt you are aware that the automatic switching of the four stereo sequences is accomplished by means of a foil contact strip. It probably wouldn't take much "bleeding" or "oozing" of the adhesive under the edges of the foil strip to cause a snag. It is claimed that when the playback machines are used for fairly long periods of time, they get quite warm. This is especially true of the machines built into the dashboards of new cars as original equipment, where lack of ventilation is alleged to cause temperatures in excess of 130 deg. If this temperature is beyond the thermal tolerance of the adhesive it naturally would aggravate the "snagging" problem. At the moment I have no way of knowing if this heat problem is valid. I rather doubt it, but my friends at 3M may be able to cast some light on the subject. Still another opinion was that with the extreme bending angles the tape in the cartridge is subjected to, it is easy to form small "kinks" in the tape which could be pulled behind the capstan shaft. Since this opinion was ventured by a rabid supporter of the cassette format, I tend to discount this possibility.

No matter what the cause, when the tape gets wound around the capstan shaft, you're in trouble. If your playback machine is of the "add on" variety, at least you can remove the unit from the car and work on it at home or transport it to a repair station. If your playback machine is original equipment built into the car, this "tape wrap" problem can be financially catastrophic. According to Wally's Tape City, a New York firm specializing in 8-track car installations, in some cars it is necessary to remove the unit, repair and check it and re-install it in the dashboard. The tab for this can reach 30 or 40 dollars!

Fortunately, there are some things you can do yourself before resorting to professional help. Let's assume you have the kind of unit you can remove from your car easily and take to a work table with a good strong light. Next thing is to get an injector-type razor blade. In a vise or between two pliers break off a piece of the blade approximately ⅜ths of an inch (avert your eyes when you do this). Now grasp the blade with long-nose pliers, insert into the tape slot and carefully cut through the layers of tape. While the shaft is hardened steel, don't take a chance of scoring it by cutting with too much pressure. After cutting the tape, use the long-nose pliers to grasp the loose ends and a good solid tug will usually free the tape from the shaft. An alternative method is to use a long screwdriver to cut the tape. However, most screwdrivers aren't sharp enough to do this without a lot of hacking, and must be given a sharp edge with a file or grindstone. Needless to say, the same techniques can be used on machine built into the dashboards, but you need a good trouble-light or flashlight, a lot of physical agility and luck. If you can't cut the tape in an "add-on" machine, it will be necessary to disassemble the unit to the point where the flywheel/capstan shaft can be lifted out of the yoke and bearing. I must warn you that there are a lot of parts packed into an 8-track cartridge player, some of which are held in tension by the casing. To tackle this job without at least an exploded diagram and considerable dexterity is a chancy business.

Can anything be done to salvage the damaged cartridge? The tape can be spliced like any reel-to-reel tape, with, of course, special attention to the splicing tape for any signs of excess adhesive. You will remember that earlier in this article I cautioned against breaking the tape by hand in order to free the cartridge. This is because most 8-track tapes are made of Mylar and pulling it results in stretching and "tubing," which would destroy more program material than merely cutting the tape. Naturally, there will be music missing at the splice point, because of the tape which was destroyed in wrapping around the capstan shaft. Since we are dealing with the 8-track, endless-loop format the music will be missing at the same point in all four sequences on the tape. Since cartridges range from 6 to 9 dollars each, perhaps you won't be too distressed by the missing material, especially if it's pop music. Hopefully, you may never have to contend with this tape problem, but it evidently happens with sufficient frequency to warrant a thorough investigation by the interested companies and followed by instituting remedial measures in either the cartridges or playback units or both.

There are other playback problems with 8-track cartridge machines. Many of the earlier units had hum and excessive motor noise, of a cyclic variety. What may be termed "second generation" units are much quieter and have better tape motion. In either case, it's a good idea to check the grounds. In machines installed in cars, not only should there be a good ground wire from the tape machine case to the car chassis, but ground points throughout the car should be checked, as they are often broken. In home stereo 8-track units, be sure to run a ground wire from the case to the input of your preamplifier.

