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BUILDING HI|FI SPEAKER SYSTEMS

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MARKETING COMMUNICATIONS
ELECTRONIC COMPONENTS AND MATERIALS DIVISION

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Foreword

In the ten years that have elapsed since the first edition of this book was published, we have seen semiconductor technology mature and the degree of integration of circuits drastically increase. We have seen high fidelity become the norm rather than the exception. And we have all come to expect more and more from our sound reproduction systems.

The principle of the moving coil loudspeaker has remained unaltered ever since its early introduction in 1925, yet today's speakers reflect all the latest advances in modern electronics technology. The computer now plays a substantial part in the design and development of loudspeakers. And the use of real-time analysers in frequency response and sound pressure measurements ensures that we know everything that there is to know about our loudspeakers before they leave our factories. We can also predict their future performance with a high degree of accuracy under their final operational conditions.

Concerning high fidelity and the standards by which to judge it, many interesting developments have taken place recently. Studies have been made on the content of modern music, particularly from the point of view of power/frequency, and results prove that the earlier Standards by which hi-fi has been judged are no longer valid. The European high fidelity Standard DIN 45500, which has been used for many years to define hi-fi loudspeakers, now looks like being changed to accomodate the requirements for modern music. Details of the latest recommendations are given in this edition.

As with earlier editions, we are introducing a number of new loudspeakers, and details of enclosures using these speakers are provided. All the new tweeters and mid-range speakers are *sealed* at the rear to prevent back-radiation and isolate them from the woofer. This enables the constructor to make a simpler enclosure because no separate compartment or cover is necessary to achieve this isolation. Basic principles of system design will also be found in the book; home constructors can avoid the mathematics, if they wish.

We have now been making loudspeakers for over 50 years. And in that time we have produced millions and millions of loudspeakers. Every one of the speakers described in this book is backed by 50 years' experience. We promise the reader an exciting and fulfilling time in building his own high fidelity speaker systems.

M. D. H.

Contents

1 Introduction

Maximum performance at minimum cost is the theme of the loudspeaker system designs in this book. No matter whether the reader has business or private reasons for wishing to adopt the recommended systems given here, full satisfaction can be assured, provided that the correct speakers and network components are employed in the recommended volume of enclosure. Within certain limits, the performance of these speaker systems remains unchanged when the dimensions of the enclosure are altered, *provided that the volume remains the same.* This gives the constructor a certain amount of flexibility in his design.

The loudspeaker has the very exacting task of converting the electrical signals from the power amplifier back into a faithful reproduction of the original sound. The rest of the equipment in the reproduction chain counts for little if the speaker is inadequate, whereas the sound quality of even the cheapest tape recorder can be greatly improved when a good quality loudspeaker system is employed.

The performance of the loudspeaker depends very largely on the enclosure, and it is vitally important that for high quality reproduction the speaker is housed in a proper cabinet. To mount a loudspeaker in any old box and expect it to give superb reproduction is inexcusable. Most of the systems recommended in this book are called *sealed enclosure* systems, since the loudspeakers are mounted on one side of an air-tight box. The air inside the box controls the bass performance of the speaker system and, for a given volume, there is a specified performance.

Before choosing a speaker and a suitable enclosure, a number of factors have to be considered. This book discusses these points in simple terms and provides the reader with sufficient information on which to base his choice. For those readers who wish to avoid the theory and concentrate on building a good quality loudspeaker system, constructional details are provided in this book of 19 different loudspeaker systems. Each of these has been fully tested using the most modern equipment, and they can be relied upon to give full satisfaction to the constructor. Alternatively, those readers who wish to develop their own systems will find that sufficient background information has been provided for them to do so.

2 Sound reproduction

2.1 The nature of sound

Hearing, like seeing and feeling, is a primary sensation. The term *sound* is used to denote the sensation received by the ear, and also to indicate the physical cause of this sensation. In every case, sound is caused by something in a state of vibration. The vibration of a body cannot directly be the cause of sound; the immediate cause must be something in contact with the ear to act as the medium through which the sound is transmitted from the vibrating body to the ear drum. This medium is normally the air; sound can be transmitted through solids, liquids and gases, but not through a vacuum.

The sensation of sound is caused by compressions and rarefactions of the air through the process of progressive undulation in the form of longitudinal oscillatory motion: that is to say, each particle oscillates about its position of rest along a line parallel to the direction of propagation. When a succession of particles, such as the molecules of the air, perform similar movements in turn, it is because the movement of each one *causes* the movement of the next, and one body can only cause the movement of another body by transferring to that body some of its own energy.

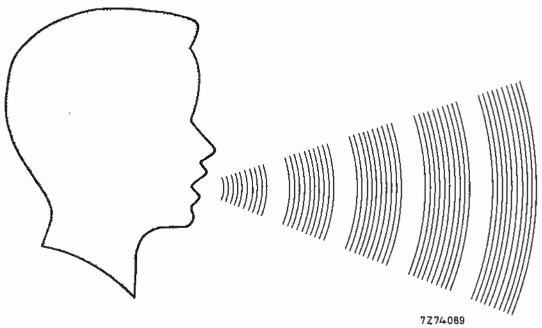


Fig. 2.1 Sound is caused by variations in air pressure.

The energy given to the particles immediately adjacent to the vibrating body is transmitted by successive influences of particles on their neighbours. In the absence of dissipation, caused in practice by losses in the air, the energy transmitted per unit area varies as the square of the distance from the source. This energy, or rather the rate at which it is transmitted, is a measure of a very important property of a sound wave: it expresses the *intensity* of the sound upon which our sensation of loudness depends.

Sound travels through the air with a constant velocity depending upon the density of the air; this is determined by the temperature of the air and the static air pressure. At normal room temperature of 22 °C (71,6 °F) and a static pressure of 0,751 m Hg (10^5 N/m^2), the density of the ambient air is 1,18 kg/m^3. Under these conditions, the velocity of sound is 344,8 m/s (1131,2 ft/s) but, for all practical purposes, the velocity can be taken to be 340 m/s. The wavelength of a sound (λ) is equal to the velocity of propagation divided by the frequency of vibration (f):

$$\lambda = \frac{340}{f} \quad \text{m.} \tag{2.1}$$

The wavelengths of sounds for different frequencies given in Table 2.1 have been calculated on this basis.

We have said that our sensation of loudness depends on the intensity of the sound. When a sound wave is propagated through the air, the pressure of the air at any point will vary above and below the normal ambient pressure. This incremental variation of the air pressure is known as the *sound pressure* and, for practical reasons, it is this which we measure in determining the loudness of sound.

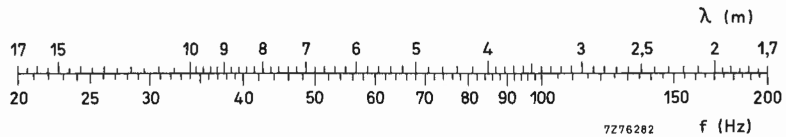


Table 2.1 Wavelength vs frequency, based on a sound propagation speed of 340 m/s. If frequency scale is multiplied by 10^n, wavelength scale must be divided by 10^n.

We can measure the sound pressure in *absolute* terms, such as so many microbars or newtons per square metre, but this does not give any indication of how loud a sound will appear. It is more useful to measure the sound in *relative* terms with reference to the level of sound at which our hearing starts to respond. Alexander Graham Bell discovered that the ear responds to sound intensity in a logarithmic way, our ears becoming less sensitive to the sound as the intensity increases. A logarithmic scale is used, therefore, to ensure that proportional changes are expressed in the same number of units. The basic unit is the *Bel* (B), named after its inventor, but as this represents rather a large change in intensity, we use *decibels* (dB) which are only one-tenth that size.

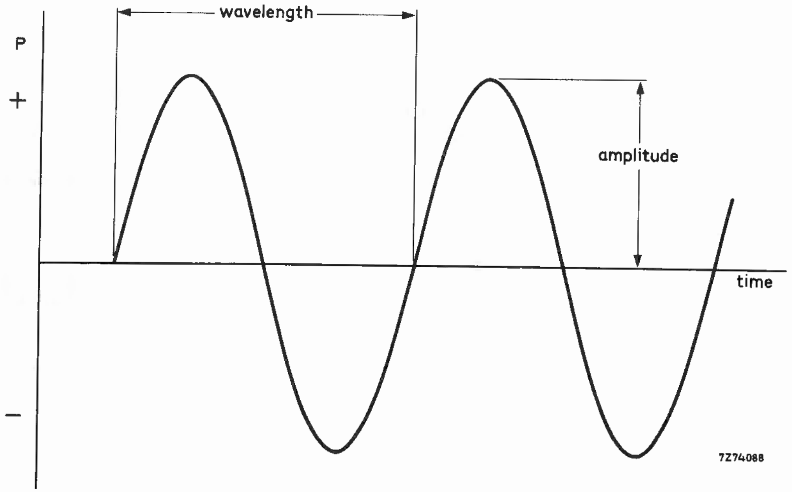


Fig. 2.2 If the sound consists of only one frequency and is of constant strength, the wave will be sinusoidal.

Since the ear responds to sound in a logarithmic way, we measure the level of the sound pressure in decibels with respect to a standard reference sound pressure representing the threshold of hearing at 1000 Hz. Sound pressure level

in decibels is defined as 20 times the logarithm to the base 10 of the ratio of the measured effective sound pressure (p) to a reference sound pressure (p_{ref}). That is:

$$\mathrm{SPL} = 20 \log \frac{p}{p_{ref}} \quad \mathrm{dB}. \tag{2.2}$$

It is important to remember that two different reference levels are in use. The first, and the one which concerns us most, is in general use for measurements dealing with hearing and for sound-level and noise measurements in the air: here, $p_{ref} = 0{,}0002$ microbar (2×10^{-5} N/m^2). The other, which has gained widespread use for calibrating transducers such as microphones, is $p_{ref} = 1$ microbar (0,1 N/m^2). The two levels are almost exactly 74 dB apart, so the reference pressure should always be clearly stated if there is likely to be any confusion.

The intensity (I) of a sound wave in the direction of propagation is given by:

$$I = \frac{p^2}{\varrho_0 c} \quad \mathrm{W/m^2}, \tag{2.3}$$

where p is the sound pressure in N/m^2,
ϱ_0 is the density of the ambient air in kg/m^3, and
c is the velocity of sound in m/s.

On the other hand, the intensity *level* of a sound in decibels is measured with respect to a standard reference level representing the intensity at the threshold of hearing at 1000 Hz. The intensity level in decibels is defined as 10 times the logarithm to the base 10 of the ratio of the intensity of a sound to a reference intensity. That is:

$$\mathrm{IL} = 10 \log \frac{I}{I_{ref}} \quad \mathrm{dB}. \tag{2.4}$$

The reference intensity in this case is taken to be 10^{-12} W/m^2; this value has been chosen to correspond to the reference pressure of 2×10^{-5} N/m^2.

The exact relation between intensity level and sound pressure level may now be found by substituting Eq. (2.3) for intensity in Eq. (2.4). Inserting values for p_{ref} and I_{ref} yields:

$$\mathrm{IL} = \mathrm{SPL} + 10 \log \frac{400}{\varrho_0 c} \quad \mathrm{dB}. \tag{2.5}$$

It will be apparent that the intensity level IL will be equal to the sound pressure level SPL in decibels when $\varrho_0 c$ has a value of 400. Certain combinations of temperature and pressure will satisfy this condition, but for a room temperature of 22 °C and an ambient pressure of 10^5 N/m², the value of $\varrho_0 c$ is 407. This means that the intensity level will be slightly less than the sound pressure level by about 0,1 dB. For all practical purposes in this book, we shall assume them to be equal.

Another interesting quantity is the *acoustic power level.* The acoustic power level of a sound source in decibels is 10 times the logarithm to the base 10 of the ratio of the acoustic power radiated by the sound source to a reference acoustic power:

$$\text{PWL} = 10 \log \frac{W}{W_{\text{ref}}} \quad \text{dB.} \tag{2.6}$$

Here, the reference acoustic power W_{ref} is taken to be 10^{-13} W. This means that a source radiating 1 acoustic watt has a power level of 130 dB. At normal temperature and pressure, the acoustic power level will be slightly less than the sound pressure level by about 0,5 dB. Again, we shall consider them to be equal for the purpose of this book.

The acoustic performance of loudspeakers is normally represented graphically with the dependent variable plotted vertically in decibels. We have just seen that there are three quantities which, for our purpose, have the same values in decibels:

- sound pressure level ($0 \text{ dB} = 2 \times 10^{-5}$ N/m²)
- intensity level ($0 \text{ dB} = 10^{-12}$ W/m²)
- acoustic power level ($0 \text{ dB} = 10^{-13}$ W).

The reader will now appreciate that three kinds of information can be obtained from one graph. In this book, where the vertical axis of a graph is marked in dB only, the reader can attach his own interpretation of its meaning within the restrictions imposed by the reference levels given above, bearing in mind that it is the sound pressure level that is actually measured. A detailed explanation of the methods of measuring the characteristics of our loudspeakers is given in Chapter 9.

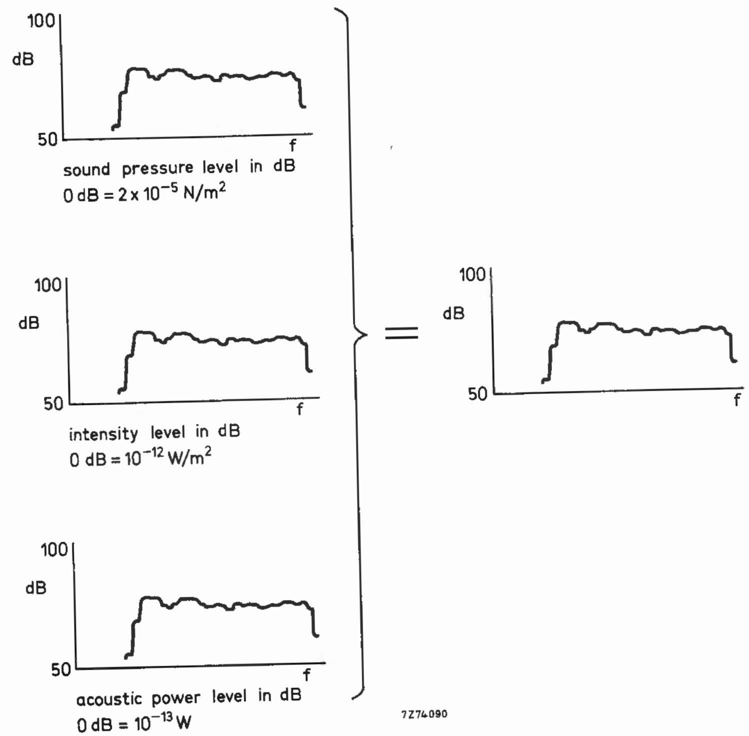


Fig. 2.3 Three performance characteristics can be expressed by the same graph.

Before we conclude our discussion on the nature of sound, we should mention two important characteristics of its behaviour: reflection and diffraction. If a sound wave encounters a body which is large compared with the wavelength, reflection of the wave occurs. When we consider a small part of a large surface, ignoring edge effects, reflection will only be complete if the surface is perfectly rigid. Acoustic rigidity can be improved by increasing the density of the material. When the material is not rigid, however, some reflection will take place, the rest of the energy of the wave being absorbed by the material. Conversely, if we wish to prevent reflections we use an acoustically absorbent material; in general, this has a low density. This is the kind of material we use to line the inside of loudspeaker enclosures to prevent internal reflections which would otherwise affect the quality of the sound.

When a sound wave encounters a small object in its path, or emerges from a small orifice, its wavefront is disturbed or distorted. By the term *small* we mean that the width of the object or orifice is less than the wavelength of the sound. In sound reproduction we are more interested in the case of the orifice; the slotted vent in a bass-reflex enclosure suggests itself. If the slot width is very large compared to the wavelength, the incident wavefront emerges virtually

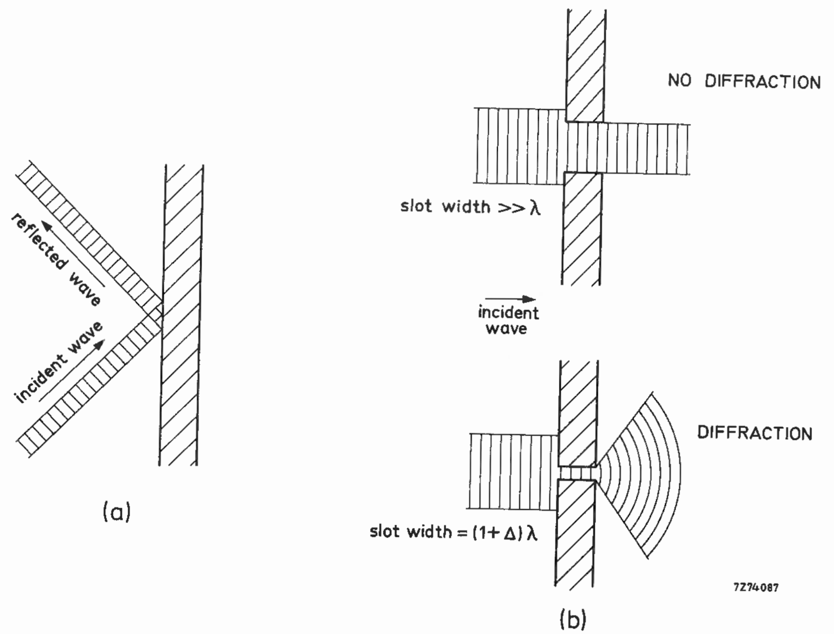


Fig. 2.4 Reflection and diffraction: (*a*) perfectly rigid body absorbs no sound and reflects complete wave; (*b*) diffraction at slot causes divergent wave when slot width approaches wavelength.

unchanged, but as the ratio of the slot width to the wavelength is reduced, the emergent wave becomes increasingly divergent. A limiting condition is reached when the slot width and wavelength are equal; the wave then diverges over an angle of 180° and the slot acts as a new source of sound waves.

In this section we have tried to explain a few important characteristics of sound. A detailed study is beyond the scope of this book, and the reader is referred to standard textbooks for further details.

2.2 Frequency range and harmonics

A musical tone consists of a *fundamental* tone with a certain frequency of vibration, accompanied by a series of *harmonics* each of which is a multiple of the fundamental frequency. The amount of energy which each harmonic contains depends on the type of instrument which produces the sound and this is what distinguishes one instrument from another. In music, frequency is referred to as *pitch*, whereas the character of a sound which depends on the proportion of harmonics it contains, is known as *timbre*. Harmonics are also known as *partials*, or *overtones*.

Mathematically, it can be shown that *all* waveforms can be broken down into a combination of sine waves consisting of a fundamental frequency together with harmonics of that frequency. This is what Fourier's analysis is all about; it is a mathematical method of analysing a complex waveform to determine the frequency, amplitude and phase of its content.

Sounds of a transient nature such as those produced by a piano, drums and cymbals must be reproduced in a crisp and life-like manner. A sudden crash of the cymbals produces a very steep-fronted waveform which, because of its sudden rise in amplitude, will contain a large proportion of higher harmonics. If these are not capable of being reproduced effectively, without distortion or loss, then the music will lack 'punch' or 'attack'.

The most important factors to be considered are the lowest and highest frequencies to be reproduced, the smoothness of the response and its permitted deviations from the horizontal, and the distribution of acoustic power over the frequency range concerned. The fundamental frequencies of the tones produced by our musical instruments range from about 16 Hz to 4186 Hz. The lowest fundamentals of a number of these instruments are given in Table 2.2.

Fig. 2.5 A complex tone consists of a fundamental frequency plus harmonics of that frequency.

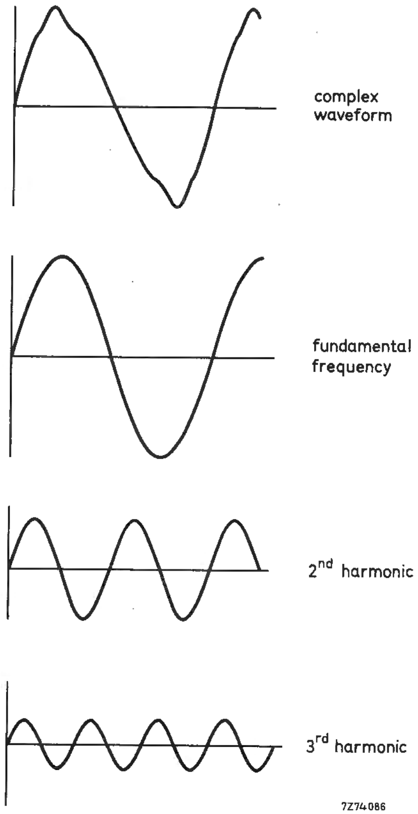


Table 2.2 Lowest fundamental frequencies of a number of musical instruments

instrument	Hz	instrument	Hz
organ	16,351	bass trombone	55
piano	27,5	french horn	61,7
contrabassoon	30,868	cello	65,14
harp	32,703	kettle drum	87,307
reed organ	36,708	bass voice	87,3
double bass	41,203	clarinet	110
tuba	43,654	trumpet	146
chimes	48,999	violin	196
bass saxophone	51,913		

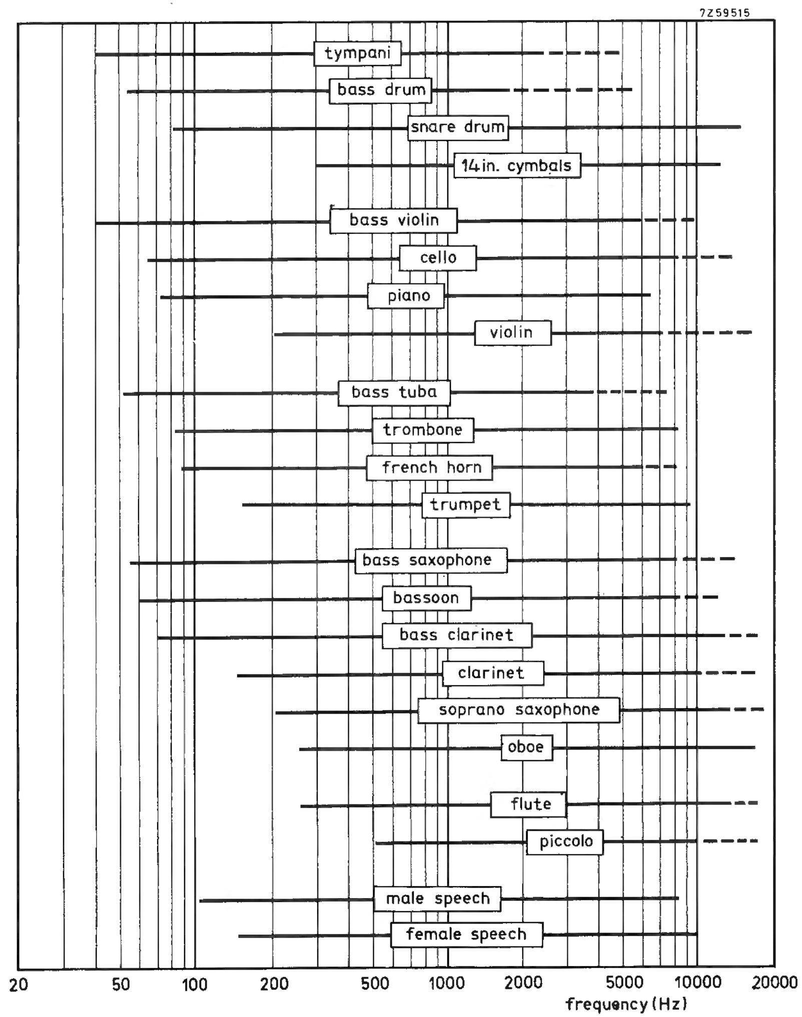


Fig. 2.6 Frequency ranges of musical instruments.

To assume from this, that recording and reproduction down to 16 Hz is necessary is not, however, true. Apart from the fact that the occurrence of such frequencies is rare, the fundamental frequencies of such low tones are considerably weaker than their second and third harmonics. Unless one is listening in a very quiet room, the fundamental frequency is inaudible except at high volume levels. The frequency of a complex vibration constituted by the harmonics gives us the impression of pitch and not the presence of the fundamental. Its presence has certain influences upon the timbre of the music but only at very high levels of sound.

From the foregoing, we can see that the response of an electro-acoustical installation does not need to go down to 16 Hz; there are no recorded sounds at this frequency in any case. But we must bear in mind that higher power is much easier to obtain in the home, today, than even a few years ago, and listening habits have changed considerably since the time when most of the basic research into listening criteria was conducted. This means that our missing fundamentals can be more easily detected than before because of the higher level at which sound is now reproduced.

In very general terms, we believe that in all cases for good quality sound reproduction, a frequency of 40 Hz should be capable of being reproduced, with no noticeable distortion for a whole octave below that, i.e. down to 20 Hz. For the discriminating listener, we recommend that he should adopt 30 Hz as his low frequency objective. How this can be achieved will be described later.

Table 2.3 Highest harmonic frequencies produced by various musical instruments

instrument	Hz
kettle drum	7 000
bass tuba	8 000
singing voices	9 400
harp	11 000
organ	13 000
xylophone	13 000
saxophone	14 000
cello	16 000
violin	16 000
cymbals	17 000

If the limit of high note response was taken as the highest fundamental note ever encountered, on the music score for the piccolo which gives up to 4186 Hz, not a single note would be missed but, because of the suppression of many harmonics, the timbre would suffer considerably. Table 2.3 lists the highest order of frequencies produced by various musical instruments. These high frequencies can be of considerable intensity; the 15 000 Hz harmonic of the cymbal is almost equal in intensity to its 300 Hz fundamental and, as we shall see, the ear can be more sensitive to the high tones than the low tones at certain levels of volume. Although the acuity of hearing falls off with age, e.g. 16 000 Hz in the twenty and thirty age groups, down to 12 000 in the forty and fifty age groups, and so on, the necessity to reproduce transient sounds with sufficient 'attack' means that no restriction should be placed on our ability to reproduce these high frequencies. Fortunately, the higher limit of the frequency range presents no problems in reproduction; loudspeakers specially designed for this task are readily available.

2.3 Intensity and dynamic range

No matter how nature produces sound, our sensitivity to that sound varies according to its frequency. We can easily prove that when we listen to sounds of the same intensity but of different frequencies, our sensation of loudness varies. At low intensities, for example, the low frequencies sound weaker than the high frequencies and very much weaker than the mid-range. As we increase the intensity, we find that the low tones and the mid-range are producing equal sensations of loudness, while our ear becomes more sensitive to the high frequencies.

For convenience, a unit of loudness level called the *phon* was introduced to take into account the variations in sensitivity of the ear at different frequencies. The loudness level in phons of a sound is numerically equal to the intensity level in decibels of a 1000 Hz pure tone which is judged by listeners to be equally loud. At 1000 Hz, therefore, the number of phons equals the sound intensity in decibels, but at other frequencies this depends on the sensitivity of the ear. Equal-loudness contours were first published by H. Fletcher and W. A. Munson in 1933. These are shown in Fig. 2.7.

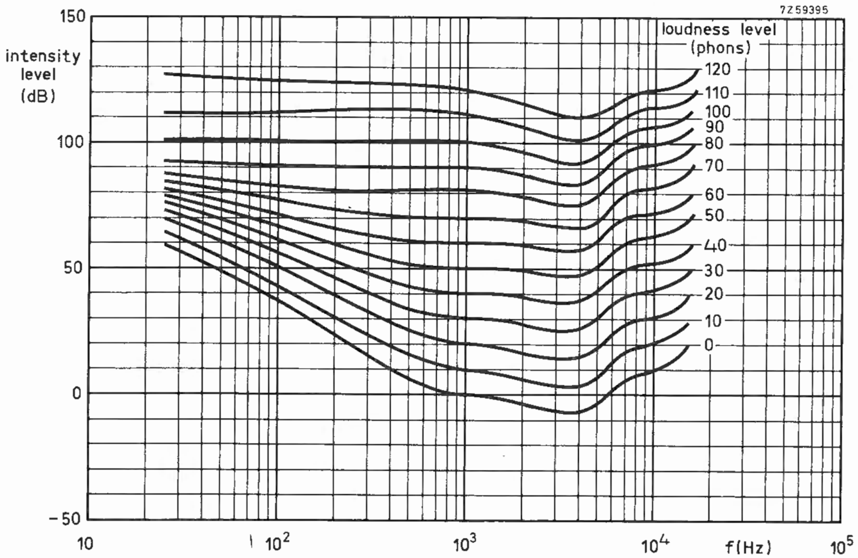


Fig. 2.7 Fletcher-Munson equal-loudness contours.

Further investigations by other researchers have not changed the fundamental work of Fletcher and Munson who pointed out that the effective loudness is substantially logarithmic above about 40 phons and semi-logarithmic below that level. How the effective loudness is related to loudness level is shown in Fig. 2.8. By taking the loudness at various frequencies for a given intensity and correcting for the modified logarithmic response of the ear, as shown in Fig. 2.8, a curve can be plotted showing the effective loudness as a function of frequency. Fig. 2.9 shows the result and clearly illustrates how a reduction in volume causes a considerable drop in bass. When we reproduce music, therefore, at a lower level than the original sound, we cannot expect to hear all the frequencies in their proper relation to one another unless we take steps to correct their amplitudes in proportion to the intensity at wish we wish to listen. How we can achieve this will be explained in Section 2.6.

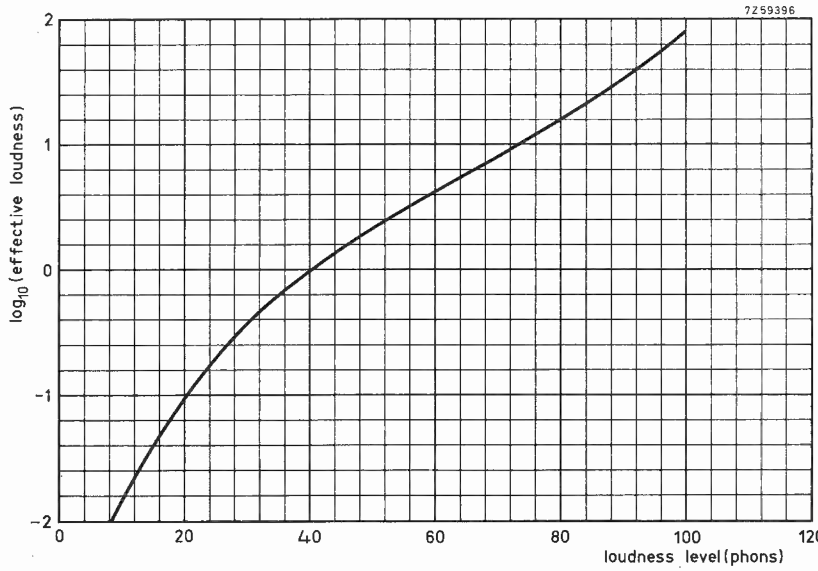


Fig. 2.8 Effective loudness as a function of loudness level.

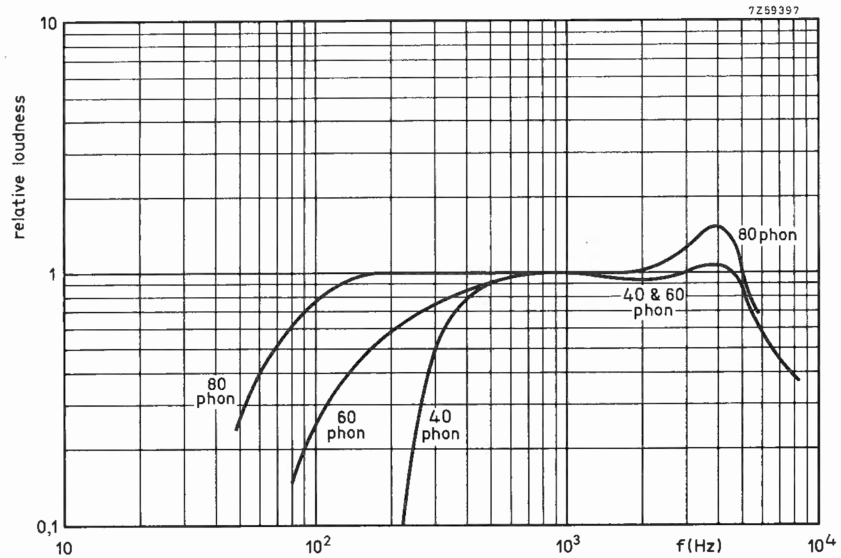


Fig. 2.9 Response of the ear as a function of loudness level.

The volume range, or *dynamic range* as we shall call it, is the ratio of maximum to minimum intensity of a sound source, expressed on a decibel basis. Considering speech and music sources only, the maximum dynamic range occurs in orchestral music. During a three-hour recording session by the Philadelphia Symphony Orchestra during which ten selections were played, the maximum ratio observed was about 74 dB; if one particular crash of the cymbals lasting only 0,1 second was excluded, the dynamic range would be down to 65 dB. The dynamic range encountered in speech is considerably lower, any single individual rarely exceeding 40 dB. Clearly, if a sound reproduction system can handle a dynamic range of 70 dB nothing of consequence will be missed, but the reader will realize that this does not take into account the *masking* effect of noise at low levels.

Masking is the reduction in the subjective loudness of one tone by the introduction of another tone; the degree of masking depends on the level and frequency of the second tone. A detailed discussion is beyond the scope of this book but we should remember that the effect of masking due to room noise is to raise our threshold of hearing; the louder the background noise in the room, the louder should the wanted sound be reproduced, otherwise its weaker levels will be inaudible.

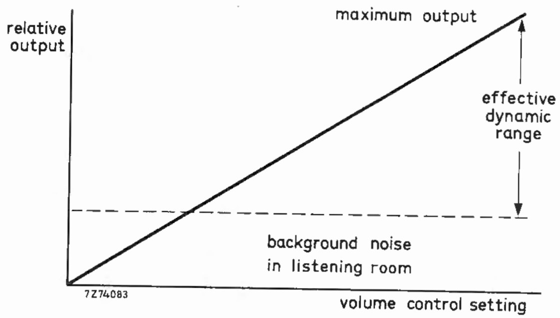


Fig. 2.10 Reduction in effective dynamic range due to masking effect of room noise.

2.4 Sources of programme material

A wide number of sources of programme material are now available to the listener:

- amplitude modulated (a.m.) radio in the long, medium and short wave bands;
- frequency modulated (f.m.) radio in the VHF band;
- TV sound;
- normal commercial disc recordings;
- open-reel magnetic tape recordings;
- tape cassette recordings using iron oxide (FeO_2) and chromium dioxide (CrO_2) coated tapes;
- microphone.

Radio and television provide sources of sound in which their electrical signals can be amplified and reproduced in the simplest way. Assuming we are using a linear power amplifier in which all the input frequencies are handled equally, with sufficient power output, no audible distortion or circuit noise, the sound quality will be limited only by the loudspeaker; that is, the full frequency and dynamic range of the programme material will be correctly reproduced. Amplitude modulated radio transmissions do not, however, provide the best quality sound. Due to the large number of radio stations which make demands on our radio frequency spectrum, it is necessary to restrict their bandwidth. In the long, medium and short wave bands, therefore, international agreement limits the total bandwidth to 9 kHz (in general). This means that for normal double sideband transmissions we are only getting 4500 Hz as our maximum transmitted audio frequency.

In addition, to limit distortion to an acceptable level in a simple low-cost receiver, the percentage modulation of the carrier frequency is restricted to around 30%. To achieve the maximum possible geographical coverage it is necessary to set this maximum level of modulation as corresponding nearer to the average sound level, rather than peak values, because the intelligence is conveyed in the sidebands. This means that the peak values of sound levels are not transmitted and *volume compression* is applied at the transmitter. The result is a maximum programme dynamic range of 45-50 dB for amplitude modulated broadcast transmissions.

During the period of about ten years before frequency modulated transmissions commenced, many ingenious methods of restoring the loss in dynamic range of a.m. broadcast transmissions were attempted. These ranged from connecting incandescent lamps in parallel with the loudspeaker, to using vari-μ tubes in special *volume expander* circuits. None of these systems has proved very successful due to the rise and fall time of its response and it was a great day for music lovers when VHF f.m. broadcast transmissions commenced and the restrictions in frequency range to 4500 Hz and in dynamic range to 45-50 dB were removed.

Before we move on to other things, we must remember that these restrictions in transmission still apply to a.m. signals; if a.m. radio is your only source of programme material, the reproduction requirements are not difficult to meet, but we shall come back to this subject at the end of the Chapter.

Frequency modulated (f.m.) broadcast transmissions on VHF offer a much better source of sound than a.m. radio. Since the carrier frequency is extremely high in relation to the deviation, a wide dynamic range of up to 60 dB can be transmitted. In general, with f.m. transmissions, there is still some volume compression, but for most orchestral works this is not apparent to the listener. The frequency range, also, is much greater, the limit being about 15 000 Hz. To prevent masking by noise and interference at the receiver, and to balance the fall-off in deviation as the modulating frequency rises, the upper audio frequencies are emphasized before transmission. This *pre-emphasis*, as it is called, is applied to all f.m. transmissions. To restore the signal, *de-emphasis* is used in the tuner, with a significant improvement in signal-to-noise ratio and, as with a.m. signals, a 'flat' response is delivered by the tuner. From our f.m. tuner, therefore, we expect an audio frequency range up to 15 000 Hz, with a dynamic range up to 60 dB.

TV sound is a very useful but rather elusive source of programme material. Very few manufacturers provide audio outlets on TV receivers. Modifying a TV receiver for this is not difficult, but although making connections across the loudspeaker terminals may be a last desperate attempt to get at the signals, it is definitely not recommended because a much 'cleaner' signal lies ahead in the TV circuit: at the volume control. The circuit noise and distortion of the output stage are avoided in this way.

Most TV sound, by international agreement on bandwidths, occupies about 25 000 Hz of its channel spectrum. In general, audio frequencies up to 12 500 Hz

are actually broadcast. This is considerably better than a.m. but not as good as f.m. radio. With TV transmissions, the dynamic range is good, being about 60 dB on f.m. sound.

Normal long-playing recordings are the best possible source of programme material at the present time as far as the frequency range is concerned. Frequencies up to 18 000 Hz are recorded, there being no restrictions other than the recording equipment and the quality of the pressing. A dynamic range of 50 dB can normally be expected to be obtainable and, provided that discs are properly handled, played and looked after, they do not deteriorate with age and give as good a response as it is possible to obtain by any other medium. Usually, the more expensive the record, the better the technical quality.

'Singles' and cheap records of 'pop' music do not come into the same quality class as the more expensive commercial recordings. The dynamic range of such records is only around 35-40 dB, while the frequency range obtainable varies enormously; some are recorded with predominantly low frequencies such as with military band and 'carnival' music, while others such as childrens' records contain mainly high tones. We are not suggesting for a moment that the reader should not buy such records, since some tunes are only recorded on cheap discs, but the reader should not *expect* to get very good quality reproduction. In fact, if these tunes are broadcast over local radio, you would probably be better off to tape a live broadcast on f.m. or TV.

Disc recordings since 1955 have been made to the RIAA and European (IEC) Standards. At low frequencies, where the energy level is high, the amplitude of the signal is reduced; at high frequencies, conversely, where the energy level is low, they are emphasized. On playback of a disc recording it is necessary, therefore, to equalize the response with the inverse of the recording characteristic. How we equalize it, however, depends on the type of pick-up cartridge we are using. An inexpensive piezo-electric (crystal) pick-up behaves as a capacitance in series with a resistance and so its frequency characterstic compensates in a very large measure for the recording characteristic; consequently, little or no equalization is needed. Although the output is high from a crystal pick-up, distortion is high too, and this type of pick-up is only used in the less expensive class of sound reproduction equipment. With a crystal pick-up, the maximum response is reached at a frequency of around 12 000 Hz.

The next best quality of pick-up is the ceramic cartridge type. This has a better response than the crystal pick-up and is free from sharp peaks, with an extended frequency range to around 15 000 Hz. Its response characteristic also balances the recording characteristic to a large extent and, again, little is needed in the way of equalization.

Undoubtedly, the magneto-dynamic pick-up is the best there is. A dynamic pick-up has a coil, with stylus to cause movement of the coil, mounted in a magnetic field and operating as a generator. With good design, this construction is virtually distortionless at low frequencies, and it has a linear output. Its output voltage is very low so, compared with the crystal pick-up, it requires an input amplifier with about an extra 15 dB gain. Equalization is also needed, since the magneto-dynamic pick-up has a linear response, which is the inverse of the recording characteristic.

For the amateur, open-reel tape recording run at 19 cm/s ($7\frac{1}{2}$ in/s) offers the best means of recording from any source, particularly f.m. radio. This is, naturally, very wasteful of tape and, if some loss in high frequencies is acceptable, 9,5 cm/s ($3\frac{3}{4}$ in/s) is certainly a more economical speed. With a good quality tape recorder, e.g. Philips N4450, a frequency response up to 20 000 Hz at 19 cm/s is reduced to 17 000 Hz at 9,5 cm/s. Still more economical use of tape is possible with the tape running at 4,75 cm/s ($1\frac{7}{8}$ in/s). This tape speed, however, only provids a frequency response up to 8000 Hz. In general, for open-reel tape recorders, pre-recorded tapes are available for running at a speed of 9,5 cm/s and provide a very good source of high quality sound. The dynamic range obtainable is of the order of 60 dB.

For the listener, the development of the compact tape cassette, a Philips' invention, has opened up new possibilities. The cassette has the special advantage that it is compact, and the tape is almost completely protected in both handling and operation. Its ease of use is unrivalled by any other medium of recording programme material. Designed to run at 4,75 cm/s, it does not, however, aim to provide a high quality sound source. But because of its popularity, and the firm belief that one day the cassette will take over from the disc recording, much effort is being put into the development of new systems which will make the cassette a reliable source of top-quality sound. While manufacturers' claims vary widely, it is certainly possible at the time of writing to obtain a fairly flat response up to about 15 000 Hz using the latest tapes with good quality cassette equipment. A dynamic range of at least 55 dB is also possible.

One special problem which arises with magnetic tape is *noise*. If the noise level is high, it masks the high tones; to reduce noise by switching-in a filter

also cuts the top response. Obviously, this method of noise reduction makes no sense for good quality reproduction, so other methods have been developed. These are, notably, the Dolby* system in which correction is applied during both the recording and the reproduction processes; and the Philips DNL system (dynamic noise limiter) which is applied on playback only, in the interests of compatibility. In addition, 'low-noise' tapes have been produced and, for high quality reproduction, the latest development is that of tapes using chromium dioxide.

Last comes the microphone. Studio microphones are of the highest quality and the reader can be assured that their sensitivity, frequency range, noise and distortion figures are all that they possibly could be for high performance. A discussion about these is beyond the scope of this book but it is important to realize that all that applies to studio microphones does not apply to the 'domestic' models supplied with tape recorders. During the last few years, however, the crystal microphone which was considered as 'standard' equipment as an accessory to a tape recorder, has given way to the moving coil, or dynamic microphone. In general, the remarks made about magneto-dynamic pick-ups also apply to dynamic microphones: a full frequency range, low distortion, and low output. An equalizing amplifier is necessary, therefore, with a dynamic microphone.

2.5 High fidelity and realism

Sound recording and reproduction became firmly established with the development of the gramophone in 1887. Ever since, enthusiasts have talked about the *fidelity* or faithfulness of sound reproduction made possible by every technological advance. The term *high fidelity* is used to describe the most realistic sound reproduction obtainable; we can never expect complete fidelity to the original sound because our listening surroundings differ from those of the original. Unfortunately, the term 'hi-fi' is also used to describe anything capable of producing a very loud sound regardless of its frequency range or the amount of distortion present.

* Dolby is the registered trade mark of Dolby Laboratories Limited.

The degree of realism now attainable is very high, but so also is its cost. It is a matter for the listener to decide how far he is willing to go in this respect, so we shall now discuss the degree of realism that can be achieved and how this is obtained. The problem can be simply illustrated by considering the dynamic range of music; let us take the case of an installation which has a dynamic range of 67 dB. If we wish to improve this to take into account peaks in orchestral works by increasing the capability to 70 dB, we require only 3 dB extra. But 3 dB increase means that a factor of two is involved and thus the power output will have to be *doubled.* If a 25 W amplifier was in use to obtain a dynamic range of 67 dB in the first place, increasing the dynamic range to 70 dB will mean raising the output power to 50 W. It can be seen that this will require much higher power reproduction equipment with a corresponding increase in cost.

The simplest sound system is the monophonic system, or *mono* for short. This is a single-channel system in which a complete electrical signal representing the total sound information is amplified and reproduced with a single loudspeaker system, or a number of separately mounted but parallel connected loudspeaker systems. Usually, one full-range loudspeaker is employed. The result is that the loudspeaker acts as a point source of sound and the overall effect is a complete lack of any sense of dimension. Frequency range and dynamic range may still be faithfully reproduced, but the whole installation lacks realism although it may be of 'hi-fi' standard.

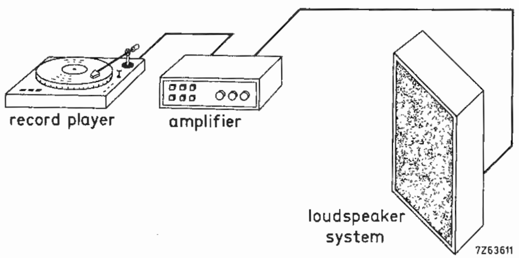


Fig. 2.11 Mono system lacks realism because loudspeaker acts as point source of sound.

The next step to realism is two-channel reproduction, or *stereophony.* Stereophony, or *stereo* for short, is now accepted as the name for two-channel reproduction although its name implies multi-channel reproduction. With stereo, two separate channels are broadcast (f.m.) or recorded (disc) in such a way that they can be replayed on mono equipment and produce the full programme sound. This is what we mean by *compatibility.* Listeners who have stereo equipment will be able to reproduce each channel separately and so enjoy the full sense of dimension available, but mono listeners are not prevented from enjoying the programme material just because they only have single channel installations. By international agreement all f.m. stereo broadcasts are transmitted so that if your tuner has a stereo decoder you will be able to separate the two channels into their left and right signals; if not, you will get the sum signal, i.e. left plus right, which represents mono. The same applies with records; the groove modulation will produce left and right signals if your system is two-channel, but with a mono pick-up, or mono system, only the composite signal is reproduced. Readers who may have acquired a stereo pick-up but are still using a mono amplifier and loudspeaker system, should take care that the two stereo outputs for the left and right channels are connected in parallel to produce the left-plus-right signal at the amplifier input.

Realism may be further improved by introducing not only the width of the sound stage as we have with stereo, but also the effects of the *depth* of the concert hall, as well. In the auditorium, and this applies to our listening room also, sound reaches our ears in two ways: direct sound from the musical instruments or vocalists, and indirect sound reflected by the walls and ceiling of the

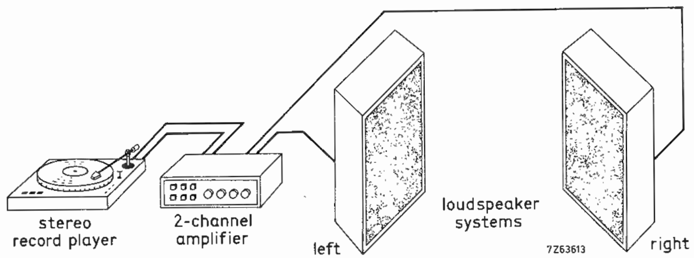


Fig. 2.12 Stereo system improves realism by introducing a sense of dimension into the reproduced sound.

auditorium. Since the indirect sounds travel a greater distance than the direct sounds, they experience a delay and, due to absorption during reflection, are usually weaker than the direct sounds. In the recording or broadcasting studio, the indirect sounds will also reach the microphones unless they are damped out. But we have no way of extracting the indirect sound, in any case, so apart from an 'echo' effect which the delay of a large concert hall produces, we are left to our imagination to simulate the true environment when only two sources of sound reproduction are employed.

However, we can improve our sense of realism by introducing a third loudspeaker behind our listening position, connected to the stereo system so that it reproduces only the difference signal of the two channels at low volume. The sound arriving at the listener will then consist of direct sound from the normal left and right loudspeakers, plus the indirect sound from the rear speaker, out of phase with the direct sound. This will enhance the *illusion* of realism by producing synthetically a blend of sounds resembling concert hall conditions. The addition of yet a further speaker, front-centre, producing the sum signal, i.e. left-plus-right, at low volume also enhances the illusion of realism. Readers who wish to try this experiment in sound should check carefully that their amplifier is capable of taking the load. A wirewound variable resistor of 20 to 30 Ω should be included in series with the rear speaker, which should be adjusted to give minimum output on a mono signal. This system provides *surround sound* in its simplest form; even discs and tapes of popular music can be heard to advantage this way when two extra speakers are provided.

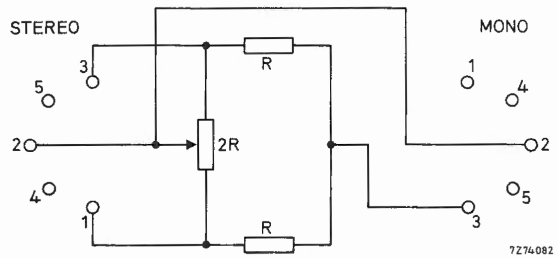


Fig. 2.13 Method of connecting a stereo source to a mono input. The value of R should be chosen to suit the source impedance.

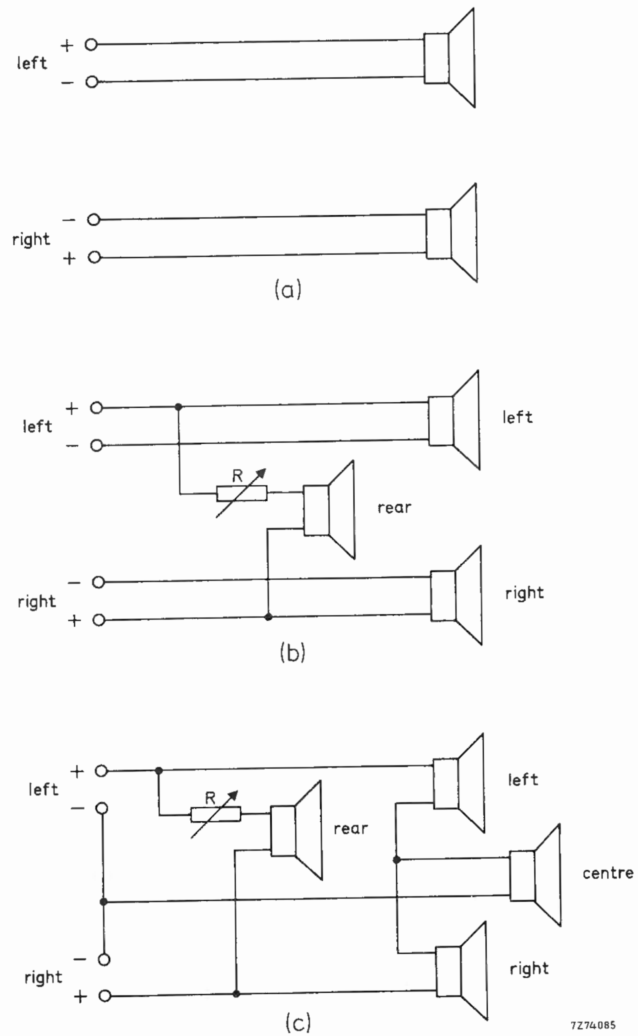


Fig. 2.14 Experiments with surround sound. Amplifier output is at left; (*a*) normal stereo, (*b*) adding rear speaker, (*c*) adding centre front speaker. Resistor R = 20 to 30 Ω and should be adjusted to give minimum signal in rear speaker on mono.

A step ahead of the simple system just described is the use of a *quadrophonic synthesizer* which produces four separate channel outputs from a normal 2-channel input; front left and right, rear left and right. The front channels normally carry the full left and right stereo signals, while the rear channels are fed with phase-modulated components of the front channels suitably processed. Although the rear signals are synthetic, the illusion of realism is extremely good, thanks to the use of two separate rear channels with phase-modulated signals incorporating definite (if somewhat exaggerated) delays.

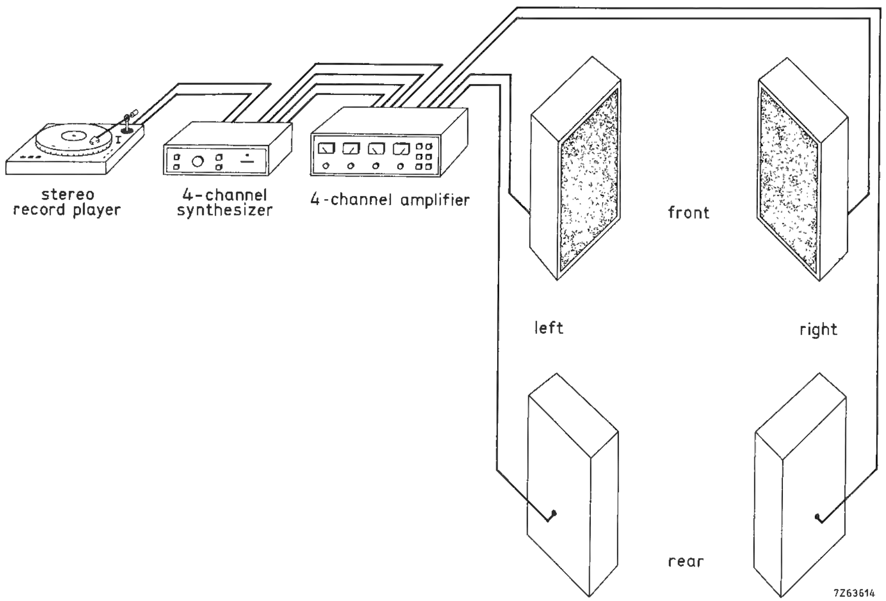


Fig. 2.15 Surround sound quadrophonic system using 4-channel synthesizer.

Finally, we come to true quadrophony, or *quadro* for short. This may take the form of four discrete channels separately recorded on tape and played back through four reproduction channels totally independent of each other except for their relative gain adjustments. Alternatively, the four channels will be suitably encoded on discs or tapes and a *matrix decoder* used in the playback system to recover the four channels which are then amplified individually. A detailed discussion of the various systems in current use for encoding and decoding the signals is beyond the scope of this book, but it important to remember that in any hi-fi reproduction system, no matter how many channels, the same high quality loudspeakers should be used on every channel.

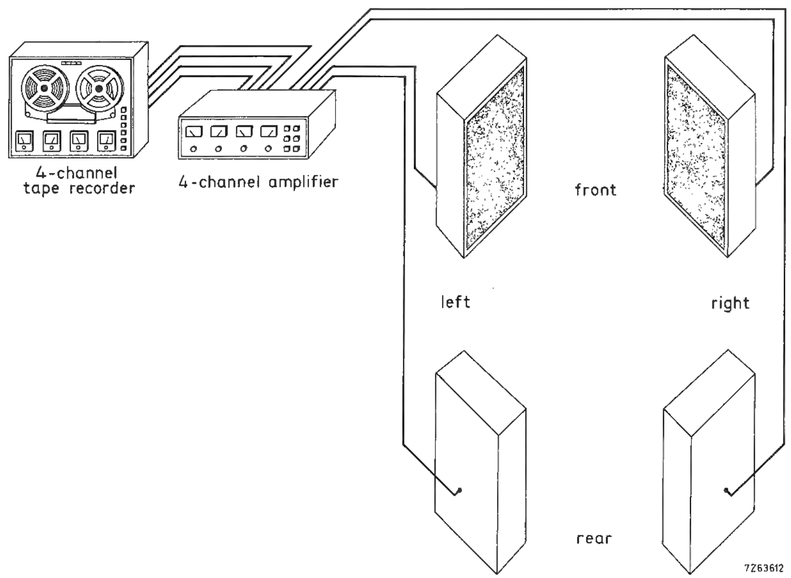


Fig. 2.16 Discrete 4-channel system.

2.6 Loudness and listening

The loudness of the reproduced sound for an amplifier of a given power output is a function of the efficiency of the loudspeaker system. Conversion of the electrical signal into sound, as we shall see in the next chapter, is a relatively inefficient process. Generally, a loudspeaker with a 15% efficiency would be considered very efficient indeed, whereas 5% would be considered nearer average. In high fidelity speaker systems, the quality of the reproduced sound is considered of paramount importance, and loudspeaker efficiency (which may be only about 1%) normally comes well down the list of technical characteristics to be considered. Loudspeaker efficiency is, however, very important to us in determining the loudness of the reproduced sound for a given amplifier output, or for finding the power output we need from our amplifier to produce a given sound pressure.

Let us decide to reproduce our music at a maximum level of 100 dB. The sound pressure in a room is a function of the acoustical energy radiated by the sound source and of the acoustical properties of the room. The actual sound pressure at a particular place in the room is the sum of the direct radiation and the reflected sound. We can therefore construct a set of curves to give us the acoustic power required to produce the required sound pressures in rooms of various volumes. How this is done is explained in greater detail in Chapter 6.

Taking our listening room as 9 m (30 ft) long by 4 m (13 ft) $\times$ 3 m (10 ft) high, we have a total volume of 108 m^3 (3900 ft^3). From Fig. 2.17 we find that an acoustic power of 0,5 W is required to produce a sound pressure level of 100 dB. Now if we already possess an amplifier capable of delivering 25 W output into the loudspeaker, then to produce an acoustic power of 0,5 W we require a loudspeaker with an efficiency of at least $(0{,}5/25) \times 100 = 2\%$. Alternatively, if we have a loudspeaker of 1% efficiency, and an amplifier that will deliver 25 W, the acoustic power produced by the loudspeaker will only be 0,25 W and the sound pressure will be lower by a half, or 3 dB, at 97 dB. To raise the sound pressure level back to 100 dB using a loudspeaker with an efficiency of 1%, in the room we are considering, would require a 50 W amplifier.

Earlier in this Chapter, we discussed the effect of masking by background noise in the listening room. Assuming a 30 dB noise level, the 100 dB sound pressure level in the previous example will produce a dynamic range of 70 dB. What you have to decide, therefore, if realism is your objective, is how far you are prepared to go in terms of complexity and the cost of achieving it.

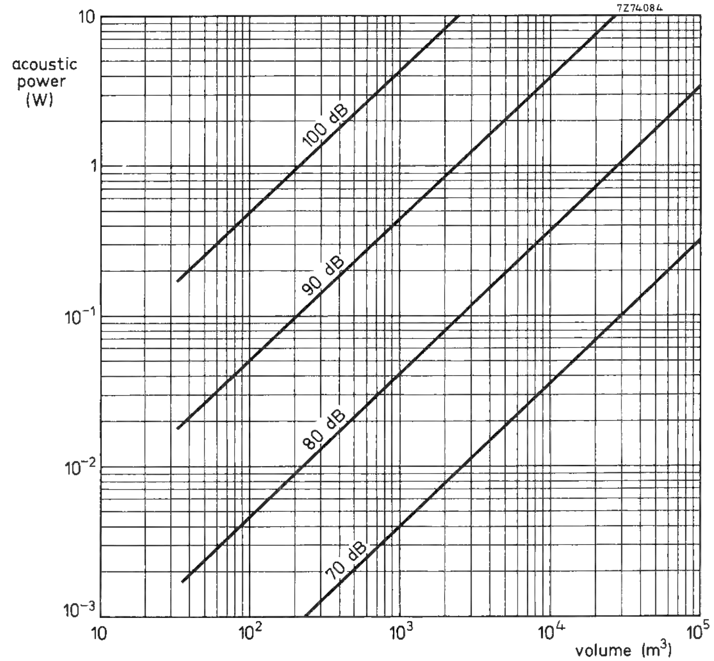


Fig. 2.17 The acoustic power required to establish a specified sound pressure level in different size listening rooms. For a room of 100 m^3 we see that to establish a sound pressure level of 100 dB we will need an acoustic power of 0,5 W. With a loudspeaker of 1% efficiency, the amplifier should be capable of delivering 50 W.

Most radio and television sets are fitted with loudspeakers that have a power handling capacity of around three watts maximum, seldom more, while transistor portables are fitted with half-watt speakers. How then do you reconcile your family to the arrival of two 50 W loudspeakers? The instant answer is to keep quiet about it! There is a definite case for a policy of 'near-silence', initially, with a very gradual rise in output over a protracted period of time.

One of the problems we have is the question of 'watts'. On the face of it there should be no confusion, but some manufacturers of amplifiers are always anxious to offer the customer more watts for less money, so the specified ratings for amplifiers have become a matter of considerable suspicion. When an electrical signal passes through an amplifier, the current drawn by the amplifier depends on the strength of the signal. Most of the current is supplied to the output stage which has a rating of so many watts depending on the capabilities of the output transistors and, also, on the type power of supply. The arrival of a sudden large signal will cause the currents in the output transistors to rise, but how far they will rise depends on whether the power supply voltage will start to fall. If a stabilized power supply is used, the voltage will remain constant and the transistor currents, and thus the output power, will remain at the same level no matter whether the music signal is of a transient nature, or is a sustained tone. We can say, therefore, that the *music power* rating is the same as the *continuous sine-wave power rating* when a stabilized power supply is employed.

When an unstabilized power supply, as is normally found in hi-fi equipment, is used, the continuous power rating is always lower than the music power rating because of the fall in supply voltage of the output stage under sustained load conditions. The sine-wave rating is usually about two-thirds of the music power with an unstabilized power supply. It follows that the sine-wave rating is a much more reliable figure to work with and, as we shall see later, our loudspeakers are specified for this condition of operation.

We have discussed earlier the problems of aural sensitivity at low volume levels. One does not always wish to listen to a music programme with the full dynamic range being reproduced; often background music at lower volume levels is desired. But we know that when the volume level is reduced, there can be a considerable loss in bass and also a small loss in treble reproduction. This was shown in Fig. 2.9. At low volume levels, therefore, realistic sound reproduction requires bass boosting and possibly some treble boosting, and to avoid the listener having to reset the tone controls each time the volume is adjusted a *physiological volume control* is sometimes used. This automatically raises the frequency response at low volume levels and, because it follows the Fletcher-Munson contours, it is often called a *contour control*. The frequency characteristic of a typical control is shown in Fig. 2.18. Another popular name for this is *loudness control* but, in general, a loudness control provides a fixed amount of boost, e.g. +12 dB at 50 Hz and +3 dB at 10 000 Hz, whereas a contour control automatically controls the amount of boost according to the volume setting.

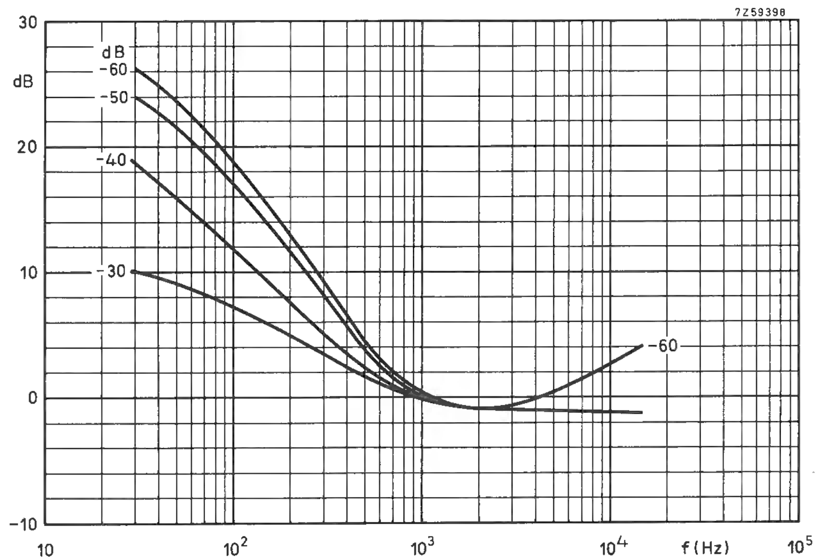


Fig. 2.18 Frequency characteristic of a contour control.

Finally, we come to the question of the neighbours. It is always difficult to define 'intolerable disturbance'. No complaints are possible when the sound of music reaches the neighbours at a level equal to the general noise level. The latter will be 30 to 45 dB above the threshold of hearing, depending on the surroundings. This means that, on average, the walls separating the listener's room from the neighbours' rooms should attenuate the sound passing through them by about 60 dB. This is more than the usual building materials are able to do and consequently it will only be possible in detached houses with closed windows to play music at natural loudness without annoying the neighbours.

The average transmission loss of an 8-inch brick wall plastered both sides is 51 dB. If, in one of two adjoining houses separated by such a wall, music is reproduced at a peak level of 100 dB, the peak levels of the disturbance in the other house will not exceed 49 dB. The average disturbance level will, of course, be lower than this and probably masked by the ambient noise to which it, of course, contributes. This may be acceptable in many cases, but the floors and ceiling can be a problem. A wooden floor on joists with a plastered ceiling below has an average transmission loss of only 43 dB. A further 5 dB might be added for carpeting, resulting in similar losses for both walls and ceilings. But concrete floors do not have such favourable sound-insulating properties. One very annoying source of interference is that caused by a lightly built loudspeaker cabinet which stands on the floor. Particularly at the lower frequencies, a lightly built cabinet will resonate and excite the floor into vibration far more efficiently than the sound waves emanating from the loudspeaker. Apart from the undesirable sounds such a cabinet produces in the listener's room, the losses of the concrete floor to this kind of sound are only 20 dB at the most, so it is essential to use a good solidly-built resonance-free cabinet not only to improve the quality of the sound but also to reduce the interference with the neighbours. Placing the cabinet on a thick layer of hair felt may improve matters.

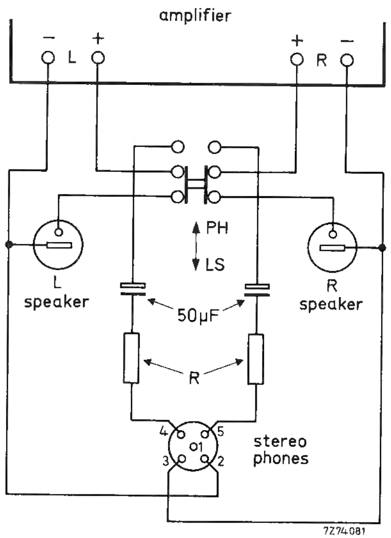


Fig. 2.19 Headphones may be connected to an amplifier as shown. Values of *R* are given in Table 2.4.

For those who are unable to enjoy the full dynamic range of their installation at all hours of the day, headphones may be used. These may be connected in series with a resistor and capacitor across the loudspeaker terminals of the amplifier as shown in Fig. 2.19. To avoid having to physically disconnect the loudspeakers, a change-over switch may be employed. Values of resistor *R* are given in Table 2.4.

Table 2.4 Values of R for different headphones

amplifier rating	headphones 8 Ω; 50 mW *R*	headphones 600 Ω; 20 mW *R*
10 W	68 Ω; 0,5 W	390 Ω; 0,25 W
25 W	180 Ω; 2 W	1800 Ω; 1 W
40 W	270 Ω; 2 W	2700 Ω; 1 W

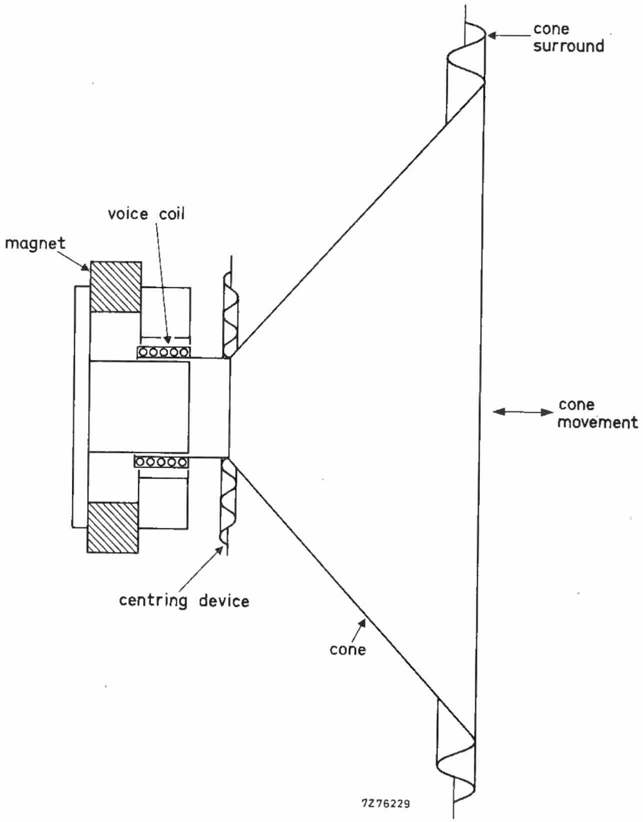


Fig. 3.1 Construction of a typical moving coil loudspeaker.

3 Moving coil loudspeakers

3.1 Principles of operation

A loudspeaker is a device for converting electrical energy into acoustic energy. There have been many forms of loudspeakers but a detail discussion of these is beyond the scope of this book; here we are concerned with the electrodynamic type, or *moving coil* loudspeaker. A loudspeaker may be considered to consist of two systems; a drive system, and an acoustic system. The acoustic system consists principally of a specially-shaped sound radiator which is made to vibrate by the drive system. The latter consists basically of a permanent magnet to produce a strong magnetic field which surrounds a coil of wire fixed to the neck of the cone. When an electrical signal passes through the coil, motion of the coil takes place at the frequency of the current, and the cone to which the coil is fixed is moved backwards and forwards in sympathy. The cone is mounted on a strong metal frame, being supported at the wide end by means of a flexible surround and at the neck end by a centring device which keeps the coil in the centre of the magnetic field. The construction of a moving coil loudspeaker is shown in simplified form in Fig. 3.1.

When a current flows in a conductor, a magnetic field is created around the conductor as shown in Fig. 3.2. If the current-carrying conductor is then placed in a magnetic field at right angles to the lines of force, the effect of the current

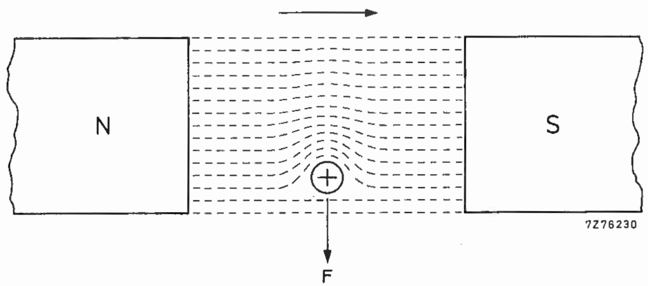
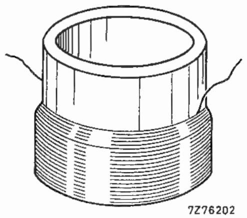


Fig. 3.2 With current flowing into the paper, the direction in which the conductor tends to move is shown by *F*.

is to concentrate the resultant magnetic field on the side where the two fields are acting in the same direction. Since the lines of force try to take the shortest path between the N and S pole of the magnet, the conductor experiences a mechanical force F in the direction shown by the arrow and movement of the conductor may result. Obviously, a larger number of conductors will produce a greater force; this is the principle of the electric motor.



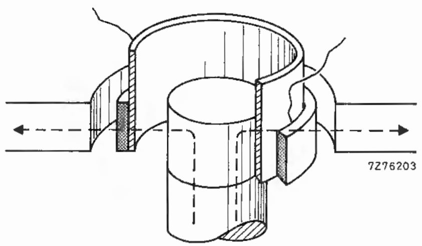


Fig. 3.3 (*a*) Typical voice coil construction. (*b*) Voice coil in magnetic field. Direction of magnetic field is shown dotted.

3.2 Magnet system

In order to apply the motor principle to a loudspeaker we have to design the magnet system so that we obtain the most efficient motion of the coil. By using a 'centre-pole' magnet system, as shown in Fig. 3.2, a very efficient design can be achieved. On the left side of the illustration the current flowing in the coil causes upward motion, and similarly, on the right side upward motion occurs because the direction of both the current and the magnetic field are reversed. The magnetic flux density in the air gap of a modern loudspeaker system would be typically 1000 mT (10 000 gauss) for a large good quality loudspeaker.

Partly to reduce the depth of a loudspeaker and partly for economy reasons, a ring magnet made of Ferroxdure has now been introduced. The cross-section of a Ferroxdure ring magnet system for a loudspeaker is shown in Fig. 3.4. Since the force which is exerted on the current-carrying conductors of the coil is dependent upon both the strength of the magnetic field as well as the strength of the current, it follows that a given force can be produced with less current if a stronger magnet is employed. As the current has to be provided by the power amplifier, it is obviously an advantage to use as strong a magnet as possible so that an amplifier with a lower power output can be used. Since the magnet system is the most expensive part of the loudspeaker, an economic limit to the strength of the magnet is soon reached. The question of efficiency is discussed later, in Section 3.6.

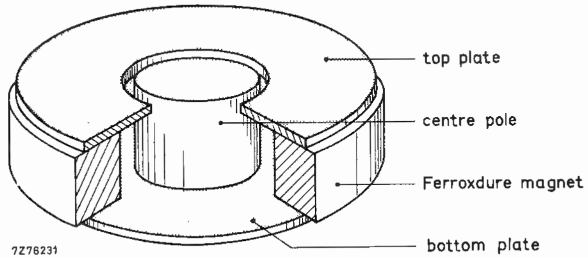


Fig. 3.4 Centre pole magnet system of modern loudspeaker uses Ferroxdure magnet.

3.3 Acoustic system

The acoustic system of a loudspeaker comprises the radiator and its suspensions. The radiator normally takes the form of a cone of compressed paper pulp but, where specially-designed loudspeakers are used to reproduce only the high frequency tones, the radiator takes the form of a plastic dome.

Where a paper cone is used as the radiator, the apex end of the cone is attached to the moving coil. Any motion of the moving coil is therefore transmitted to the cone. The cone and coil assembly have now to be attached to the frame of the loudspeaker so that the coil is accurately positioned in the magnetic field and the whole assembly is free to move under the influence of the current in the coil, returning to a neutral position in the absence of any current.

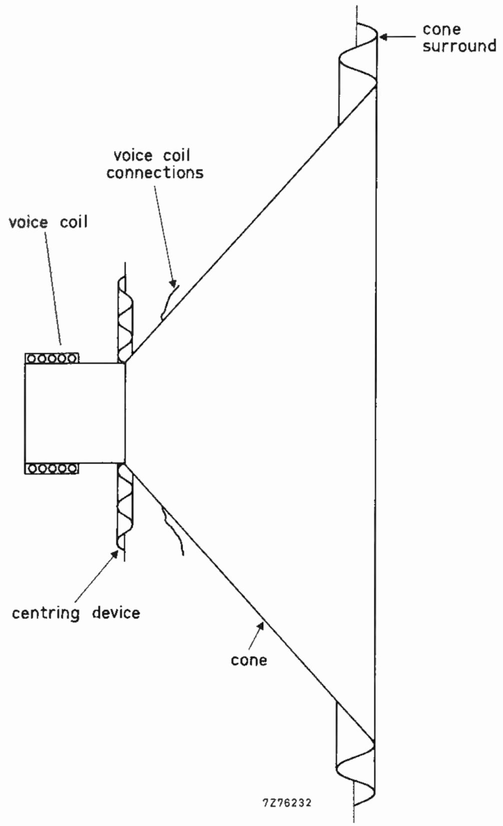


Fig. 3.5 Moving system of loudspeaker. The acoustic system comprises the coil and its suspensions.

The cone and coil assembly is normally supported at the apex end of the cone by a centring device made of stiff, impregnated cloth in which corrugations have been pressed. The outer end of the cone is supported in the frame by a similar flexible suspension which may either be the end of the cone itself in which corrugations have been pressed or, where large cone motions are required, a butyl-rubber surround which has one side fixed to the outer edge of the cone and the other side cemented to the speaker frame. The rubber surround allows much more flexibility and is preferred at low-frequencies where greater power, and hence cone motion, is required.

When an alternating current flows in the coil, the coil oscillates backwards and forwards in the magnetic field. The part of the cone which is attached to the coil also moves in sympathy with the coil. The remainder of the cone, however, can only vibrate in sympathy when it remains *rigid.* At low frequencies this is generally the case, but as the frequency increases a point is reached at which the wider end of the cone cannot follow the vibrations of the apex of the cone unless the cone is extremely stiff. This is known as *cone break-up* and results in linear distortion of the reproduced sound due to standing waves in the material of the cone.

Even when a very stiff cone is used, as in loudspeakers designed for reproduction of the full frequency range, the cone material is stretched and compressed in such a way that little or no vibration occurs at the outer end of the cone at high frequencies and it is only the part of the cone near the coil that is actually producing sound. This causes a loss in high note response and to improve the high frequency output an additional small cone, stiff and lightweight, may be attached to the apex of the main cone. In addition, the coil can be made very light in weight by winding it, for example, with aluminium wire. A loudspeaker can, therefore, reproduce a wide frequency range successfully.

Loudspeaker design is a compromise, and for producing both good bass and good treble the requirements conflict. A lightweight cone of small diameter is needed for the high frequencies, whereas a large and robust cone is needed for the bass. Whilst a detailed description of all the factors affecting loudspeaker design is beyond the scope of this book, the reader will soon realize how the mechanical properties of a loudspeaker affect its electrical characteristics and, hence, its acoustical performance.

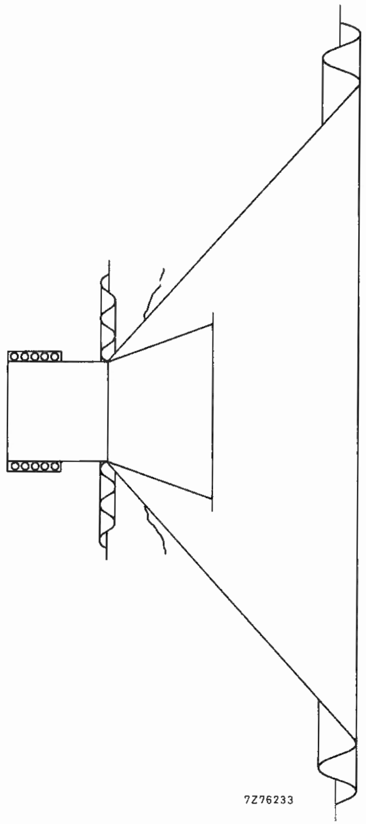


Fig. 3.6 Addition of small cone increases high tone output.

3.4 Electrical impedance

The electrical behaviour of a loudspeaker over its entire frequency range is very complex. It is usual to study this behaviour by means of an equivalent electrical circuit but, for simplicity, it is best to consider the behaviour over a small portion of frequency range at a time. Let us consider the low end of the frequency range.

It is well known that a body can be excited into vibration when its mechanical dimensions are equal to the wavelength of the sound field in which the body is placed. In a similar way, when the mass of the cone and the stiffness of its suspensions are related to the frequency of vibration, these mechanical properties produce the effect of an electrical parallel resonant circuit in series with the moving coil. Let us compare the two cases.

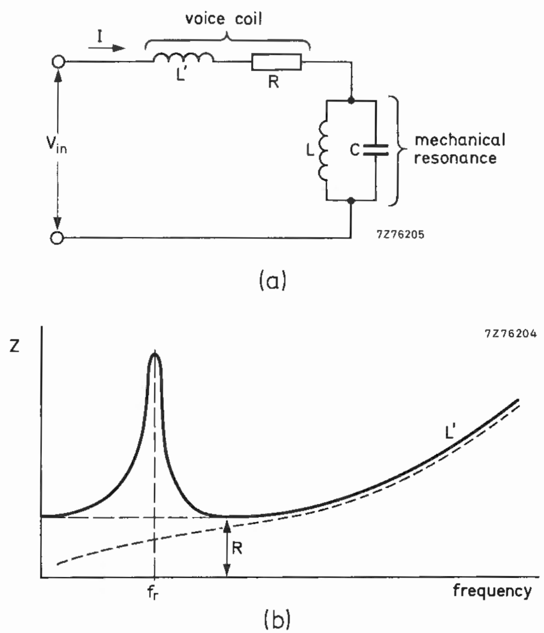


Fig. 3.7 (*a*) Equivalent circuit of a loudspeaker at resonance frequency without a baffle. (*b*) Impedance of loudspeaker without a baffle.

The resonance frequency of a parallel electrical circuit comprising an inductance and a capacitance is given by the well-known equation:

$$f_r = 1/2\pi \sqrt{(LC)},$$

where L is the inductance and C is the capacitance.

When we consider a loudspeaker, the resonance frequency is given by:

$$f_r = \frac{1}{2\pi}\sqrt{\frac{S_s}{M_d}},$$

where S_s is the stiffness of the suspensions and M_d is the dynamic mass.

If we think of the suspensions in terms of their ease of bending, or *compliance*, rather than their stiffness, we can substitute compliance (C_s) for stiffness in the above equation, where

$$\text{compliance} = 1/\text{stiffness} = C_s = 1/S_s.$$

Thus we can write

$$f_r = 1/2\pi \sqrt{(M_d C_s)}.$$

We can now see that the dynamic mass of the moving system is behaving like an electrical inductance, and the compliance like a capacitance.

At resonance, a parallel electrical circuit exhibits a high impedance across its ends; in the same way with a loudspeaker, at resonance frequency the impedance due to the effect of the dynamic mass and the compliance rises to a maximum acting in series with the resistance and inductance of the moving coil. Around the resonance frequency, which is normally low (around 50 Hz) for a full-range loudspeaker, the inductance of the moving coil has little reactance and the only significant impedance is that due to the resistance of the wire with which the coil is wound. At higher frequencies, however, the inductance of the moving coil becomes effective and the impedance of the coil begins to rise. The rated impedance of the loudspeakers described in this book is taken as the lowest value of the impedance occurring above the resonance frequency.

3.5 Frequency characteristic

When a constant amplitude electrical signal is applied to an *unmounted* loudspeaker, the sound pressure begins to fall off at the rate of 6 dB/octave below a point at which the half-wavelength of the sound produced is equal to the distance from the front of the speaker to the rear, as the frequency is lowered. This effect is known as acoustic *short-circuiting* and depends on the loudspeaker dimensions. A further attenuation of 12 dB/octave occurs when the resonance frequency of the loudspeaker is reached. This is due mainly to the inflexibility of the suspension, and makes the total 18 dB/octave. Over the middle frequency range, a fairly uniform response is obtained but, when the inertia of the moving mass becomes too great at high frequencies, the response starts to fall off at 12 dB/octave.

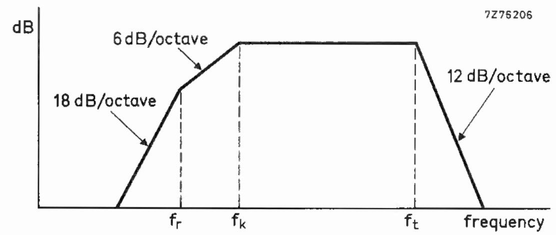
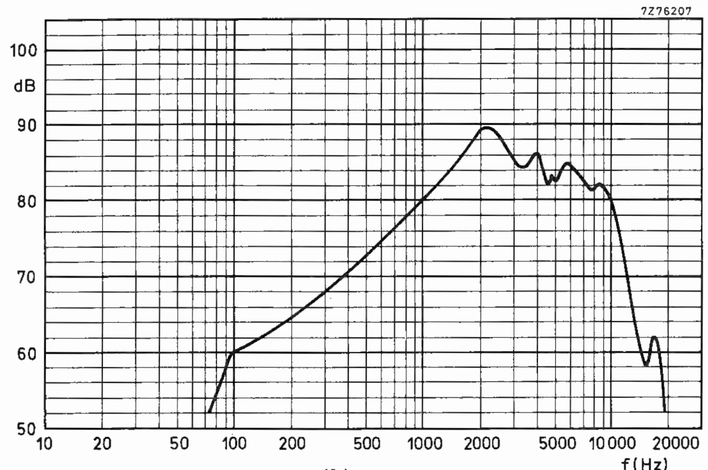


Fig. 3.8 Frequency response characteristic of ideal loudspeaker without baffle. Low frequency roll-off starts where acoustic cancellation occurs at f_k. Below resonance frequency f_r, a further 12 dB/octave is added to roll-off. At high frequencies above f_t, voice coil inductance takes control.

We can see now that below resonance, the performance of a loudspeaker is compliance controlled, and above resonance it is mass controlled. Hence, reducing the stiffness by using a highly compliant surround is necessary for good bass response and a stiff cone of low mass is needed for good treble.

In order to obtain a good bass response we can see that the uniform response of the middle frequencies should continue downwards in frequency as far as possible before roll-off occurs. As we shall see in the next chapter, the initial

(*a*)



(*b*)

Fig. 3.9 (*a*) Frequency response curve of typical full-range loudspeaker. 0 dB $= 2 \times 10^{-4}$ μbar SPL. (*b*) Useful part of response curve above 52 dB.

drop in response of 6 dB/octave can be avoided by preventing the acoustic short-circuiting taking place. Since this was due to acoustic cancellation by out-of-phase sound waves radiated from both the front and rear of the cone, all that is necessary is to prevent the rear radiation by mounting the loudspeaker in a hole on a large panel or a box. The roll-off at 12 dB/octave in response below resonance is another matter, however, as it is a fundamental characteristic of all moving coil loudspeakers, so it is desirable to have as low a resonance frequency as possible.

When we consider our equation for the resonance frequency of an unmounted loudspeaker:

$$f_r = 1/2\pi \sqrt{(M_d C_s)},$$

we see that increasing the mass of the moving system will lower the resonance frequency. This means large and heavy cones should be employed for good bass response. At the same time, since the resonance frequency must be low, the compliance should be high and the motion of the cone restricted as little as possible by the suspension. We have thus defined the requirements that apply at low frequencies.

3.6 Acoustic radiation and polar response

Low frequencies are always well diffused in the listening area and the acoustic radiation from the cone is only very slightly directional (up to about 500 Hz for a 10-inch speaker) and, in practice, low frequency directivity can usually be ignored. When the frequency is increased, we know that the cone no longer behaves as a rigid piston and vibration tends to take place nearer and nearer to the apex of the cone as the frequency goes higher and higher. The remainder of the cone surface produces a 'beaming' effect and tends to concentrate the acoustic radiation along the axis of the cone, the sound pressure falling off considerably as the angle from the axis increases. This beaming effect considerably detracts from realism and the quest for non-directional diffusion accounts for the trend to multi-speaker systems, which will be described later.

The directivity of the acoustic radiation is normally determined by measuring the *polar response*. For this, the loudspeaker is mounted on a turntable in an anechoic room and a constant voltage signal at a particular frequency is applied to the moving coil. A recording microphone is held a specified distance away from the loudspeaker and the turntable is slowly rotated. The test is usually repeated at different frequencies, the results being recorded on polar co-ordinate graph paper.

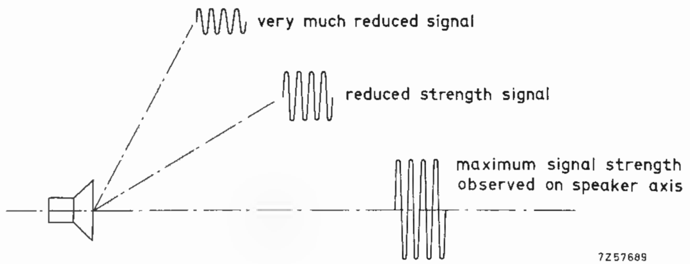


Fig. 3.10 The strength of the high frequencies decreases as the angle to the axis increases.

3.7 Power considerations

Energy is required to produce sound, the sound pressure level due to a loudspeaker being a function of the cone motion which, in turn, depends upon the electrical power delivered to the moving coil. There are three different power ratings to be considered:

- operating power
- power handling capacity
- music power.

Each of these serves a different purpose and there is little direct relationship between them, although an experienced engineer can roughly estimate any two of them from the other one.

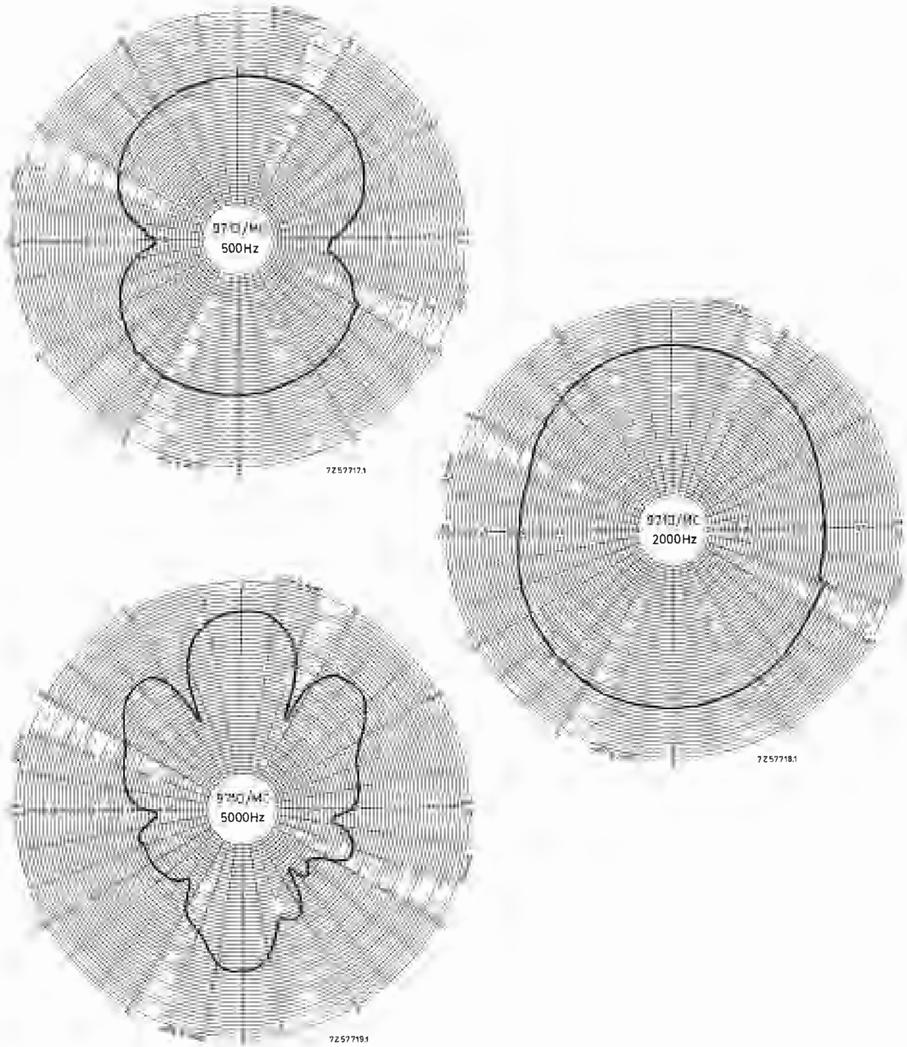


Fig. 3.11 The polar response of a typical unmounted loudspeaker at different frequencies. Note the fall in output at 90° and 270° on the 500 Hz curve due to acoustic short-circuiting.

Operating power (for the loudspeakers described in this book) can be defined as the power input required to produce a sound pressure of 12 μbar at 1 m distance along the axis of the loudspeaker (or 4 μbar at 3 m). Taking a sound pressure of 2×10^{-4} μbar as the reference level (0 dB), 12 μbar = 96 dB SPL (4 μbar = 86 dB SPL). This simplified definition gives us an excellent reference for all acoustical calculations. The operating power is, naturally, in electrical watts and is simply determined by increasing the electrical input to the loudspeaker until the required sound pressure at the appropriate distance is reached.

A sound pressure level of 96 dB represents a loud sound. In Section 2.6, we discussed sound pressure levels and their relationship to loudness and listening. Clearly, 96 dB would be a sound pressure level which many listeners would not wish to exceed in their homes, while a few enthusiasts who like to feel the music rather than listen to it would consider 96 dB only a 'good average'.

In either case, specifying the operating power in this way gives a very clear idea of the capabilities of a loudspeaker. For example, if the operating power of a loudspeaker is quoted as 1 W, we now know that this will produce a sound pressure level on axis at 1 m from the loudspeaker of 96 dB.

But one thing which the specification of the operating power does not tell us is how much power a loudspeaker can withstand before it fails to work properly, or is damaged. There are two ways in which this can be specified:

- power handling capacity
- music power rating.

Let us consider our loudspeaker with an operating power of 1 W. Suppose we wish to take account of those higher level sounds around 100 dB. This is 4 dB above the sound pressure level of 96 dB and represents an increase of about 2,5 times. Our electrical power requirement has now risen to 2.5 W. But what happens if we want to give some bass boost, or use a loudness control, with a further 10 dB increase? This represents a ten-fold increase in the power which the loudspeaker has to handle, and the total becomes 25 W.

We can now see that the operating power on its own is insufficient to completely specify the loudspeaker and, in addition to knowing how much power we need to produce a given sound pressure level, we also need to know how much power our loudspeaker is capable of handling. This is what we mean by the *power handling capacity*; for the loudspeakers mentioned in this book, it represents the maximum *continuous* power the loudspeaker is designed to withstand.

There is another way of specifying the power handling capabilities of loudspeakers, namely, the music power rating. This is usually measured in terms of pulsatory loading representing music and speech at the low frequency end of the response curve, where distortion is not so readily heard, and is the maximum power which may be applied without observing a rattling, buzzing, etc., below 250 Hz. Due to the large number of variables which may occur in defining the overall performance of a sound reproduction system, it is much more reliable to use the continuous power rating throughout, i.e. sine-wave power for the amplifier, and power handling capacity for the loudspeaker. This point was mentioned in Section 2.6. When these ratings are used, there will be no doubt that the loudspeaker and amplifier will be correctly chosen for power considerations. While still discussing power considerations, it is useful to consider what happens when a loudspeaker of a different power rating to the amplifier is used. If the loudspeaker has a power handling capacity *greater* than the maximum continuous sine-wave rating of the amplifier, no damage will occur to the loudspeaker and, since there will be no overloading, distortion will be minimum. However, if the loudspeaker has a power handling capacity *lower* than the continuous (sine-wave) rating of the amplifier, when the volume control is turned fully up damage may be done to the loudspeaker. It is unlikely that any serious listener would do this, because an untolerable level of distortion will be reached before the conditions for damage occur, but the risk is still there, nevertheless.

3.8 Distortion and damping

Distortion in any loudspeaker can be caused by non-linearities in the cone suspension system and also by the cone itself. Additionally, lack of uniformity of the magnetic field in which the moving coil vibrates can also cause distortion.

The action of the suspension should be linear out to the maximum excursion of the cone, so that the cone motion is directly proportional to the force applied. With large cone movements, this is sometimes difficult to achieve and non-linear distortion occurs. Most loudspeakers employ paper pulp for the cone material, moulded to suit the required configuration. This material can be considerably non-linear, especially as its thickness is reduced.

Unless the magnetic field in which the coil moves is uniform, the cone motion will be non-linear. Two methods are used to overcome this non-linearity. If a short coil is used, coil movement in the fringe area at the ends of the gap is avoided; if a large coil is used, one end of the coil moves into a region of higher flux density as the other end of the coil is moving into a region of lower flux density, the product (turns $\times$ flux cut) remaining constant.

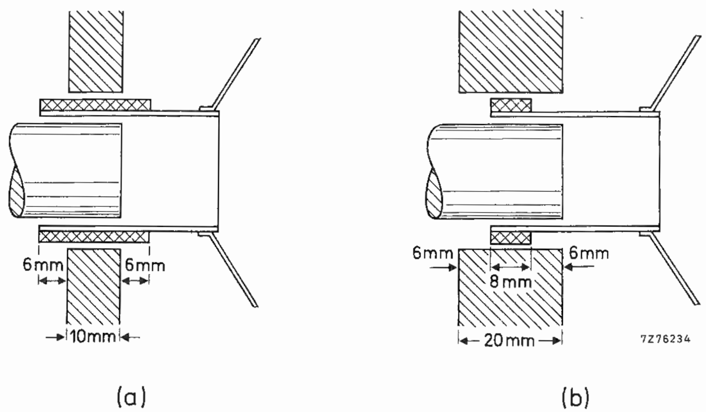


Fig. 3.12 The construction of the voice coil: (*a*) shows a long coil in a short magnetic field; (*b*) shows a short coil in a long field. Both methods are used to overcome non-linearity in the field strength which cuts the turns of the voice coil.

In addition to the non-linear distortion arising for the reasons so far described, there is one particularly annoying form of distortion; namely, *transient distortion.* This is the inability of the loudspeaker to respond to a short duration pulse without distortion of the waveshape and, particularly, without the *addition* of any frequencies. Good transient response requires a smooth frequency characteristic and this is not easy to obtain in a complex mechanical system. After removal of the driving pulse, the moving elements, excited by the coil but not necessarily rigidly coupled to it, continue to oscillate on their own. It follows that some form of damping is therefore necessary.

A detailed discussion on damping is not within the scope of this book. However, it is important to remember that at resonance frequency, when the mass

reactance of the moving system equals the compliance reactance of the suspension and the mechanical components behave as a parallel tuned circuit in series with the moving coil, there is a magnification of the energy within the system and a tendency to increased self-oscillation at the resonance frequency. In addition, it should be remembered that the restoring force on the moving system is provided by the suspension, and where a very compliant suspension is employed there will be a greater tendency to continued oscillation and the moving system will not accurately follow the electrical signal. In the latter case, the sound from the loudspeaker would lack 'attack' and distortion on transients would be unacceptable.

The magnification of the response at resonance is similar to the circuit magnification factor (or quality factor), Q, of a parallel resonant circuit. We can, therefore, speak of the Q of a loudspeaker at its bass resonance frequency. To restrict the Q of the speaker to an acceptable level we have to introduce some form of *damping*. This is normally obtained electrically by the internal resistance of the amplifier which acts as a parallel resistance across the equivalent resistance of the moving coil. Modern solid-state amplifiers have a very low output resistance which acts as the source resistance for the loudspeaker. The *damping factor*, which is the ratio of load impedance to source resistance can be easily as high as 200.

In view of the low internal resistance of the amplifier, it is important that the resistance of the speaker cables which run from the amplifier to the loudspeaker does not significantly reduce the damping factor. Since damping is vital in the control of transient performance, due regard should be given to this aspect.

An interesting consequence of the effect of source resistance is shown in Fig. 3.13. Two curves are shown of the frequency response of a 5-inch speaker mounted in a 7-litre box filled with glass wool. One curve shows the response with a constant voltage input, the other with a constant current input. The constant voltage condition corresponds to a source resistance of zero, whereas in the constant current condition the source resistance can be taken as infinity. The effect of varying the source resistance between zero and infinity is clearly shown, a high Q resulting in the case of a high source resistance. Since a modern solid-state amplifier offers a low source resistance to the speaker, and corresponds to a nearly constant voltage generator, the underdamped condition shown in Fig. 3.13 does not normally apply, assuming the effect of speaker cable resistance can be neglected.

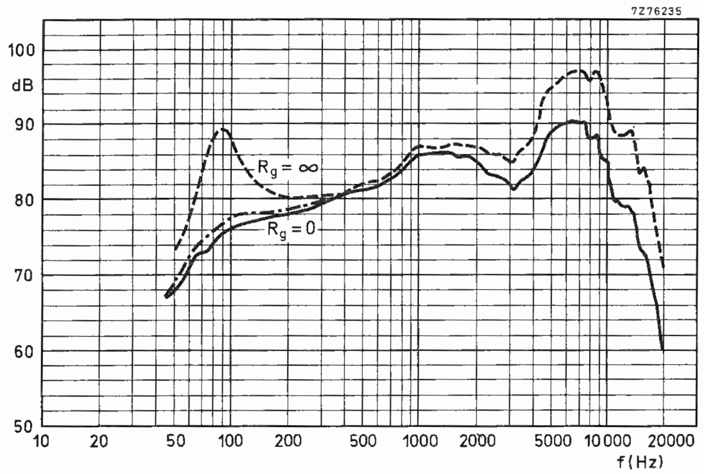


Fig. 3.13 Effect of source resistance on the speaker response characteristic. Dotted line shows constant current condition, where source resistance $R_g = \infty$; full line indicates constant voltage condition where $R_g = 0$. The chain dotted line gives the response with a typical solid-state amplifier.

3.9 Practical loudspeakers

We are now in a position to discuss how best we can meet the requirements for high quality sound reproduction. So far we have assumed that we have a loudspeaker for producing the full frequency range with equal quality and we have examined its requirements and its behaviour, but we have not said exactly how we meet all the requirements at the same time. The answer is that it is economically impossible to meet such a specification, and there is also another very good reason why it is unnecessary to do so.

The relationship between the force exerted on the moving system and the corresponding displacement is not linear. This gives rise to distortion, which is worst when the cone displacement is greatest. If a low tone which gives rise to a large cone displacement has to be reproduced together with a high tone which causes a small displacement, the tops of the waves will be distorted. This effect is very noticeable and gives the sound a disagreeable harshness. It is called

modulation distortion. Obviously, this is a very good reason for reproducing the high tones separately from the low tones, using speakers specially designed for each part of the frequency range.

From our earlier discussions on the differing requirements for high and low frequencies, we know that a speaker for low frequencies should have a large and heavy cone, and a speaker for high frequencies a small and light one. This is exactly what we provide to obtain high quality sound. A speaker specially designed to reproduce low frequencies is known as a *woofer*, and one specially designed for the high frequencies is known as a *tweeter*.

Loudspeakers system employing both a woofer and a tweeter are called *two-way systems*. Two-way systems are very popular and offer an excellent solution to providing high quality sound at a resaonable cost. The electrical division of the frequency spectrum is normally carried out by means of a filter network as shown in Fig. 3.14. A more advanced system may be employed in which the

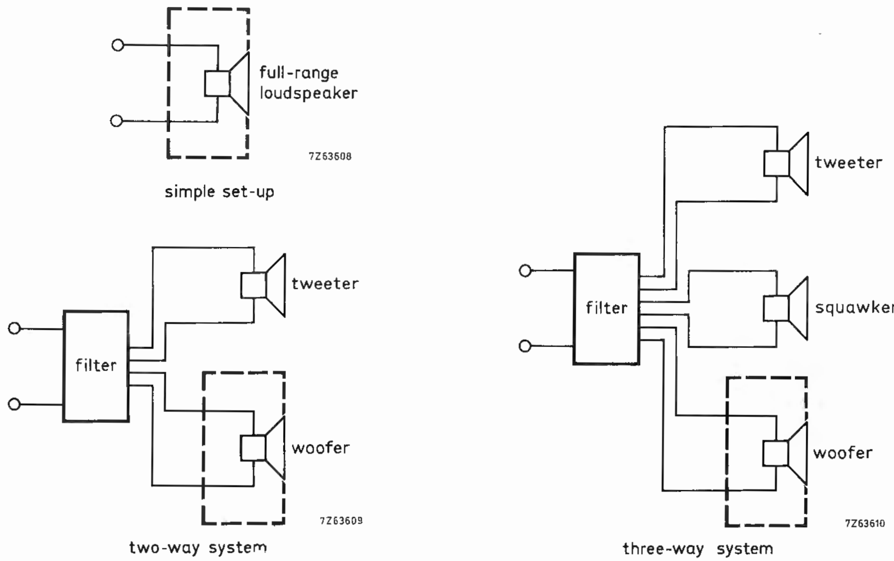


Fig. 3.14 Different methods of covering the audio spectrum. The dotted line around the woofers indicates a sealed enclosure.

frequency range is split up into three groups of frequencies. This is known as a *three-way system* and uses a woofer for the bass reproduction, a tweeter for the treble and a third speaker for the mid-range tones. This third speaker is known as a *squawker.* A three-way system incorporating a woofer, a squawker and a tweeter will give the most perfect coverage of the whole of the audio frequency spectrum and, although it is obviously the most expensive, the results make additional costs more than worthwhile.