One other aspect of the newer 8-track playback machines is that they have an output impedance of 8 ohms, rather than the 2- to 4-ohms impedance of the first playback units. This enables a much greater choice in speakers and the opportunity to employ some really high - quality speakers. Installation space varies among the various car models, but the high fidelity 8-in. speakers made by Wharfedale, Jim Lansing, University, Jensen, and Electro-Voice are shallow enough to make their use possible. I know they will fit into the storage wells in my Corvette, and I intend to investigate these speakers at the earliest opportunity. Bass response remains the major problem in car installations. In most cases the lack of bass is an affront to the musical balance and the ear. The worst situation is where tiny "pre-packaged" speaker/baffles are installed under the dash or on the kick panels or above the sun visors. The sound coming from these abominations is a shrill caricature of the original recording. The car doors are a favorite place for speaker installation and will certainly afford improved bass. Make sure there is plenty of fiberglass stuffed behind the speakers. This seems to help the bass response and also helps to subdue the metallic resonances the speakers excite in the doors of certain cars.

Another aspect of bass response will have to be resolved by the manufacturers of the playback units. I refer to their lamentable practice of furnishing their machines with the type of tone control that apparently increases bass response by attenuating treble response. I would like to see them adopt a fixed standard treble equalization (most car interiors don't vary much in terms of reflection and absorption) and furnish a separate bass control. To anticipate a question, compensatory equalization of the recording to suit the environment of the car has been tried, and is being used to some extent. For a variety of reasons it is only marginally successful, usually more trouble than it's worth . . . the result is generally mishmash. . . .

Like everything else, the playback problems of the 8-track stereo tape cartridge will be resolved by time and money. For the most part, considering the difficult, even exotic configuration of the 8-track cartridge format, it works remarkably well, with a high degree of consistency. The problems that have been discussed are minor in scope, considering the millions of cartridges in use. Nonetheless, these shortcomings do exist and ignoring them won't make them go away.

One final note—and certainly a dubious testimony to the success and popularity of the 8-track tape cartridge—is that the playback units have become the favorite targets of car burglars. I have been a victim and I understand quite a few RCA people have had machines stolen from their cars. A check with my insurance agent reveals this to be a common and ever-growing occurence. It would appear that when an 8-track stereo tape cartridge system is installed in a car, it is prudent to install a burglar alarm as well.

Off-the-air Recording

Some months ago, Mr. Norman Racusin, Vice President and General Manager of RCA Victor, made a speech at a tape symposium in which he voiced alarm at the growing practice of "off the air" recording by means of the stereo tape cassette. His contention was that the cassette has made this type of recording so simple that it can be done by anyone including the teenagers who constitute the majority of the record-buying public. He saw this as a definite threat to the future of the record business. He cited the figures for the cassette business in Britain, 50 per cent of sales being for blank tape cassettes. He stated that if this "off the air" recording trend became really widespread, who would pay for recordings produced by the record companies? In his conclusion, Mr. Racusin said that in order to combat this problem, it might be desirable for engineers to devise some technique which could make a broadcast signal either impossible or extremely difficult to record, without, of course, altering the musical content of the program.

Is Mr. Racusin being unduly concerned, or is his speech a timely warning? There will be those who will raise a supercilious eyebrow about this matter since Mr. Racusin's company introduced the 8-track stereo tape cartridge, which, as you know, is almost exclusively a playback medium. Well, it is not my function to impugn Mr. Racusin's motives. As far as I am concerned, he raises some interesting points that merit serious consideration.