WOOFERS

Six sizes of woofer loudspeakers are currently available in our range: 4-inch, 5-inch, 7-inch, 8-inch, 10-inch and 12-inch nominal diameter. In addition, a 15-inch diameter speaker is now in development. The 4-inch AD4060/W is intended for use in bass reflex enclosures up to 15 W, or sealed enclosures up to 30 W, and the 5-inch AD5060/W is for up to 10 W in small sealed enclosures up to three litres. 7-inch speakers include four types for sealed enclosures up to 40 W system power. Eight 8-inch types offer the greatest choice with speakers for up to 70 W systems and lower frequency responses. Two 10-inch and four 12-inch types complete the current range of woofers with rated frequency responses down to 30 Hz and system powers to 100 W. The woofers are summarised in Table 3.1.

Table 3.1 Woofers

nominal radiator diameter (in)	type number	power handling capacity in sealed enclosure (W)	enclosure volume (litres)	resonance frequency (Hz)	operating power (W)
4	AD4060/W	30	2	60	12
5	AD5060/W	10	3	60	2
7	AD70601/W	30	7	45	6,3
7	AD70610/W	30	15	45	12
7	AD70611/W	30	15	45	12
7	AD70650/W	40	7	45	4
8	AD80601/W	50	25	42	5
8	AD80602/W	50	25	42	5
8	AD80603/W	50	25	38	6
8	AD80605/W	30	25	50	4
8	AD80651/W	60	25	39	3,8
8	AD80652/W	60	25	39	3,8
8	AD80671/W	70	25	35	9
8	AD80672/W	70	25	35	9
10	AD10650/W	30	35	20	5
10	AD10100/W	40	50	25	2,5
12	AD12600/W	40	80	28	4
12	AD12650/W	60	80	22	4
12	AD12200/W	80	80	22	5
12	AD12250/W	100	80	24	2,9

SQUAWKERS

For the mid-range frequencies there are eight squawkers: three 5-inch cone types with paper cones, and five 2-inch dome versions of which four have textile domes and one has paper. The domed types provide a more uniform pattern of acoustic radiation than the cone types which are considerably more directional. Used singly, they are suitable for system powers up to 80 W. All squawkers are *sealed* at the rear to isolate them from the woofer when they are mounted in the enclosure. Table 3.2 gives the main characteristics of the squawkers.

Table 3.2 Squawkers

nominal radiator diameter (in)	type number	type of radiator	power handling capacity (at squawker) (W)	resonance frequency (Hz)	operating power (W)
2	AD0210/Sq	paper dome	60	350	5
2	AD0211/Sq	textile dome	60	350	5
2	AD02110/Sq	textile dome	80	340	5
2	AD02150/Sq	textile dome	80	340	3
2	AD02160/Sq	textile dome	80	320	5
5	AD5060/Sq [1])	paper cone	40	210	4
5	AD5061/Sq [2])	paper cone	80	680	2
5	AD5062/Sq [1])	paper cone	60	220	4

[1]) AD50600 will replace AD5060/Sq and AD5062/Sq.
[2]) AD50601 will replace AD5061/Sq.

TWEETERS

With a total of no less than 22 different tweeters to choose from, the reader will easily find one to meet his particular requirements. He can select from $2\frac{1}{4}$-inch cone, 2-inch cone, 1-inch and $\frac{3}{4}$-inch dome types. The 1-inch dome tweeter range covers a choice of three dome materials: polycarbonate, textile and paper — all with different frequency characteristics. In addition, there is a choice of front plates, round or square, for non-exposed, semi-exposed and exposed domes. Three types are embellished with aluminium trim rings. The main characteristics of the tweeters are given in Table 3.3.

All the loudspeakers so far mentioned are available with rated impedances of 4 Ω and 8 Ω. In addition, all tweeters except the $2\frac{1}{4}$-inch types are also available in 15 Ω versions.

Before we bring this Chapter to a close we would like to mention our $8\frac{1}{2}$-inch loudspeaker type 9710/M8. This is an extremely sensitive speaker which, over a number of years, has become the most popular type for hi-fi enthusiasts. It has an exceptionally smooth response from 45 Hz to 19 kHz. Power handling capacity is 20 W in a sealed enclosure of up to 30 litres volume, and up to 10 W in bass-reflex enclosures over 30 litres an example of which is given in Chapter 7. Full details of the 9710/M8 are given in Chapter 9.

Table 3.3 Tweeters

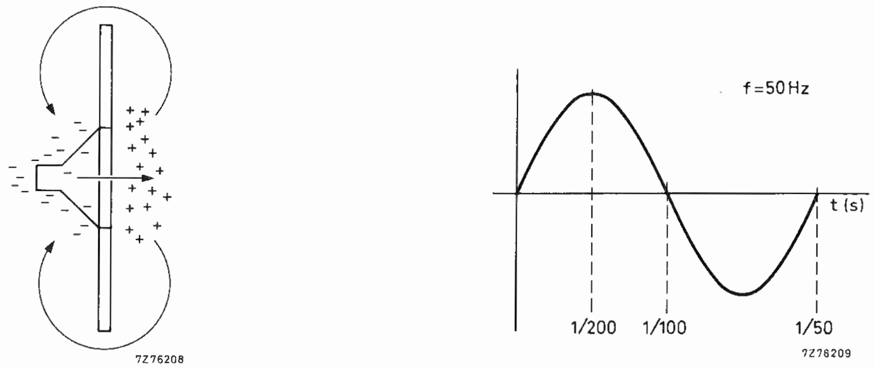
nominal radiator diameter (in)	type number	type of radiator	system power (W) cross-over 2000 Hz	4000 Hz	resonance frequency (Hz)	mechanical design
$\frac{3}{4}$	AD00400/T	textile dome		75 +	1500	SQ N
$\frac{3}{4}$	AD00800/T	textile dome		75 +	1000	SQ N
1	AD0140/T	polycarbonate dome	20	40	1200	R N
1	AD0141/T	textile dome	20	50	1450	R N
1	AD0162/T	polycarbonate dome	20	50	1000	R N
1	AD0163/T	textile	20	50	1300	R N
1	AD01411/T	textile dome	20	50	1450	SQ N A P
1	AD01420/T	paper dome	50	70	950	SQ N P
1	AD01421/T	paper dome				SQ N A P
1	AD01430/T	textile dome	50	70	1100	SQ N P
1	AD01431/T					SQ N A P
1	AD01630/T	textile dome	20	50	1300	SQ N
1	AD01632/T	paper dome	50	70	1300	SQ N
1	AD01631/T	textile dome	20	50	1300	SQ N A
1	AD01633/T	paper dome	50	70	1300	SQ N A
1	AD01610/T	textile dome	20	50	1250	SQ S
1	AD01600/T	textile dome	20	50	1250	SQ E
1	AD01605/T	textile dome	20	50	1250	SQ E A
2	AD2096/T	paper cone	10		1300	R
2	AD2296/T	paper cone	10		1300	SQ
$2\frac{1}{4}$	AD2273/T	paper cone	10		1000	SQ
$2\frac{1}{4}$	AD2274/T	paper cone	10		1000	SQ

R = round, SQ = square, E = exposed dome, S = semi-exposed dome, N = non-exposed, A = aluminium trim rings, P = with damping pot.
+ System power for cross-over frequency 5000 Hz.

4 Loudspeaker enclosures

4.1 The infinite baffle

In Section 3.5, we briefly described acoustic short-circuiting. Let us now consider this question more fully. When the cone moves forward, compression of the air takes place in front of it and a rarefaction takes place behind it. When the loudspeaker is mounted on a relatively small baffle board the compressed air spills around the edge of the baffle into the zone of rarefaction still present at the rear, thereby inhibiting the excursion of the cone. This is shown in Fig. 4.1.



(a) (b)

Fig. 4.1 (*a*) Effect of small baffle causing acoustic short-circuit. (*b*) Time taken for half a vibration at 50 Hz is 1/100th of a second.

This *acoustic short-circuit* as it is called, worsens towards low frequencies owing to the period of these vibrations being relatively long compared with the treble tones. Let us consider what effect this has on a 50 Hz tone. The period of a single complete vibration is one-fiftieth of a second and that of *one-half* of a vibration (the time it takes for the air to be compressed and rarefied) 1/100th of a second. In this time the wave travels a distance of $1/100 \times 340$ m = 3,40 m.

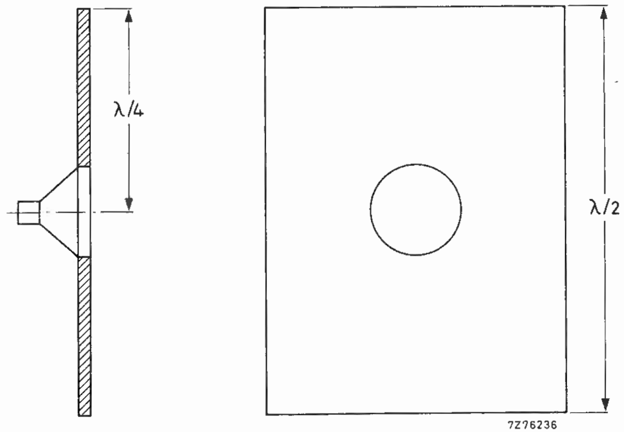


Fig. 4.2 Minimum baffle size is one-half the wavelength of a given tone.

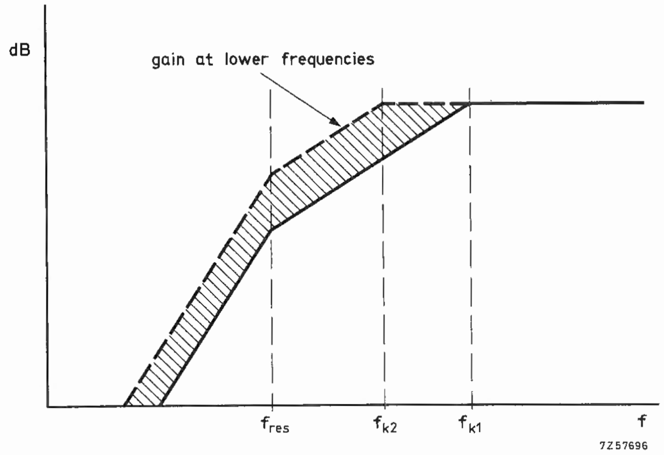


Fig. 4.3 Showing how a baffle board improves the bass response (dotted line, with baffle; solid line, without).

To prevent the air compression on one side of the baffle from having any appreciable effect on the rarefaction on the other side, the distance from the centre of the compression or rarefaction to the edge of the baffle must therefore be *at least* half of 3,40 m, for a 50 Hz tone.

From this we conclude that the minimum length of the side of a baffle to prevent acoustic short-circuiting of a given tone will be half the wavelength of that tone. For a 50 Hz tone, the baffle will have an area, therefore, of $3{,}40 \times 3{,}40 = 11{,}56\ \mathrm{m}^2$.

Obviously, the larger the baffle, the lower the acoustic cancellation frequency becomes. If the baffle is made infinitely large, the bass roll-off does not commence before resonance frequency and the response then falls at 12 dB/octave as the frequency is reduced. It follows that for obtaining the best bass response from a loudspeaker, an infinite baffle is desirable.

4.2 Sealed enclosure systems

The purpose of the baffle was to prevent acoustic cancellation of the radiated sound. But the infinitely large baffle is only a theoretical concept and practical limitations very quickly reduce the usefulness which a baffle can achieve. The same result, however, can be obtained by folding the baffle around the back of the loudspeaker to form a closed box.

Although a totally enclosed cabinet and an infinite baffle are often considered synonymous, there is in fact one major difference between them, namely that the air in the enclosure is compressed when the cone moves in and expands when the cone moves out. This is not, of course, the case with the baffle. The varying pressure of the air inside the enclosure has the effect of an expanding and contracting spring attached to the cone, with the result that the stiffness of the 'spring' changes the effective resonance frequency of the loudspeaker. The degree of the change depends on the volume of the air inside the enclosure.

Let us now consider the effect of the enclosure in greater detail. In the last Chapter we saw that the resonance frequency of a loudspeaker was given by

$$f_r = \frac{1}{2\pi \sqrt{(M_d C_s)}}, \tag{4.1}$$

where M_d is the dynamic mass and C_s is the compliance of the suspensions.

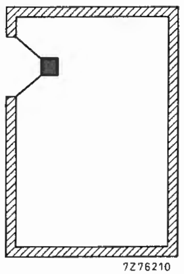


Fig. 4.4 A sealed-enclosure system is an airtight box.

The dynamic mass, $M_d = M_c + M_a$, where M_c is the mass of the moving parts and M_a is the mass of the air moved on both sides of the cone. Equation (4.1) applies only to an unmounted speaker under 'free space' conditions, normally achieved only in an anechoic room.

From equation (4.1), the compliance of the speaker may be calculated:

$$C_s = \frac{1}{4\pi^2 f_r^2 M_d}. \qquad (4.2)$$

The method of determining both the compliance and the dynamic mass is to take two measurements. First, the resonance frequency (f_r) is found by applying a controlled signal to the unmounted loudspeaker in an anechoic room. A known mass m is then applied to the cone and the new lower resonance frequency (f_m) is determined. From equation (4.2):

$$C_s = \frac{1}{4\pi^2 f_m^2 (M_d + m)}. \qquad (4.3)$$

Since the value of the compliance C_s was the same during both measurements, equations (4.2) and (4.3) may be combined, from which

$$4\pi^2 f_m^2 (M_d + m) = 4\pi^2 f_r^2 M_d \qquad \text{and} \qquad M_d = \frac{m f_m^2}{f_r^2 - f_m^2}. \qquad (4.4)$$

The value of M_d obtained from equation (4.4) may be substituted in equation (4.2) and hence the compliance C_s calculated.

When the speaker is mounted in a sealed enclosure, at low frequencies the internal volume of air will act as a stiffness which must be added to the stiffness of the loudspeaker suspension system, i.e. the total stiffness becomes $S_s + S_b$, where S_b is the stiffness of the air in the box. Now, compliance is the reciprocal of stiffness, hence

$$S_s + S_b = \frac{1}{C_s} + \frac{1}{C_b} = \frac{C_s + C_b}{C_s C_b}. \qquad (4.5)$$

From this, we see that equation (4.1) can now be modified to include the effect of the enclosure and the new resonance frequency for the combination of the loudspeaker in a sealed enclosure becomes:

$$f_{sys} = \frac{1}{2\pi}\sqrt{\frac{C_s + C_b}{M_d \times C_s \times C_b}}. \qquad (4.6)$$

This equation for the system resonance frequency neglects certain radiation effects and the change in air loading when the loudspeaker is mounted in an enclosure.

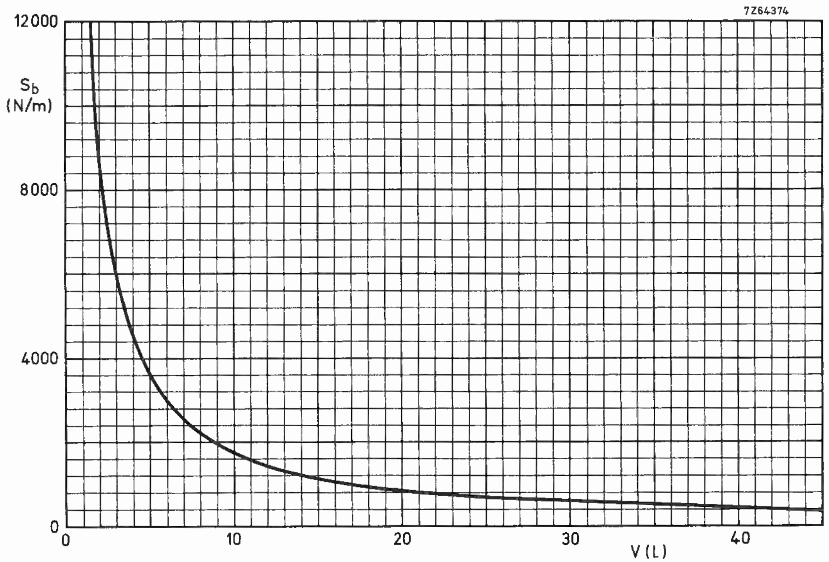


Fig. 4.5 The stiffness of a sealed enclosure rises rapidly at low volumes. This curve gives the stiffness of an enclosure fitted with a 7-inch woofer.

If we now combine equations (4.6) and (4.1), the proportional increase in resonance frequency becomes:

$$\frac{f_{sys}}{f_r} = \sqrt{\left(1 + \frac{C_s}{C_b}\right)}. \tag{4.7}$$

If the loudspeaker occupies less than one-third the area of the baffle board on which it is mounted, the ratio of the resonance frequency of the loudspeaker system with the sealed enclosure (f_{sys}) to the resonance frequency of the unmounted loudspeaker (f_r) is approximately:

$$\frac{f_{sys}}{f_r} = \sqrt{\left\{0{,}87\left(1 + \frac{C_s}{C_b}\right)\right\}}. \tag{4.8}$$

This equation may be freely used in designing normal hi-fi sealed enclosures. The value for the compliance of the loudspeaker suspensions may be calculated as described earlier in this section or, for the range of woofer loudspeakers used in the systems of Chapter 7, may be obtained from Table 4.1.

Table 4.1

Type	effective 'piston' dia. (mm)	cone mass (g)	air mass (g)	dynamic mass (g)	compliance (mm/N)	resonance freq. (Hz)
AD4060/W	85	4,9	0,45	5,35	0,9	60
AD70601/W4	125	11,7	1,5	13,2	1,2	45
AD70610/W4	125	10,5	1,5	12	1,1	45
AD80602/W4	160	11	3	14	1,12	42
AD80603/W4	160	15	3	18	1,16	38
AD80652/W4	160	14,5	3	17,5	1,02	39
AD10650/W4	200	32,4	5	37,4	1,82	20
AD12650/W4	250	42	10	52	1,34	22
AD12200/W4	250	57	10	67	0,8	23

The stiffness of the air within the sealed enclosure depends on the enclosure volume and also on the area of the equivalent 'piston'. Expressing this in terms of compliance, the compliance of the enclosure is given by:

$$C_b = \frac{V}{\varrho c^2 A^2}, \tag{4.9}$$

where V = enclosure volume
ϱ = density of the air
c = velocity of sound
A = area of equivalent 'piston'.

For practical purposes, this formula may be simplified to:

$$C_b = 0{,}72 \times 10^{-3} \times \frac{V}{A^2} \quad \text{m/N}, \tag{4.10}$$

where V is the enclosure volume in cm^3, and A is the area of the 'piston' in cm^2.

As an example, let us determine the new resonance frequency of a woofer loudspeaker when it is mounted in a 40-litre enclosure. Data for the woofer are as follows:

resonance frequency (unmounted)	28 Hz
compliance	$1{,}3 \times 10^{-3}$ m/N
effective cone radius	7,5 cm.

From equation (4.10),

$$C_b = 0{,}72 \times 10^{-3} \times \frac{40 \times 10^3}{\pi^2 \times 7{,}5^4} = 0{,}923 \times 10^{-3} \text{ m/N}.$$

Substituting in equation (4.8), we get

$$\frac{f_{sys}}{f_r} = \sqrt{\left\{0{,}87\left(1 + \frac{1{,}3 \times 10^{-3}}{0{,}923 \times 10^{-3}}\right)\right\}} = \sqrt{(0{,}87 \times 2{,}4)} = 1{,}44$$

and

$$f_{sys} = f_r \times 1{,}44 = 28 \times 1{,}44 = 40{,}3 \text{ Hz}.$$

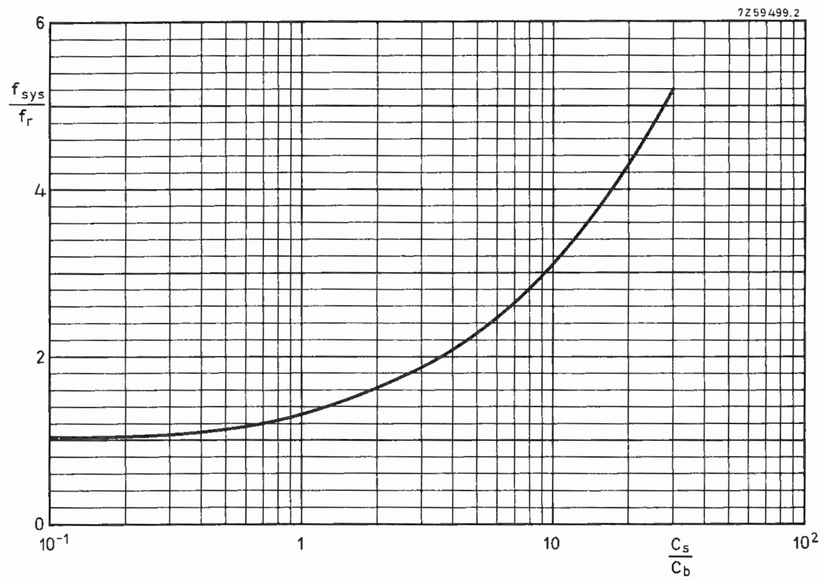


Fig. 4.6 Proportional rise in the resonance frequency of a loudspeaker when fitted in a sealed enclosure.

To avoid having to extract square roots, the reader may obtain the approximate proportional rise in resonance frequency from Fig. 4.6, when the ratio of the compliances has been found.

We now see the importance of the size of the enclosure in determining the bass response. When a small enclosure is used, the bass resonance frequency of the system can very quickly become double that of the speaker alone; for this reason, the unmounted loudspeaker should have a very low basic resonance frequency when it is intended for sealed enclosure service.

In our earlier discussions of the general properties of loudspeakers, we considered operating power and power handling capacity. If we now relate these characteristics to the performance of a sealed enclosure, we see that the increased stiffness of the moving system will reduce the cone excursions for the same power input to the voice coil. In other words, *more power is required* to produce the same sound pressure level. Put it another way, and say the *efficiency* is

reduced. It is extremely important, therefore, that the *conditions under which the power is measured are clearly stated.* For example, if the power handling capacity of a loudspeaker in a 35-litre enclosure is given as 40 W, the unmounted speaker might only be capable of handling 10 W at most without damage.

In principle, there are no special restrictions in the design of the enclosure except that, if the enclosure is unlined, the depth should be *less than one-eighth* of the wavelength at resonance frequency to avoid trouble from standing waves. At higher frequencies, however, standing waves can still occur and, although these are less troublesome than those at resonance, it is usual to damp them out by using a sound absorbent material. To prevent internal reflections, therefore, the enclosure should be lined with a suitable damping material. Glass wool (handle with rubber gloves), which is obtainable everywhere, is ideal.

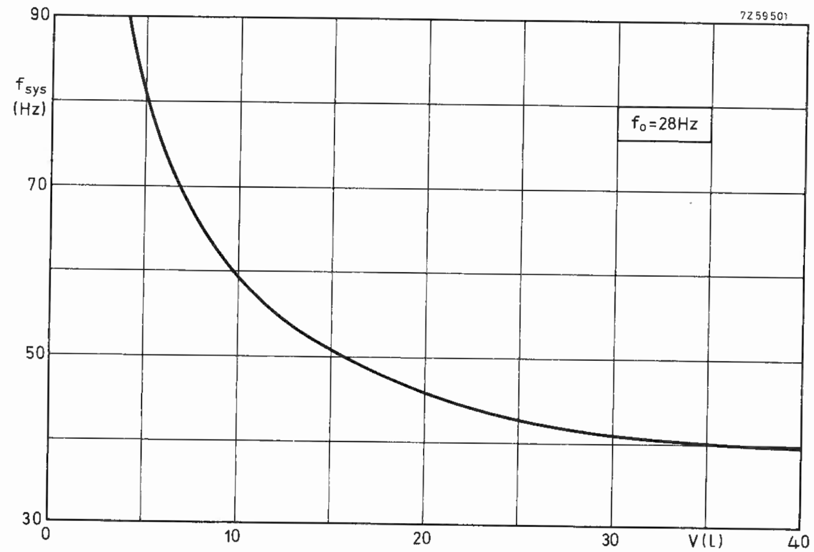


Fig. 4.7 Variation in system resonance frequency with enclosure volume for a 7-inch woofer with a basic resonance frequency of 28 Hz.

The enclosure should be really air-tight otherwise the bass response will be adversely affected. All joints should be glued and screwed, with plenty of hard-setting glue used in the construction. Special attention should be given to the cable entry to make sure that this is air-tight. Self-adhesive polyester foam tape (draught excluder) should be used between the loudspeakers and the baffle board to avoid leaks; if the baffle board is intended to be removable, plastic foam tape should also be used between the baffle board and the enclosure battens to which it is screwed.

In view of the need to make the enclosure air-tight, and also absorb the rear radiation from the loudspeaker cone, the reader will now appreciate why it is necessary to acoustically isolate the tweeter and squawker from the woofer. All our tweeters and squawkers are of sealed construction and require no additional air-tight covers. Full constructional details of sealed-enclosure systems are given in Chapter 7.

4.3 Bass-reflex enclosures

At low frequencies, the radiation from the rear of the cone represents half the total radiated power. The bass-reflex loudspeaker system makes use of this radiation. To do so involves reversing the sense of the air-particle motion at the rear of the cone before adding it to the vibration at the front. The enclosure takes the form of a closed box with the loudspeaker mounted on the baffle, and a hole, or vent, cut in the baffle board to allow the rear radiation to escape. Reversal of the direction of particle motion is achieved by the resonance effect associated with the vented cabinet.

As with a Helmholz resonator, resonance is due to the compliance of the enclosed air and the inductance of the air-mass in the vent, or neck. The particle velocity in the vent is magnified more than the particle pressure, relative to the input velocity and pressure. This corresponds to the input impedance (presented to the rear of the cone) being higher than that of the vent.

The increased low frequency output depends on the phase angle between the resonator input and output quantities. When the enclosure is resonant, this angle is approximately 90° and thus, allowing for the opposite senses of front and rear radiations, the output at the vent is also 90° out of phase with the front cone radiation. At frequencies above enclosure resonance, the vent output

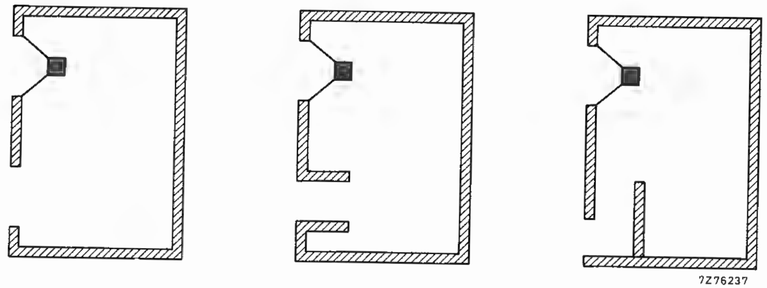


Fig. 4.8 Different forms of construction of a bass-reflex enclosure.

phase moves towards that of the front cone radiation, and the cone radiation is increased. At frequencies below resonance, the vent output phase is such that the cone radiation is reduced.

The coupling of the resonator to the cone also modifies the electrical impedance characteristic. If the enclosure is made to resonate at the cone resonance frequency, the rise in impedance we have previously mentioned may be almost entirely suppressed. At a frequency above resonance, the cone is mass-controlled (inductive) and the enclosure is compliance-controlled (capacitive). At a frequency below resonance, the reverse takes place. Thus there are two possibilities for resonance of the system as a whole. This is shown by the occurrence of two peaks in the impedance curve of Fig. 4.9.

The 'capacitance' of the enclosure varies as the volume; the 'inductive' component is proportional to the ratio of the length to the area of the vent, and is usually varied by forming a duct or tunnel behind the vent so as to allow the vent area to be similar to that of the cone. To allow for end correction on a rectangular duct, the length/area factor is increased by $1/\sqrt{\text{area}}$. The vent area is usually made equal to the loudspeaker cone area, so that the volume required for a given resonance frequency is a function of the length of the tunnel. A long tunnel has the advantage that the cabinet volume is reduced for a given resonance frequency. As a general rule, the tunnel should not be longer than one-twelfth of the wavelength at the resonance frequency. With equal vent and cone areas, a high mechanical impedance is offered to the rear of the cone and most of the output comes from the vent.

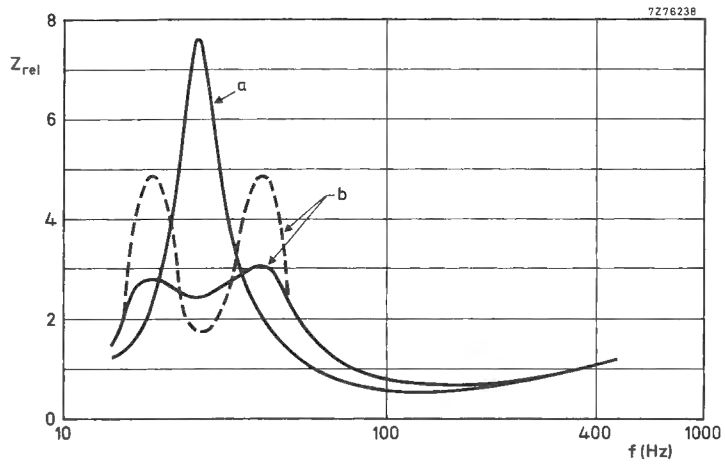


Fig. 4.9 Typical effect of vented enclosure on loudspeaker impedance. (*a*) 'Free air' impedance. (*b*) 'Enclosed' impedance; peak frequencies and magnitudes depend on system tuning and Q factors.

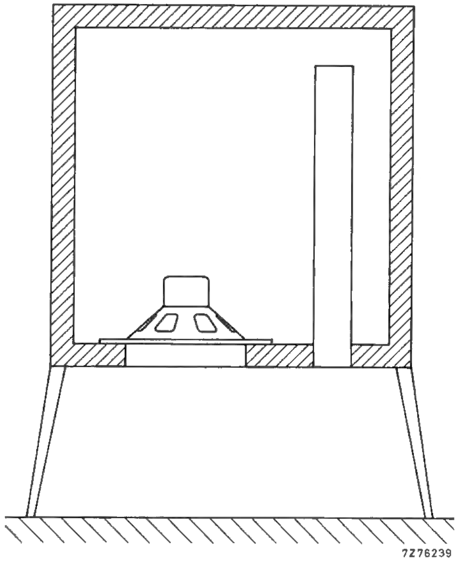


Fig. 4.10 Duct loaded, free-suspension type reflex enclosure, mounted on legs to control spacing from floor to optimize radiation.

Following on the development of the sealed enclosure systems, further work was carried out on the bass-reflex system to reduce its size and maintain its competitiveness. Instead of the radiation being approximately equal from the front and back of the cone at resonance, a newer small duct-loaded system was introduced which produces its major radiation from the duct.

In this case, the combination of the reactances, the compliance and the mass of the duct, act as an impedance transformer as well as a phase reversal coupling from the rear of the cone. This means that the back of the cone sees a stiffness comparable to its own suspension, or possibly even higher, in a similar way to the sealed enclosure system. The difference is that this stiffness is transformed by means of the volume of the box and the mass of the duct into high amplitude radiation at the mouth of the duct.

The overall result is that over quite a range of low frequencies, large amplitude bass may be achieved with a relatively small cone movement. One disadvantage is that the radiation is still restricted by the size of the unit. Some types vent the duct towards the floor to optimize radiation.

4.4 Open-back cabinets

The open-back cabinet commonly used in older models of table and console radios behaves partly like a folded baffle. It has, however, a resonance peak in its response characteristic which is mainly a function of the cabinet although it is also influenced by the loudspeaker characteristics. The height of the peak is from 3 to 6 dB for shallow cabinets and about 6 to 10 dB for deep cabinets.

Open back cabinets are undesirable for good fidelity but, if they must be used, they should be as shallow as possible and with no acoustical obstruction at the rear. They should be mounted at least 15 cm (6 inches) from the wall.

4.5 Summary of enclosure characteristics

So far we have considered only methods of mounting the loudspeaker in an enclosure. Let us now compare the characteristics of these different methods.

The simple baffle board is easiest to make and the least expensive. When it is suspended in the corner of a room with its top edge against the ceiling, pointing downwards, reasonably good quality sound can be obtained down to a frequency determined by the size of the board. The operating power is low and so, also, is the power handling capacity.

Similarly, the open-back cabinet can produce reasonable quality sound but, because of its restricted size and peaky response, it does not make the cost of its construction worthwhile in comparison with other alternatives.

The bass-reflex enclosure is an old favourite and for good bass response it takes a lot of beating. The main objection, however, in this age of miniaturization is the size necessary to obtain the bass frequencies. The operating power is low and the power handling capacity is high. One disadvantage is that unless the constructor is building a design that has already been made and tested, he is faced with a laborious tuning procedure, even if he has the instruments and the knowledge to do the job.

Last comes the sealed enclosure. This is easy to make and probably the design which has the most predictable results. A reasonable performance can be obtained in a very small volume compared with other designs and, from a constructor's point of view, good results can be guaranteed. Superb quality can be obtained with a sealed-enclosure, and the efforts of construction will be amply repaid. The operating power is reasonably low and the power handling capacity high.

It is reasonable to conclude, therefore, that for a wide frequency range of reproduction with a good dynamic range, the sealed enclosure and the bass-reflex enclosure are the best. The simpler construction and smaller volume of the sealed enclosure, coupled with superb performance, have made it the most popular design of all time.

In concluding this Chapter on enclosures, it should be stressed that all enclosure walls should be rigid. Although this may be stating the obvious, it is necessary to point out that our ears receive sound through vibrations in the air: air which has been made to vibrate by the loudspeaker, *and not by the walls of the enclosure*. Vibration of the enclosure introduces *new frequencies* that were not in the original sound, and such distortion is intolerable. Lightly place your finger tips on the top of an enclosure. If you detect any vibration when the volume is advanced the panels are not sufficiently thick.

5 Multi-way speaker systems

5.1 Principles of frequency division

Multi-way speaker systems, as we have seen, provide the best possible acoustic performance, each loudspeaker being specially designed for covering a limited part of the total frequency spectrum. To obtain such performance, one must ensure that the electrical signal input to the different loudspeakers is divided in such a way that each speaker receives only the signals it is required to reproduce. In a two-way system, the signal is divided into two parts; the low frequencies to the woofer, the high frequencies to the tweeter. In a three-way system, where the signal is divided into three parts, the squawker receives the mid-range frequencies, while the woofer and tweeter signals are of a narrower bandwidth.

In some speaker systems, division of the frequency spectrum into as many as five parts has been employed in the past, but with today's designs of loudspeakers this is no longer necessary.

To maintain a continuous response throughout the audio range, the individual frequency characteristics of the separate loudspeakers must overlap each other slightly. The point at which they cross one another is called the *cross-over point* and the frequency at which this occurs is called the *cross-over frequency*. When determining a cross-over frequency and the degree of attenuation either side of it, the frequency characteristics of the loudspeakers must be taken into account, since it is important for each loudspeaker to generate the correct amount of acoustic energy for the part of the audio spectrum it is handling.

It is also equally important that the loudspeakers themselves are chosen with due regard to their sensitivities and power handling capabilities when used in combination with each other.

5.2 Energy requirements

The most important factor in the design of any multi-way speaker system is the *energy* which each loudspeaker has to radiate. Unless the correct balance of power is maintained in the combined acoustic output of all the speakers in the system, the final result will be unacceptable.

For many years, tests have been carried out to determine the power levels of both individual instruments and full orchestras. The results of these tests have been the establishment of what have now become clearly defined standards throughout the world. Although slight differences occur in these 'standards', they are all based, in principle, on the energy content of music.

In Europe, the IEC/DIN Standards clearly define the noise spectrum which is to be used for testing loudspeakers and loudspeaker systems. This is shown in Fig. 5.1 and is based on the energy levels required to be produced throughout the audio spectrum for the correct reproduction of music.

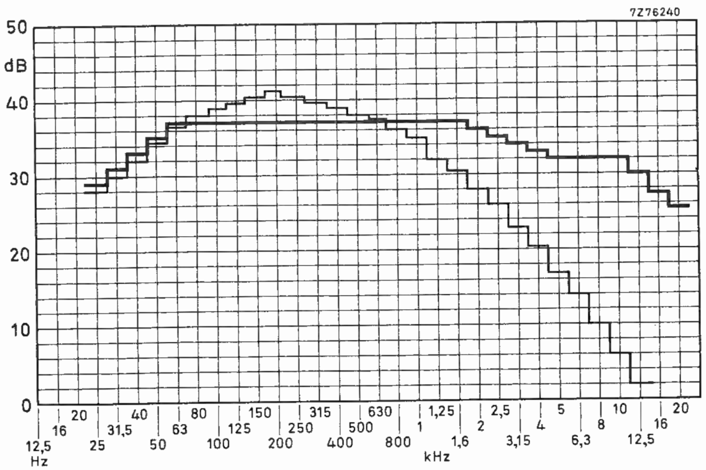


Fig. 5.1 Thin line: IEC/DIN noise spectrum for testing loudspeakers. Bold line: new noise spectrum recommended for testing loudspeakers takes modern music into account.

The development of these standards has, however, resulted from measurements made principally with orchestral music. Recent popular scores have now been measured, from which different conclusions may be drawn. Pop music *does* have a different energy content to classical orchestral works, as the latest results now prove, much greater energy being present in frequencies above about 800 Hz than was originally measured.

In order to test loudspeakers and loudspeaker systems in such a way that their performance will be satisfactory for *all* types of music, including modern pop music, the test signal has to be modified to include the effects of greater energy at high frequencies. The new test spectrum is also shown in Fig. 5.1 and the details of the intermediate filter in Fig. 5.2.

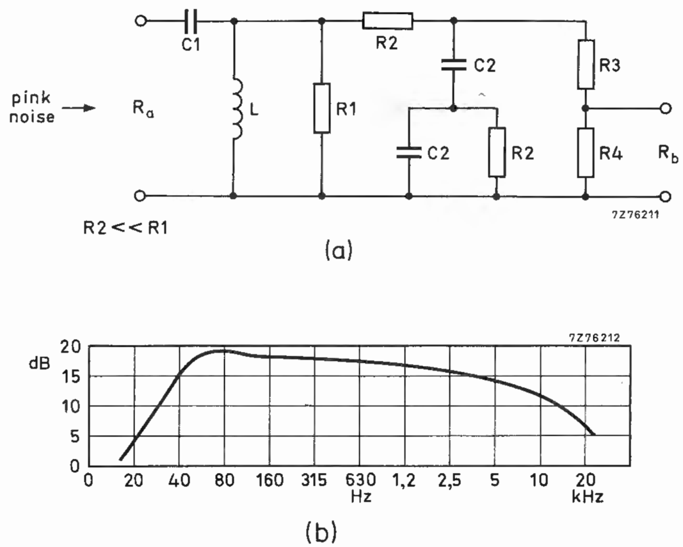
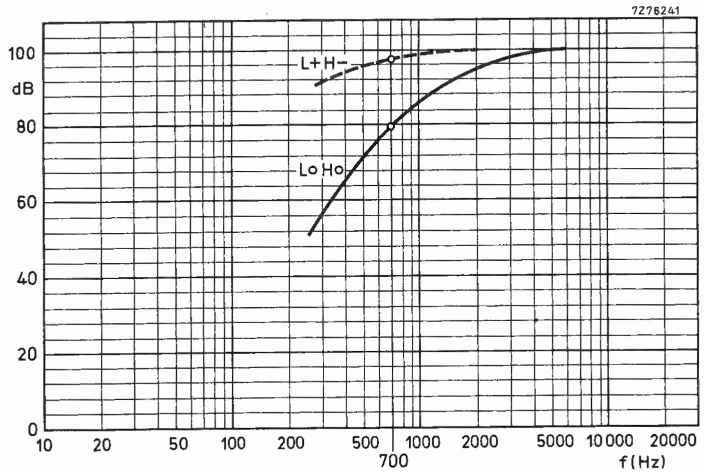


Fig. 5.2 Details of intermediate filter to produce new test signal: (*a*) circuit details, (*b*) frequency characteristic of filter.

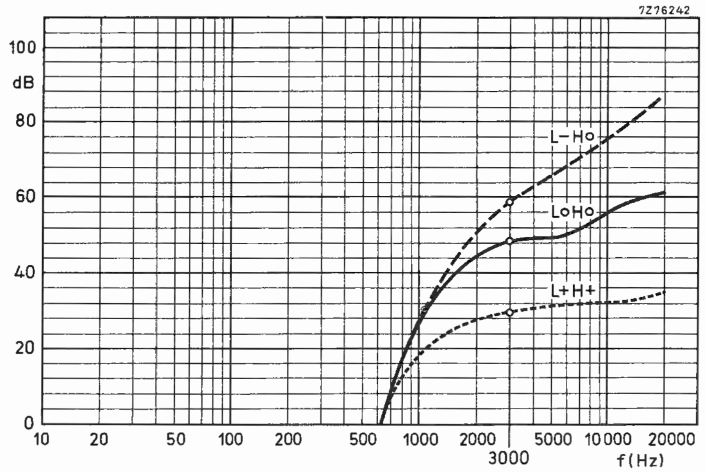
The energy content is of special interest to us here, since this is of considerable importance in the design of a multi-way speaker system. Fig. 5.3 gives one of the results of tests carried out on modern music. A power amplifier was used with separate variable bass and treble tone controls, the bass control providing $\pm$ 16 dB variation, and the treble control $\pm$ 12 dB. In the curves of Fig. 5.3, various combinations of settings of these controls are indicated, since it must be assumed that the loudspeaker system shall adequately handle the amplifier signal under all conditions of control settings.

From Fig. 5.3(*a*), we see that the energy to the woofer at 700 Hz can lie between 80% and 98% of the total; from Fig. 5.3(*b*), that the energy to the squawker may be between 30% and 60% of the total over the band 650 Hz to 3000 Hz; and from Fig. 5.3(*c*), that the tweeter may have to handle from 18% to 70% of the total power at 3000 Hz.

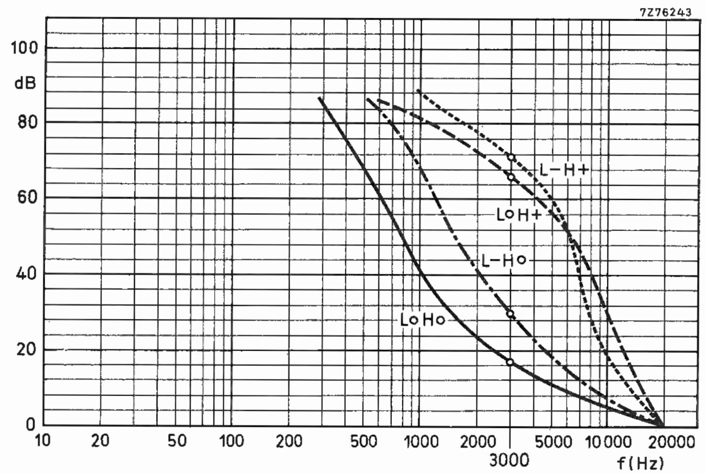


(*a*) Energy at woofer.

Fig. 5.3 Results of typical test on modern music. Amplifier tone control settings are indicated as follows: L+, bass max.; LO, bass flat; L—, bass min.; H+, treble max.; HO, treble flat; H—, treble min.



(*b*) Energy at squawkers.



(*c*) Energy at tweeter.

It is thus possible to combine these results in such a way that we can easily find our power requirements at any frequency. In Fig. 5.4, which results from a detailed study of many recordings, a cross-over frequency of 1200 Hz, for example, shows that 75% of the energy is being fed to the woofer and, if we consider only a two-way system, 25% is fed to the tweeter. Thus for 20 W input, the woofer must handle 15 W and the tweeter 5 W.

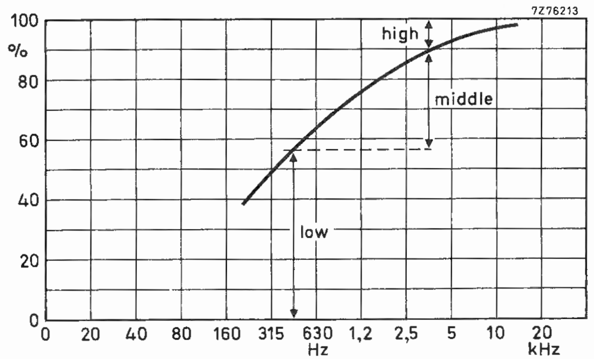


Fig. 5.4 Power distribution using new test signal.

If we consider a 3-way 40 W system with cross-over frequencies of 630 Hz and 2500 Hz, the woofer power will be 25,6 W (64%), that of the squawker 8,8 W (22%), and that of the tweeter 5,6 W (14%). Clearly, our choice of cross-over frequency does not depend only on the frequency characteristics of the speakers. We must consider the power requirements, as well. If our power requirements cannot be met at the cross-over frequency of our choice, and moving the frequency will not give us the margin we need, then we must either use a different loudspeaker with a higher power rating, or use two or more of the same type of loudspeaker suitably connected with regard to impedance.

5.3 Cross-over filter networks

The choice of cross-over frequency is a compromise between the frequency and power responses of the individual speakers and the smoothness of transition

between one speaker and another as the audio signal is divided. There are two basic methods of signal division:

- electronic cross-over systems
- passive filter networks.

Where an electronic cross-over system is employed, individual adjustment of signal levels to each speaker is normally provided and correct tonal balance may be easily achieved. With passive filter networks the smoothness of transition between speakers cannot be of the same high order as with electronic systems. But we are primarily concerned with passive networks on grounds of economy.

There are two basic types of circuits used as passive filter networks. The first type consists of separate high-pass and low-pass filters arranged in series or parallel to provide a four-terminal output from a two-terminal input. The second type, known as a *constant resistance network*, looks identical to the first type but has different component values. The advantage of the constant resistance types is that not only does the input impedance remain constant over the frequency range but, in the case of two-way speaker systems, similar components can have similar values.

The 'classic' approach to filter design is based on transmission line theory, using a fictitious iterative impedance and iterative parameters. Iterative impedance is rather like the characteristic impedance of a transmission line; terminating a filter with this impedance causes an identical impedance to appear, reflected, at the input. But in practice, characteristic impedance always has a real or complex value, and can even be made constant or frequency independent. Iterative impedance, on the other hand, cannot be simulated by any real impedance; at cut-off frequency it can be zero or infinity, in a passband it is real and resistive and varies in value, and in a stop band it is an imaginary, positive or negative reactance.

In view of the similarity of the two quantities, it is mistakenly assumed that terminating the last section of a classic filter in a constant resistance will cause the correct impedance to be reflected back through any number of filters designed for the right value. However, in the vicinity of cut-off this is no longer true, and the use of conventional half-section low-pass and high-pass filters of the *m*-derived type has given way to the constant resistance types in high fidelity applications.

5.4 Constant resistance networks for two-way sytems

Cross-over filter networks for high fidelity applications are characterized by the following features in their transfer response:

- attenuation at the cross-over frequency is 3 dB;
- the slope of the transfer characteristic at the cross-over frequency is half the ultimate slope;
- the ultimate slope is asymptotic to a straight line drawn through zero level at the cross-over point having a slope of 6 dB/octave multiplied by the number of reactive elements, as shown in Fig. 5.5;
- when two filters having complementary responses are fed from a common source and the two outputs are correctly terminated, the *total* power at the outputs will be constant over the passband;
- when two complementary filters are correctly terminated, the impedance presented at their common input will be a constant resistance equal to each terminating resistance;
- the phase transfer response at the cross-over frequency is half the ultimate value;
- the phase difference between the complementary *outputs* is constant, depending on the number of reactive elements.

The transfer characteristics of multi-reactance networks are clearly illustrated in Fig. 5.5. The basic performance of the constant resistance type of network in the low-pass section of a cross-over filter is shown for different numbers of reactive elements.

Constant resistance networks are derived from the circuits given in Fig. 5.6. If the component values are chosen to make $R_o = \sqrt{(L/C)}$, the impedance presented at the input terminals is constant and equal to R_o at all frequencies. At frequencies below $f_o = 1/2\pi\sqrt{(LC)}$, all the input power is delivered to terminals 3 and 4; at frequencies above f_o, all the input power is delivered to terminals 4 and 5. At either side of frequency f_o, the slope of the attenuation characteristic approaches 6 dB/octave. This is normally too low to be of any real value, and can be improved by increasing the number of reactive elements in the filter section. Many loudspeakers of current design normally require filters with an attenuation characteristic of 12 dB/octave for high fidelity applications.

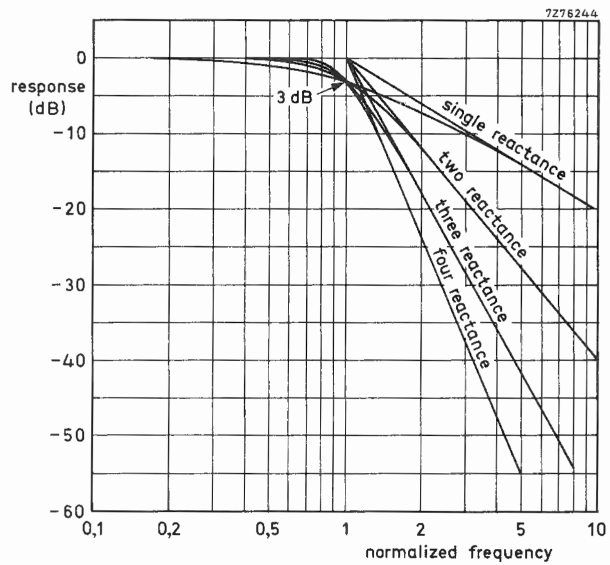


Fig. 5.5 Basic form of 'constant resistance type' response in a low-pass section of a cross-over filter, according to the number of reactance elements employed.

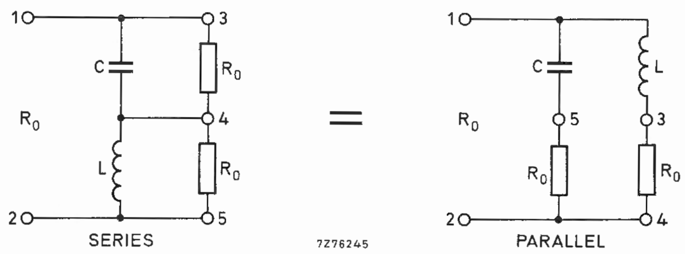


Fig. 5.6 When component values are selected to make $R_o = \sqrt{(L/C)}$, the impedance presented at the input terminals is a resistance R_o.

The values of the inductances and capacitances can be determined in the simple case of the 6 dB/octave filter by multiplying R_o and f_o:

$$R_o = \sqrt{\frac{L}{C}} \qquad f_o = \frac{1}{2\pi\sqrt{(LC)}} \qquad R_o f_o = \sqrt{\frac{L}{C}} \times \frac{1}{2\pi\sqrt{(LC)}}$$

whence

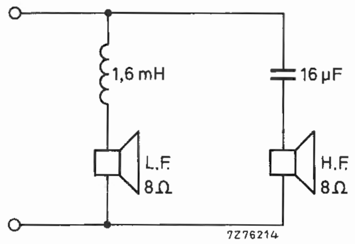
$$C = \frac{1}{2\pi f_o R_o} \qquad \text{and} \qquad L = \frac{R_o}{2\pi f_o}$$

from which

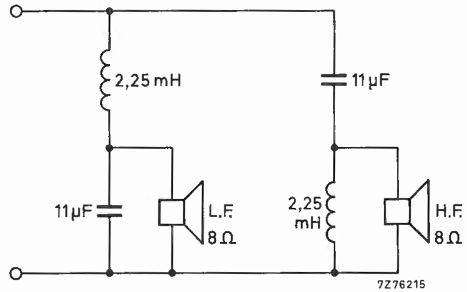
$$C = \frac{159000}{f_o R_o}\ \mu\text{F}, \qquad \text{and} \qquad L = \frac{159 R_o}{f_o}\ \text{mH},$$

where f_o is in hertz and R_o is in ohms.

In the case of single reactance filters, therefore, the reactance of each component is made equal to R_o at the cross-over frequency. For filters having two reactances per section (12 dB/octave types), the components have values that make their reactances equal to $\sqrt{2}$ times R_o in the parallel case and $1/\sqrt{2}$ times R_o in the series case. This means that both inductances have the same value, and both capacitances have the same value in the same filter. Figure 5.7 shows two practical circuit arrangements where the cross-over frequency is 1000 Hz. Figure 5.8 shows the arrangements for cross-over filters for two-way systems; component values are given for different cross-over frequencies in Table 5.1 for 6 dB/octave filters, and in Table 5.2 for 12 dB/octave filters.

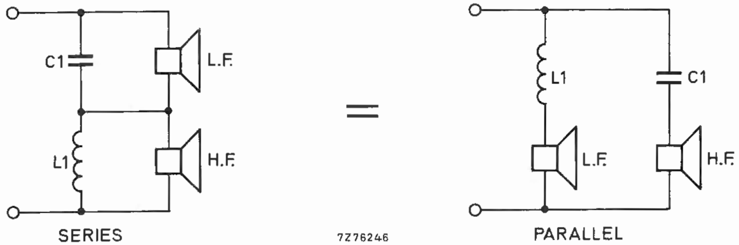


(a)



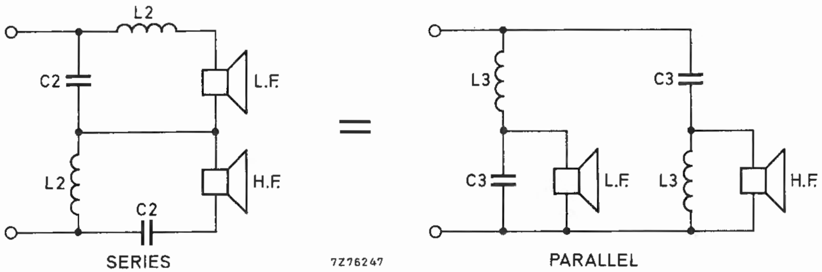
(b)

Fig. 5.7 (a) Cross-over filter for 2-way system. Attenuation is 6 dB/octave, symmetrical; cross-over frequency is 1000 Hz. (b) Cross-over filter for 2-way system. Attenuation is 12 dB/octave, symmetrical; cross-over frequency is 1000 Hz.



6 dB/octave

$$C_1 = \frac{1}{2\pi f_0 R_0} \qquad L_1 = \frac{R_0}{2\pi f_0}$$



12 dB/octave

$$C_2 = \frac{\sqrt{2}}{2\pi f_0 R_0} \qquad C_3 = \frac{1}{2\pi f_0 R_0 \sqrt{2}}$$

$$L_2 = \frac{R_0}{2\pi f_0 \sqrt{2}} \qquad L_3 = \frac{R_0 \sqrt{2}}{2\pi f_0}$$

Fig. 5.8 Constant resistance cross-over filter networks for 2-way systems.

Table 5.1 Component values for the 6 dB/octave filters of Fig. 5.8.

f_0 (Hz)	R_0* (Ω)	L_1 (mH)	C_1 (μF)
500	5	1,6	64
	10	3,2	32
	20	6,4	16
700	5	1,1	45
	10	2,3	23
	20	4,5	11
1000	5	0,8	32
	10	1,6	16
	20	3,2	8
1200	5	0,7	26
	10	1,3	13
	20	2,6	7
1600	5	0,5	20
	10	1,0	10
	20	2,0	5
2000	5	0,4	16
	10	0,8	8
	20	1,6	4
2400	5	0,3	13
	10	0,7	7
	20	1,3	3

Table 5.2 Component values for the 12 dB/octave filters of Fig. 5.8.

f_0 (Hz)	R_0* (Ω)	L_2 (mH)	C_2 (μF)	L_3 (mH)	C_3 (μF)
500	5	1,1	90	2,2	45
	10	2,2	45	4,5	22
	20	4,5	22	9,0	11
700	5	0,8	64	1,6	32
	10	1,6	32	3,2	16
	20	3,2	16	6,4	8
1000	5	0,5	45	1,1	22
	10	1,1	22	2,2	11
	20	2,2	11	4,5	5,5
1200	5	0,47	37	0,94	19
	10	0,94	19	1,87	9,4
	20	1,87	9	3,75	4,7
1600	5	0,35	28	0,7	14
	10	0,7	14	1,4	7
	20	1,4	7	2,8	3,5
2000	5	0,28	22	0,56	11
	10	0,56	11	1,1	5,5
	20	1,1	5,5	2,2	2,8
2400	5	0,23	19	0,47	9,4
	10	0,47	4,7	0,94	4,7
	20	0,94	9,4	1,87	2,3

* Corresponding to nominal loudspeaker impedances of respectively 4 Ω, 8 Ω and 16 Ω.

5.5 Constant resistance networks for three-way systems

Cross-over filters for three-way systems may use band-pass filters for the mid-range frequencies. Their design, however, is a compromise because when a band-pass filter is used with two single-ended (one high-pass and one low-pass) filters, the reactances of the band-pass filter components interact with the reactances of the components of the associated high and low-pass filters. This can be compensated by bringing the design frequencies for the band-pass filter closer together; the interaction between the reactances then spreads the frequencies apart to their correct places.

In a band-pass filter the circuit elements are designed to resonate at the geometric mean frequency $f_m = \sqrt{(f_1 f_2)}$. The series circuit has zero impedance at resonance, whilst the shunt circuit has an infinitely high impedance.

Above resonance the series reactance is positive (inductive), and the shunt reactance is negative (capacitive) and the network acts as a low-pass filter; below resonance the series reactance is negative (capacitive), and the shunt reactance is positive (inductive) and the network behaves as a high-pass filter. Fig. 5.9 shows the principles of design for a band pass filter. The two cross-over frequencies are first established and from these the frequency band ratio is obtained. Because of the interaction of the high and low-pass filters upon the band-pass section, the design frequencies are brought closer together; the design ratio used for the band-pass filter is one less than the frequency band ratio, i.e. design ratio = frequency band ratio minus one. If

f_1 = lower cross-over frequency
f_2 = upper cross-over frequency
f_3 = lower design frequency
f_4 = upper design frequency,

then

f_2/f_1 =frequency band ratio

and

f_4/f_3 = design ratio.

Since the design ratio is made equal to the frequency band ratio minus one, we may write

$$\frac{f_4}{f_3} = \frac{f_2}{f_1} - 1. \tag{5.1}$$

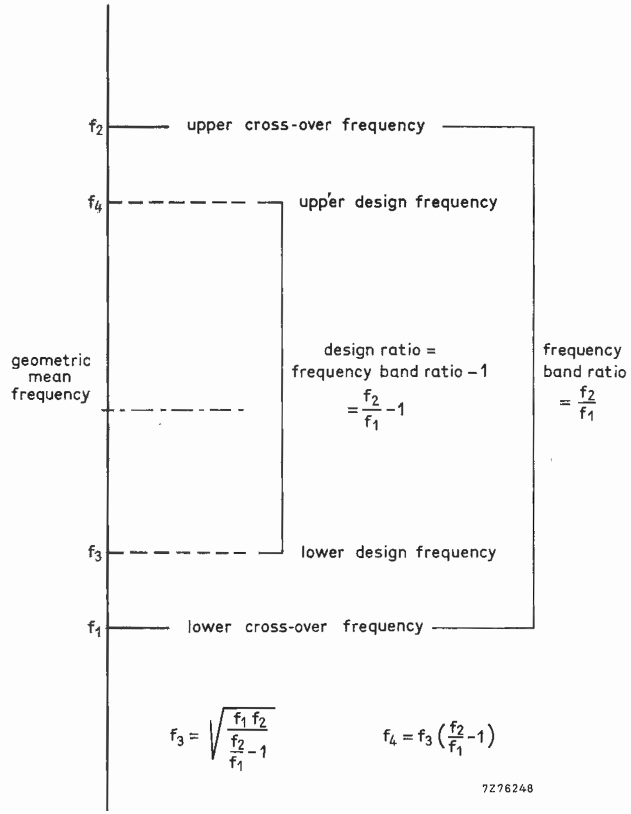


Fig. 5.9 Calculation of design frequencies for a band-pass filter section from the required cross-over frequencies.

As the 'centre frequency' of the cross-over network will lie at the geometric mean of the cross-over frequencies, so this frequency will also be the geometric mean of the band-pass filter design frequencies. Hence,

$$\sqrt{(f_1 f_2)} = \sqrt{(f_3 f_4)} \tag{5.2}$$

from eq. (5.1),

$$f_4 = f_3\left(\frac{f_2}{f_1} - 1\right) \tag{5.3}$$

Substituting this value for f_4 in eq. (5.2), we get

$$f_1 f_2 = f_3^2\left(\frac{f_2}{f_1} - 1\right)$$

from which

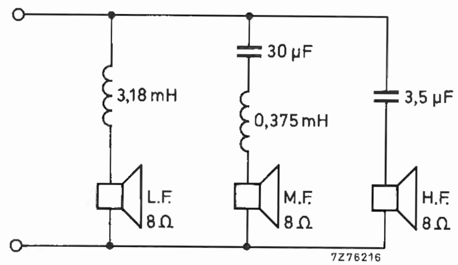
$$f_3 = \sqrt{\frac{f_1 f_2}{f_2/f_1 - 1}}\,. \tag{5.4}$$

Table 5.3 gives an example of how the design frequencies are obtained.

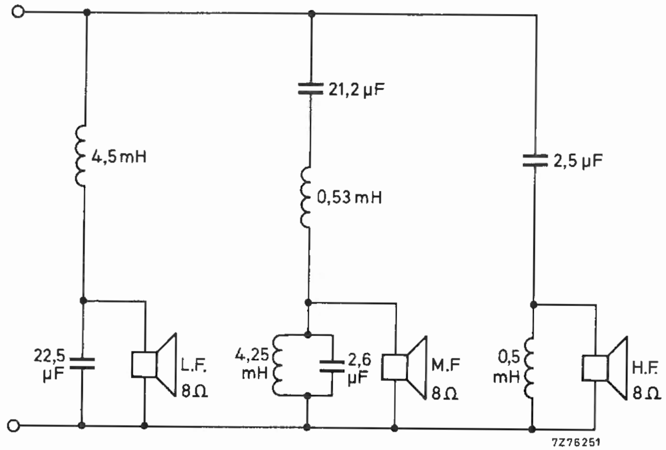
Table 5.3 Design data for examples of Fig. 5.10.

nominal loudspeaker impedance	8 Ω
effective impedance	10 Ω
cross-over frequencies	f_1 = 500 Hz; f_2 = 4500 Hz
frequency band ratio	$f_2/f_1 = 9$
design ratio	$(f_2/f_1) - 1 = 8$
design frequencies	f_3 = 530,3 Hz, f_4 = 4242,6 Hz
geometric mean frequency	f_m = 1500 Hz

Two practical circuit arrangements are shown in Fig. 5.10. Component values for 6 dB/octave and 12 dB/octave cross-over filters for three-way systems are obtained using the data given in Fig. 5.11.

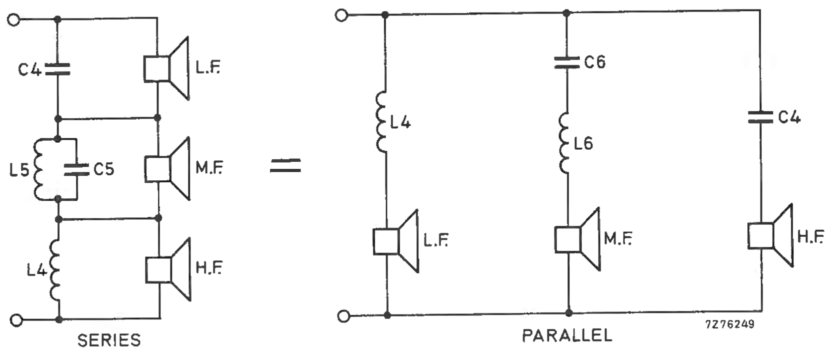


(a)



(b)

Fig. 5.10 (*a*) Cross-over filter for 3-way system. Attenuation is 6 dB/octave, symmetrical; cross-over frequencies 500 Hz and 4500 Hz. (*b*) Cross-over filter for 3-way system. Attenuation is 12 dB/octave, symmetrical; cross-over frequencies 500 Hz and 4500 Hz.



6 dB/octave

$$C_4 = \frac{1}{2\pi f_2 R_0}$$

$$C_5 = \frac{1}{2\pi f_4 R_0}$$

$$L_4 = \frac{R_0}{2\pi f_1}$$

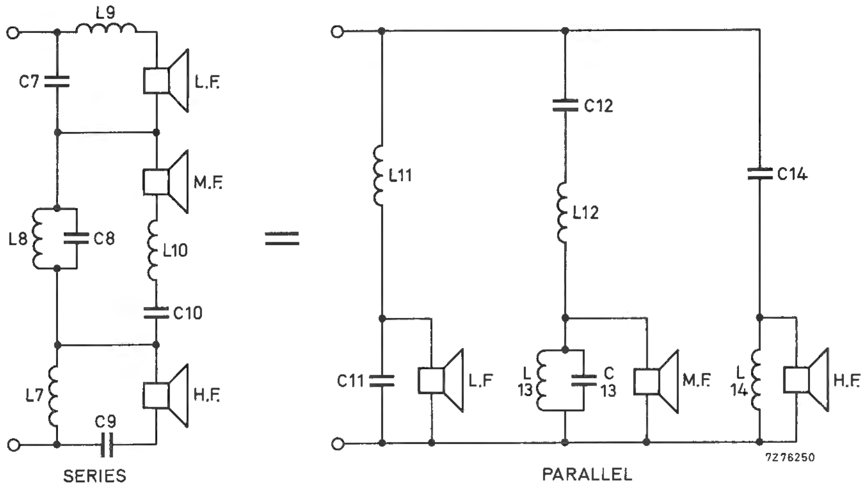
$$L_5 = \frac{R_0}{2\pi f_3}$$

$$C_4 = \frac{1}{2\pi f_2 R_0}$$

$$C_6 = \frac{1}{2\pi f_3 R_0}$$

$$L_4 = \frac{R_0}{2\pi f_1}$$

$$L_6 = \frac{R_0}{2\pi f_4}$$



12 dB/octave

$$C_7 = \frac{\sqrt{2}}{2\pi f_0 R_1} \qquad L_7 = \frac{R_0}{2\pi f_1\sqrt{2}} \qquad C_{11} = \frac{1}{2\pi f_1 R_0\sqrt{2}} \qquad L_{11} = \frac{R_0\sqrt{2}}{2\pi f_1}$$

$$C_8 = \frac{\sqrt{2}}{2\pi f_4 R_0} \qquad L_8 = \frac{R_0}{2\pi f_3\sqrt{2}} \qquad C_{12} = \frac{1}{2\pi f_3 R_0\sqrt{2}} \qquad L_{12} = \frac{R_0\sqrt{2}}{2\pi f_4}$$

$$C_9 = \frac{\sqrt{2}}{2\pi f_2 R_0} \qquad L_9 = \frac{R_0}{2\pi f_1\sqrt{2}} \qquad C_{13} = \frac{1}{2\pi f_4 R_0\sqrt{2}} \qquad L_{13} = \frac{R_0\sqrt{2}}{2\pi f_3}$$

$$C_{10} = \frac{\sqrt{2}}{2\pi f_3 R_0} \qquad L_{10} = \frac{R_0}{2\pi f_4\sqrt{2}} \qquad C_{14} = \frac{1}{2\pi f_2 R_0\sqrt{2}} \qquad L_{14} = \frac{R_0\sqrt{2}}{2\pi f_2}$$

Fig. 5.11 Constant resistance cross-over filter networks for 3-way loudspeaker systems.

5.6 Effect of loudspeaker impedance

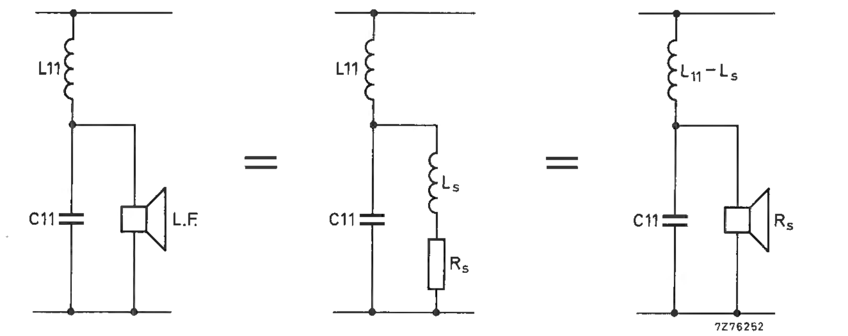
So far, we have assumed that the loads on the outputs of the cross-over filters are purely resistive and constant in value. In practice, when moving coil loudspeakers are connected to a filter, the load presented to the filter will vary with frequency owing to the overall inductance of the voice coil.

Whilst filters designed on the 'classical' basis require correct termination at both ends, constant resistance filters are not critical of input termination. If the outputs are correctly terminated, the input impedance is a constant resistance and the response will be unaffected by the source impedance. But constant resistance networks are critical of output termination and the effect of mismatch at the output depends on how the input impedance is affected.

In a 6 dB/octave filter using single elements, a parallel circuit gives a dip in the reflected impedance around cross-over if the termination is high. For a 12 dB/octave filter this is the case for the series circuit. The effect on the transfer response is dependent on the input source impedance, so if the input matching is improved by the change of output termination there will be a rise in the transfer response. If the matching becomes worse, there will be a fall.

It is preferable to arrange for correct termination in the region of the cross-over frequencies. If a circuit configuration can be used which has a series inductance in the low frequency output, then part of this inductance can be the inductance of the voice coil. This brings about a reduction in the value of the filter component, as shown in Fig. 5.12, and the constant resistance properties of the cross-over filter can be maintained.

In the case of a 3-way system, each of the three speakers has its own resonance frequency. Whilst the resonance frequency of the woofer is of no account in this discussion, the resonance frequencies of the mid-range and tweeter loudspeakers are important because of their effect upon the impedance characteristic. The impedance characteristic for a typical 3-way system is shown in Fig. 5.13. With the exception of the left-hand peak, the other peaks in the curve do not occur at the speaker resonance frequencies because of the action of the filter.



low-frequency section of 12 dB/octave parallel filter for 3-way system (Fig. 5.11)	loudspeaker impedance consists of voice coil inductance L_s and resistance R_s	series inductor L_{11} reduced by amount L_s to maintain constant resistance behaviour

Fig. 5.12 Using the voice coil inductance as part of the circuit design reduces the size of the filter component and maintains constant resistance performance.

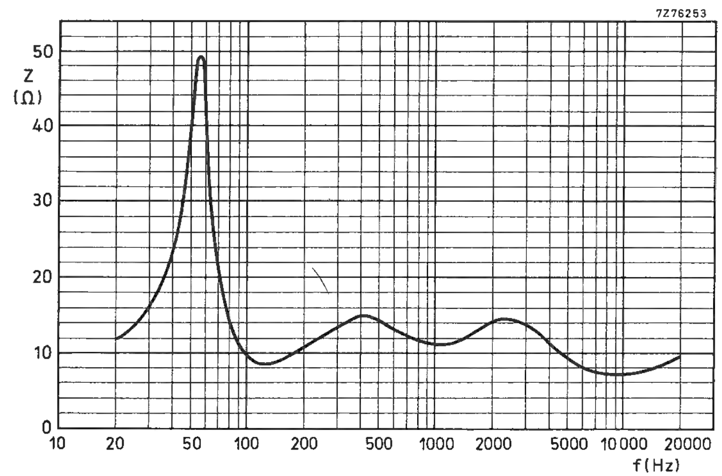


Fig. 5.13 Impedance characteristic of a typical 3-way system. Resonance frequencies of mid-range and tweeter loudspeakers are 210 Hz and 1000 Hz respectively; cross-over frequencies are 500 Hz and 4500 Hz.

5.7 Phase transfer response

It is essential that the phase transfer response through a cross-over filter is carefully considered. From first principles, we know that the relative phase of the backward radiation from the loudspeaker cone can cause cancellation of the forward radiation, and hence we must use some form of baffle. The cross-over network composed of reactive elements introduces phase changes into the system and unless due regard is given to the relative phase of the signal outputs, cancellation due to anti-phase conditions is very likely.

In the simplest case of single-element sections, as the frequency *increases*, the phase change between the input and the low frequency output approaches —90° (lag), and the high frequency output tends to become in phase with the input. As the frequency *decreases*, the phase of high frequency output approaches +90° (lead) relative to the input, whilst the low frequency output tends to become in phase with the input.

The phase transfer response of the outputs relative to the input for single element sections (6 dB/octave) is shown in Fig. 5.14.

In section 5.4 we described the properties of constant resistance networks. From Fig. 5.14 it will now be seen that the phase transfer response at the cross-over frequency is half the ultimate value, and also that the phase difference between complementary outputs is constant; in this case the phase transfer

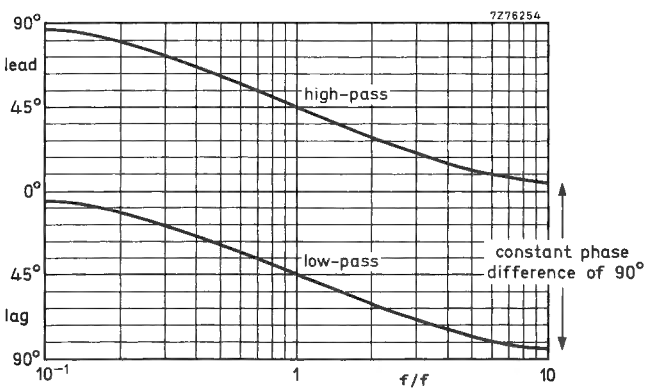


Fig. 5.14 Phase transfer response of outputs relative to input for single-element sections (6 dB/octave).

response at cross-over frequency is 45° and there is a constant 90° phase difference between the outputs.

In the case of 12 dB/octave filters employing two-element sections, the low-pass section introduces on ultimate phase change of —180° and the high-pass section a phase change of +180°. This is shown in Fig. 5.15 and it can be seen that a phase difference of 90° occurs between the input and the outputs at cross-over frequency, the outputs being a constant 180° out of phase throughout the frequency range.

Something must be done therefore to maintain a constant difference of 0° between the outputs throughout the audio frequency range, and with a two-section network having a 180° phase difference between its outputs, it is a simple matter to reverse the connections to one of the speakers as shown in Fig. 5.16. Electrically, the voice coils will be fed in anti-phase, but since one is reversed the cone motions are in phase.

Since the matter of correct phasing is of such importance, all our loudspeakers have one voice coil terminal indicated with a *red dot*. When a d.c. voltage is applied to the voice coil terminals such that the red connection is positive, the voice coil will move outwards.

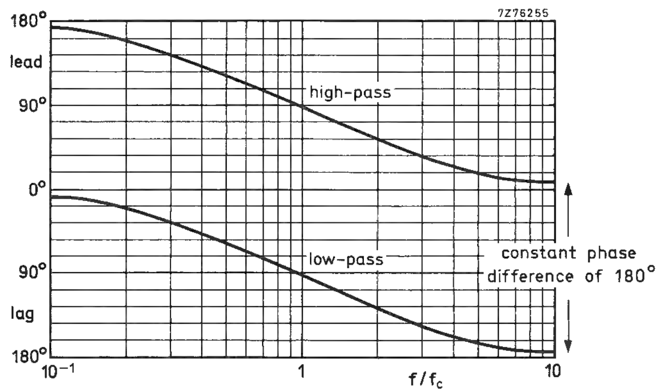


Fig. 5.15 Phase transfer response of outputs relative to input for two-element sections (12 dB/octave).

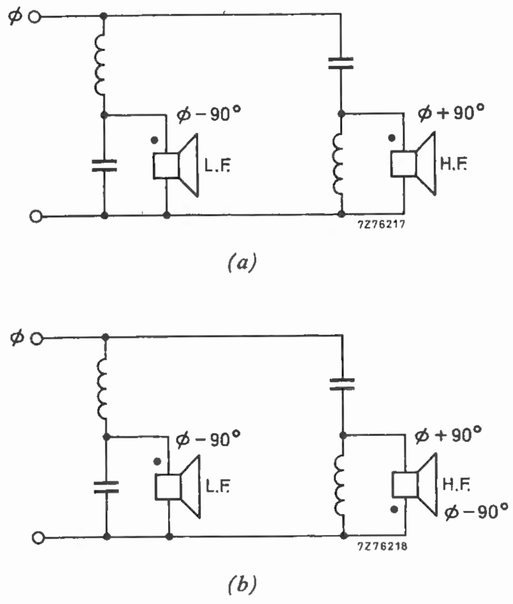


Fig. 5.16 Reversal of speaker connections to one of the speakers brings acoustic outputs in phase with 12 dB/octave network. (*a*) Speaker outputs 180° out of phase. (*b*) Speaker outputs in phase.

5.8 Choice of loudspeaker characteristics

The most important characteristics which determine the choice of loudspeakers for use in multi-way systems are

- equal sensitivity at the proposed cross-over frequency
- extended frequency response beyond the required roll-off point.

In order to maintain a smooth transition from one speaker to another, it is essential to select two speakers with equal sensitivities, or sensitivities which are within about 2 dB of each other, at the cross-over frequency. A greater difference in sensitivity will cause an audible step in the overall response, as shown in Fig. 5.17.

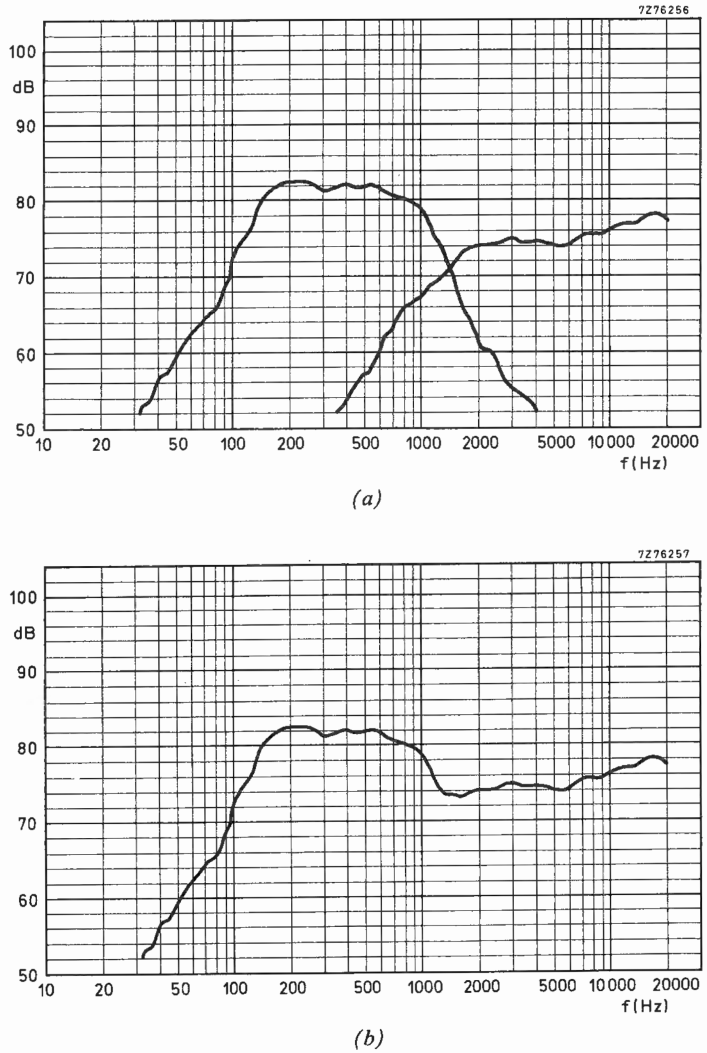
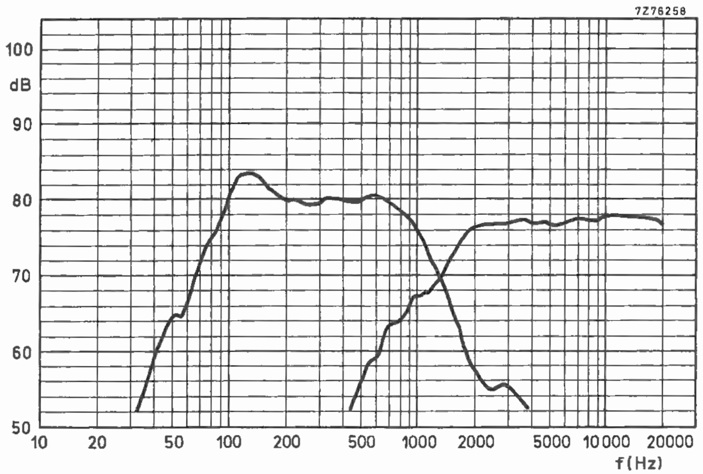
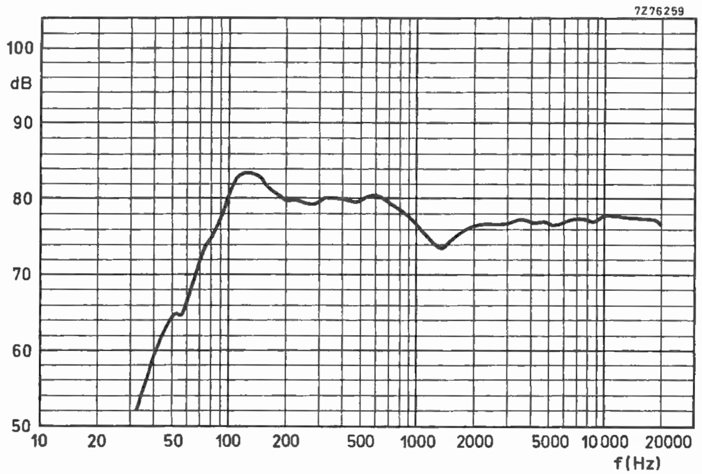


Fig. 5.17 (*a*) Frequency response of individual speakers having widely different sensitivities. (*b*) Combined response of two speakers having different sensitivities. Note the pronounced step in the curve.



(a)



(b)

Fig. 5.18 (*a*) Frequency response of two speakers having insufficient overlap in their characteristics. (*b*) Combined frequency response of speakers with insufficient overlap. Note dip in the curve at cross-over.

It is also necessary to ensure that one speaker is capable of taking over the signal from the other as soon as possible as the frequency is swept over the range. If two speakers have responses which do not overlap appreciably, a gap will be created in the overall response when they are connected through a filter which inherently introduces a further 3 dB attenuation at cross-over. This is shown in Fig. 5.18.

Although what has been described may be considered as common sense, it is surprising how unbalanced many commercial units are in terms of speaker sensitivities and also how many woofers have started their high frequency roll-off before there is any sign of response from the tweeters.

5.9 Asymmetric filters

The design of a cross-over network cannot be considered from a purely objective point of view for first class results. A properly engineered loudspeaker system is capable of very high quality reproduction and this is only arrived at by patient investigation of all the criteria affecting the sound output.

The variation of loudspeaker impedance with frequency is an important feature which clearly upsets the predictability of results using objective analysis only. At resonance, the impedance is high, falling sharply as the frequency is increased and then rising slowly again. The effect of the changing impedance on the filter transfer response, coupled with the frequency response of the loudspeaker, would lead one to believe that a smoother overall transition could be obtained by making the woofer roll of at 6 dB per octave and the tweeter roll on at 12 dB per octave. A filter that provides for roll-off and roll-on at different slopes is said to be *asymmetric*.

Ideally, a cross-over network should be designed in conjunction with the speakers and the enclosure to be used. After objective analysis of the problems involved and production of a tentative design, the speakers should be mounted in the proposed enclosure and the system tested with the cross-over network outside the enclosure. Changes in the design of the network can then be made, or the speaker connections reversed, between tests until best results are obtained.

5.10 Passive radiators

Before concluding this chapter, we should like to mention passive radiators. In effect, a passive radiator is a loudspeaker without a magnet system or speech coil. In all respects other that these, it resembles a normal loudspeaker. The passive radiator is normally the same diameter as the woofer and shares with it the enclosed volume of air of the enclosure, being mounted on the baffle board near to the woofer.

The effect of including a passive radiator is to extend the bass response of the loudspeaker system, and this effect can clearly be seen in Fig. 5.19 which gives the measured sound pressure with and without the passive radiator.

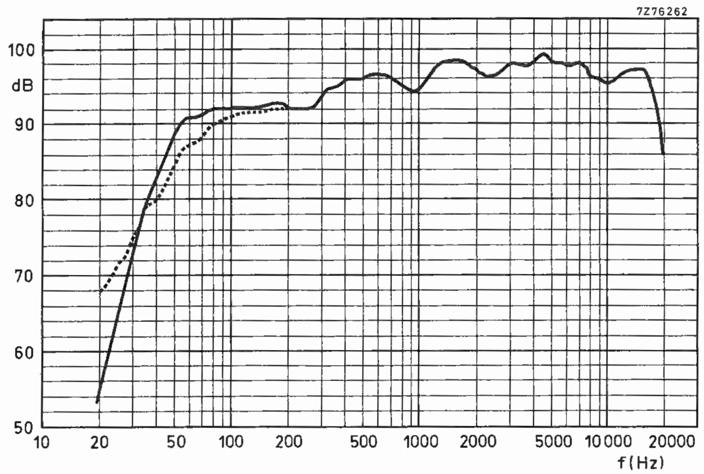


Fig. 5.19 The effect of using a passive radiator; dotted line shows measured sound pressure with woofer only, the combined result is shown by the full line.

6 Listening room acoustics

6.1 Absorption and reverberation

For most readers the opportunity to influence the *distribution* of the sound output from the speaker is very limited, but it is quite important to understand the factors which affect it, nevertheless. In a living room, the sound waves can travel only a short distance before they strike some object. They are reflected and some of the energy contained in them is dissipated by absorption so that the energy in the wave gets progressively less and less. The time taken for the sound energy to fall to one-millionth of its original value is known as the 'reverberation time'. It depends on the size of the room, but about half a second is considered satisfactory for listening to reproduced music; this applies to the middle frequencies.

Not only do the various materials of the floor, walls and ceilings absorb energy, but so also do the normal furnishings of the room and the listeners themselves. Modern furnishings tend to utilize materials which do not absorb sound readily, particularly glass, which is one of the worst materials acoustically. Large windows are, nevertheless, very desirable.

Only limited improvements can be made to the living room acoustics. Thick carpet with foam rubber underlay is probably the best for floor covering and solid floors are better than wood boards supported on joists. Where greater absorption in the high and middle frequency ranges is desirable, acoustic tiles can be fixed in strategic positions on ceilings and walls. The advice of an expert on acoustics should be sought before this is undertaken. Any experiments with listening room acoustics should be done with the tone controls of the equipment set at 'flat'. When no further improvement in the sound quality can be made by modifying the room acoustics, the effect of the tone controls can then be tried.

6.2 Room resonances

Where any dimension of the room is one-half the wavelength of the sound, resonant vibrations will be excited in the air. At a frequency of 40 Hz, the wavelength of the sound is 8,5 metres, so the length of the room which will resonate at that frequency will be half that, that is, 4,25 metres (14 feet). In addition there are harmonics of this frequency; the second harmonic at 80 Hz,

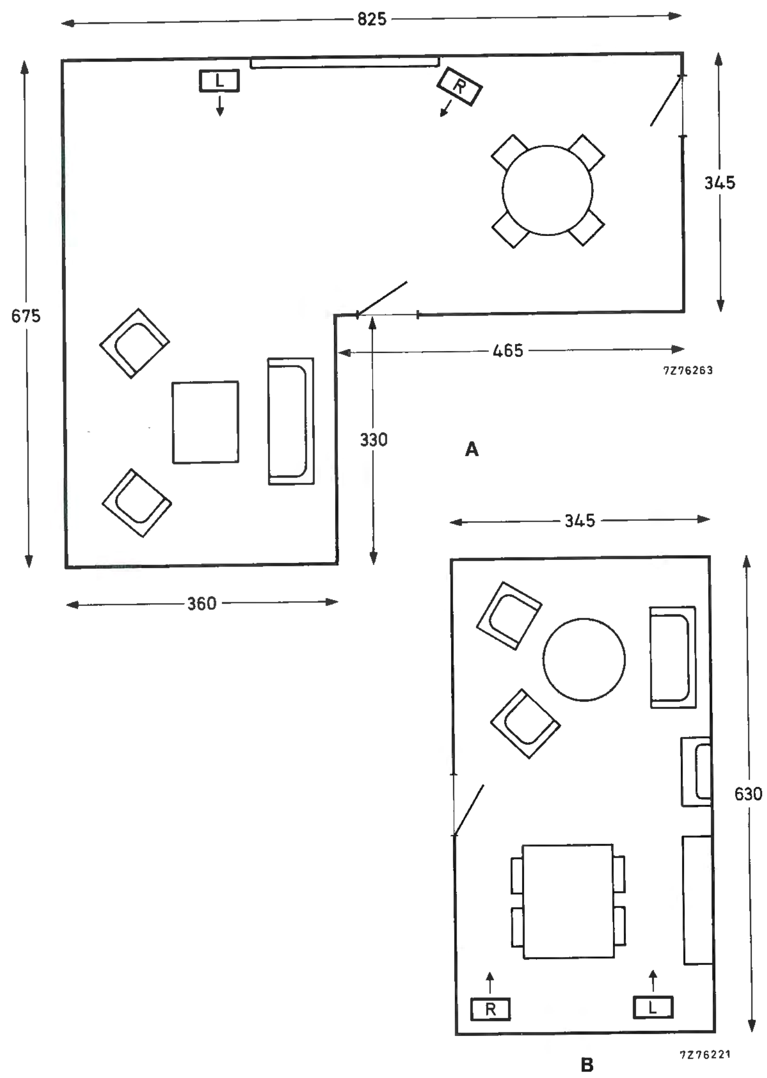


Fig. 6.1 Two of the living rooms in which the tests were made.

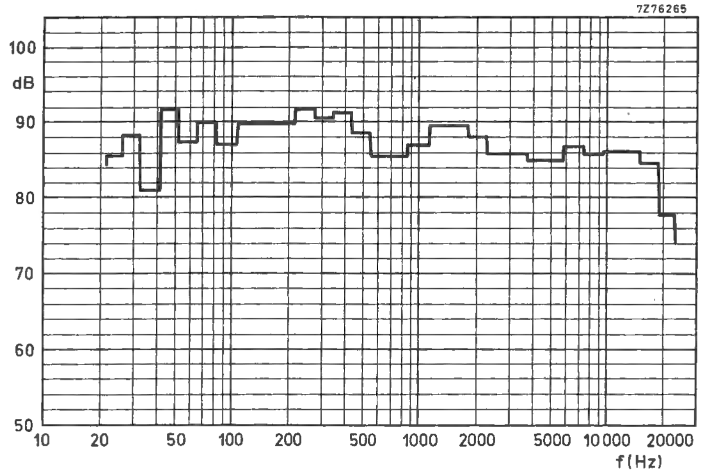
and the third harmonic at 120 Hz. Not only does this apply to the length of the room, but resonances will occur at many frequencies due to the height and width, and diagonal dimensions as well. Where one of the dimensions of the room is nearly an exact multiple of the other, the fundamental resonance frequency of one dimension of the room will modulate a harmonic of the resonance frequency of the other dimension and the whole picture becomes very complex. Standing waves occur which create increases in sound pressure at certain regions in the room.

Obviously little can be done, without calling the builders to change the shape of the room, to alleviate the resonances but, whatever the proportions of the room, any broken surfaces or projections will give some improvement. Advantage can often be taken of the use of 'room-dividers' and free-standing bookshelves.

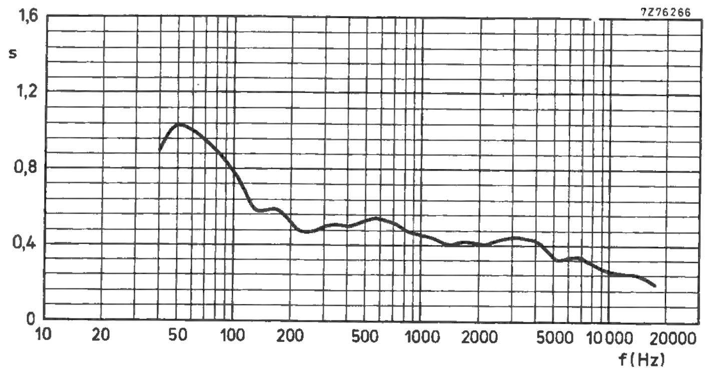
To avoid room resonances, moving the chair may take the listener out of a region of pressure maximum and overcome a certain 'boominess' which might be blamed on the speaker. Similarly, moving a speaker may excite different room resonances and the result may be preferable to the original positioning.

In order to determine the behaviour of loudspeakers in practice, a large number of measurements in living rooms have been carried out making use of highly sophisticated equipment. The most important conclusion to emerge from these investigations is that the acoustic properties of all living rooms are very similar above about 600 Hz. Below this frequency, the dimensions of the room and the furniture it contains play a part.

The loudspeaker boxes were placed where the sound system would normally be situated in the rooms shown in Fig. 6.1. Using a microphone amplifier and a real-time analyser with a storage oscilloscope and an *X-Y* recorder, the average sound pressure level was measured over each one-third octave band at different places in the listening area and the reverberation time was also measured. The results are shown in Figs 6.2 and 6.3. For comparison, the sound pressure levels and reverberation times measured in six other rooms are given in Fig. 6.4.

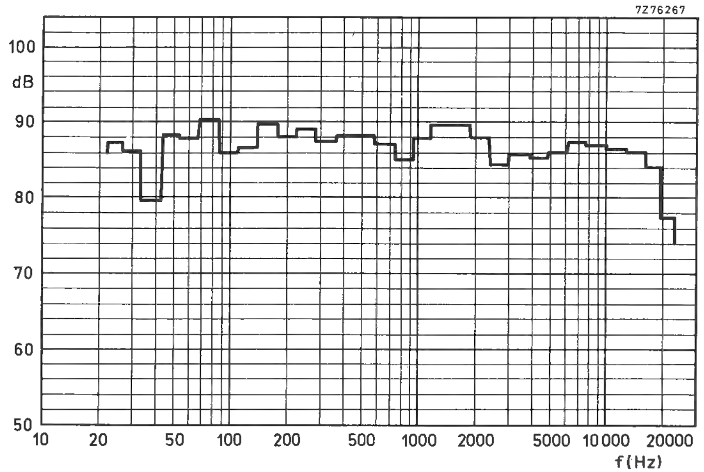


(a)

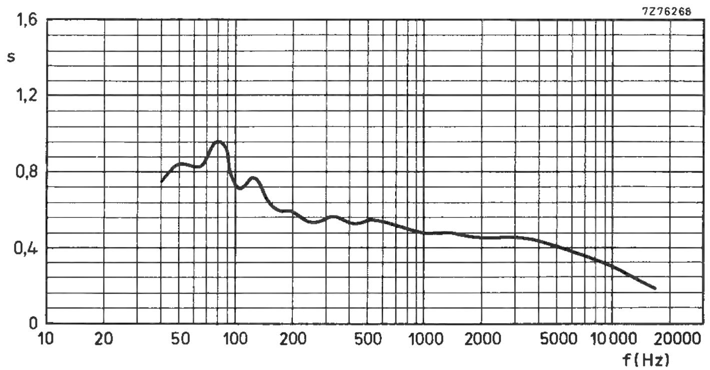


(b)

Fig. 6.2 Living room A: (*a*) the sound pressure levels, (*b*) the reverberation time.



(a)



(b)

Fig. 6.3 Living room B: (*a*) the sound pressure levels, (*b*) the reverberation time.

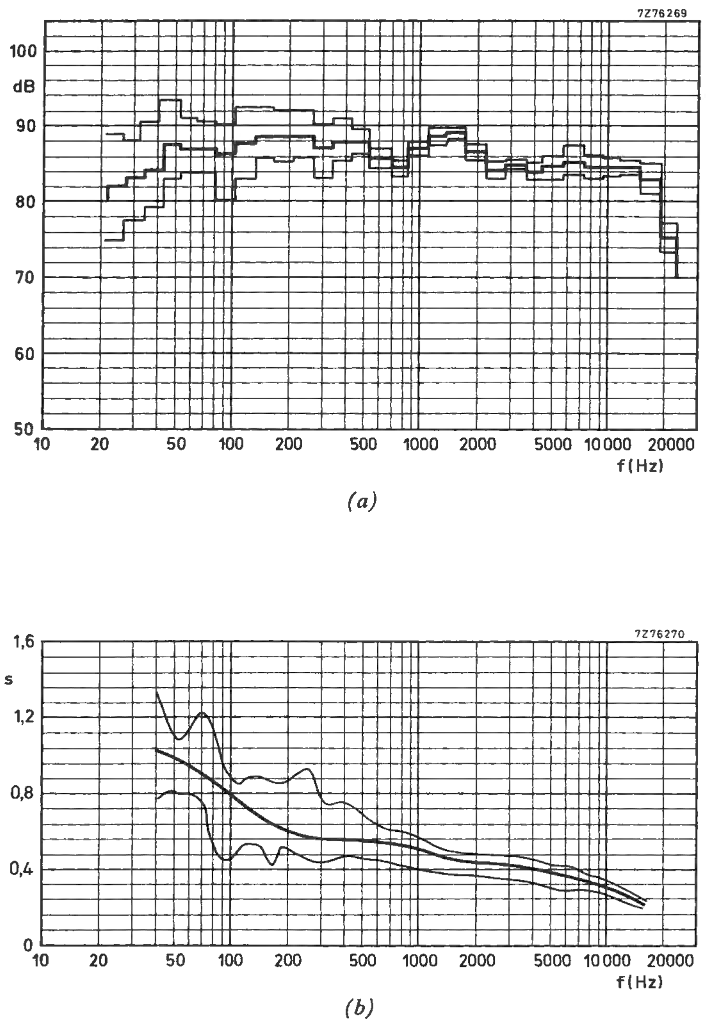


Fig. 6.4 (*a*) Extreme and average sound pressure levels for six different living rooms. The results are very similar above about 600 Hz. (*b*) Extreme and average reverberation times of six living rooms.

6.3 Sound level in the living room

Proper adjustment of the gain control of the amplifier is essential if the reproduction is to sound natural. This rule is only too often violated in practice, mostly by turning the gain control too high, which inevitably results in unnatural timbre owing to undue bass emphasis. Moreover, intelligibility is worsened, not enhanced, because the treble tones are masked by the bass. This sometimes gives rise to unjustified complaints about loudspeaker systems which have really nothing wrong with them but have merely been abused.

Outside the laboratory, it is customary to measure the sound pressure level through 'weighted' instruments. These simply filter the sound received by the measuring instrument to simulate the natural characteristics of the human ear. There are various weighting scales, but the one most commonly used is known as the A scale; most measurements of sound pressure are in dBA units and sound level meters give a direct dBA read-off in which 0 dB still corresponds to the threshold level of 2×10^{-5} N/m^2.

Customary reference points are a just audible sound (0 dBA), a quiet bedroom in the middle of the night (35 dBA), a busy shop (60 dBA), a noisy workshop or factory (85 or 90 dBA), an underground train entering a station (95 dBA), and a jet plane at 200 m (120 dBA). More relevant in the present context are the following:

– A television receiver or radio at normal listening level will be between 60 and 65 dBA. If it is loud, it will probably reach 70 dBA. A live performance of Beethoven's 9th can produce an occasional 105 dBA in the front row.

– Normal cinema levels are in the dBA 70's.

– Pop groups regularly produce 110 dBA and can often reach peaks of 120 and 130 dBA. Monitoring in pop recording studios may be at 120 dBA and levels of 128 dBA are not unknown. The sound pressure at the mouth of some reproducer horns used by pop groups is 140 dBA, which is physically dangerous to anyone close to them.

In some countries there is a move to limit the sound level in discos to prevent young people suffering permanent damage to their hearing. Whatever the outcome, regardless of any legislation that is passed, a cheap hi-fi set can easily produce levels of 100 dBA at the ears of a listener if headphones are used, and the pop fan will still be able to deafen himself without disturbing anyone else in the room.

6.4 Positioning of loudspeakers

It is extremely difficult, if not impossible, to accurately relate the response curve for a loudspeaker system to how such a system will sound in a living room, especially as most response curves are produced, for reasons of standardization, in an anechoic room.

The positioning of a loudspeaker enclosure in the room has considerable effect on the low frequency response. As we have shown, above about 600 Hz, most normal living rooms are acoustically very similar. There are a large number of practical possibilities for the listener in mounting his loudspeaker boxes. Some of these are described here and the results of measurements taken using pink noise, with a real-time analyser, can be compared.

The room in which the measurements were made is shown in Fig. 6.5. The single loudspeaker box used for the tests was first placed in an anechoic room and a response curve was taken for reference; this is shown in Fig. 6.6. The loudspeaker was then taken into the living room and mounted in a free-standing position 60 cm above the floor, 80 cm from the rear wall, and 2 m from the side wall and the response curve test repeated. The effect of the living room can clearly be seen in Fig. 6.7.

The speaker was then placed on the floor and the distances from the two adjacent walls was varied for a series of tests. In Fig. 6.8 the results (solid line curves) are compared with the free-standing result (dotted line). Finally, the speaker was mounted 60 cm above the floor and a further series of tests was made; these results are given in Fig. 6.9.

The differences in response compared with the free-standing position are shown in Fig. 6.10. The conclusions reached for this series of tests were that when the loudspeaker box stood on the floor, the sound pressure between 150 and 400 Hz improved by 3 to 9 dB, and between 500 and 1000 Hz by 2 dB. With the loudspeaker against the wall, for frequencies from 40 to 100 Hz there was an improvement from 4 to 5 dB, whereas between 120 and 300 Hz the sound pressure was 1 dB lower. When the box was stood in the corner, for frequencies between 30 and 100 Hz the pressure was increased from 5 to 9 dB, and between 120 and 400 Hz was reduced by 2 dB.

If the reader will now refer to Fig. 6.6 (page 110), which was recorded in an anechoic room, he will realize how little idea such a response curve gives of what the final result will be like when the loudspeaker box is installed in the living room.

One final golden rule for positioning speakers is *do not obscure the baffle*. It is useless to construct a high quality enclosure and then stand it on the floor behind a chair and expect it to perform satisfactorily.