I was sent a copy of Mr. Racusin's speech and in addition, a reprint of an advertisement by Harman-Kardon, the message of which would give palpitations to any record company executive. The ad concerned the Harman-Kardon SC2520, a modular system which combines a stereo record player, stereo FM receiver and a built-in stereo cassette recorder. Emblazoned across the top of the ad was this eyecatcher: YOU MAY NEVER BUY ANOTHER RECORD! The ad copy told how easy and simple it was to record programs off the air, or from a friend's disc recording via the cassette recorder, all with high fidelity quality. I believe a half-dozen blank tape cassettes were being offered as a bonus for purchasing the system. It is easy to appreciate the shock value of the ad blurb, but at a selling price of $479, this isn't going to constitute such a vast market that it is a threat to the record industry.

Let's look at the other end of the scale. It is possible to buy an AM radio with a built-in monophonic cassette recorder for as little as $49. There are AM/FM portable radios with recorders for $99 and on up to 4 band portables with recorders for $149. It is obvious that even for the affluent American teenager, this is "a lot of bread, man!" The ubiquitous transistor radios which seem permanently grafted to the ears of the kids, like some monstrous Orwellian fantasy, range between $5 and $15. This is what they buy for themselves. The jazzier units are usually presents from parents, and it would appear that "dear old dad" would have to furnish the wherewithal for these radio/cassette combinations. While North Americans are probably the most doting and indulgent parents on earth, it seems hard to conceive that they would buy these for their offspring in such quantity as to upset the record market. However, let's keep an open mind and delve a little deeper into the teenage economy.

The kids buy their pop/rock 45-rpm recordings for an average cost of 69 to 79 cents each. There is one selection on each side averaging 3 minutes in duration. So for their money they get about 6 minutes of music. Now you can buy a 30-minute blank tape cassette as cheap as 99 cents. If the kids record it properly they can get five times as much music—at least 10 selections—for 20 or 30 cents more than they paid for a single 45-rpm recording. Consider too, that most of the pop/rock music the kids listen to is very ephemeral in nature—a big popular hit for a few months, even weeks—and then it fades into limbo. Thus when the kids tire of their material or it is displaced from the "top forty," they can record new hits on the same cassette. The way the kids devour records, the cassette approach could save them quite a bit of money and the smarter kids (and their parents) might be encouraged to make the "larger-than-usual" purchase of the radio/cassette recorder figuring that it would soon pay for itself.

This all sounds very rosy for the young'uns, but what about the sound quality of the "off the air" cassette recordings? I've listened to these cheap radio/cassette units and, speaking personally, I could be very crass and give all I've heard a blanket indictment as utter sonic horrors. About the kindest thing that could be said about the better units, is that they have somewhat less distortion than the others, which affords a little better clarity. Actually the cassette recorders are mirroring the source quite well, the playback offering the same quality of music and distortion as the radio. You've heard this kind of sound before . . . tinny, peaky, shrill, nasal, and utterly devoid of any bass response below 150-200 hertz. We must not forget however, that kids are used to this sonic assault. Whether the 45-rpm records they buy sound better through their parents' "mahogany monstrosities"—the typical living room consoles—than their cassette recordings, is a moot point. It is sad in a way—in spite of the rock content, most of which I can't stand, you would be surprised how much high-quality sound you can get from the 45's made by the better labels, when played back over a good component system. In any case, one tends to think that the kids would be satisfied with the quality of the cassette off-the-air recordings. But even conceding this and spelling out the monetary advantages, I still can't see this growing to sufficient volume to depress the record market.

However, for the sake of argument, let's assume that the off-the-air recording grew to epidemic proportions, what could be done to stop the practice? Mr. Racusin's suggestion of some technique to alter the broadcast signal to render it unrecordable, seems impractical. I have discussed this point with AUDIO's own Len Feldman and with Murray Crosby, both eminent radio engineering authorities, and they cannot conceive of anything that would work. In any case, there would be quite a legal and political ruckus involving the FCC and the public if such signal "doctoring" were attempted.