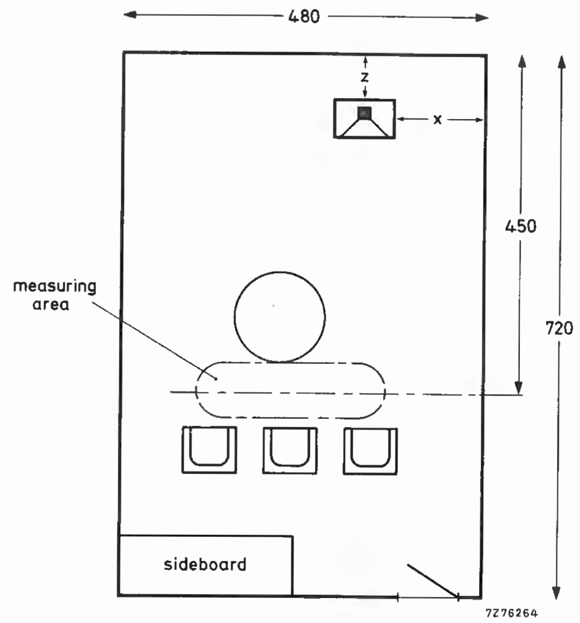


Fig. 6.5 Living room in which tests were made on speaker positioning.

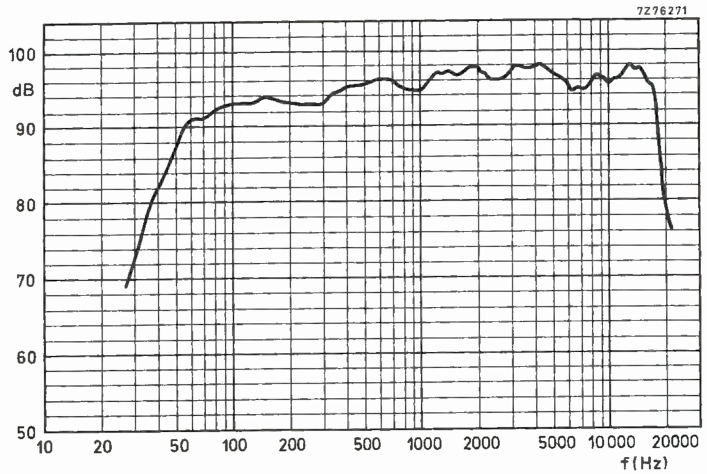


Fig. 6.6 Response curve of loudspeaker enclosure measured in the anechoic room.

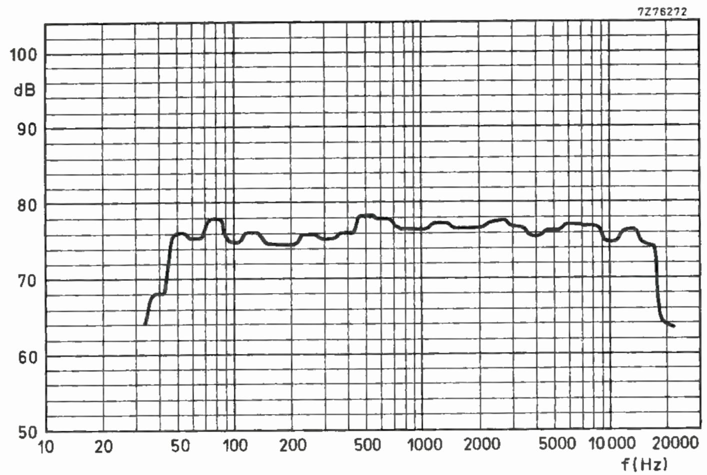


Fig. 6.7 Response curve of loudspeaker enclosure in living room when mounted in a free-standing position 60 cm above the floor, 80 cm from the rear wall and 2 m from the side wall.

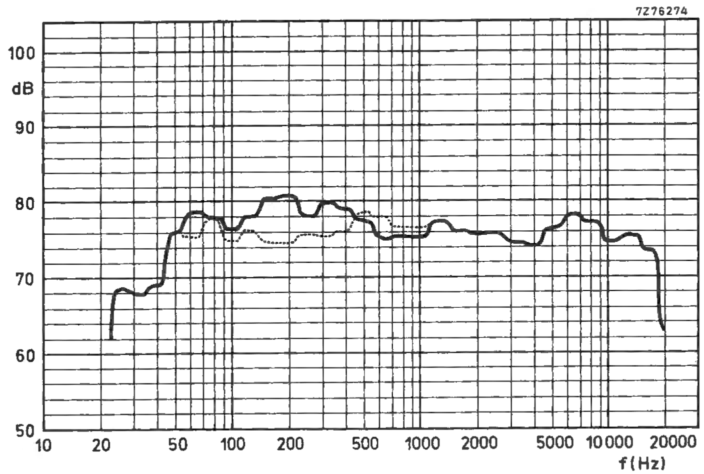


Fig. 6.8 (*a*) Response with enclosure on the floor, 80 cm from rear wall and 2 m from side wall.

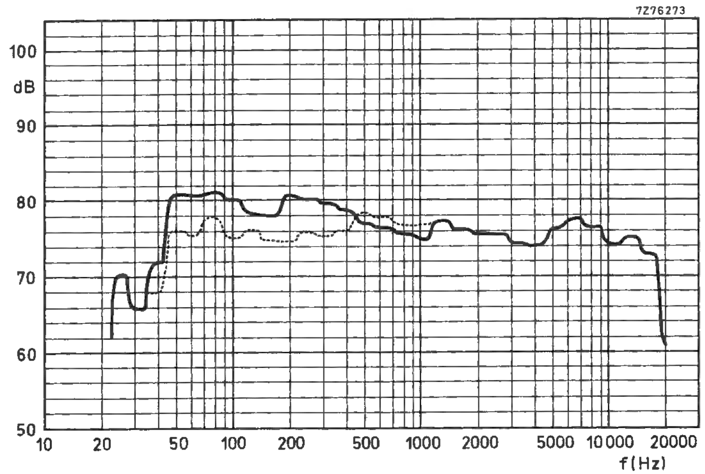


Fig. 6.8 (*b*) Response with enclosure on the floor, 30 cm from rear wall and 2 m from side wall.

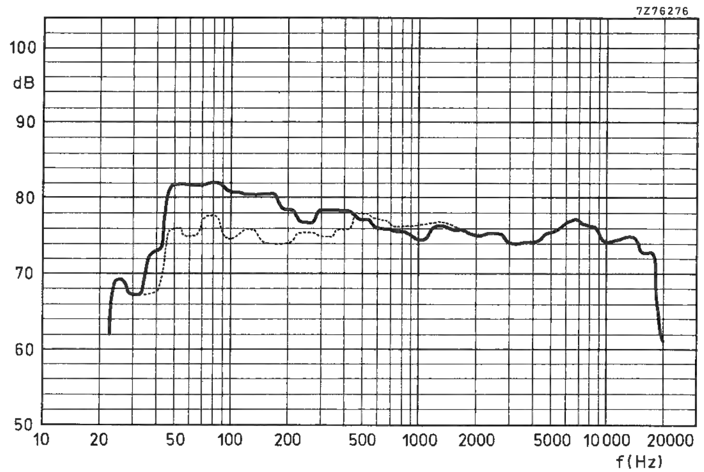


Fig. 6.8 (*c*) Response with enclosure on the floor against the rear wall and 2 m from the side wall.

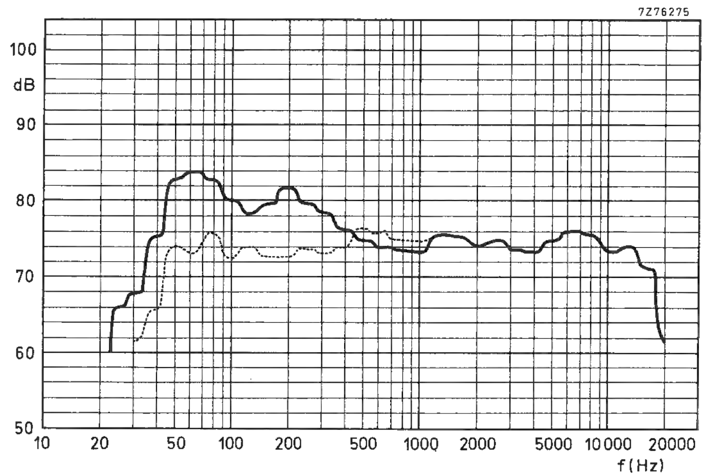


Fig. 6.8 (*d*) Response with the enclosure in the corner.

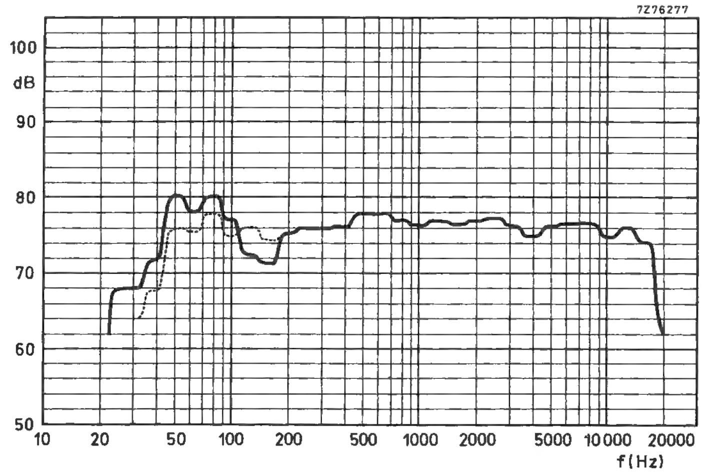


Fig. 6.9 (*a*) Response with the enclosure 60 cm off the floor, 30 cm from the rear wall and 2 m from the side wall.

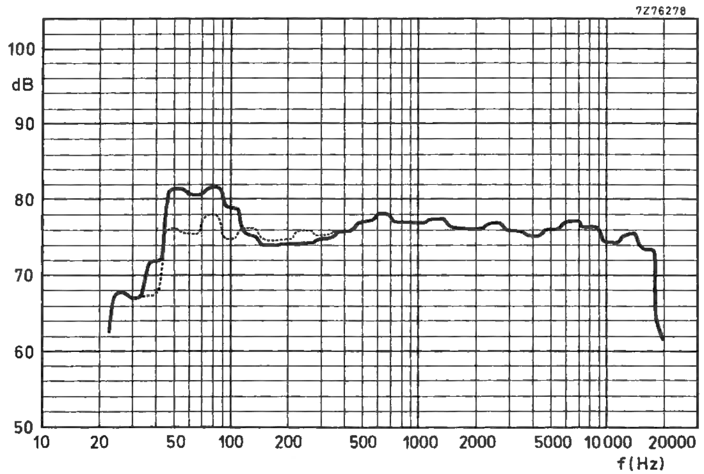


Fig. 6.9 (*b*) Response with the enclosure 60 cm off the floor, back to the rear wall and 2 m from the side wall.

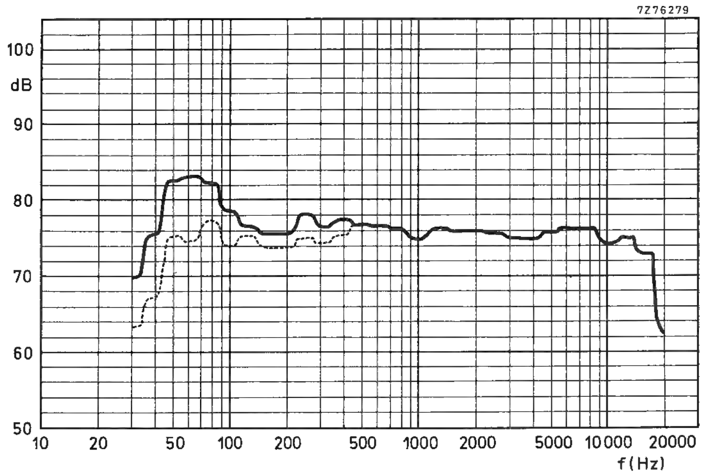
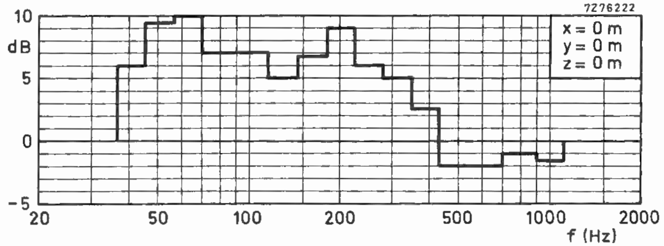
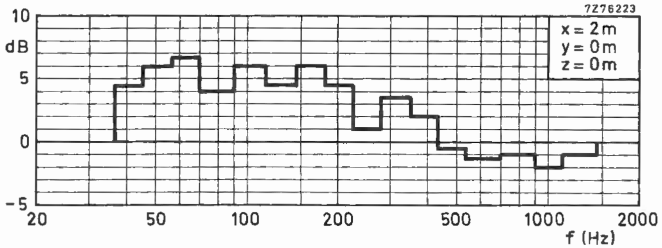


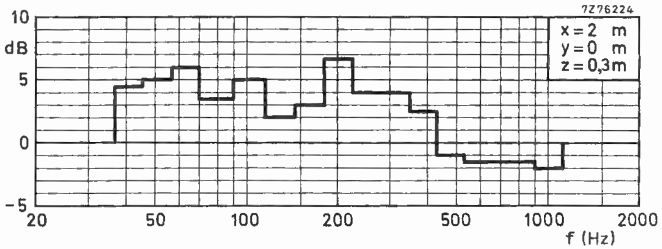
Fig. 6.9 (*c*) Response with the enclosure 60 cm off the floor in the corner.



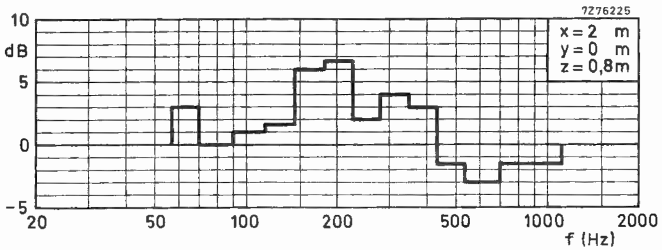
(a)



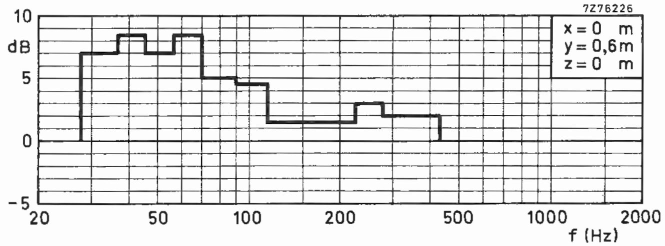
(b)



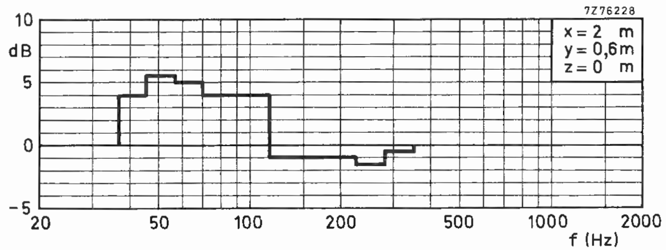
(c)



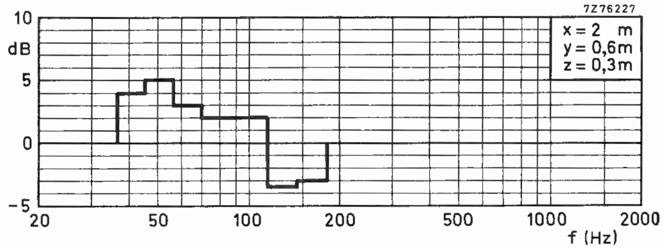
(d)



(e)



(f)



(g)

Fig. 6.10 Difference in the responses from the free-standing position of Fig. 6.7. (x = distance from side wall, y = height off floor, z = distance from rear wall.) Note the increased bass due to corner mounting on the floor.

6.5 Multi-channel systems

A listener equidistant from two loudspeakers, some distance apart but reproducing the same sound simultaneously and with equal intensity, hears them as one source, not two. The apparent source of 'virtual sound image' is then midway between the two loudspeakers. But if one speaker reproduces the sound sooner or louder than the other, the image will appear closer to it. A time lead of 3 ms or a preponderance of 15 dB will bring the image exactly to the dominant speaker; relatively minor differences in time and intensity act individually or cumulatively to produce correspondingly smaller displacements. However, shifts due to time delay can be compensated by inducing corresponding differences in intensity.

Our auditory facility for merging two sounds into a single acoustic image, whose position within certain limits depends on the differences in intensity and time of arrival between these sounds, affords the basis of stereophonic sound reproduction.

When the listener is further from one loudspeaker than the other, the difference in distance causes an extra difference in time of arrival, thereby shifting the stereophonic sound image to the nearer speaker. This shift can be kept within reasonable bounds by suitably positioning the speakers in the living room.

When the sound from two loudspeakers arrives with a time difference of more than 3 ms, all of it appears to come from whichever source is heard first; the acoustic image cannot be moved even by intensifying the sound from the other loudspeaker, and the stereophonic effect of the reproduction is lost. With a delay of about 20 ms, the lagging speaker can be stronger by up to 10 dB before the listener becomes aware that it is producing any sound at all. Nevertheless, it does affect the quality of the sound by making this not only louder, but also 'fuller' in tone, despite the delay. When the time difference exceeds 50 ms, the trailing sound, unless it is the weaker of the two by 10 dB or more, is heard separately as an irritating echo. This only applies to halls with notably bad acoustics.

To recapture the sound of the concert hall in the home, it is necessary to understand how the sound behaves in the concert hall, first. Most of the audience do in fact receive more sound by reflection than directly from the stage. Because indirect sound is diffuse, its intensity is almost constant throughout the auditorium, whereas that of the direct sound is inversely proportional to the square of the distance from the source. The relative intensities of these sounds give the

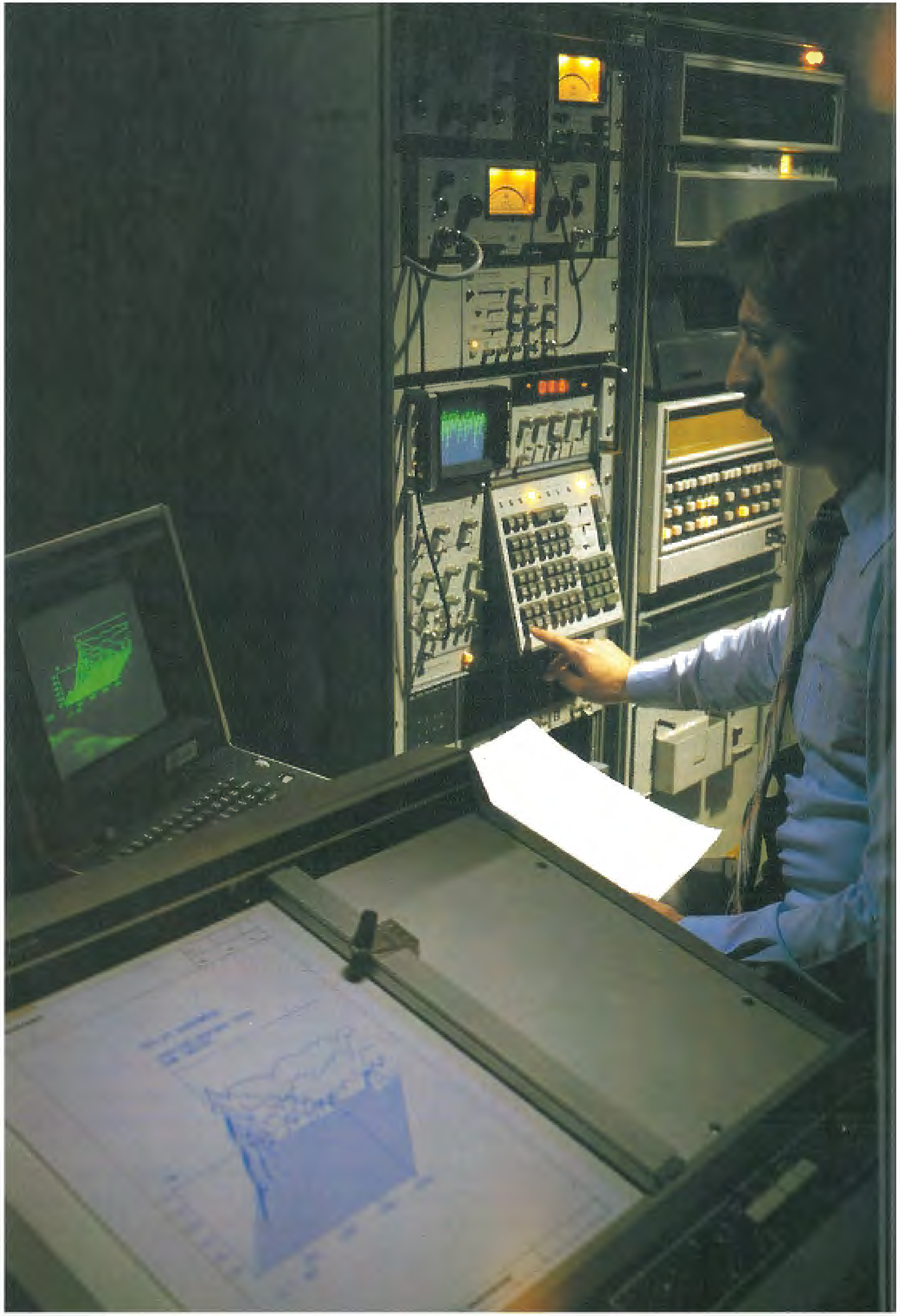
listener a clue to his position in the concert hall. Speech needs the extra loudness supplied by indirect sound which improves the intelligibility if it arrives within about 50 ms of the direct sound.

We can summarize the aspects of auditorium acoustics upon which our own room acoustics and the sound installation should be based:

- The fundamental difference between direct and indirect sound. Direct sound supplies the stereophonic sensation of whatever is happening on the stage or concert platform; it also lends 'distinction' to the music. The timbre, diffuseness and reverberation of indirect sound create a background impression of the auditorium, give the sound a more spacious quality and enrich the tone of the music.
- The indirect sound delay imparts a sense of the size of the auditorium.
- The relative intensities of the direct and indirect sound govern the sensation of distance.

From the foregoing the reader will appreciate that there is much to be gained in reproducing 'indirect' sound at the rear of the listening area in the living room. Just a single extra loudspeaker box will give an enormous improvement in the realism. It can be connected as shown in Fig. 2.14(*b*). Obviously, the level at which the rear sound is reproduced will have to be found by the listener to suit his own listening room and the effect he wants to create.

Quadrophonic systems which take into account the various aspects of sound delay to enhance the realism of the reproduction in the living room can produce some very excellent results indeed.



← *Outside the anechoic room, instruments record the response of a loudspeaker under test.*

7 Constructional details of 19 tested speaker systems

This Chapter gives the constructional details of 19 selected speaker systems that have been fully tested in our laboratories. A wide choice is offered, ranging from a small and simple system using a single full-range loudspeaker and capable of handling a maximum of 10 W input to a 100 litre enclosure with a power handling capacity of 100 W.

In all cases, the frequency range of the recommended systems has an upper limit of 20 000 Hz; the bass response depends on the woofer used and the enclosure volume, as described earlier. In choosing a system, bear in mind that just because a particular system is listed as having, say, 40 W power handling capacity, it does not mean that you *must* have 40 W available at your amplifier.

Table 7.1 Summary of the loudspeaker systems described in this Chapter.

system number	enclosure volume (litres)	power handling capacity (watts)	frequency range (Hertz)	remarks
1	3	10	90-20000	full range speaker
2	12	15	50-20000	full range speaker
3	9	20	45-20000	2-way sealed
4	40	25	40-20000	simple bass-reflex
5	7	15	40-20000	2-way bass-reflex
6	13	35	45-20000	2-way bass-reflex
7	13	40	50-20000	2-way sealed
8	23	50	37-20000	2-way sealed
9	25	50	37-20000	2-way sealed
10	17	40	45-20000	2-way bass-reflex
11	45	50	32-20000	2-way sealed
12	25	50	42-20000	3-way bass-reflex
13	32	50	32-20000	3-way sealed
14	25	35	35-20000	3-way sealed
15	60	60	26-20000	3-way sealed
16	50	80	27-20000	3-way sealed
17	60	80	23-20000	3-way sealed
18	80	100	23-20000	3-way sealed
19	100	100	20-20000	3-way sealed

The operating power stated gives you the number of watts you will need to achieve a sound pressure level of 96 dB (which is not exactly quiet), and anywhere in between the operating power and the power handling capacity will increase your dynamic range. If you have an amplifier capable of, say, 40 W output, then choose a system capable of handling it. You will then get the full value out of your complete system. Exceeding the power handling capacity of your speaker system will result in distortion and, if the additional power is much too great, it will obviously damage your speakers.

7.1 Choice of impedance

All the systems described in this Chapter are intended for use with both 4 Ω and 8 Ω amplifiers except for System 4 which is only for 8 Ω, and System 18 which is only for 4 Ω. The loudspeaker type numbers given on the pages which follow apply to 4 Ω systems; a few simple changes are all that is required to adapt the systems for 8 Ω. These changes in no way effect the performance of the described systems, and the published curves (with the exception of the impedance curves, which should have their vertical scales multiplied by two) remain valid in all cases.

The changes amount to nothing more than changing the loudspeaker impedances and altering the values of the components in the cross-over networks. The last digit of a loudspeaker type number indicates its impedance in ohms, thus: AD12650/W4 is the 4 Ω version of the woofer AD12650/W; the 8 Ω version has the type number AD 12650/W8. Similarly, the AD01630/T8 is the 8 Ω version of the tweeter AD01630/T; AD01630/T15 is the 15 Ω version.

To adapt the described systems for 8 Ω operation, the following changes should be made:

- 4 Ω loudspeakers should be replaced by 8 Ω versions:
- 8 Ω loudspeakers should be replaced by 15 Ω versions, except in the case of System 18 where there is no 15 Ω version of the woofer; only a 4 Ω version of System 18 is therefore possible. This also applies to System 4 which is for 8 Ω only:
- filter capacitances and inductances should be as given for 8 Ω systems in Table 7.2

Table 7.2 Cross-over network component values

4 Ω systems

system	L_1 (mH)	L_2 (mH)	L_3 (mH)	L_4 (mH)	C_1 (μF)	C_2 (μF)	C_3 (μF)	C_4 (μF)	R_1 (Ω)	R_2 (Ω)	R_3 (Ω)
3	0,5	0,2			8						
5					4,7						
6					4,7						
7	0,5	0,2			6,8						
8	0,8	0,2			4,7				2,2		
9	0,8	0,6			6,8				1,8		
10					6,8						
11	0,8	0,6			6,8				2,7		
12					12	6,8					
13	1,0	0,6	0,1		47	22	6,8		1,8	1,2	1,8
14	0,35				12	4,7					
15	0,6	0,35			10	3,3			2,2	1,8	
16	1,0	0,6	0,2		66	20	4,7		10	1,2	2,2
17	1,5	0,1	1,0	0,2	47	22	3,3		3,3	2,2	
18	4,0	0,8	0,35		15	3,3			6,8		
19	1,0	1,0	0,2	0,2	100	33	4,7	4,7	3,3	0,68	1,8

8 Ω systems

system	L_1 (mH)	L_2 (mH)	L_3 (mH)	L_4 (mH)	C_1 (μF)	C_2 (μF)	C_3 (μF)	C_4 (μF)	R_1 (Ω)	R_2 (Ω)	R_3 (Ω)
3	0,8	0,5			3,3						
5					2,2						
6					2,2						
7	0,8	0,5			3,3						
8	1,5	0,5			3,3				4,7		
9	1,5	1,2			3,3				3,9		
10					3,3						
11	1,5	1,2			3,3				5,6		
12					6,8	3,3					
13	2,1	1,2	0,2		22	10	3,3		3,9	2,2	3,9
14	0,8				6,8	2,2					
15	1,2	0,8			4,7	1,5			4,7	3,3	
16	2,1	1,2	0,35		33	10	2,2		18	2,7	4,7
17	3,0	0,2	2,1	0,5	24	12	1,5		6,8	3,9	
18	2,1	0,35	0,2		6,8	1,5			3,3		
19	2,1	2,1	0,35	0,35	47	15	2,2	2,2	6,8	1,2	3,3

7.2 Sizes of baffle holes

In order to keep the baffle board drawings as simple as possible, only the centre lines of the holes are dimensioned. Thus, where you see a hole marked, say, B, this refers to the B diameter given in Table 7.3 (i.e. 142,5 mm). In the case of the tweeter AD0162/T, a drawing is given of the hole because it is necessary to make two small cut-outs on opposite sides of the hole to allow for the connections.

7.3 Methods of measurement

Frequency response curves

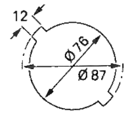
Two types of frequency response curves are given for each system. First, the frequency response and harmonic distortion have been measured in an anechoic room. This gives us a clear picture of the unmodified performance of the speaker system without any effects of the living room surroundings. Input to the system in this case is the operating power which produces a sound pressure level of 96 dB. The test microphone is mounted a distance of 1 m away in front of the enclosure.

Second, the frequency response has also been measured in a typical living room. Pink noise has been used for the tests at the operating power level of the system (see data for each system) and the measurements were made using a real-time analyser with an integration time of 16 seconds. The living room in which the tests were made was 4,8 m wide by 7,1 m long by 2,84 m high (volume 97 m^3).

Energy response

Here again pink noise was used for the tests at the system operating power level and a real-time analyser was used to take the measurements. Integration time was 16 seconds. However, the energy response was taken in a reverberant room, with the loudspeaker standing in a corner. The volume of the reverberant room was 192,5 m^3.

Table 7.3 Baffle hole sizes for loudspeakers used in the recommended systems.

system number	loudspeaker	hole size	diameter (mm)
1 12	AD5061/M AD5062/Sq	A	107,5
2 3 6	AD7063/M AD70601/W AD70610/W	B	142,5
3, 5, 7	AD2296/T	C	43,5
4	9710/M	D	193
5	AD4060/W	E	94,5
6	AD01430/T	F	87
11 7, 8 10, 12 9, 11, 13	AD8001 AD80602/W AD80603/W AD80652/W	G	180,5
10, 12 18, 19 8 9, 13, 16, 17	AD01420/T AD01605/T AD01610/T AD01630/T	H	85,5
11	AD0162/T	J	
13	AD0211/Sq	K	113,5
14	AD10650/W	L	230
14, 15	AD5061/Sq	M	96,5
14 15	AD0140/T AD0141/T	N	75
19 17, 18, 19 15, 16	AD1200 AD12200/W AD12650/W	P	280
16, 17 18, 19	AD02110/Sq AD02160/Sq	Q	122

7.4 System details

Just a few points before you embark on building your system:

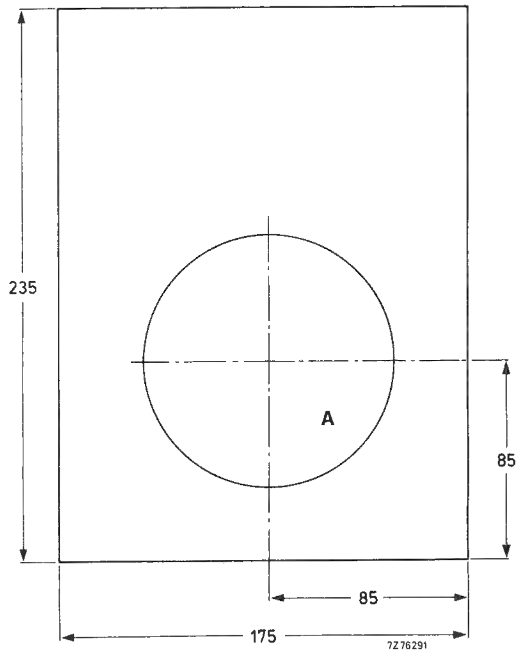
- The black dot on the filter circuit diagrams corresponds to the red dot on one of the loudspeaker terminals (in order to get the phasing correct).
- The published results for each enclosure were obtained with the damping material fixed only to the back wall of the enclosure unless otherwise stated. But the best match to your own listening surroundings may be obtained with additional damping material on the top, bottom, or side walls, so don't hesitate to experiment.
- If you are building a sealed enclosure system, make sure that it's airtight; particularly around the cable entry at the rear. Use polyether foam draught-excluder self-adhesive tape around the loudspeakers when you mount them. And make sure that the joints of the enclosure don't leak air — use plenty of glue.

System 1

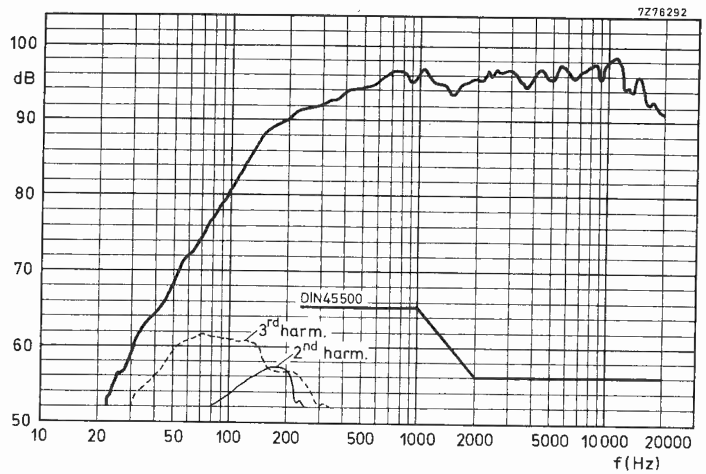
90 - 20 000 Hz

Full range speaker AD5061/M4
M8

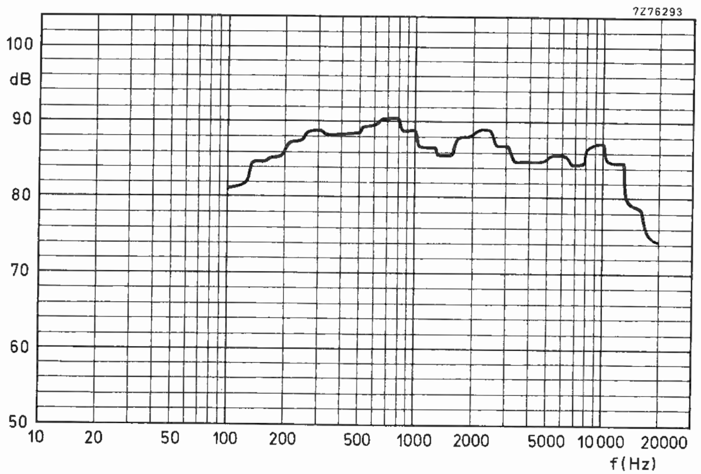
Power handling capacity	= 10 W
Rated impedance	= 4 Ω / 8 Ω
Operating power	= 3 W
Resonance frequency	= 175 Hz
Frequency range	= 90-20 000 Hz
Enclosure volume	= 3 litres
Internal dimensions	= 240 × 180 × 65 mm
Internal depth of enclosure	= 65 mm
Material thickness	= 10 mm
Damping material	= glass wool 180 × 240 × 30 mm



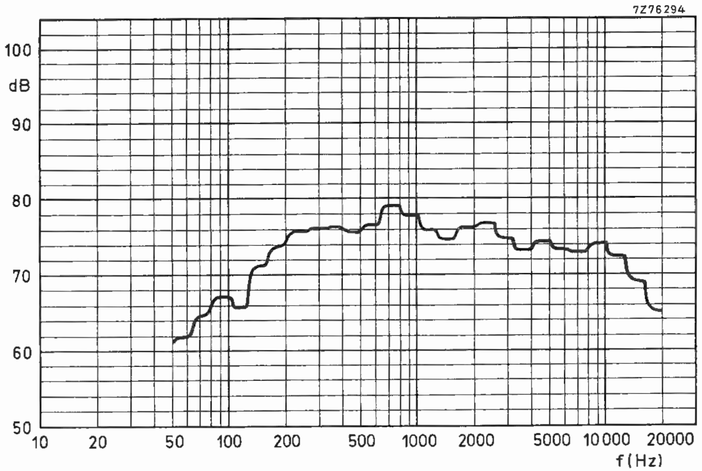
Baffle board layout
(see Table 7.3 for hole size)



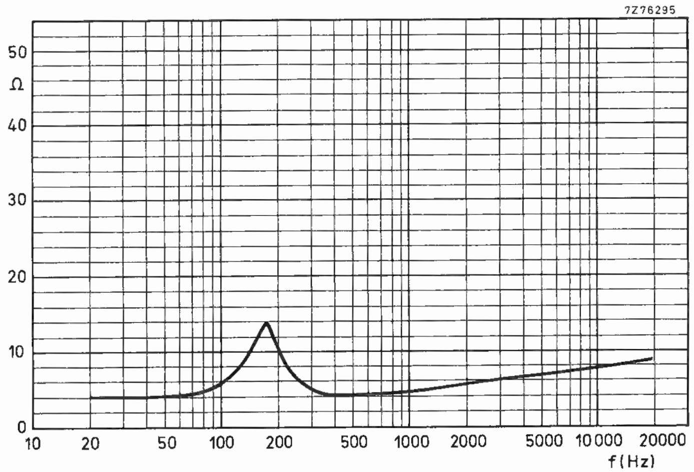
Frequency response and distortion measured in anechoic room.



Energy response in reverberant room.



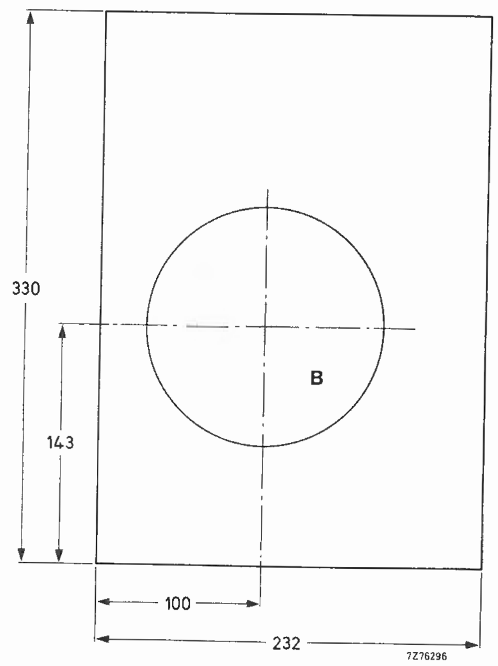
Frequency response measured in living room.



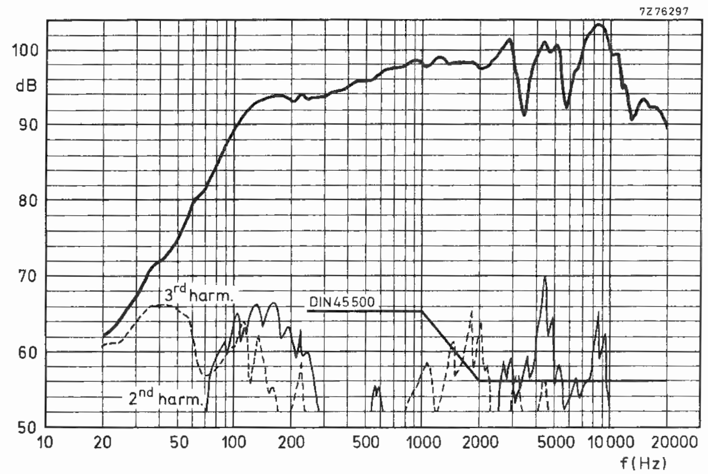
Impedance curve.

Full range speaker AD7063/M4
M8

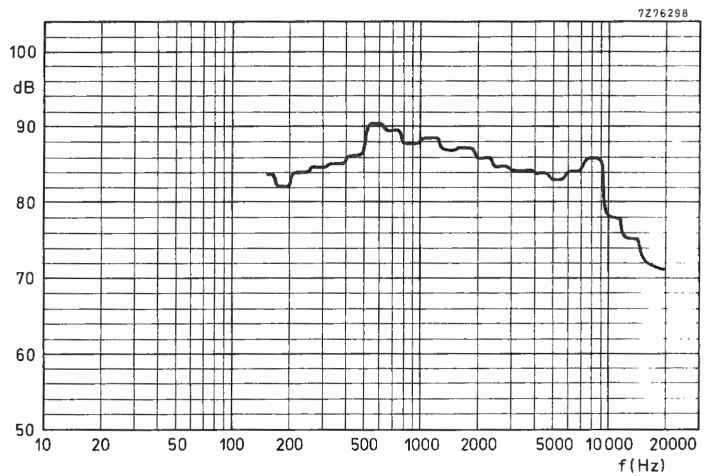
Power handling capacity	=	15 W
Rated impedance	=	4 Ω / 8 Ω
Operating power	=	4 W
Resonance frequency	=	105 Hz
Frequency range	=	50-20 000 Hz
Enclosure volume	=	12 litres
Internal dimensions	=	335 × 237 × 150 mm
Internal depth of enclosure	=	150 mm
Material thickness	=	16 mm
Damping material	=	polyurethane foam 325 × 235 × 16 mm



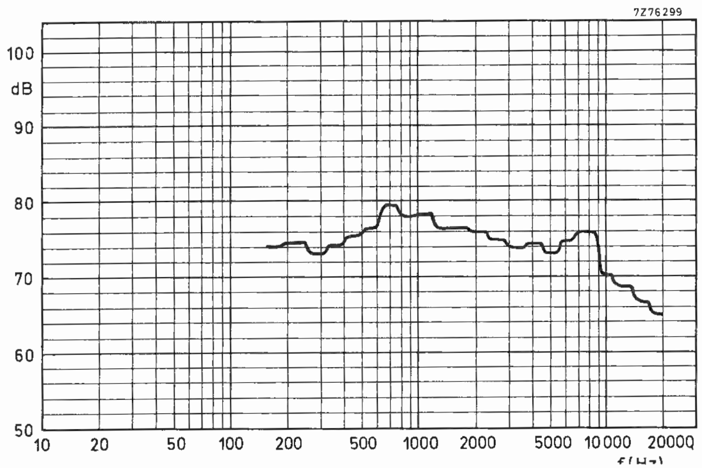
Baffle board layout
(see Table 7.3 for hole size)



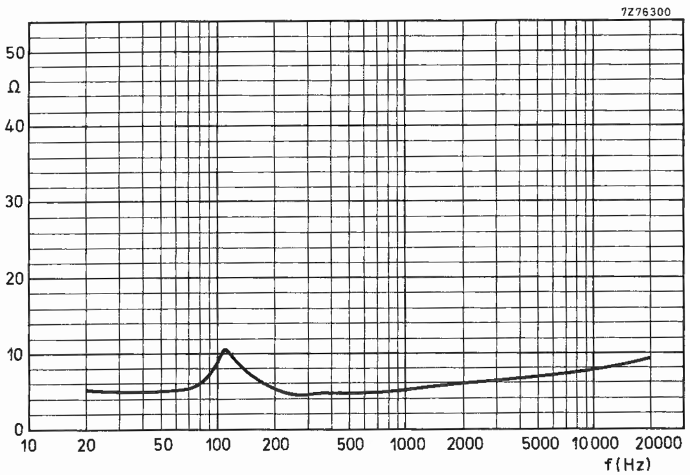
Frequency response and distortion measured in anechoic room.



Energy response in reverberant room.



Frequency response measured in living room.



Impedance curve.

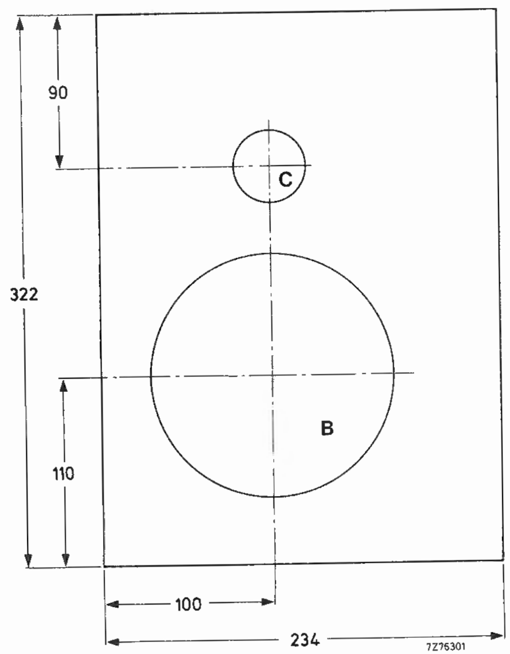
System 3

45 - 20 000 Hz

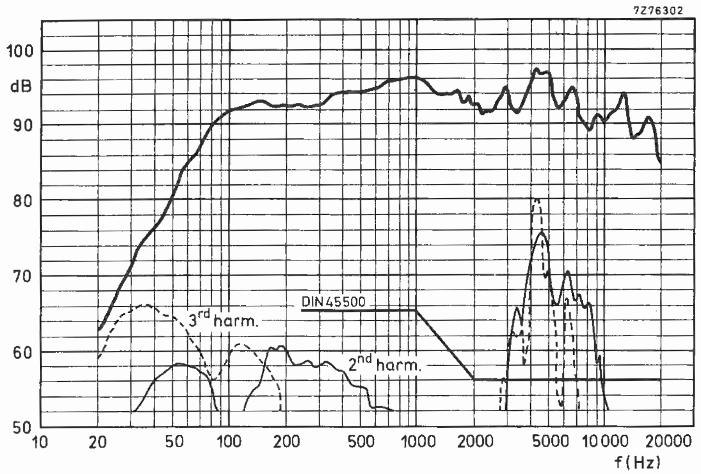
Woofer	AD70601/W4 / W8
Tweeter	AD2296/T4 / T8

Power handling capacity	= 20 W
Rated impedance	= 4 Ω / 8 Ω
Operating power	= 9 W
Resonance frequency	= 85 Hz
Frequency range	= 45-20 000 Hz
Enclosure volume	= 9 litres
Internal dimensions	= 327 × 239 × 125 mm
Internal depth of enclosure	= 125 mm
Material thickness	= 16 mm
Damping material	= glass wool 4 layers 325 × 235 × 40 mm

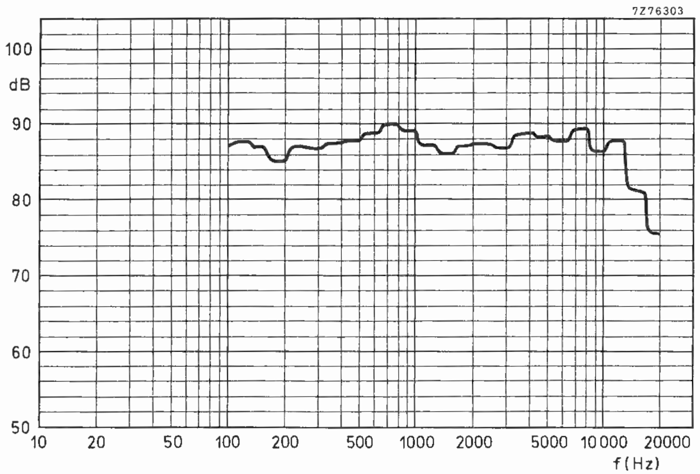
Note: Installing baffle board compresses excess glass wool into enclosure.



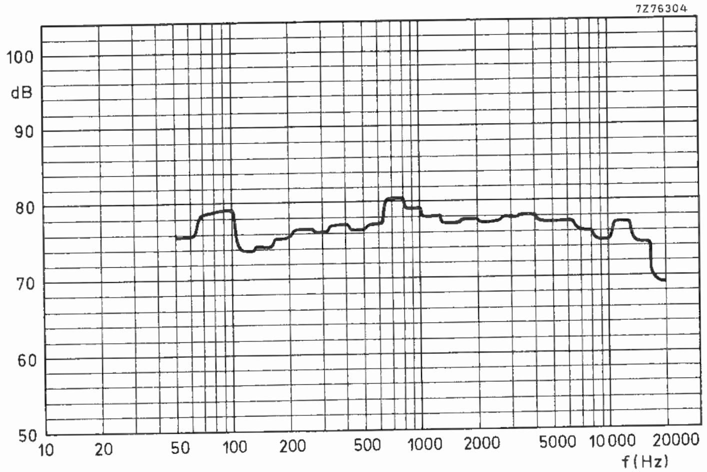
Baffle board layout
(see Table 7.3 for hole sizes)



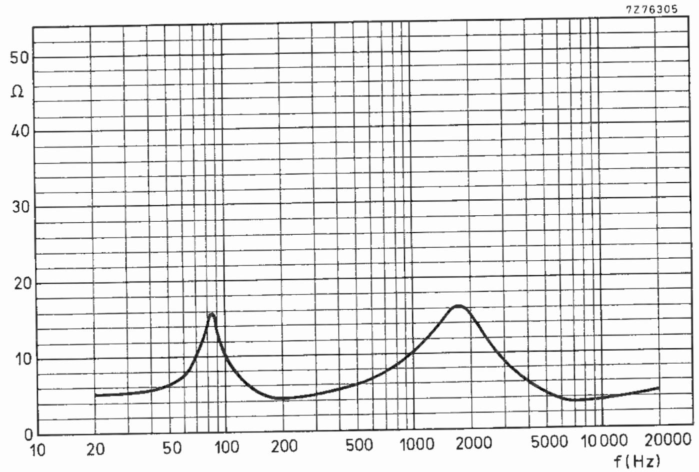
Frequency response and distortion measured in anechoic room.



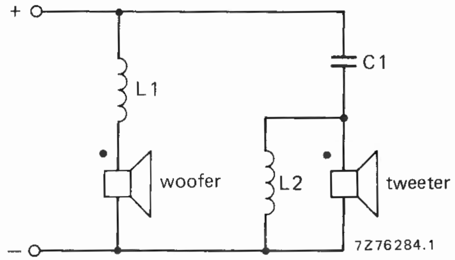
Energy response in reverberant room.



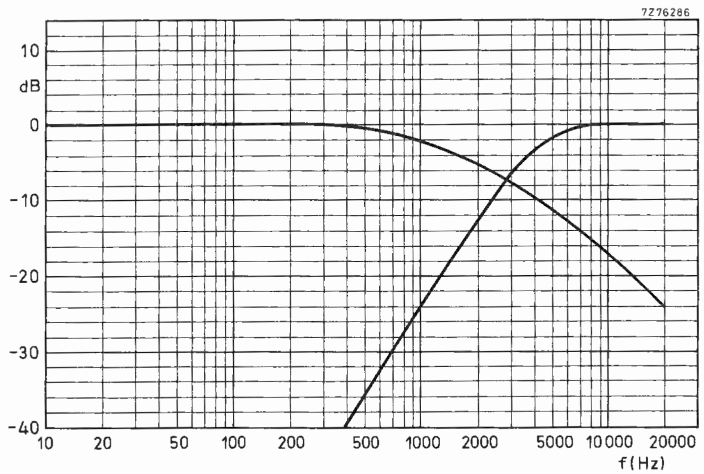
Frequency response measured in living room.



Impedance curve.



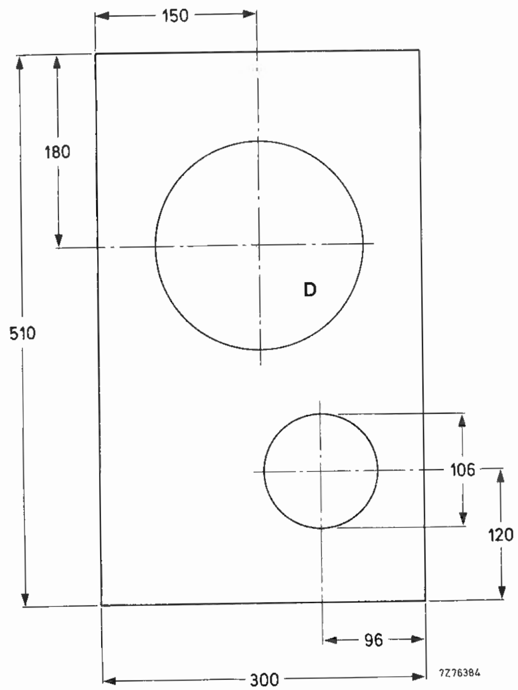
Circuit of filter (see Table 7.2)



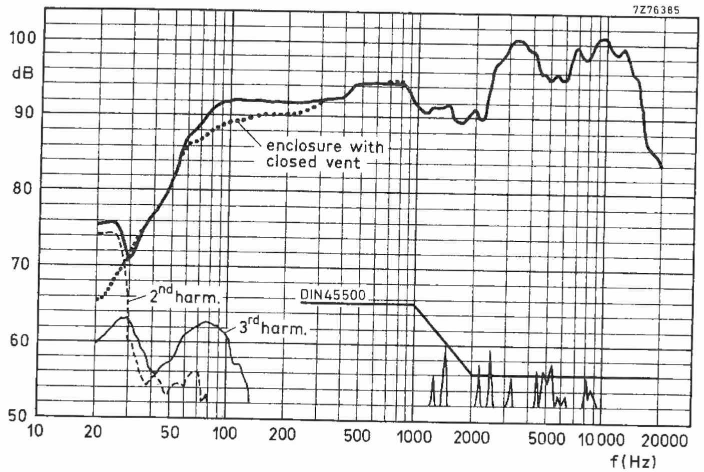
Filter characteristic. Cross-over frequency 3000 Hz (resistor-loaded).

Full range speaker 9710/M8

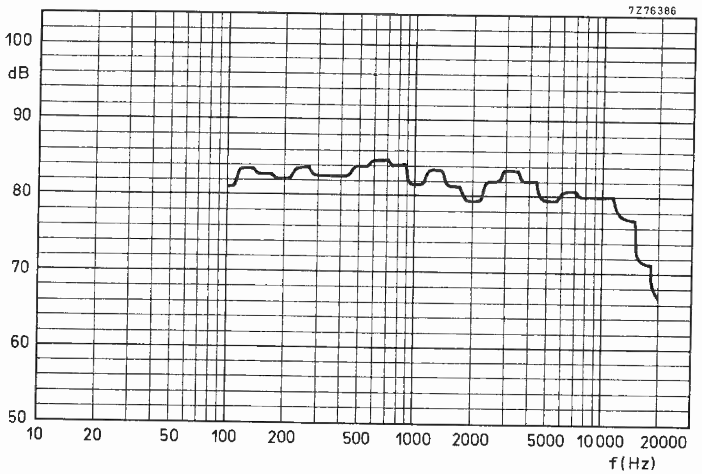
Power handling capacity	= 25 W
Rated impedance	= 8 Ω
Operating power	= 1,1 W
Resonance frequency	= 23 Hz/83 Hz
Frequency range	= 40-20 000 Hz
Enclosure volume	= 40 litres
Internal dimensions	= 515×305×280 mm
Internal depth of enclosure	= 280 mm
Material thickness	= 27 mm
Damping material	= glass wool 2 layers on back wall 510×300×40 mm
Reflex port	= 106 mm o.d. 100 mm i.d. 161 mm long (e.g. PVC or card-board tube)



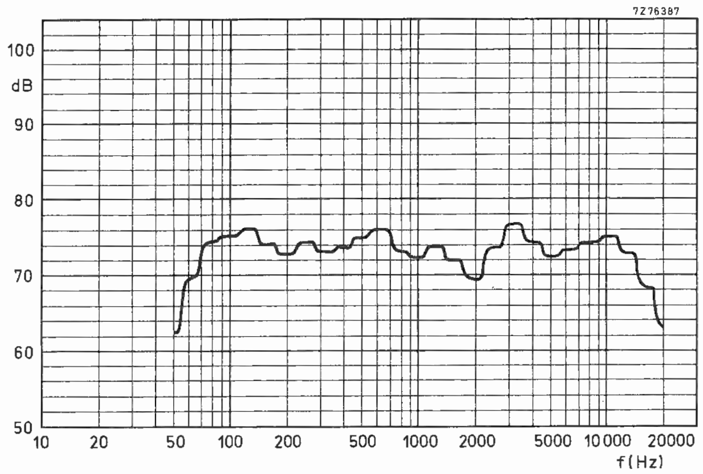
Baffle board layout
(see Table 7.3 for hole size)



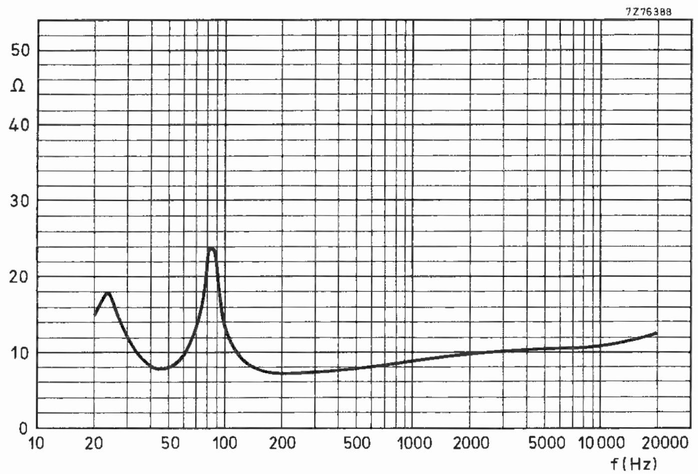
Frequency response and distortion measured in anechoic room.



Energy response in reverberant room.



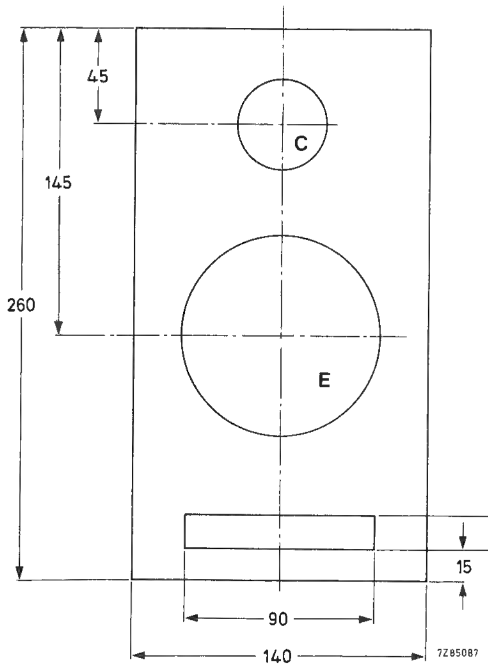
Frequency response measured in living room.



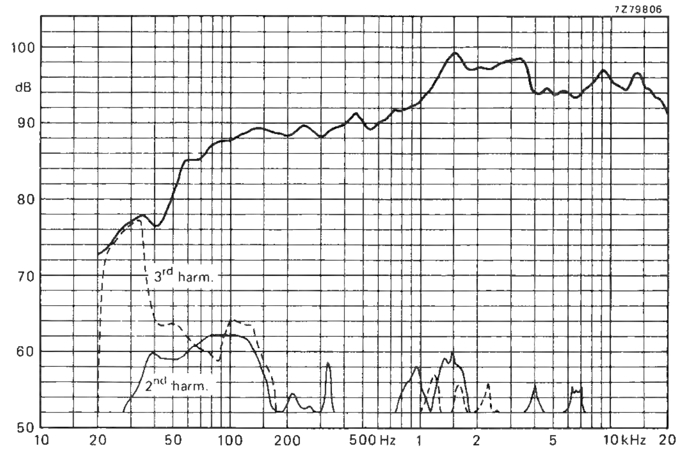
Impedance curve.

Woofer AD4060/W4
Tweeter AD2296/T8

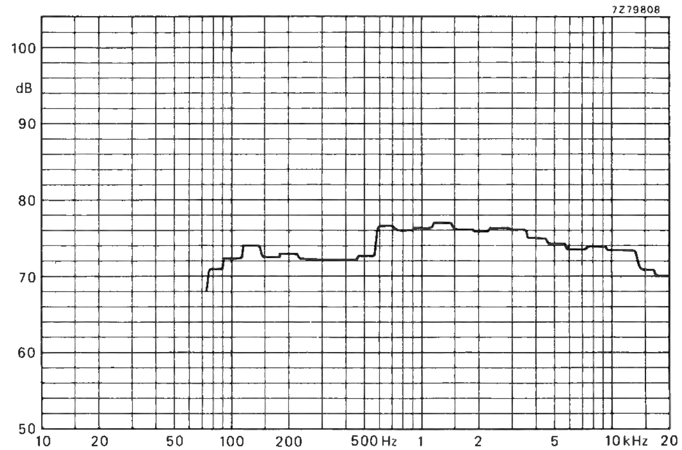
Power handling capacity	=	15 W
Rated impedance	=	4 Ω
Operating power	=	8 W
Resonance frequency	=	33 Hz/150 Hz
Frequency range	=	40-20 000 Hz
Enclosure volume	=	7 litres
Internal dimensions	=	260×192×140 mm
Internal depth of enclosure	=	192 mm
Material thickness	=	12 mm
Damping material	=	60 g glass wool lining
Reflex port	=	160 mm high
	=	90 mm wide
	=	170 mm deep



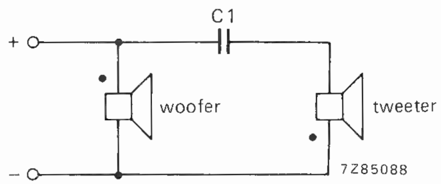
Baffle board layout
(see Table 7.3 for hole sizes)



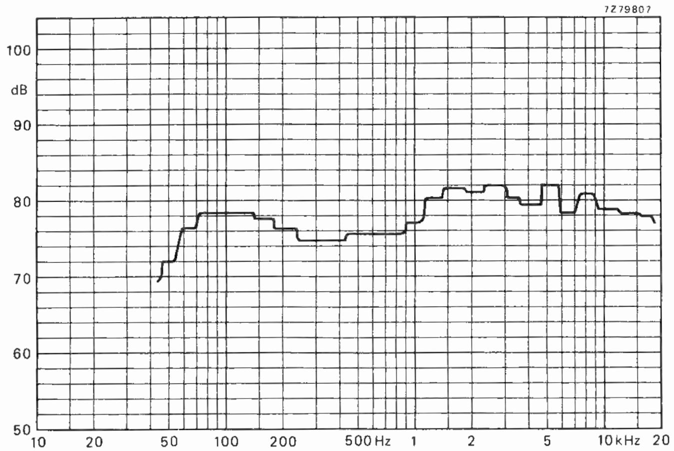
Frequency response and distortion measured in anechoic room.



Energy response in reverberant room.



Circuit of filter (see Table 7.2)

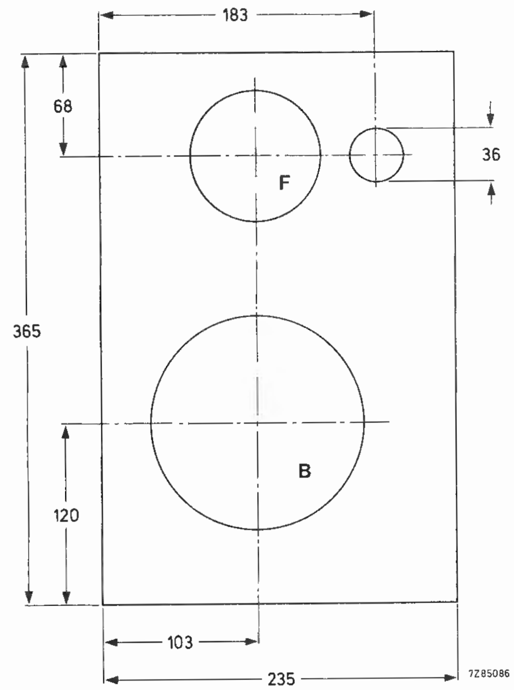


Frequency response measured in living room.

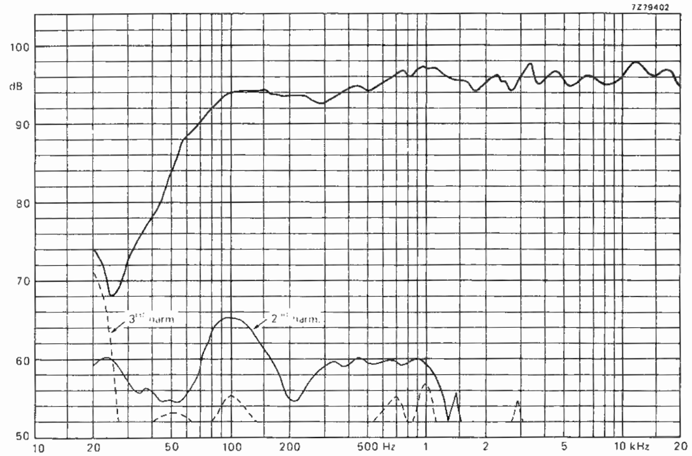
System 6

Woofer AD70610/W4
Tweeter AD01430/T8

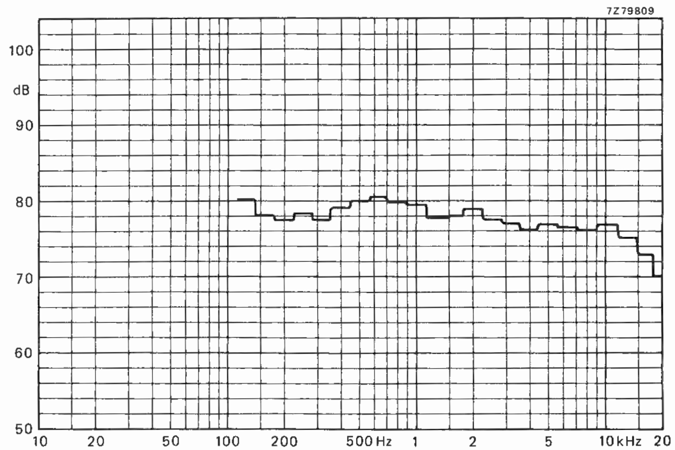
Power handling capacity	=	35 W
Rated impedance	=	4 Ω
Operating power	=	9 W
Resonance frequency	=	20 Hz/86 Hz
Frequency range	=	45-20 000 Hz
Enclosure volume	=	13 litres
Internal dimensions	=	365×235×151 mm
Internal depth of enclosure	=	151 mm
Material thickness	=	18 mm
Damping material	=	50 g glass wool
Reflex port	=	36 mm i.d.
	=	100 mm long



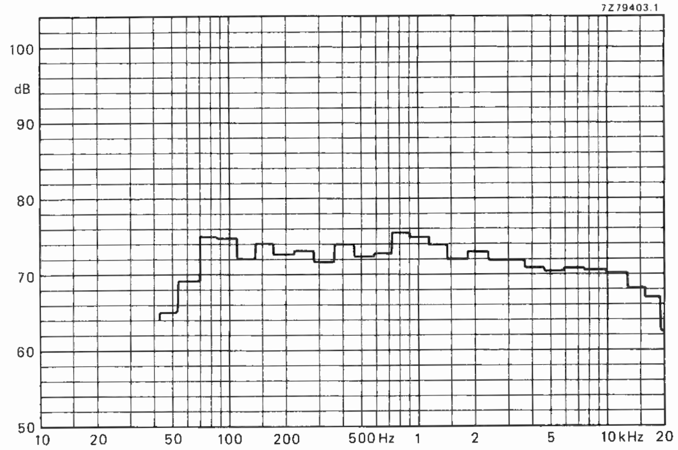
Baffle board layout
(see Table 7.3 for hole sizes)



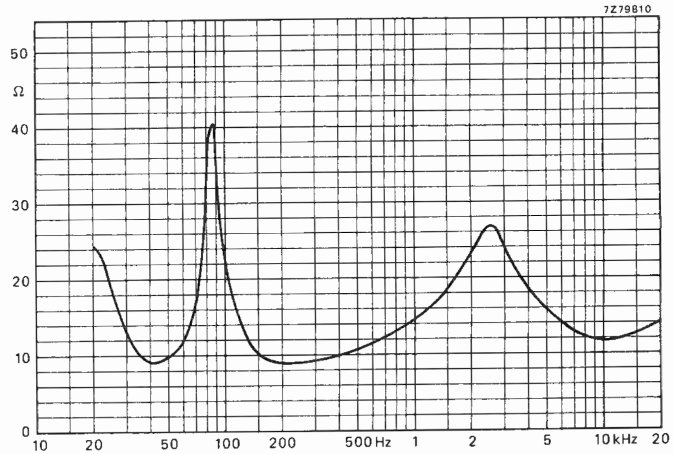
Frequency response and distortion measured in anechoic room.



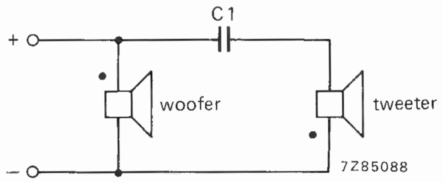
Energy response in reverberant room.



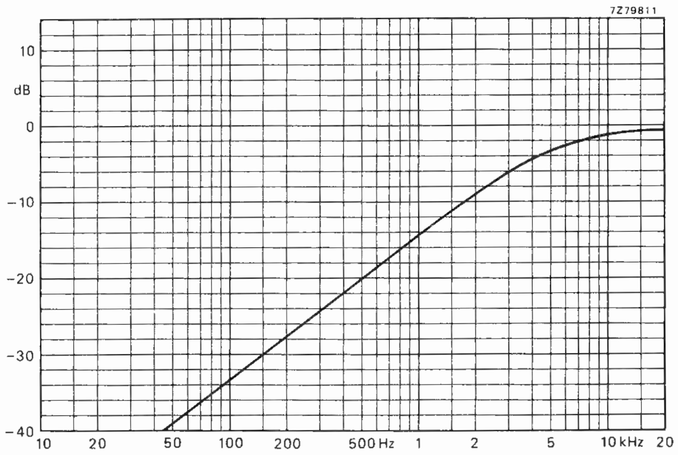
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)



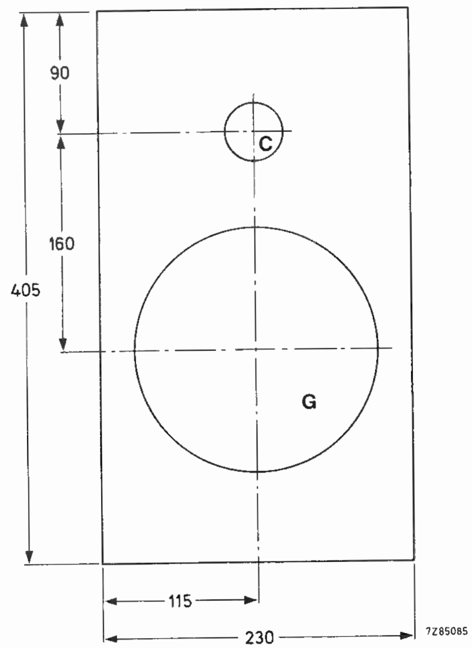
Filter characteristic (measured on resistive load).

System 7

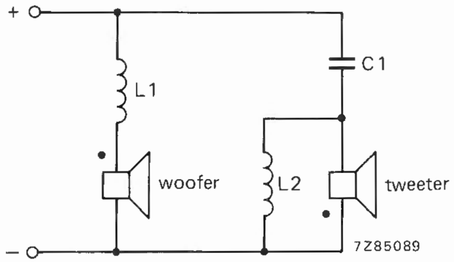
50 - 20 000 Hz

Woofer AD80602/W4
Tweeter AD2296/T4

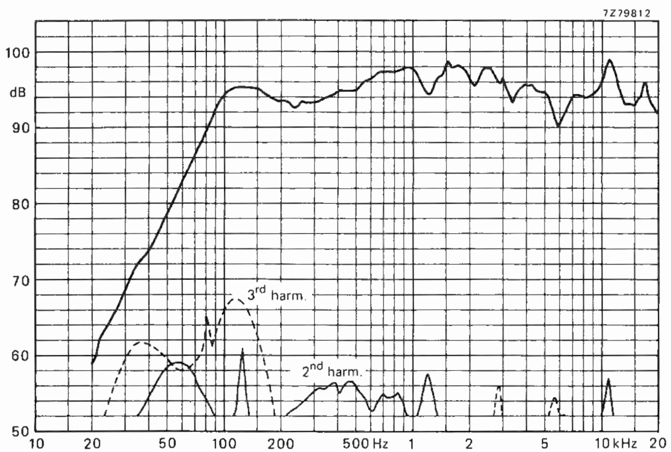
Power handling capacity	=	40 W
Rated impedance	=	4 Ω
Operating power	=	9 W
Resonance frequency	=	100 Hz
Frequency range	=	50-20 000 Hz
Enclosure volume	=	13 litres
Internal dimensions	=	405×230×135 mm
Internal depth of enclosure	=	135 mm
Material thickness	=	18 mm
Damping material	=	20 g polyester fibre



Baffle board layout
(see Table 7.3 for hole sizes)



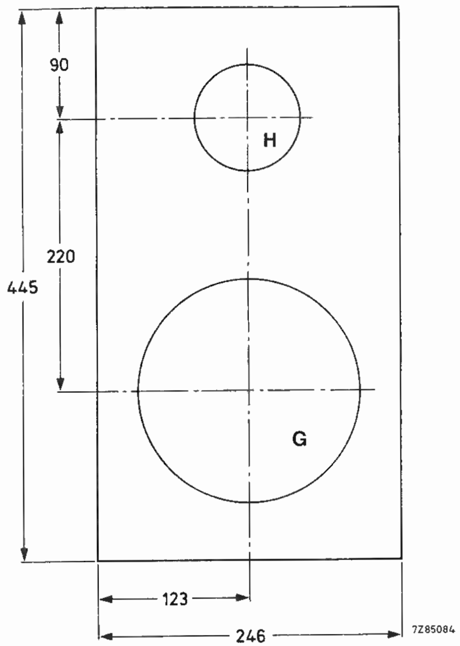
Circuit of filter (see Table 7.2)



Frequency response and distortion measured in anechoic room.

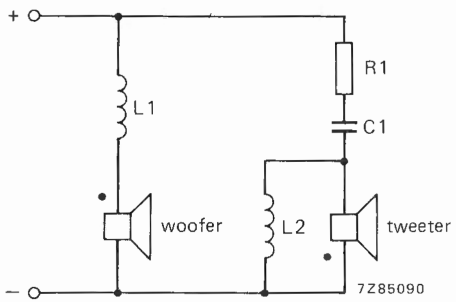
Woofer AD80602/W4
Tweeter AD01610/T8

Power handling capacity	=	50 W
Rated impedance	=	4 Ω
Operating power	=	9 W
Resonance frequency	=	74 Hz
Frequency range	=	37-20 000 Hz
Enclosure volume	=	23 litres
Internal dimensions	=	445×245×210 mm
Internal depth of enclosure	=	210 mm
Material thickness	=	18 mm
Damping material	=	30 g polyester fibre

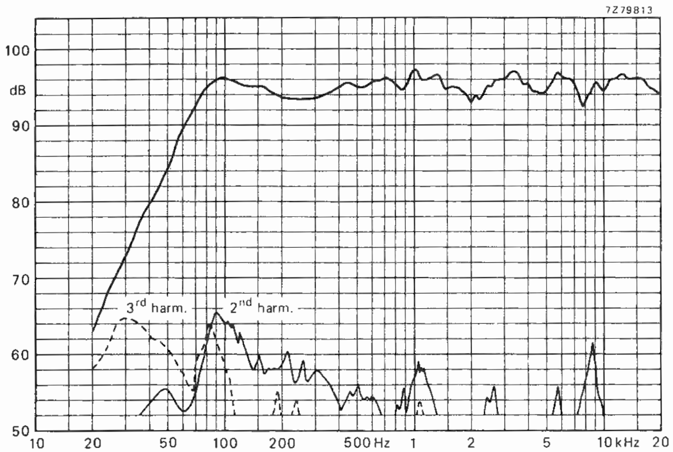


Baffle board layout
(see Table 7.3 for hole sizes)

Two-way sealed system



Circuit of filter (see Table 7.2)



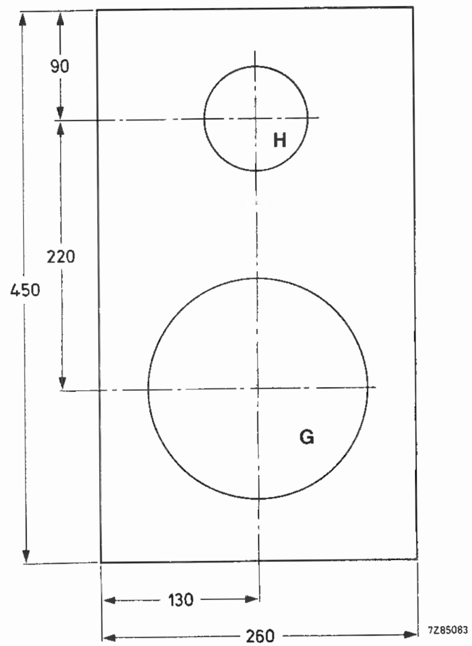
Frequency response and distortion measured in anechoic room.

System 9

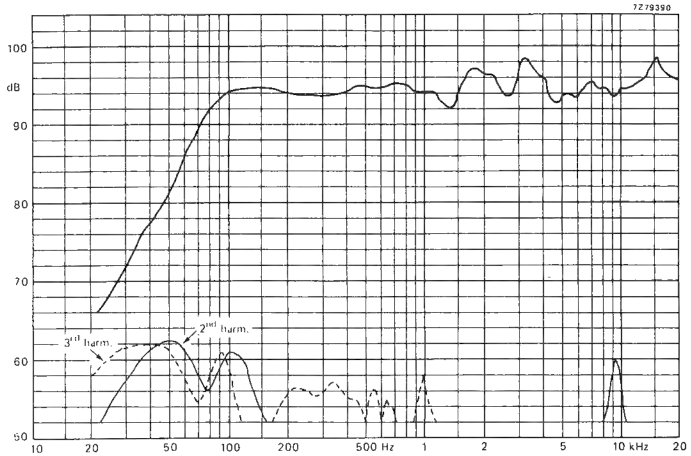
37 - 20 000 Hz

Woofer AD80652/W4
Tweeter AD01630/T8

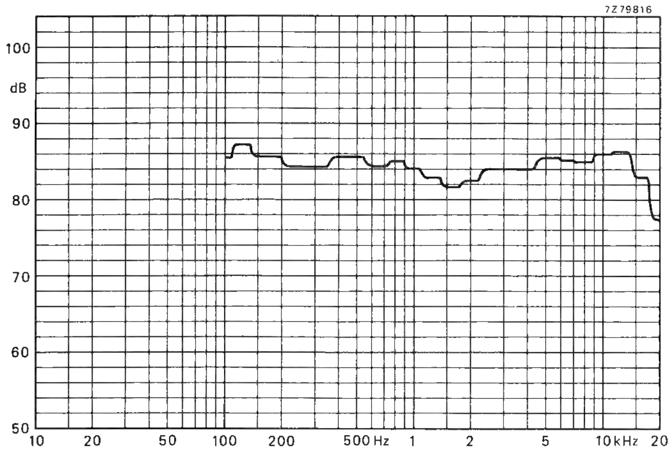
Power handling capacity	=	50 W
Rated impedance	=	4 Ω
Operating power	=	6 W
Resonance frequency	=	73 Hz
Frequency range	=	37-20 000 Hz
Enclosure volume	=	25 litres
Internal dimensions	=	450×260×210 mm
Internal depth of enclosure	=	210 mm
Material thickness	=	18 mm
Damping material	=	75 g glass wool (one layer on back wall)



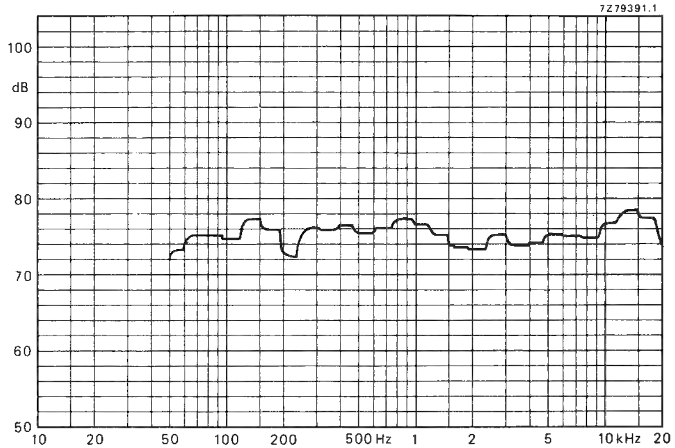
Baffle board layout
(see Table 7.3 for hole sizes)



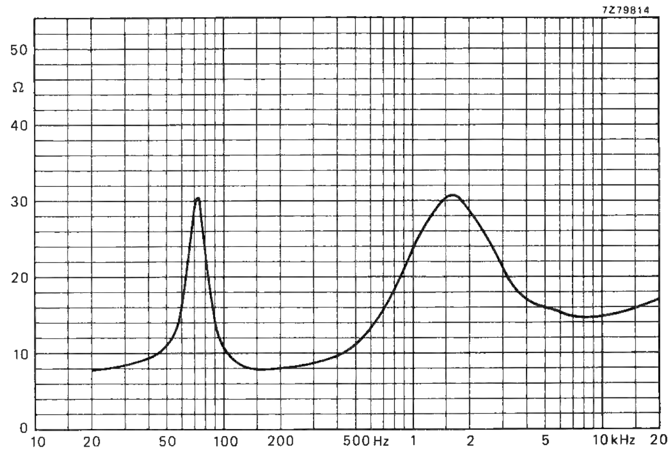
Frequency response and distortion measured in anechoic room.



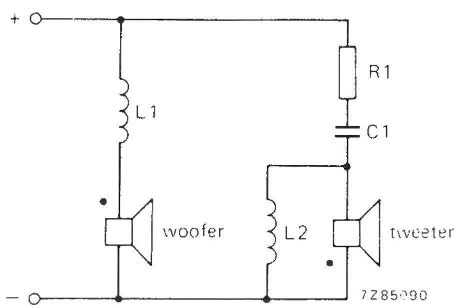
Energy response in reverberant room.



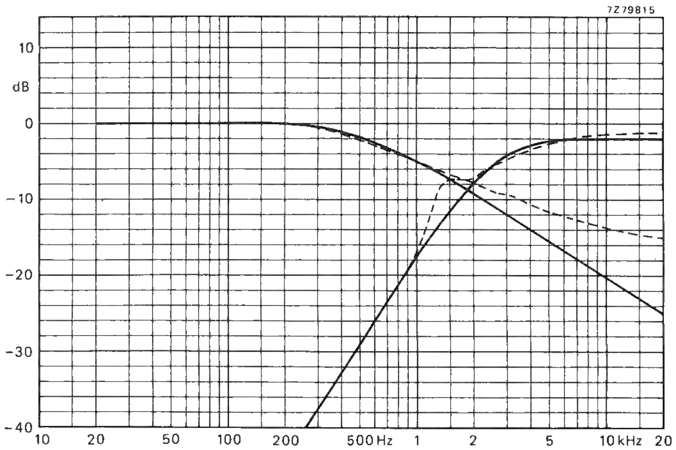
Frequency response measured in living room.



Impedance curve.



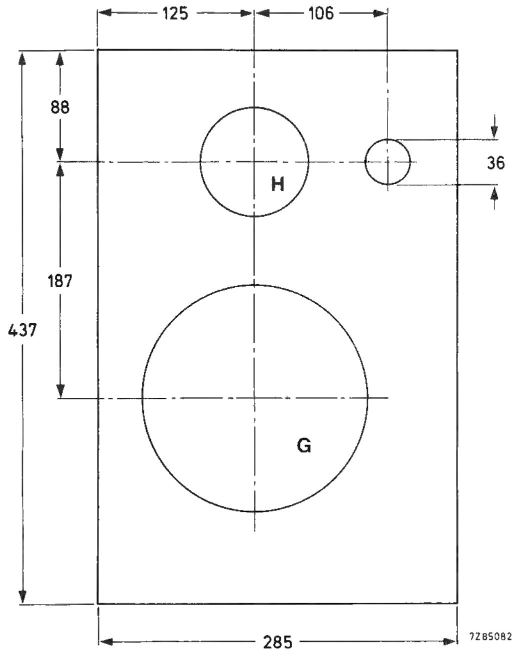
Circuit of filter (see Table 7.2)



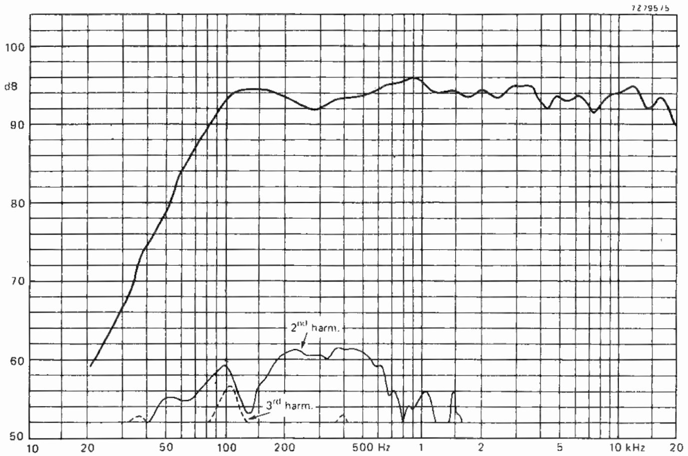
Filter characteristic. Cross-over frequencies 1800 Hz; full line — resistor-loaded, dotted line — speaker-loaded.

Woofer	AD80603/W4
Tweeter	AD01420/T4

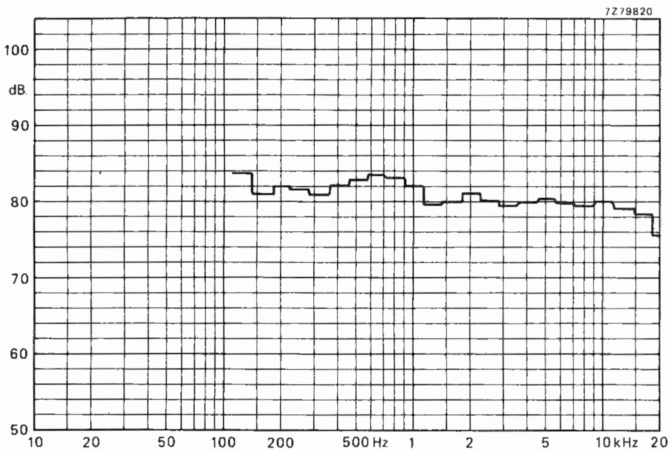
Power handling capacity	= 40 W
Rated impedance	= 4 Ω
Operating power	= 5 W
Resonance frequency	= 18 Hz/88 Hz
Frequency range	= 45-20 000 Hz
Enclosure volume	= 17 litres
Internal dimensions	= 437×285×153 mm
Internal depth of enclosure	= 153 mm
Material thickness	= 18 mm
Damping material	= 65 g glass wool
Reflex port	= 36 mm i.d. 115 mm long



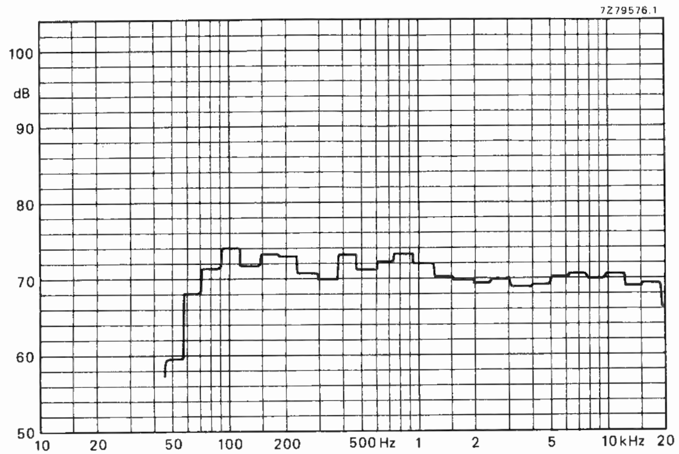
Baffle board layout
(see Table 7.3 for hole sizes)



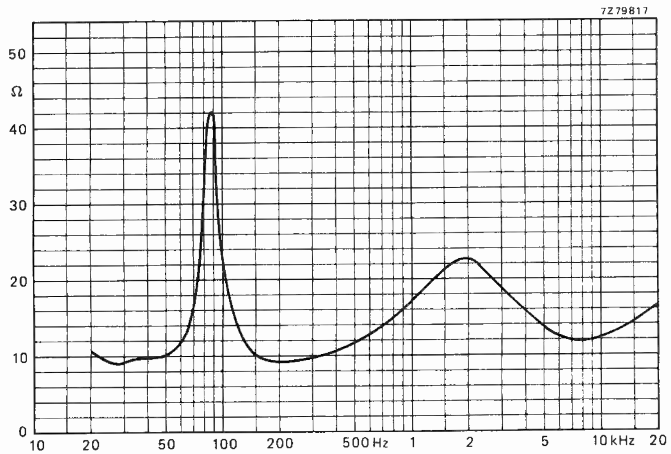
Frequency response and distortion measured in anechoic room.



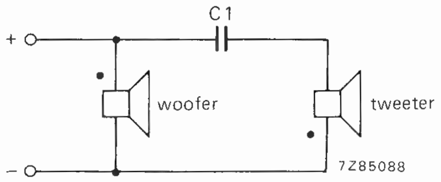
Energy response in reverberant room.



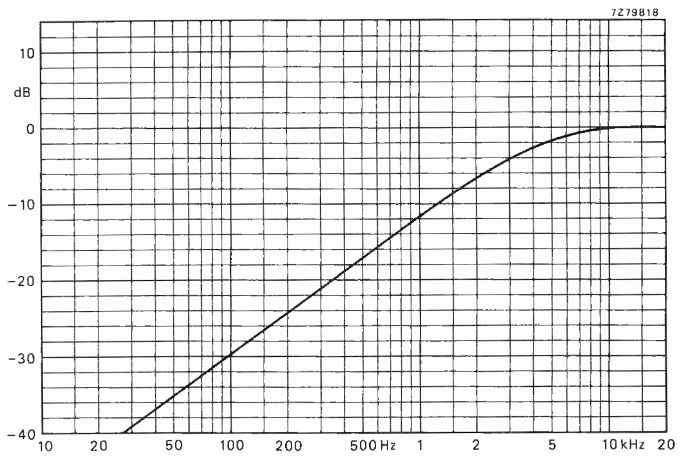
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)

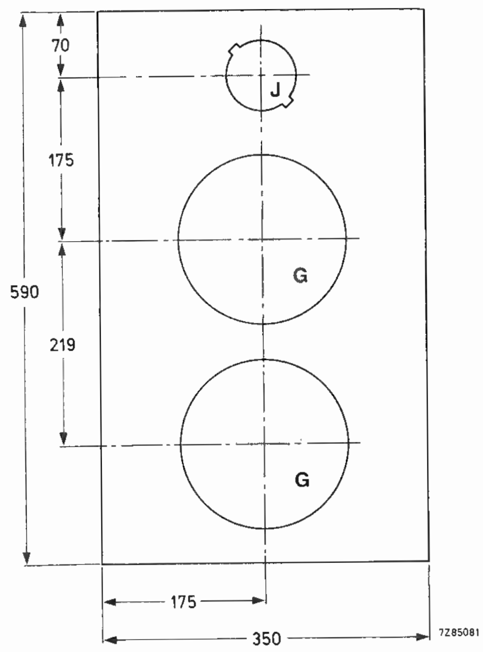


Filter characteristic. Cross-over frequency 3800 Hz (resistor-loaded).

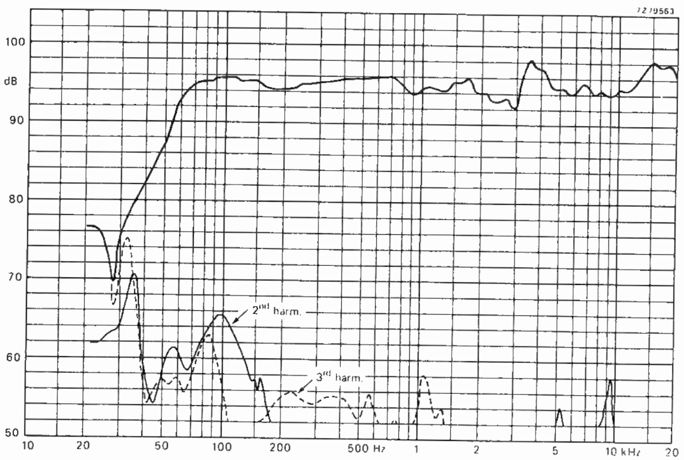
System 11

Woofer AD80652/W4
Passive radiator AD8002
Tweeter AD0162/T8

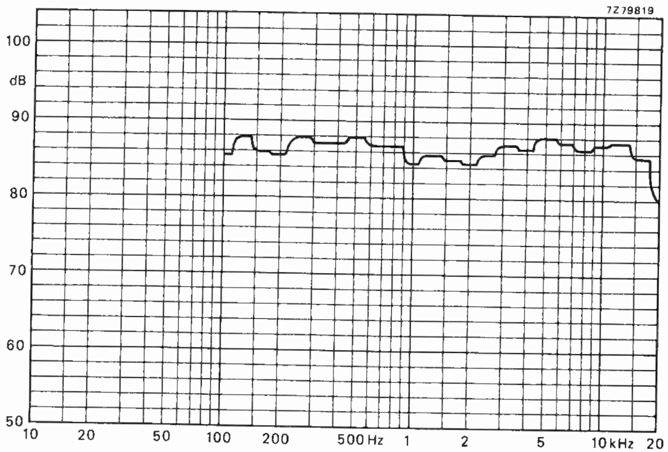
Power handling capacity	=	50 W
Rated impedance	=	4 Ω
Operating power	=	6 W
Resonance frequency	=	27 Hz/65 Hz
Frequency range	=	32-20 000 Hz
Enclosure volume	=	45 litres
Internal dimensions	=	590×350×220 mm
Internal depth of enclosure	=	220 mm
Material thickness	=	22 mm
Damping material	=	120 g glass wool (one layer on back wall)



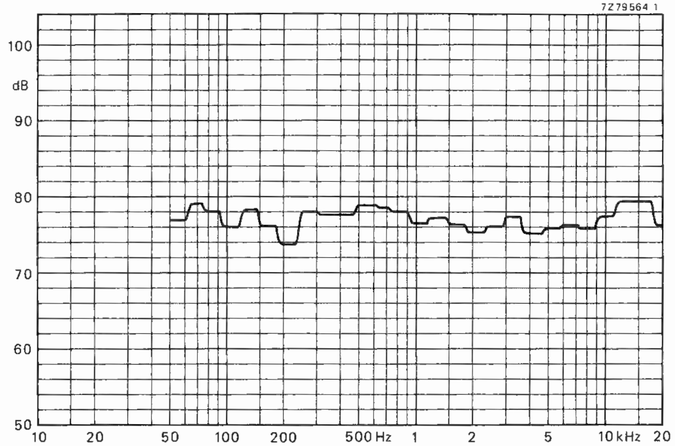
Baffle board layout
(see Table 7.3 for hole sizes)



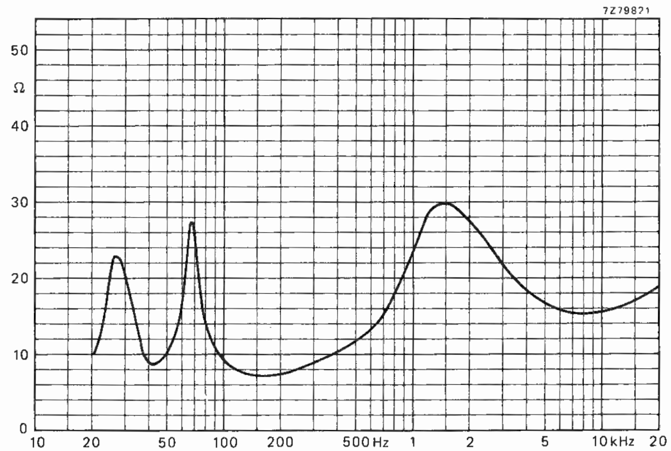
Frequency response and distortion measured in anechoic room.



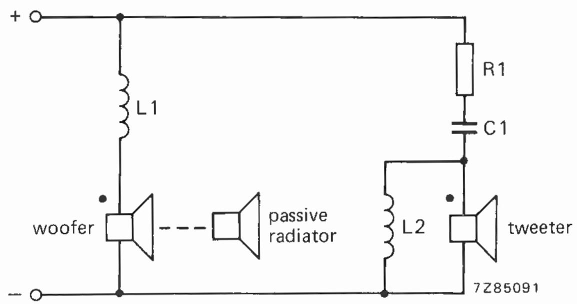
Energy response in reverberant room.



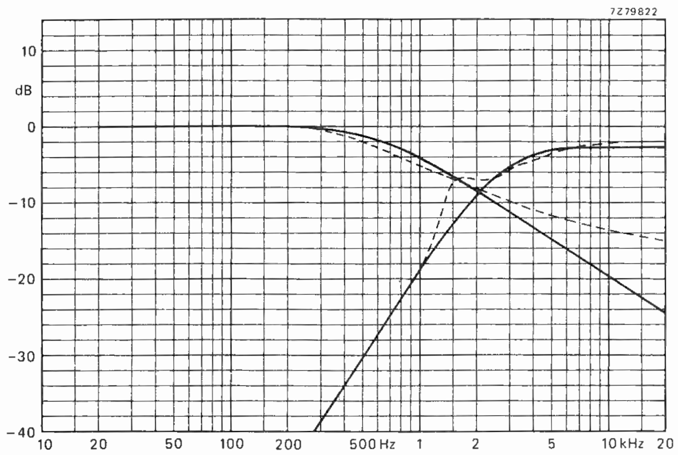
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)

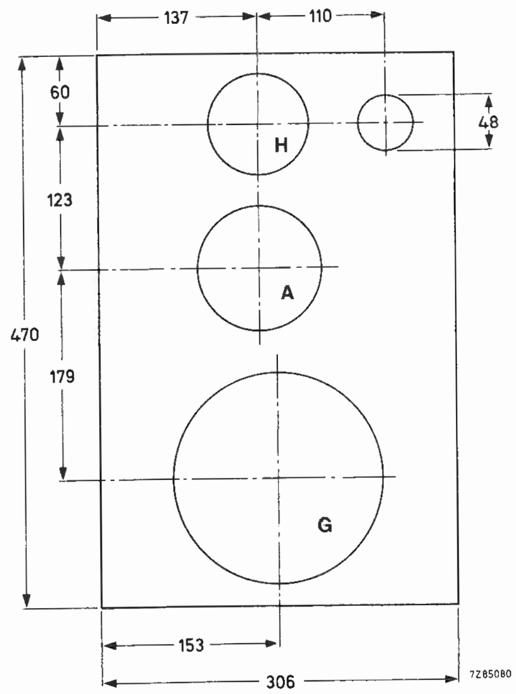


Filter characteristic. Cross-over frequencies 2000 Hz. Full line — resistor-loaded; dotted line — speaker-loaded.

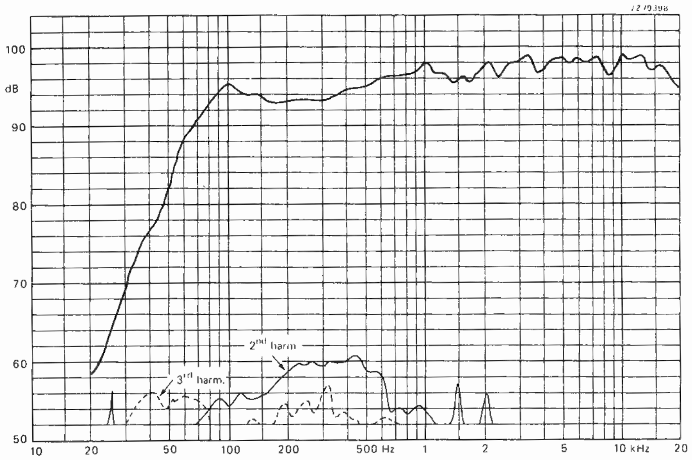
System 12

Woofer	AD80603/W4
Squawker	AD5062/Sq4
Tweeter	AD01420/T4

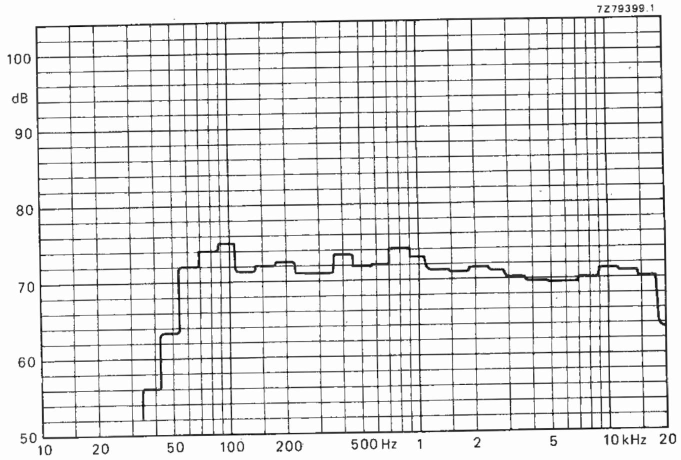
Power handling capacity	=	50 W
Rated impedance	=	4 Ω
Operating power	=	3 W
Resonance frequency	=	19 Hz/80 Hz
Frequency range	=	42-20 000 Hz
Enclosure volume	=	25 litres
Internal dimensions	=	470×306×175 mm
Internal depth of enclosure	=	175 mm
Material thickness	=	20 mm
Damping material	=	85 g glass wool
Reflex port	=	48 mm i.d.
	=	60 mm long



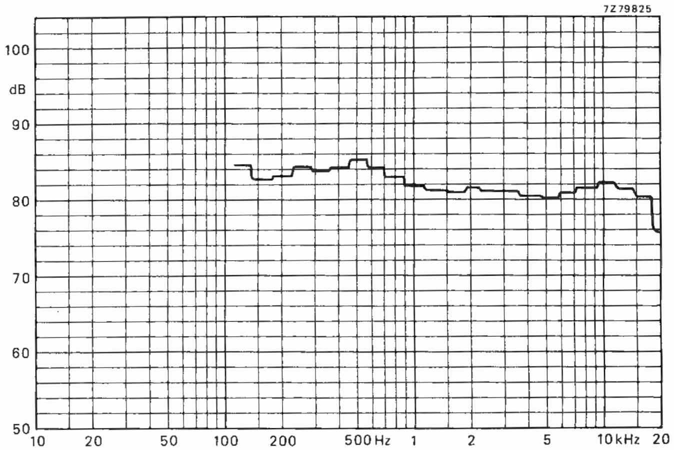
Baffle board layout
(see Table 7.3 for hole sizes)



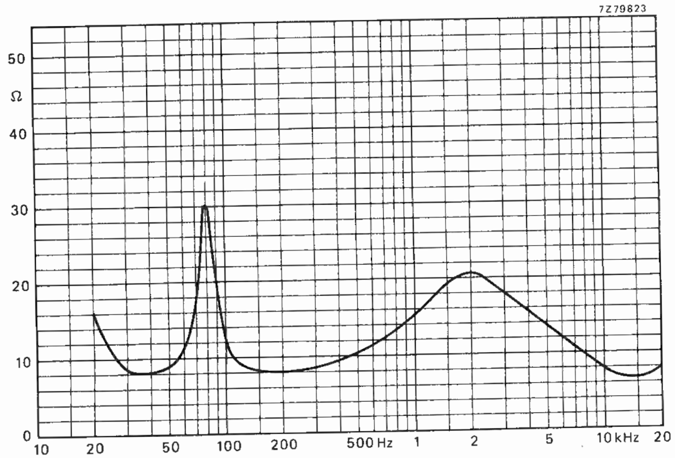
Frequency response and distortion measured in anechoic room.