As you know, off-the-air recording via (open) reel-to-reel machines has been going on for years. Mr. Racusin tends to downgrade the reel-to-reel practitioners as being too few, because the techniques involved are considerably more complicated than obtained with cassette recorders. I feel there is more money being lost to the reel-to-reel enthusiasts, because of their ability to make relatively high-quality recordings. This is especially the case with albums of stereo mood music. Given a good quality receiver or FM stereo tuner and a stereo recorder, one can record this type of music *ad infinitum*, with a tremendous variety of program material from which to choose, and at substantially lower cost than buying the equivalent disc recordings. Many also record the classical material, but here the quality differential is on the side of the disc, a fact recognized by most off-the-air recordists.

The legal aspects of off-the-air recording have never been clearly defined. There is a sort of tacit understanding that any recording made off-the-air intended for personal use and not offered for sale is legal. Copying of pre-recorded tapes is technically illegal and some tape boxes proclaim this warning. But whether it is cassette or reel-to-reel, off-the-air or commercial tape, and it is all declared illegal, how on earth could you ever police such activities? It is patently impossible.

In summation, if off-the-air recording ever did become an economic threat to the record business, the record companies would have only themselves to blame. They have sown the seeds of their own destruction by not charging the broadcasters a fee for using their recordings. They have even aided and abetted their downfall by giving free records to the stations. This country is one of the few where stations can broadcast recordings without paying for the privilege. I know this is heresy, but the record companies think they

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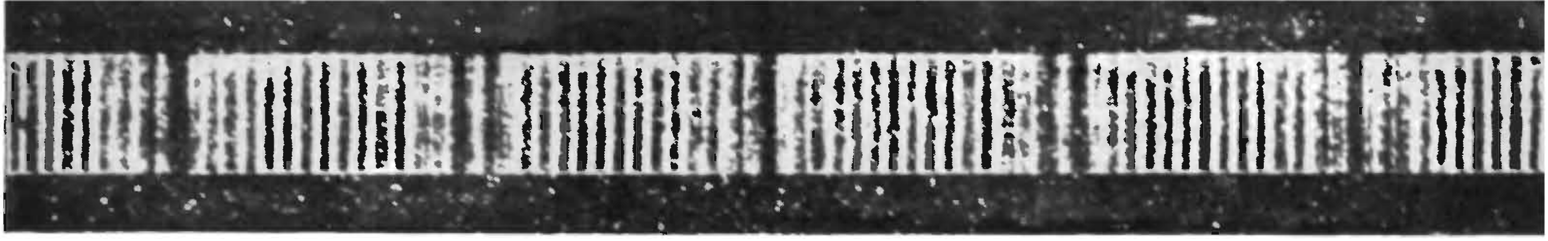
must have exposure for their products in order to ensure sales. Stop and think a minute. What would one helluva lot of stations use for program material if they didn't have free access to records? The high school marching band? Local barbershop quartet? Any two-bit station can acquire the technical facilities and then dip into this ready-made program pool, with little more expense than some under-paid deejays.