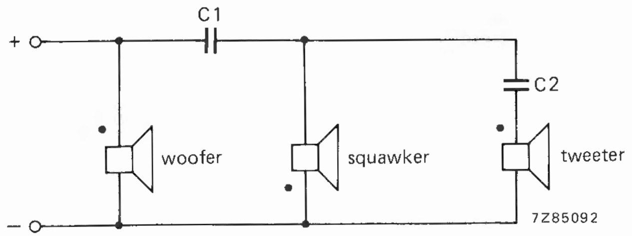
Frequency response measured in living room.



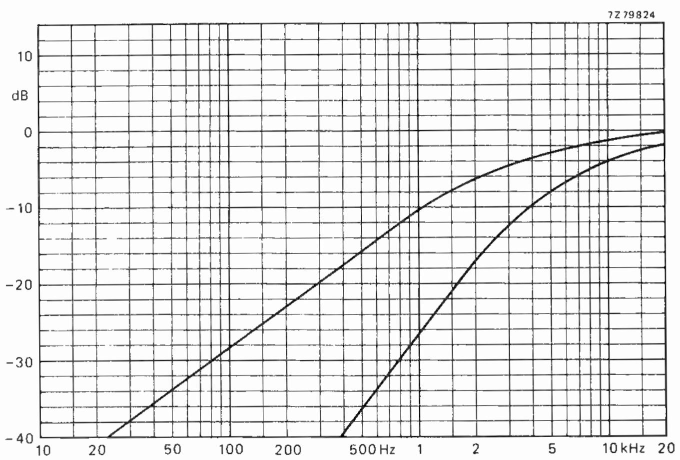
Energy response in reverberant room.



Impedance curve.



Circuit of filter (see Table 7.2)

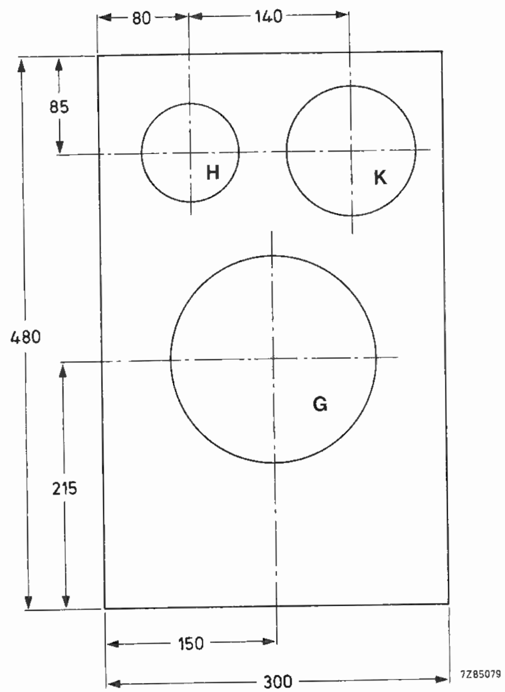


Filter characteristic. Cross-over frequencies 4000 Hz and 9000 Hz, measured on resistive loads.

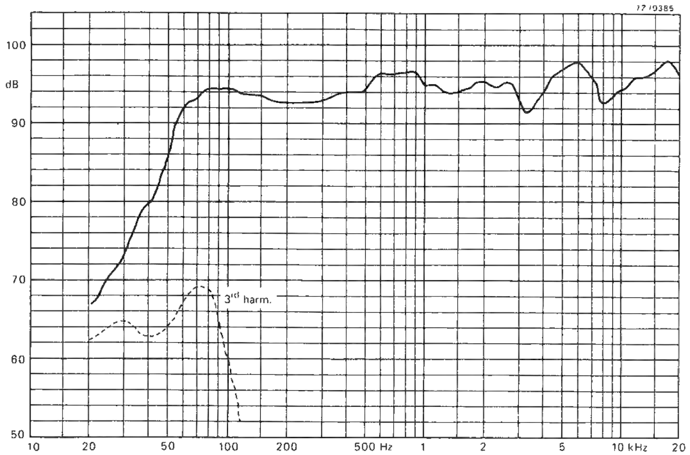
System 13

Woofer AD80652/W4
Squawker AD0211/Sq4
Tweeter AD01630/T4

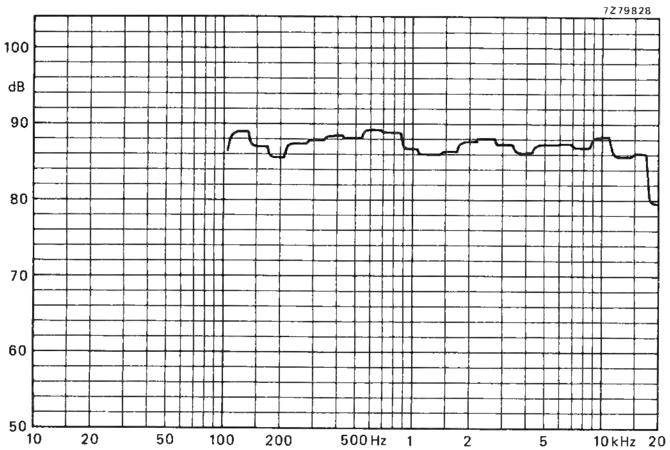
Power handling capacity	=	50 W
Rated impedance	=	4 Ω
Operating power	=	7 W
Resonance frequency	=	63 Hz
Frequency range	=	32-20 000 Hz
Enclosure volume	=	32 litres
Internal dimensions	=	480 × 300 × 220 mm
Internal depth of enclosure	=	220 mm
Material thickness	=	20 mm
Damping material	=	100 g glass wool (one layer on back wall)



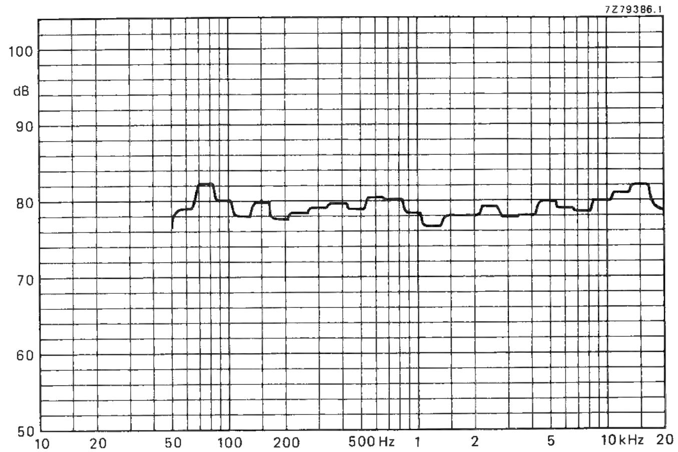
Baffle board layout
(see Table 7.3 for hole sizes)



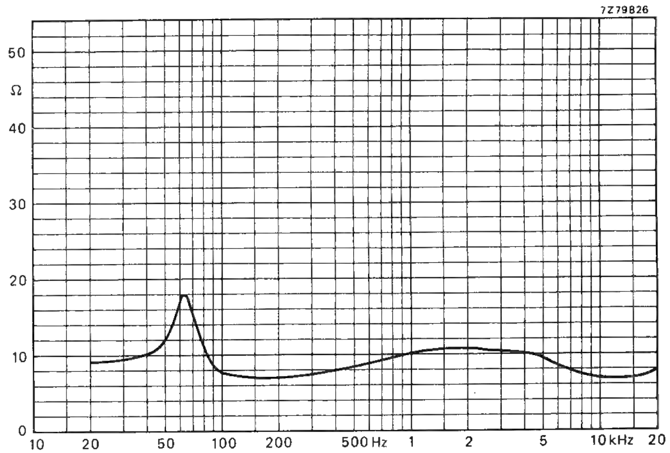
Frequency response and distortion measured in anechoic room.



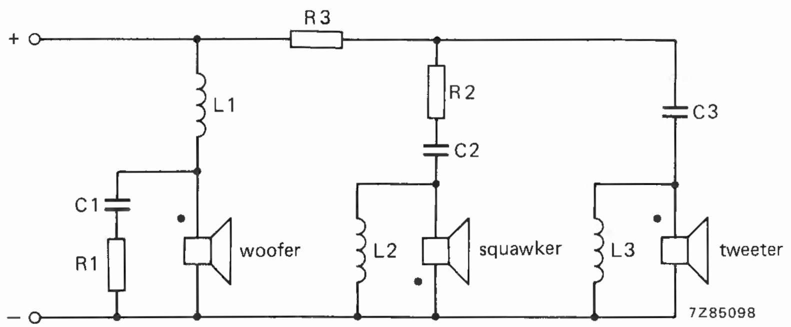
Energy response in reverberant room.



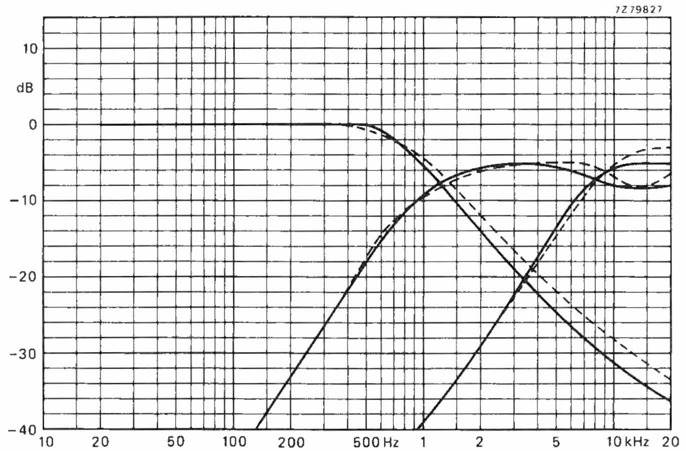
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)

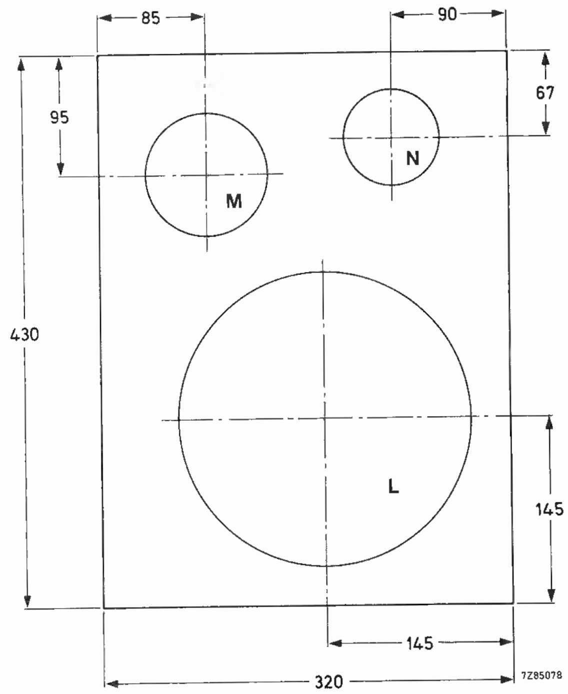


Filter characteristic. Cross-over frequency 1200 Hz and 7000 Hz; full line — resistor-loaded, dotted line — speaker loaded.

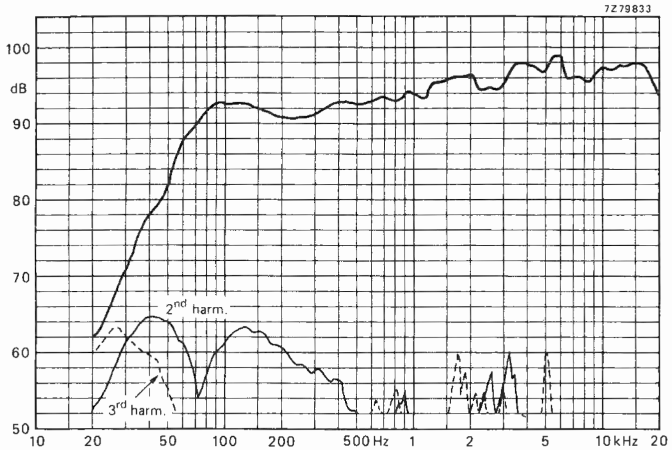
System 14

Woofer	AD10650/W4
Squawker	AD5061/Sq4
Tweeter	AD0140/T4

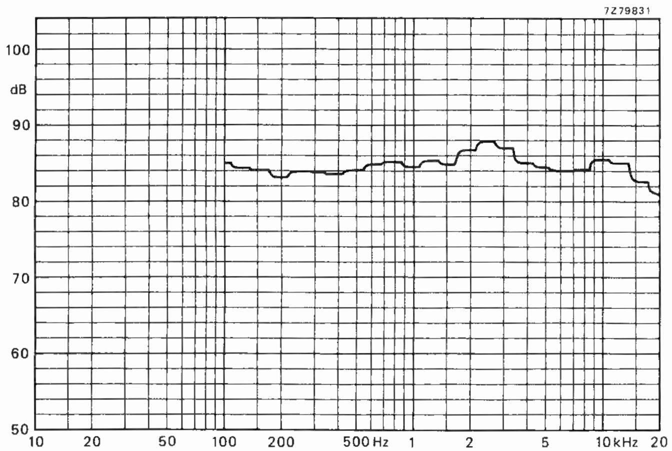
Power handling capacity	= 35 W
Rated impedance	= 4 Ω
Operating power	= 3 W
Resonance frequency	= 69 Hz
Frequency range	= 35-20 000 Hz
Enclosure volume	= 25 litres
Internal dimensions	= 430×320×180 mm
Internal depth of enclosure	= 180 mm
Material thickness	= 18 mm
Damping material	= 130 g glass wool (two layers on back wall)



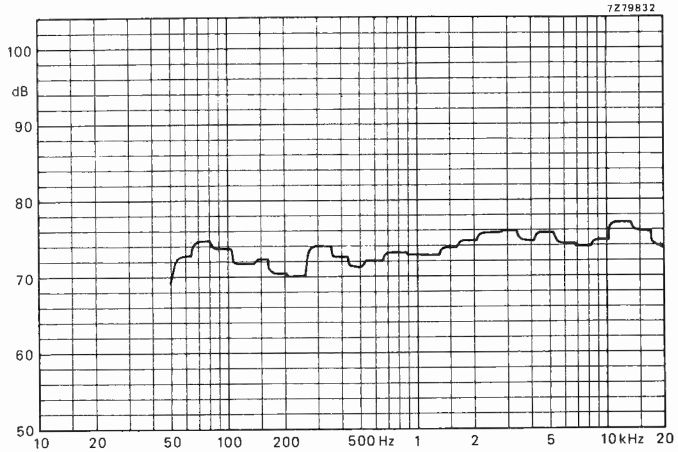
Baffle board layout
(see Table 7.3 for hole sizes)



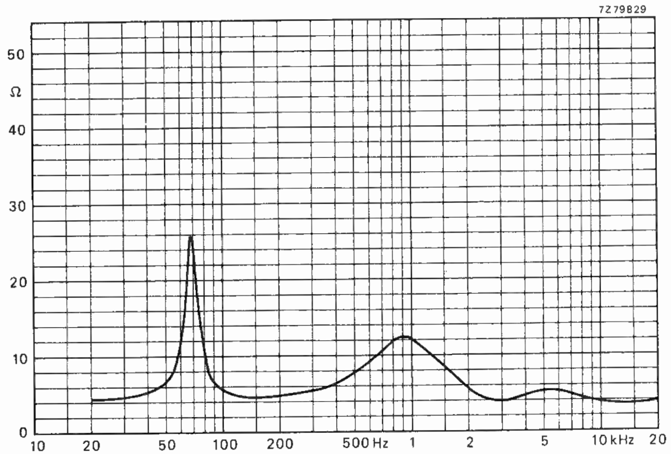
Frequency response and distortion measured in anechoic room.



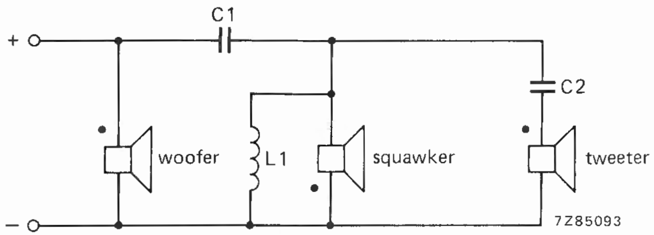
Energy response in reverberant room.



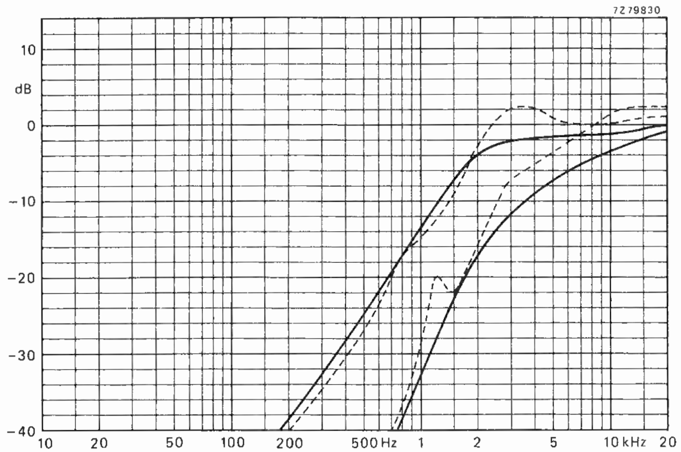
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)



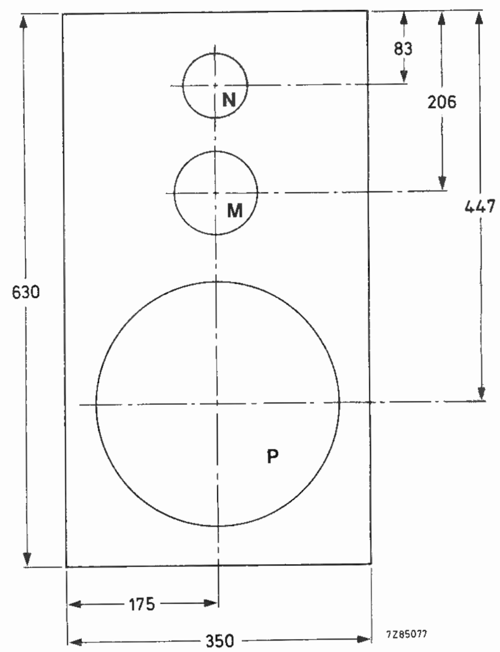
Filter characteristic. Cross-over frequencies 1600 Hz and 8000 Hz; full line — resistor loaded, dotted line — speaker-loaded.

System 15

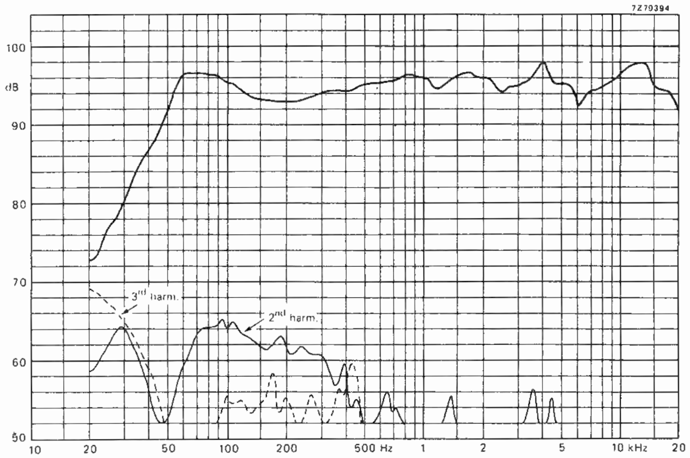
26 - 20 000 Hz

Woofer	AD12650/W4
Squawker	AD5061/Sq4
Tweeter	AD0141/T4

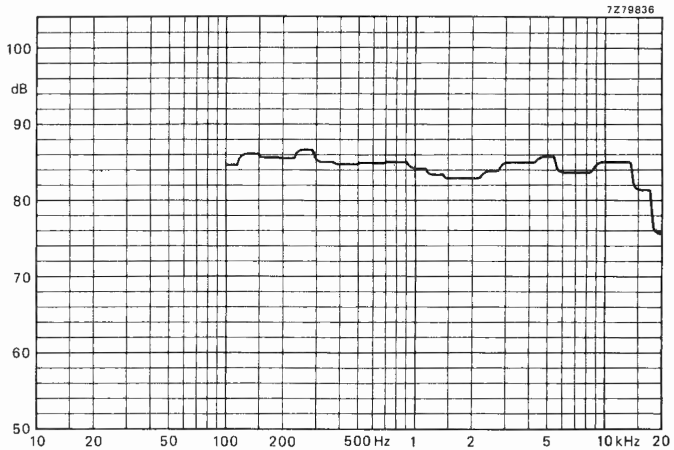
Power handling capacity	= 60 W
Rated impedance	= 4 Ω
Operating power	= 6 W
Resonance frequency	= 53 Hz
Frequency range	= 26-20 000 Hz
Enclosure volume	= 60 litres
Internal dimensions	= 630×350×272 mm
Internal depth of enclosure	= 272 mm
Material thickness	= 20 mm
Damping material	= 800 g glass wool



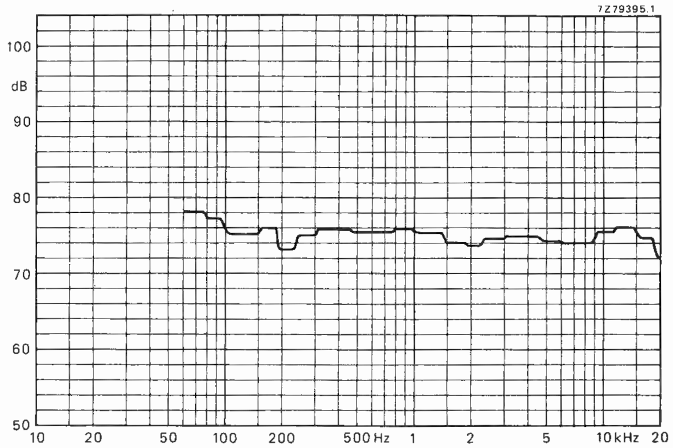
Baffle board layout
(see Table 7.3 for hole sizes)



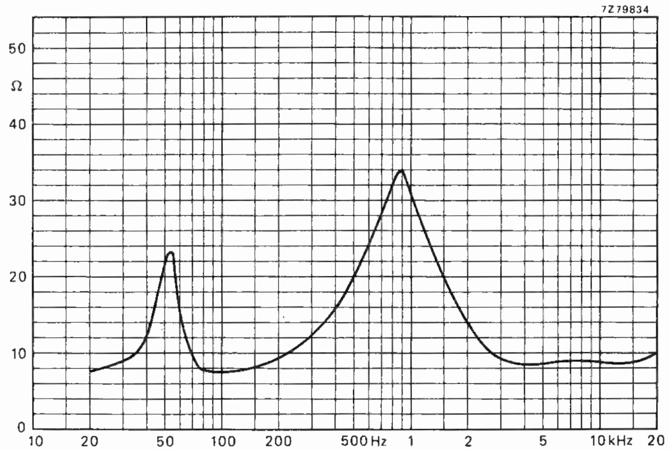
Frequency response and distortion measured in anechoic room.



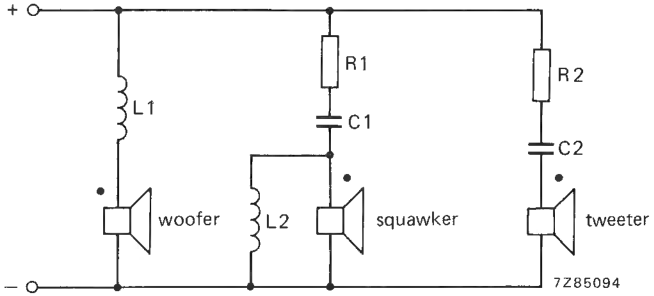
Energy response in reverberant room.



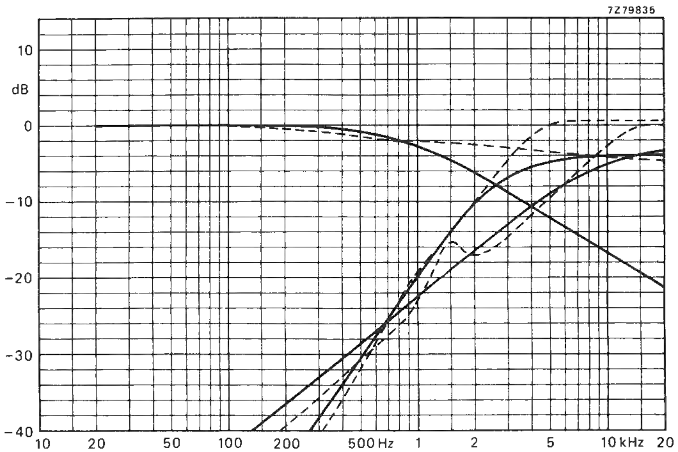
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)

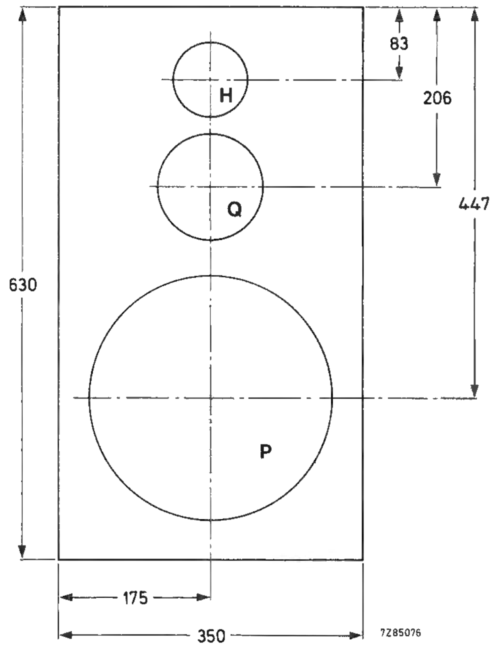


Filter characteristic. Cross-over frequencies 2500 Hz and 8000 Hz; full line — resistor loaded, dotted line — speaker-loaded.

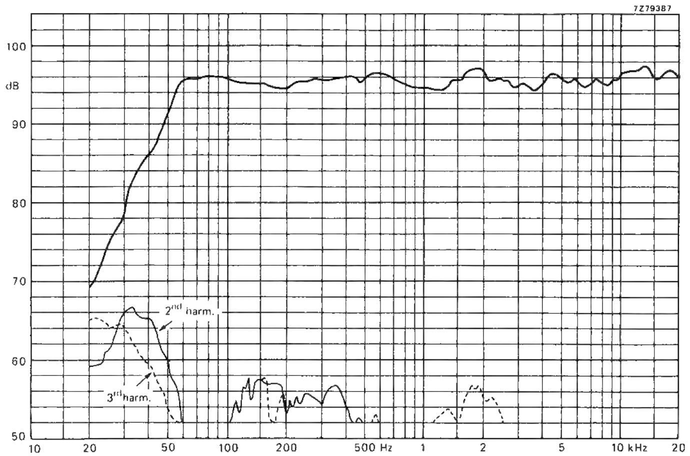
System 16 26 - 20 000 Hz

Woofer	AD12650/W4
Squawker	AD02110/Sq4
Tweeter	AD01630/T8

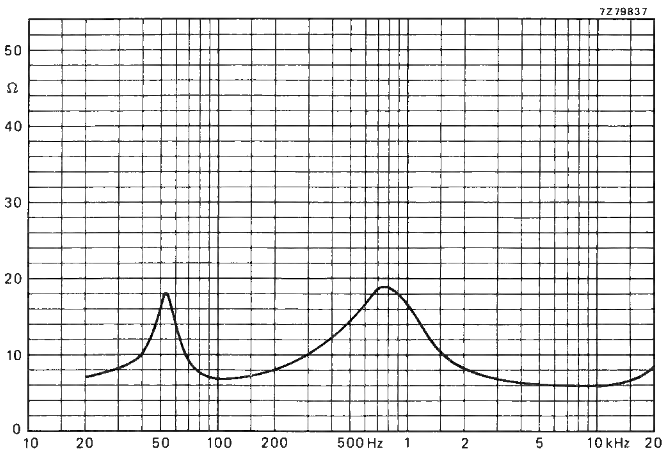
Power handling capacity	=	80 W
Rated impedance	=	4 Ω
Operating power	=	5 W
Resonance frequency	=	53 Hz
Frequency range	=	27-20 000 Hz
Enclosure volume	=	50 litres
Internal dimensions	=	630×350×240 mm
Internal depth of enclosure	=	240 mm
Material thickness	=	20 mm
Damping material	=	800 g glass wool



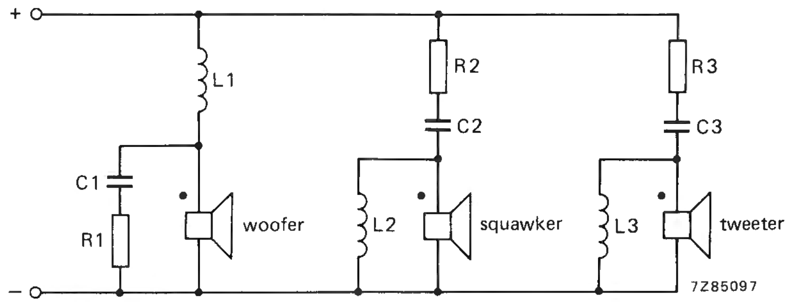
Baffle board layout
(see Table 7.3 for hole sizes)



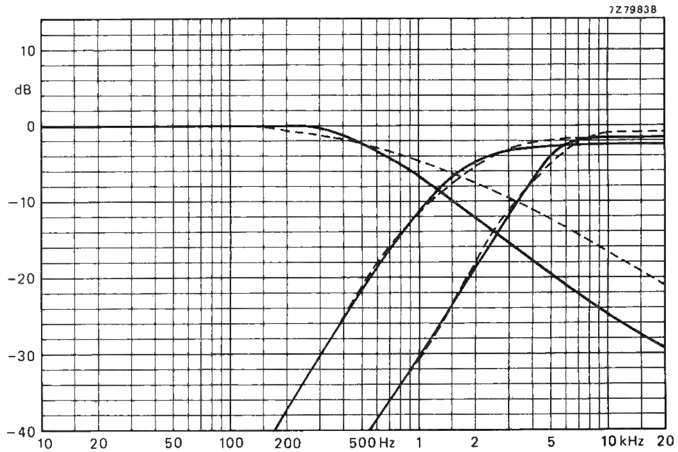
Frequency response and distortion measured in anechoic room.



Impedance curve.



Circuit of filter (see Table 7.2)

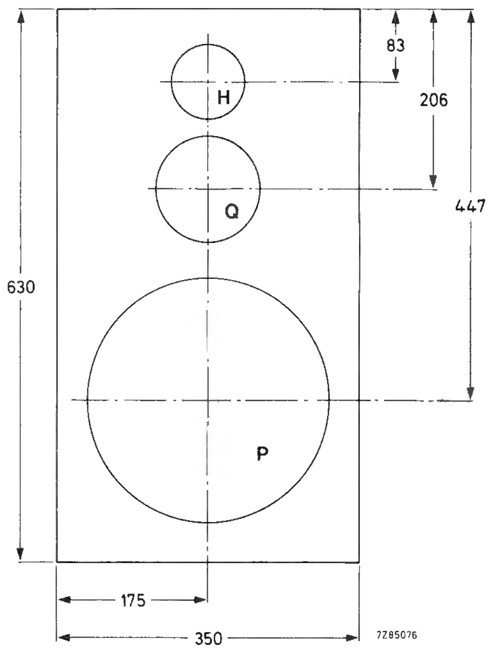


Filter characteristic. Cross-over frequencies 1200 Hz and 6000 Hz; full line — resistor loaded, dotted line — speaker-loaded.

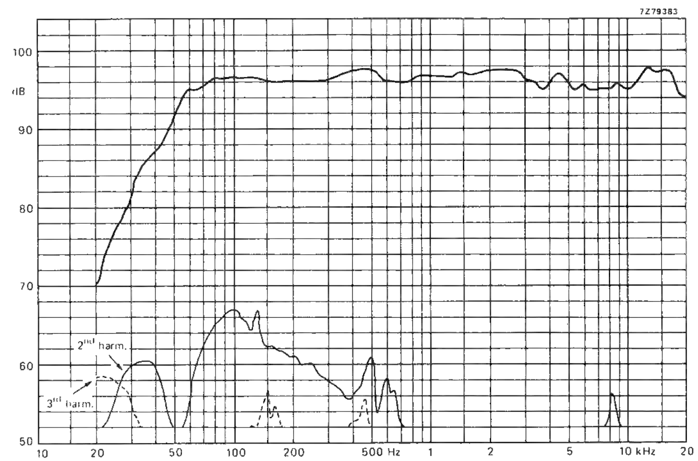
23 - 20 000 Hz

Woofer AD12200/W4
Squawker AD02110/Sq4
Tweeter AD01630/T8

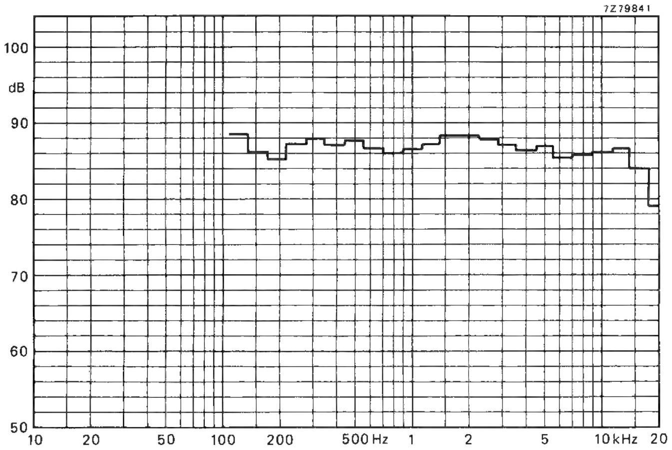
Power handling capacity	=	80 W
Rated impedance	=	4 Ω
Operating power	=	7 W
Resonance frequency	=	46 Hz
Frequency range	=	23-20 000 Hz
Enclosure volume	=	60 litres
Internal dimensions	=	630×350×272 mm
Internal depth of enclosure	=	272 mm
Material thickness	=	20 mm
Damping material	=	440 g glass wool (three layers on back wall)



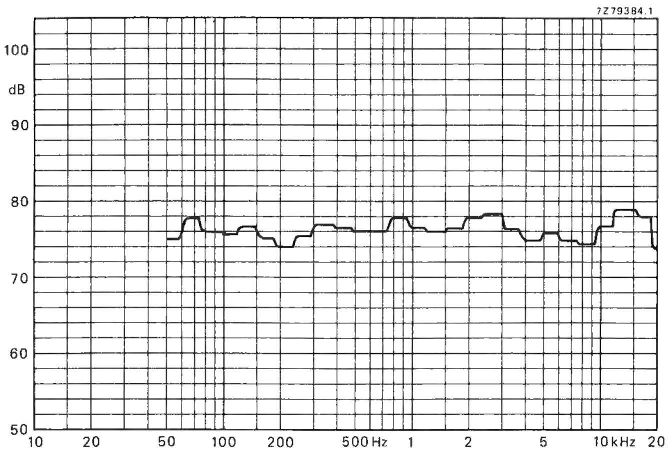
Baffle board layout
(see Table 7.3 for hole sizes)



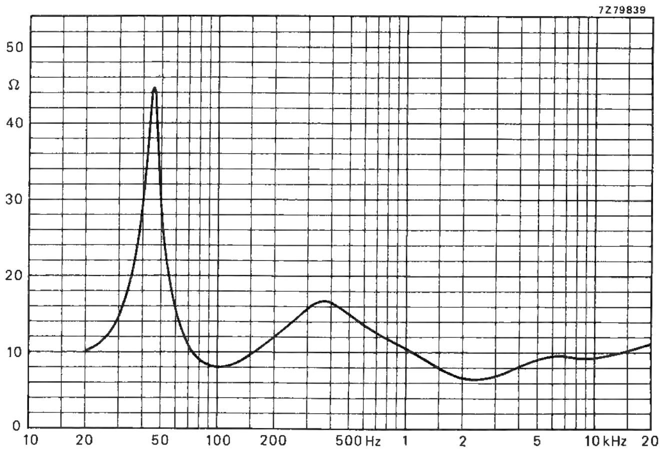
Frequency response and distortion measured in anechoic room.



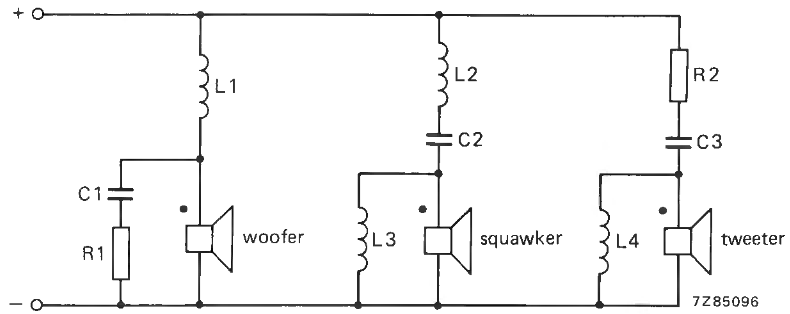
Energy response in reverberant room.



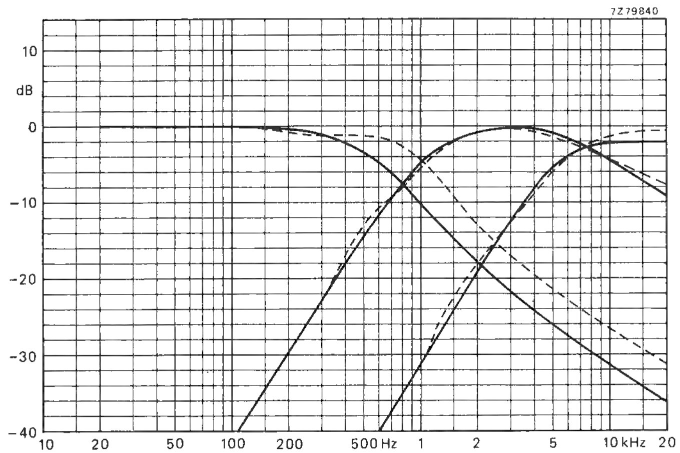
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)

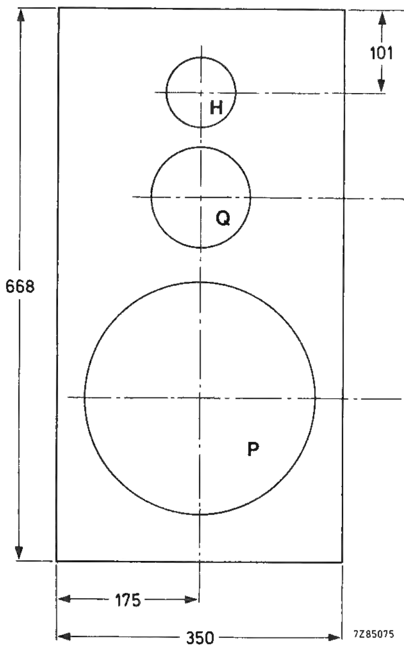


Filter characteristic. Cross-over frequencies 800 and 7000 Hz; full line — resistor-loaded, dotted line — speaker-loaded.

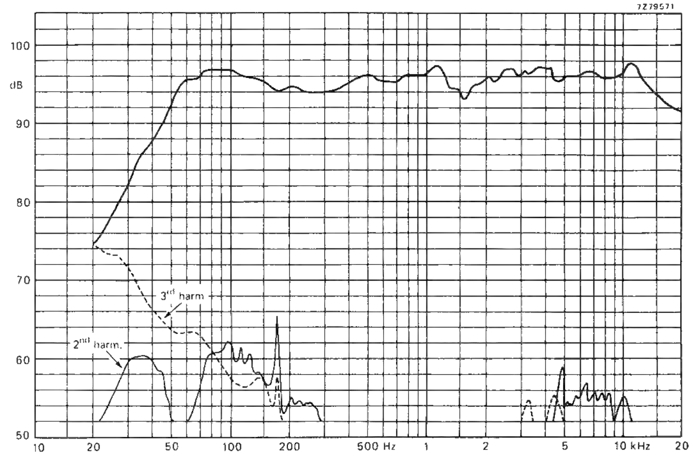
System 18 — 23 - 20 000 Hz

Woofer	AD12200/W8
Squawker	AD02160/Sq4
Tweeter	AD01605/T4

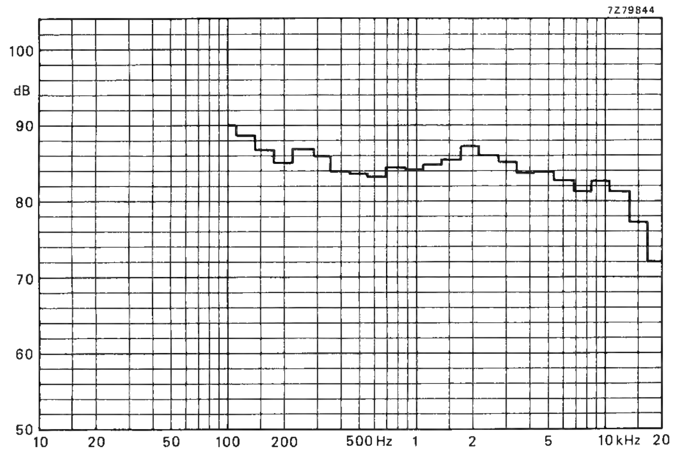
Power handling capacity	= 100 W
Rated impedance	= 4 Ω
Operating power	= 12 W
Resonance frequency	= 46 Hz
Frequency range	= 23-20 000 Hz
Enclosure volume	= 80 litres
Internal dimensions	= 668×350×342 mm
Internal depth of enclosure	= 342 mm
Material thickness	= 22 mm
Damping material	= 430 g glass wool (back and side walls)



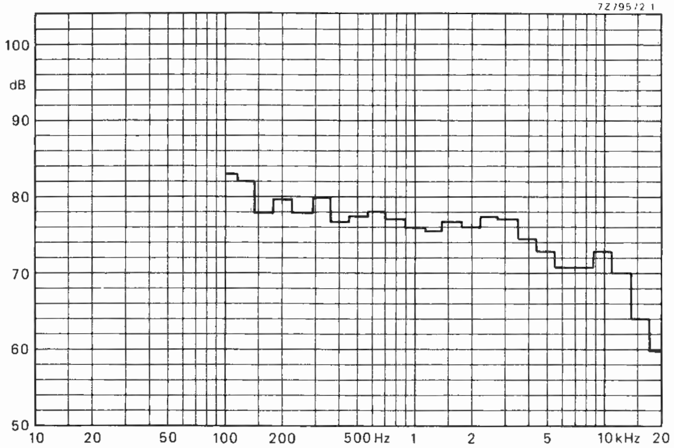
Baffle board layout
(see Table 7.3 for hole sizes)



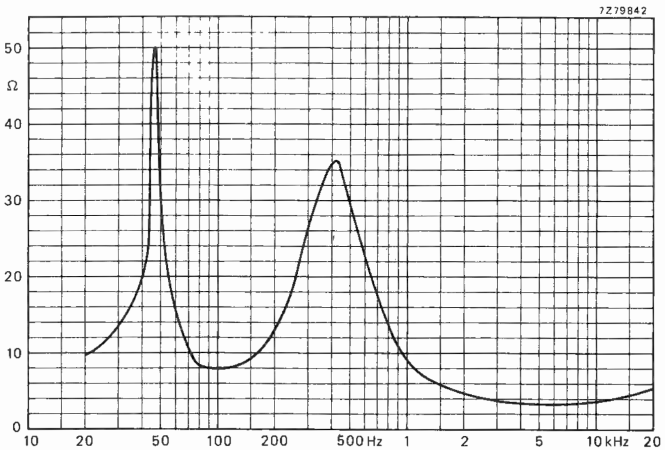
Frequency response and distortion measured in anechoic room.



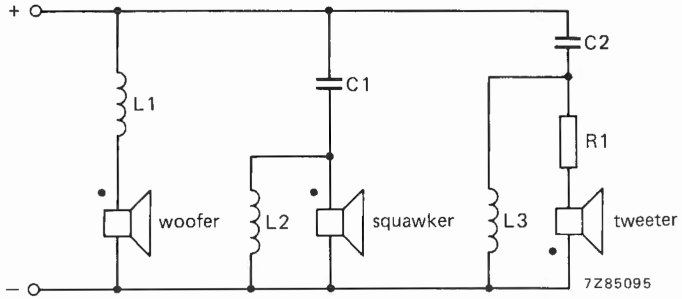
Energy response in reverberant room.



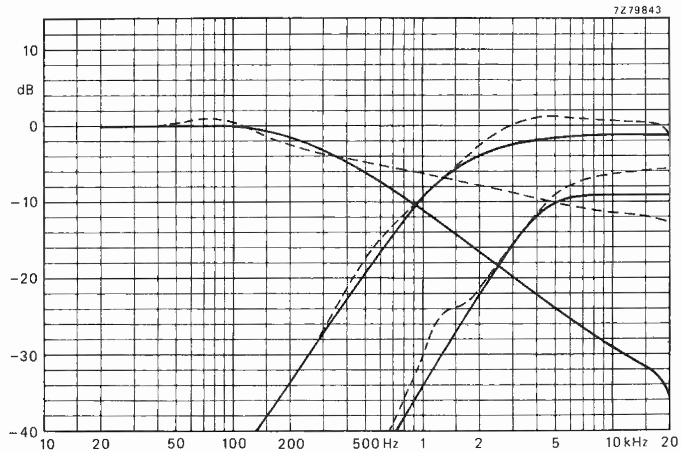
Frequency response measured in living room.



Impedance curve.



Circuit of filter (see Table 7.2)



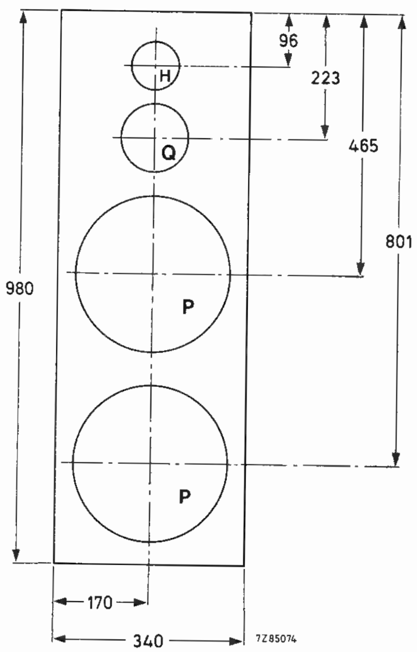
Filter characteristic. Cross-over frequencies 900 and 4000 Hz; full line — resistor-loaded, dotted line — speaker-loaded.

System 19

20 - 20 000 Hz

Woofer AD12200/W4
Squawker AD02160/Sq4
Tweeter AD01605/T4
Passive AD1200 radiator

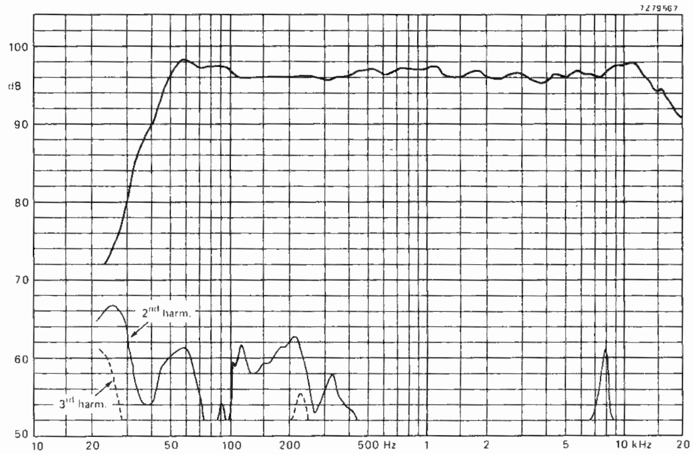
Power handling capacity	= 100 W
Rated impedance	= 4 Ω
Operating power	= 5 W
Resonance frequency	= 20 Hz/52 Hz
Frequency range	= 20-20 000 Hz
Enclosure volume	= 100 litres
Internal dimensions	= 980×340×325 mm
Internal depth of enclosure	= 325 mm
Material thickness	= 36 mm
Damping material	= 450 g glass wool (one layer on back wall)



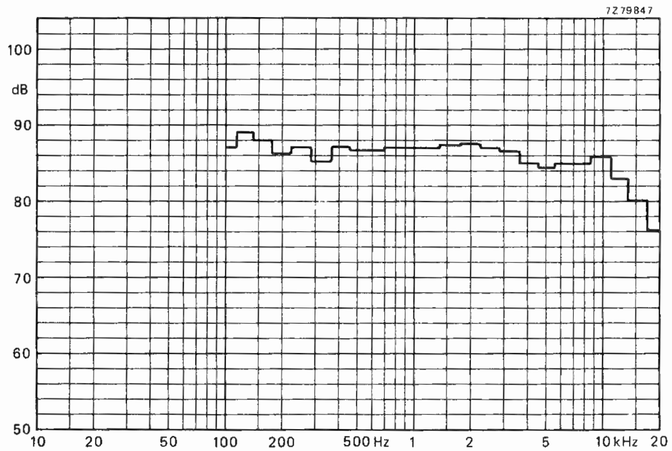
Baffle board layout
(see Table 7.3 for hole sizes)

System 19 (cont.)

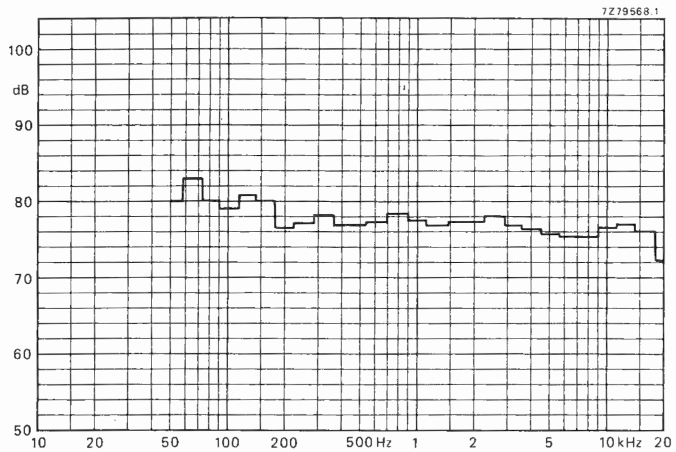
100 litres



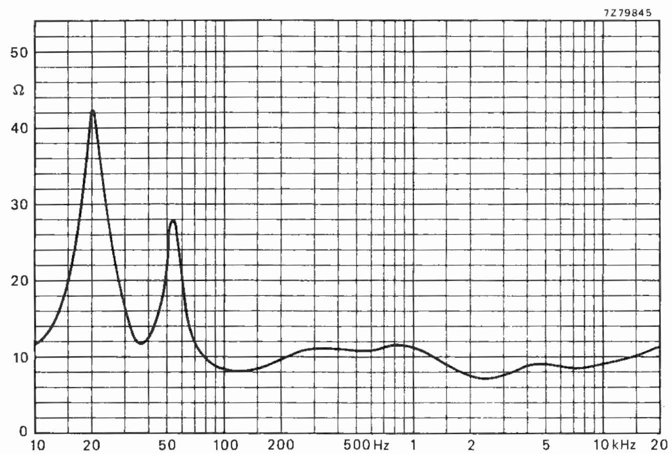
Frequency response and distortion measured in anechoic room.