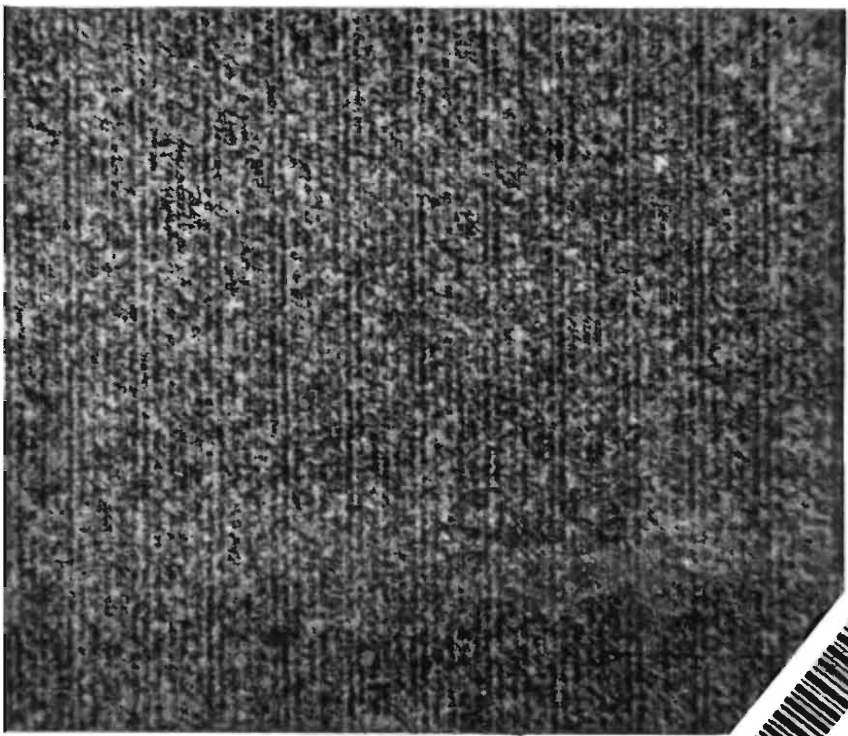
The whole thing is simple. The record companies are doing a big volume of business, but due to the mad discounting that is prevalent, profits are declining. No profits, no record company. And with no record companies, no one will have to worry about the encroachment of off-the-air recordists. The record companies need the broadcast fees to improve their product and their artist rosters, to continue research and development, to underwrite the static classical market, commission new works, aid musical education, open new recording possibilities for many more musicians and orchestras. Of course they are not philanthropic institutions, and no one expects them to be. But they are intimately woven into the musical fabric of the country and, if they are to fulfill any of these objectives and prosper, they must re-assess their relationship with the broadcasters and move in the direction of performance fees. Æ



The beat note resulting from a high level recording of the two frequencies, 200 and 230 cps, is seen in this five times enlargement.

Making Magnetic Recordings Visible

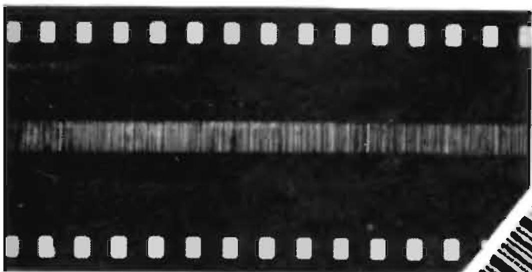
The technique used in making visible the sound tracks shown on this page was described in an article entitled, "Alignment of Magnetic Recording Heads" by B. F. Murphey and H. K. Smith in the January 1949 issue of AUDIO ENGINEERING. For some purposes, where several inches or feet of tape are to be visibly examined (as for editing), Mr. Robert Herr of Minnesota Mining & Mfg. Co., who has supplied these pictures, reports a more convenient and less messy method. The carbonyl iron is suspended by shaking in a volatile liquid, such as heptane (which will not dissolve the tape) and the tape is dipped in this suspension for a few seconds. Upon removal, the liquid will dry quickly and the track becomes visible. The carbonyl iron may be removed by wiping it off. This method allows some flocculation of the particles and does not yield quite so good resolution as the suspension in a more viscous medium, but it is simpler and adequate for examination by the naked eye.



Above: A constant tone modulated by a vibrating head is shown here. Magnification X 60.

Below: The word "tape" was recorded with a full width 1/4" track, using an Ampex machine at 30 inches per second. Enlergement, X 1 1/4.

Below: 26-times enlargement of a 0.1-inch wavelength signal recorded on black oxide tape illustrates the fringing effect. In contrast to the other photo of a 100-mil treck, no modulation noise is evident.



Above: Music recorded on oxide-coated 35 mm film is illustrated by this photo. Ready means for editing and track location is provided by making the track visible. No enlargement.

Below: A 0.1-inch wave length recorded on a 0.1 inch track using noisy tape shows residue of modulation noise between the prominent poles of this strongly recorded signal. Magnification, X 20. Also lamination faults in head are visible.