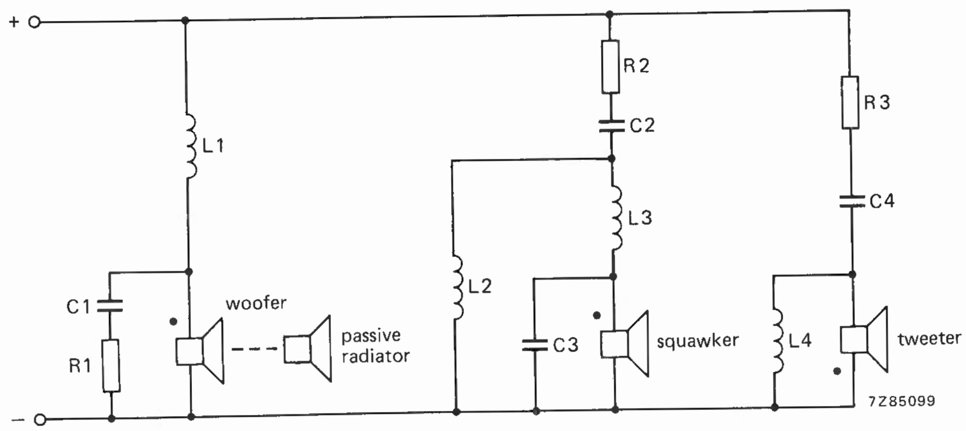
Energy response in reverberant room.



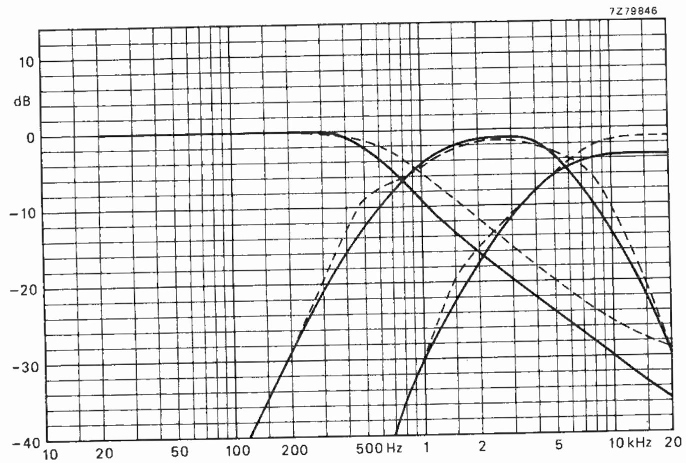
Frequency response measured in living room.



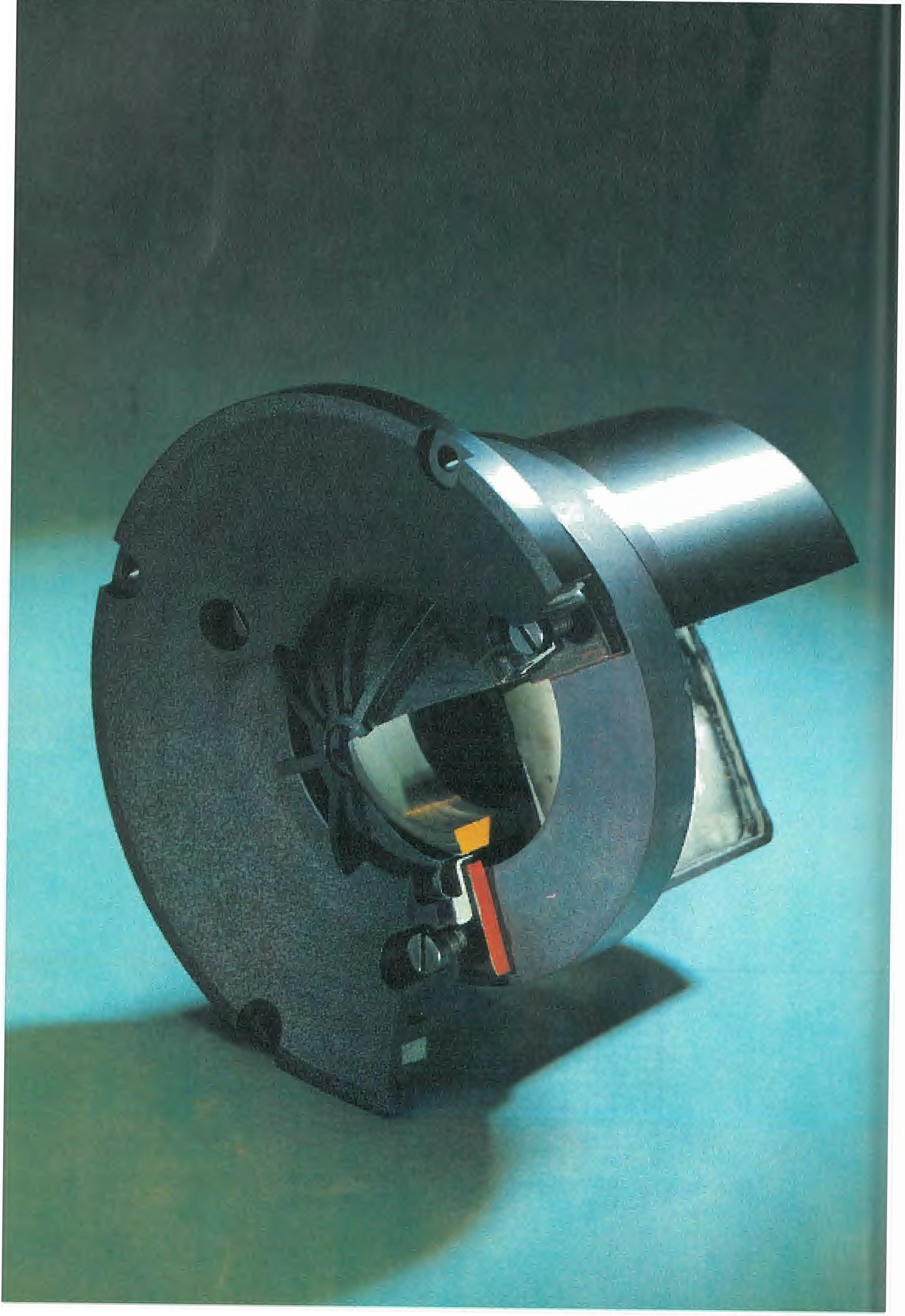
Impedance curve.



Circuit of filter (see Table 7.2)



Filter characteristic. Cross-over frequencies 800 Hz and 6000 Hz; full line — resistor loaded, dotted line — speaker-loaded.



8 Building a three-way speaker system – in pictures

The following series of pictures was taken during the construction of a three-way system. The method shown applies in general to all the systems described in this book. The baffle board assembly was made first; the enclosure last. Plywood was used for the panels, with deal for the battens and the grille cloth frame. The material used for the grille cloth was one with a suitable open weave so that the sound would not be affected. In the photographs here, you will see that a recess has been cut on the front of the baffle board around each baffle hole; when the speakers are fitted into the holes and screwed down they are flush with the surface of the baffle board. This effect can also be produced by cutting flange-size holes in a sheet of hardboard and sticking it to the front of the baffle.

Self-adhesive foamed-plastic draught-excluder tape is fitted to the rear of the speaker flanges.

The baffle board is stained black and the speakers are screwed to it. Long connecting leads can be avoided by rotating the speakers to bring their tag connections near to the cross-over filter.

Connections to the filter network may be soldered, or push-on connectors may be used. Input cable can be attached temporarily for testing the baffle board assembly at low volume.

Stapling the glass wool damping material to the inside walls.

Positioning the battens using a carpenter's square to get the back flush with the sides.

Fitting self-adhesive sealing tape to the front of the forward battens.

Close-up of applying sealing tape.

Fitting the grille cloth frame inside the front recess.

French-polishing the enclosure.

The completed sealed enclosure speaker system.

9 Technical data

9.1 Measurement of characteristics

All our loudspeakers are measured under carefully controlled conditions. In this chapter, details are given of the most important characteristics of the loudspeakers recommended in this book for use in the various system designs. To ensure that the reader clearly understands how the data is specified, the methods of measurement used are described here.

TERMS AND DEFINITIONS

Anechoic room — A room lined with acoustically absorbent material to prevent reflections and thus simulate the acoustical conditions of free space.
Unmounted — Supported in such a way that the support does not influence the radiation characteristics of the loudspeaker. This normally takes the form of a slender clamp around the magnet.
Baffle (board) — A panel for mounting a loudspeaker.
Enclosure — A box one side of which is a baffle board. In the case of measurements where an enclosure is specified, the enclosure is always a *sealed* enclosure.
Mounting — Supporting a loudspeaker in a hole in a baffle board by means of its flange so that the flange is on the front of the baffle, or flush with it.
Operating power — The sine-wave power input to a loudspeaker which produces a sound pressure level of 96 dB with respect to 2×10^{-5} N/m^2 at a microphone distance of 1 m. The sound pressure level is the average over the loudspeaker frequency range.

ATMOSPHERIC CONDITIONS

The atmospheric conditions under which the measurements are carried out are:

temperature	15 to 35 °C
relative humidity	45 to 75%
pressure	860 to 1060 mbar.

METHODS OF MEASUREMENT

Impedance — Impedance is measured at the lowest value which occurs just above the bass resonance frequency of the loudspeaker. This is determined by measuring the voltage across a 1 Ω resistor connected in series with the loudspeaker when the latter is fed with a signal from a low impedance constant voltage source. The signal is obtained from a linear amplifier, with an output impedance not greater than one-third of the rated impedance of the loudspeaker and an output power of about one-tenth the power handling capacity of the speaker, driven by a signal generator variable over the frequency range 0 to 20 000 Hz and delivering a constant voltage output. An electronic voltmeter is used to measure the voltage across the 1 Ω resistor.

Voice coil resistance — This is measured directly across the loudspeaker with a d.c. ohm-meter.

Resonance frequency — This is the frequency at which the impedance first rises to a maximum as the test signal frequency is increased from zero using the test set-up described above for measuring impedance. This measurement is carried out after applying to the loudspeaker for five seconds a signal equal to that used to test the power handling capacity.

Power handling capacity — The power handling capacity is the nominal power which the loudspeaker will satisfactorily handle during an accelerated life test. It is determined by feeding a white noise signal to the speaker for 100 hours continuously at a voltage level corresponding to the specified power handling capacity. Such a signal is obtained from a linear power amplifier, having an output impedance not greater than one-third of the rated impedance of the loudspeaker, fed from a white noise signal generator via a filter which gives a frequency distribution as described in Chapter 5. The loudspeaker has to function properly at the end of the test period. Woofers are measured in enclosures of specified volume: squawkers and tweeters are measured unmounted via specified filter networks to restrict the frequency content of the test signal to that of their operating frequency range. The test voltage in this case is measured at the input terminals of the network.

Frequency response — The frequency response is obtained by measuring the sound pressure that is produced by a loudspeaker when it is fed with a constant voltage sine-wave signal as the frequency is swept from zero to 20 000 Hz. Bruel & Kjaer measuring apparatus is employed and the test voltage $V = \sqrt{(WZ_r)}$, where W is the power required in watts and Z_r is the rated impedance in ohms. See *Response curves* below for details of set-up.

Non-linear distortion — Second and third harmonic distortion are measured separately by carrying out frequency response measurements on the loudspeakers while filtering out the fundamental frequency content during recording. To express the distortion as a percentage of the total sound pressure produced, find the difference in dB between the total signal and the distortion at the frequency required, and convert this to a percentage using the graph below.

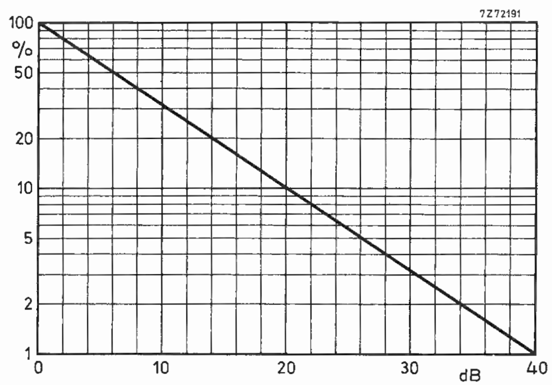


Fig. 9.1 Difference in dB converted into % distortion.

Response curves —

Curve *a* — Frequency response curve showing total sound pressure measured in anechoic room with test microphone on axis of loudspeaker at a distance of 50 cm with 50 mW input. Loudspeaker unmounted.

Curve *b* — Frequency response curve showing total sound pressure measured in anechoic room with test microphone on axis of loudspeaker at a distance of 1 m with the specified operating power applied. For the speakers described in this book, woofers are mounted in an 80 litre enclosure having a baffle board measuring 640 mm × 540 mm and filled with 1 kg of glass wool: squawkers and tweeters are mounted on a baffle board measuring 50 cm × 50 cm.

Curve c_{d2} — Second harmonic distortion content of the total sound output. Conditions are as for curve *b*.

Curve c_{d3} — Third harmonic distortion content of the total sound output. Conditions are as for curve *b*.

DIRECTION OF MAGNETIZATION

The magnet is so magnetized that the centre pole is *south* for systems with a ring magnet, and *north* for systems with a slug magnet.

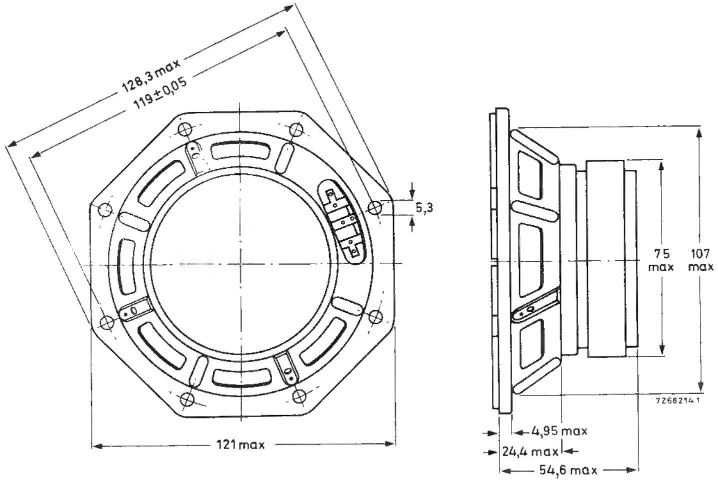
POLARITY

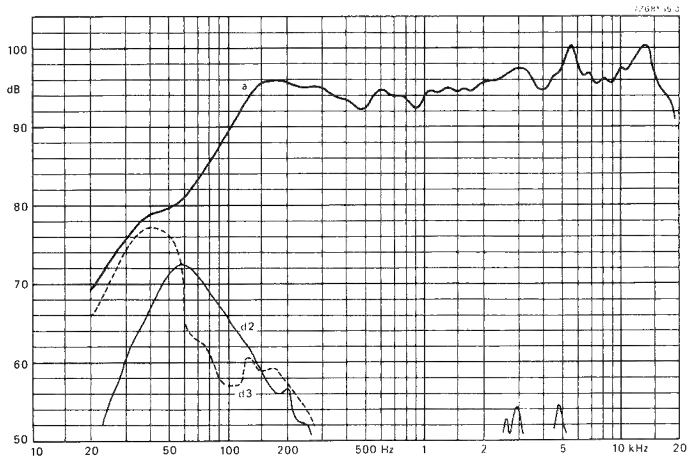
The cone of the loudspeaker will move outward when a d.c. voltage is applied to the speaker terminals so that the terminal marked red is positive.

9.2 Loudspeaker data

In the outline drawings all dimensions are in millimetres: dimensions marked [1]) indicate clearance required at rated power.

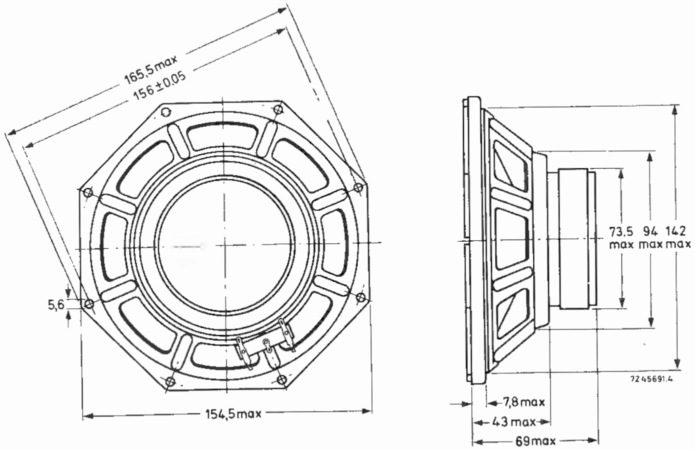
	M4	M8	
Rated impedance	4	8	Ω
Voice coil resistance	3,2	7	Ω
Resonance frequency	85		Hz
Power handling capacity, mounted in 7 litre sealed enclosure	15		W
Operating power	2	3	W
Weight	0,665		kg

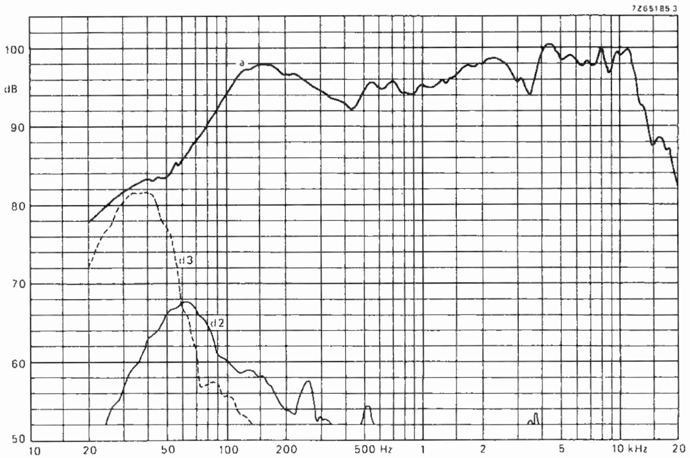




7 inch high power full range

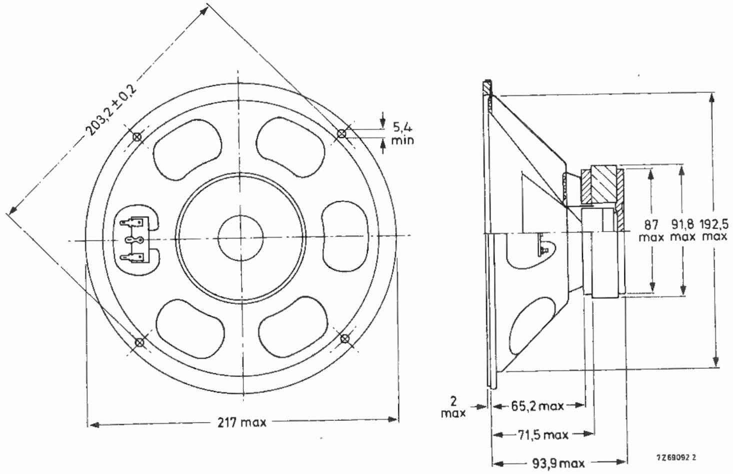
	M4	M8	
Rated impedance	4	8	Ω
Voice coil resistance	3,2	7	Ω
Resonance frequency	60		Hz
Power handling capacity, measured without filter, unmounted	10		W
Operating power	2,2		W
Weight	0,745		kg

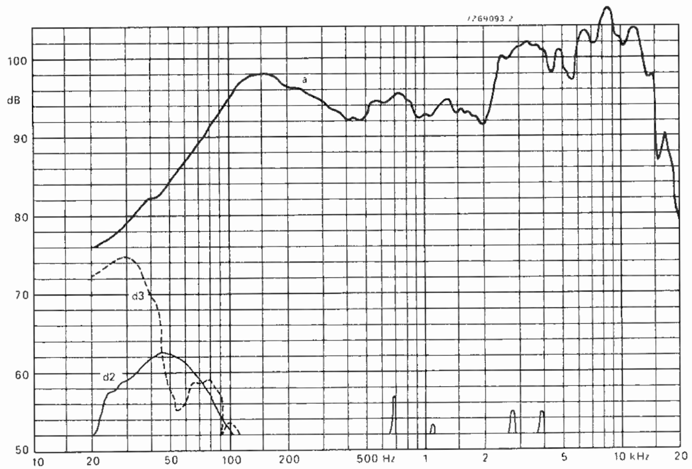




8½ inch high power full range

Rated impedance	8	Ω
Voice coil resistance	6,4	Ω
Resonance frequency	50	Hz
Power handling capacity		
mounted in sealed enclosure <30 litres	20	W
mounted in sealed enclosure >30 litres	10	W
Operating power	1,3	W
Weight	1,32	kg

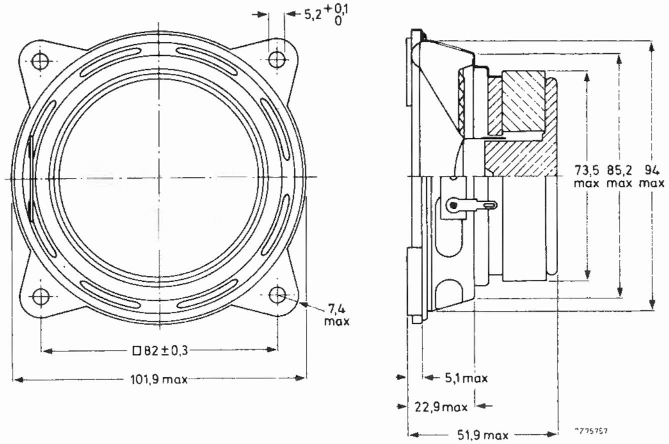


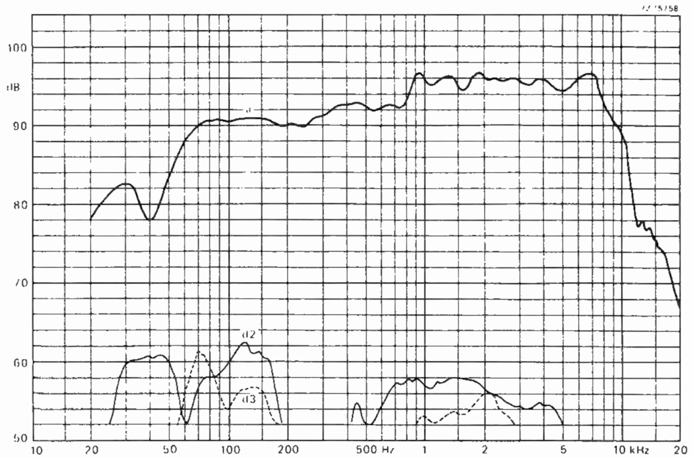


AD4060/W

4 inch high power woofer

	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,8	6,7	Ω
Resonance frequency	60		Hz
Power handling capacity			
mounted in 2 litre sealed enclosure	30		W
mounted in 5 litre bass reflex	15		W
Operating power	12		W
Weight	0,62		kg

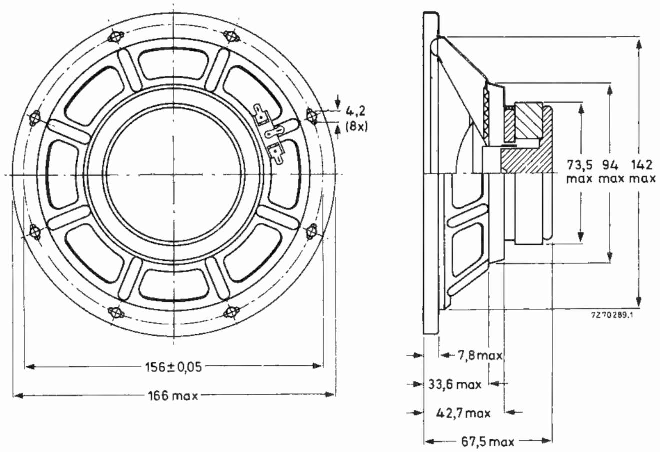


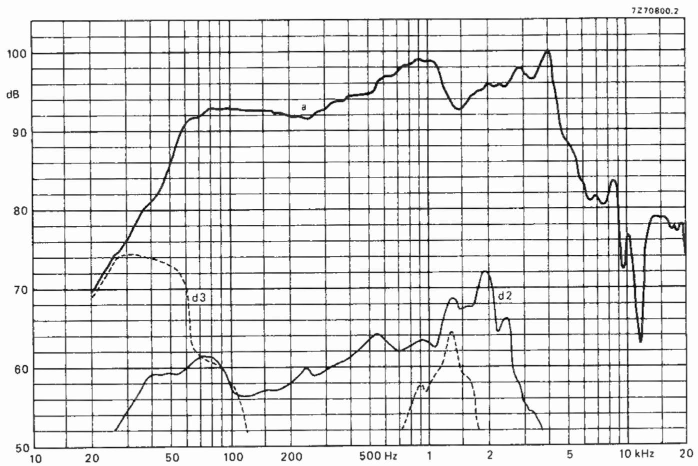


AD70601/W

7 inch high power woofer

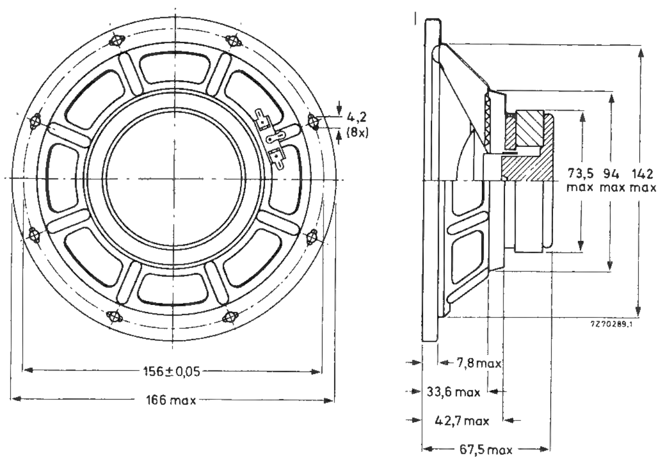
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,8	7,5	Ω
Resonance frequency	41		Hz
Power handling capacity mounted in 15 litre sealed enclosure	30		W
Operating power	12,5		W
Weight	0,68		kg

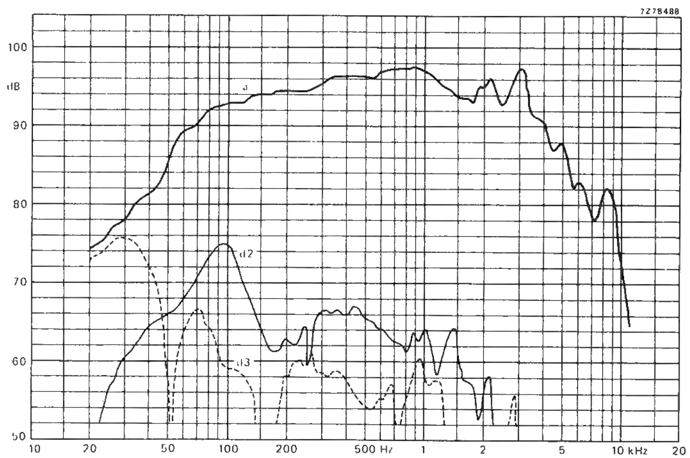




AD70610/W — 7 inch high power woofer

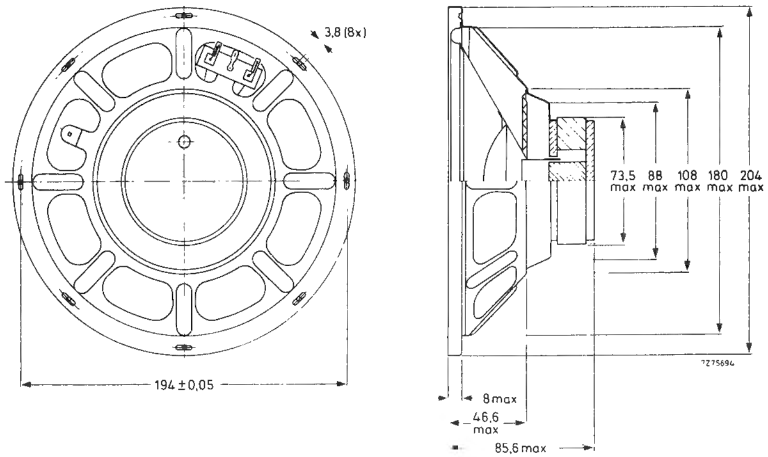
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	4	8,5	Ω
Resonance frequency	45		Hz
Power handling capacity mounted in 15 litre sealed enclosure	30		W
Operating power	12		W
Weight	0,67		kg

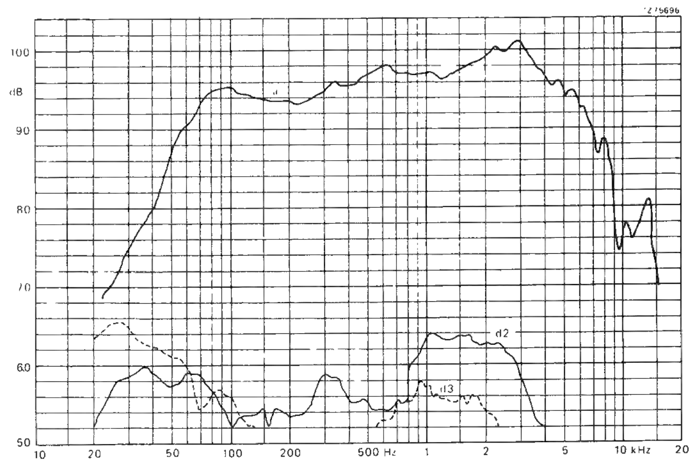




AD80602/W — 8 inch high power woofer

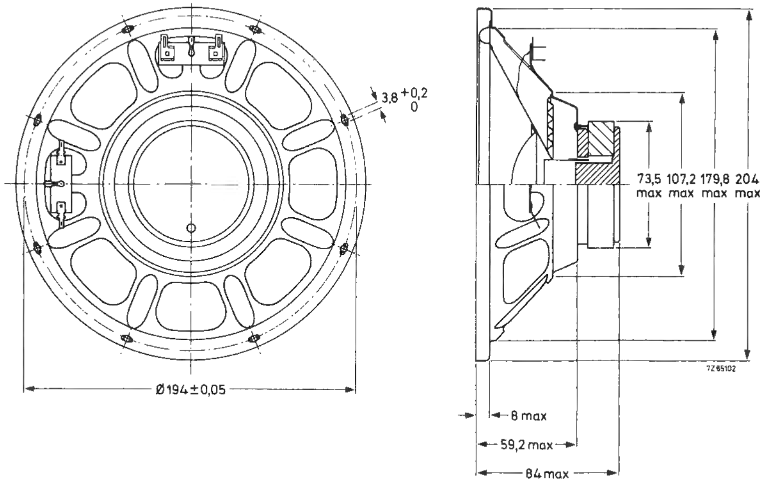
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,8	7,5	Ω
Resonance frequency	42		Hz
Power handling capacity mounted in 25 litre sealed enclosure	50		W
Operating power	5		W
Weight	0,8		kg

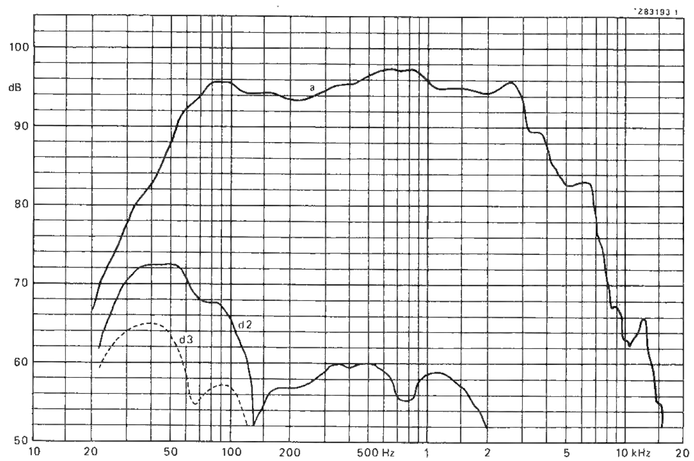




AD80603/W — 8 inch high power woofer

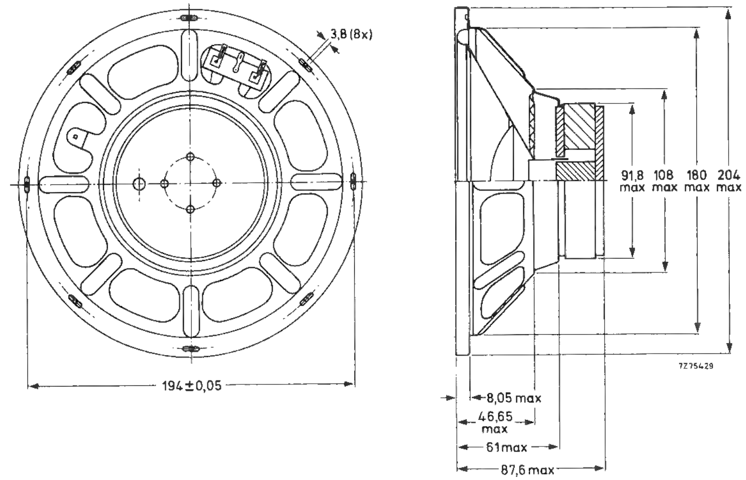
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,9	Ω
Resonance frequency	36	38	Hz
Power handling capacity mounted in 80 litre sealed enclosure	50		W
Operating power	6		W
Weight	0,77		kg

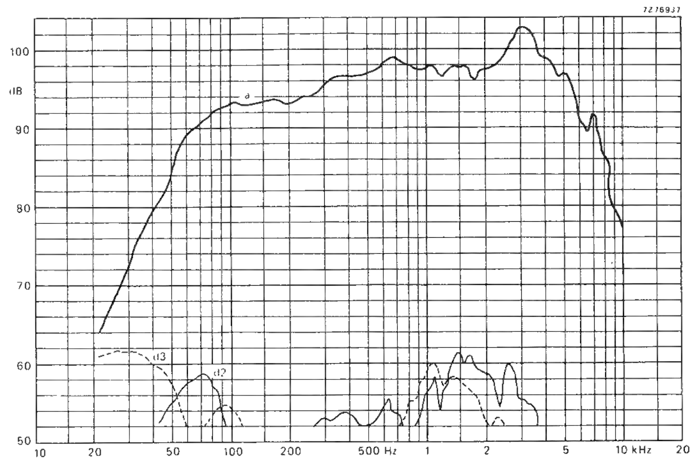




AD80652/W — 8 inch high power woofer

	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,8	7	Ω
Resonance frequency	39		Hz
Power handling capacity mounted in 25 litre sealed enclosure	50		W
Operating power	3,8		W
Weight	1,15		kg

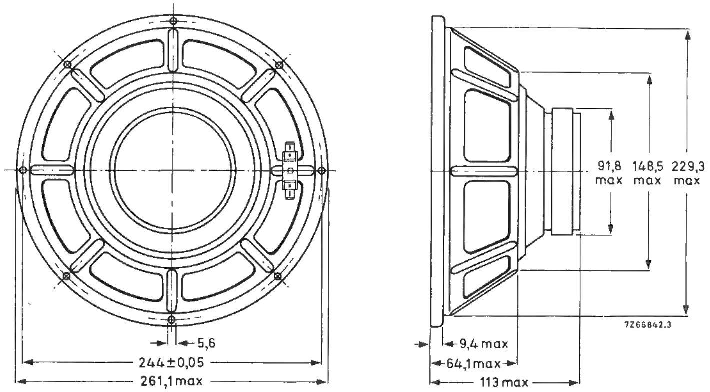


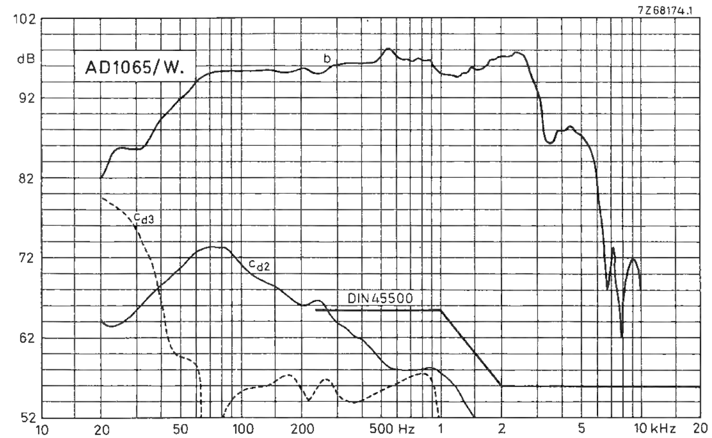


AD10650/W

10 inch high power woofer

	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,2	6,8	Ω
Resonance frequency	20		Hz
Power handling capacity, mounted in 35 litre sealed enclosure	50		W
Operating power	5		W
Weight	1,8		kg

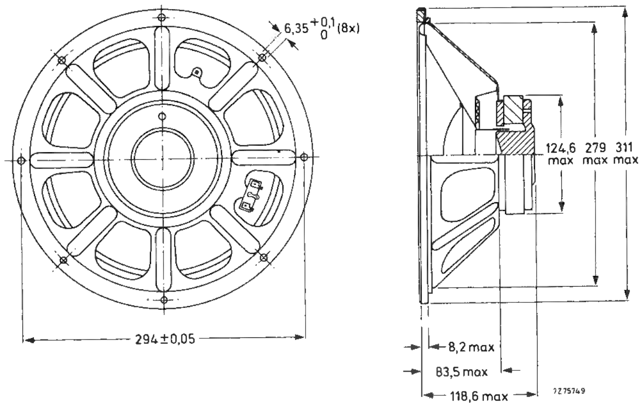


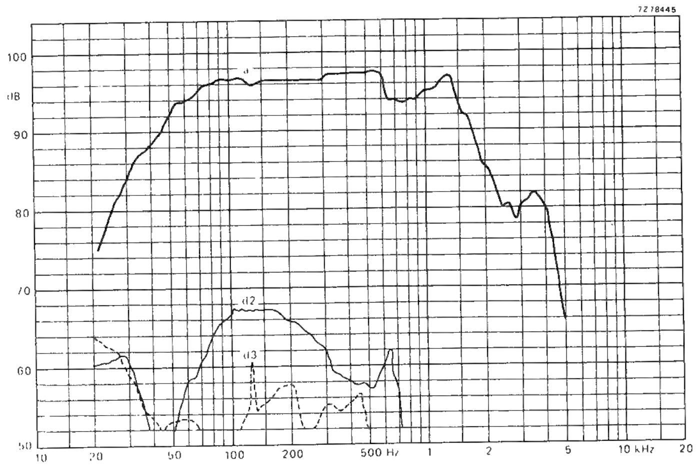


AD12200/W

12 inch high power woofer

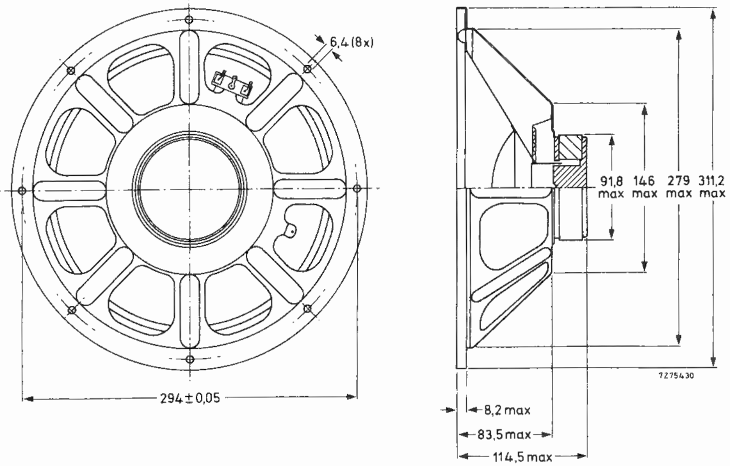
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3,3	6,7	Ω
Resonance frequency	22	24	Hz
Power handling capacity mounted in 80 litre sealed enclosure	80		W
Operating power	5		W
Weight	3		kg

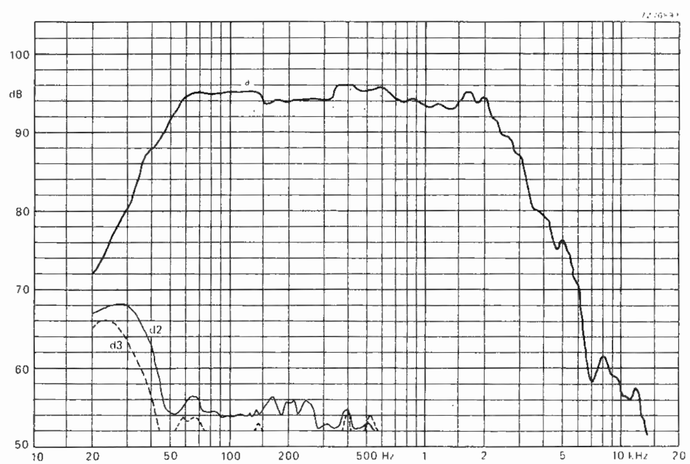




AD12650/W — 12 inch high power woofer

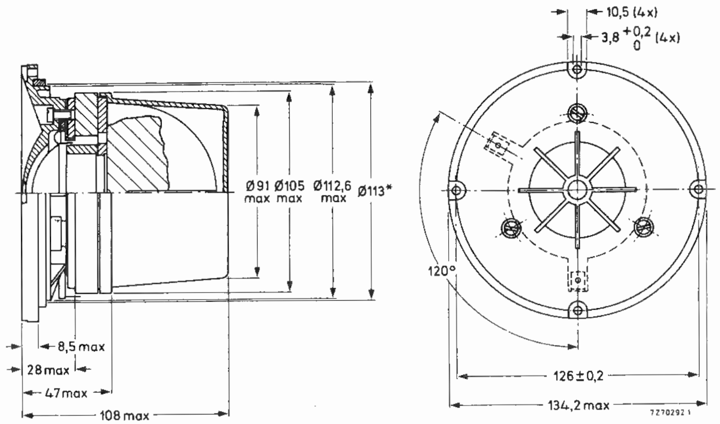
	W4	W8	
Rated impedance	4	8	Ω
Voice coil resistance	3	5,9	Ω
Resonance frequency	25	26	Hz
Power handling capacity mounted in 80 litre sealed enclosure	60		W
Operating power	5	4	W
Weight	1,8		kg

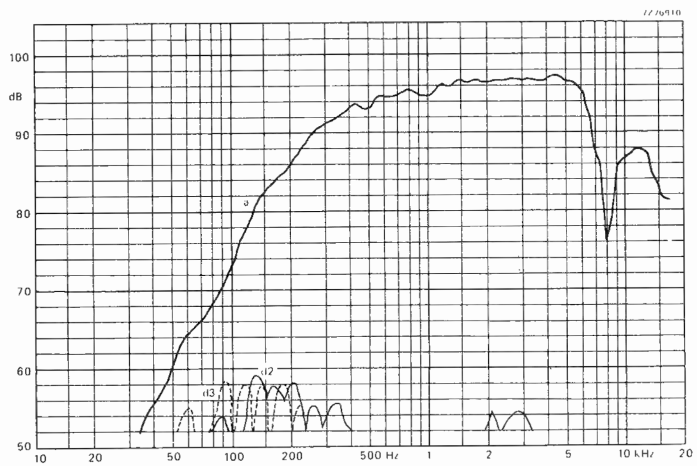




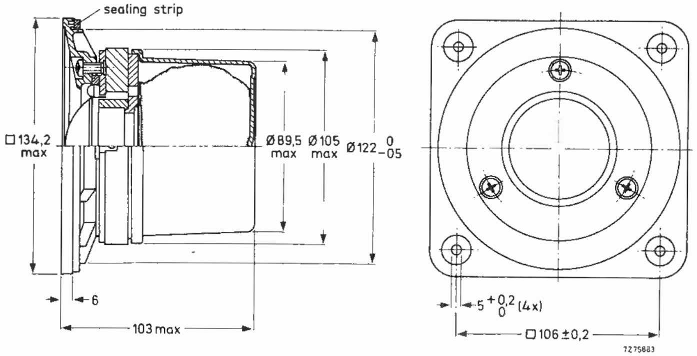
AD0211/Sq — 2 inch high power dome squawker

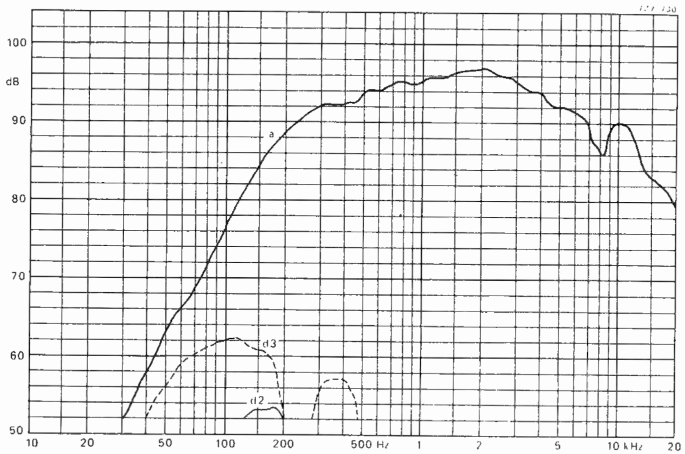
	Sq4	Sq8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,6	Ω
Resonance frequency	340	370	Hz
Power handling capacity, unmounted measured with filter 50 μF - 1,6 mH	60		W
24 μF - 3,2 mH		60	W
Operating power	5		W
Weight	1		kg



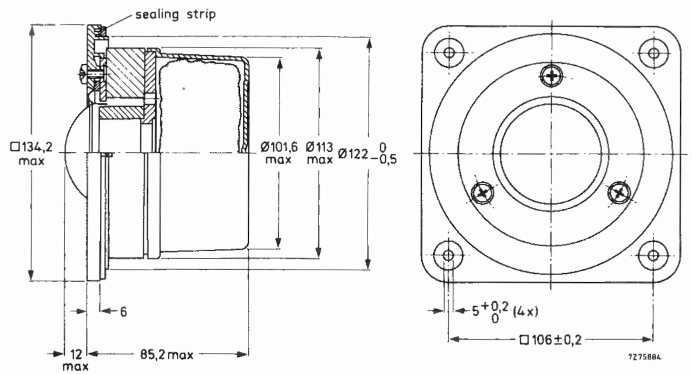


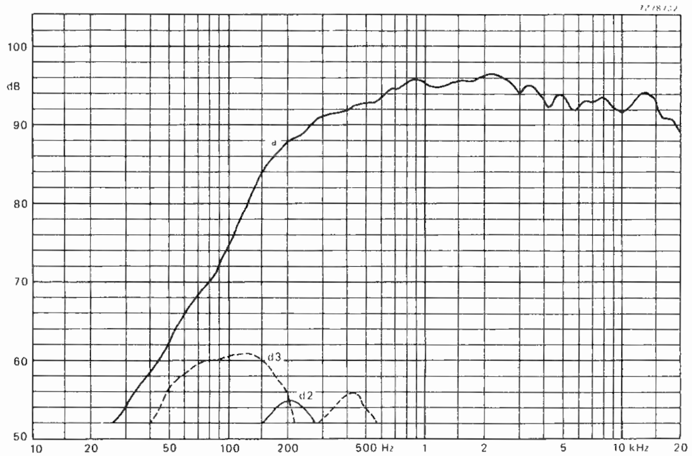
	Sq4	Sq8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,9	Ω
Resonance frequency	340		Hz
Power handling capacity, on IEC baffle measured with filter: 36 μF, 1,2 mH	30		W
18 μF, 2,4 mH		30	W
Operating power	5		W
Weight	1		kg



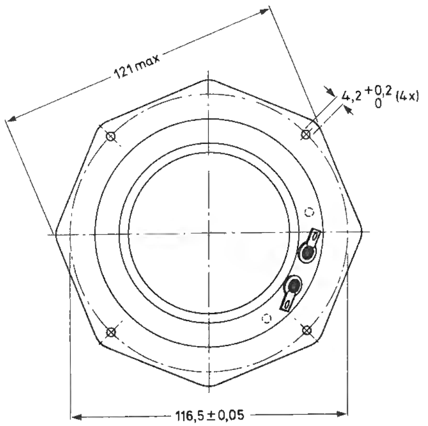


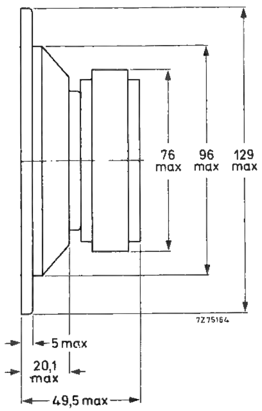
	Sq4	Sq8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,9	Ω
Resonance frequency	320		Hz
Power handling capacity, on IEC baffle measured with filter: 36 μF, 1,2 mH	30		W
18 μF, 2,4 mH		30	W
Operating power	5		W
Weight	1,5		kg

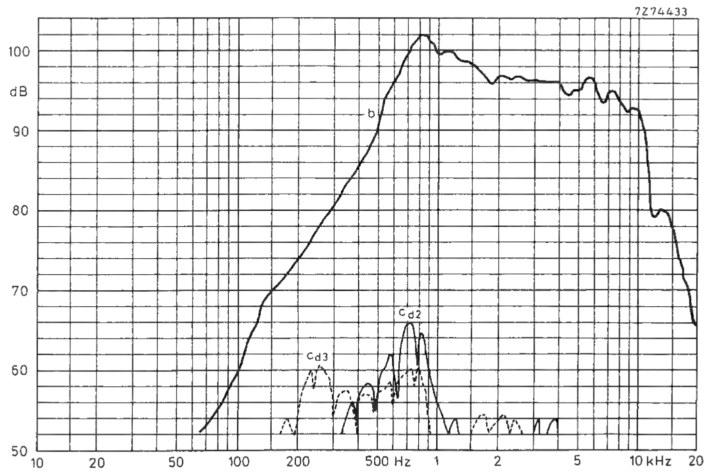




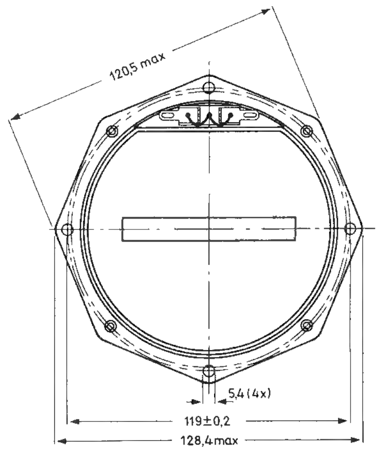
	Sq4	Sq8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	7	Ω
Resonance frequency	680		Hz
Power handling capacity, unmounted			
measured with filter 24 μF - 0,4 mH	10		W
12 μF - 0,8 mH		10	W
Operating power	2		W
Weight	0,8		kg

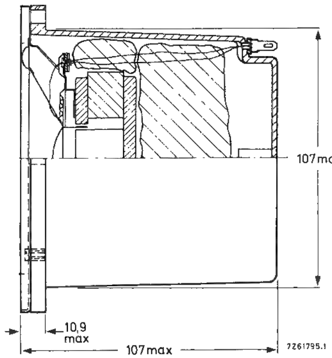


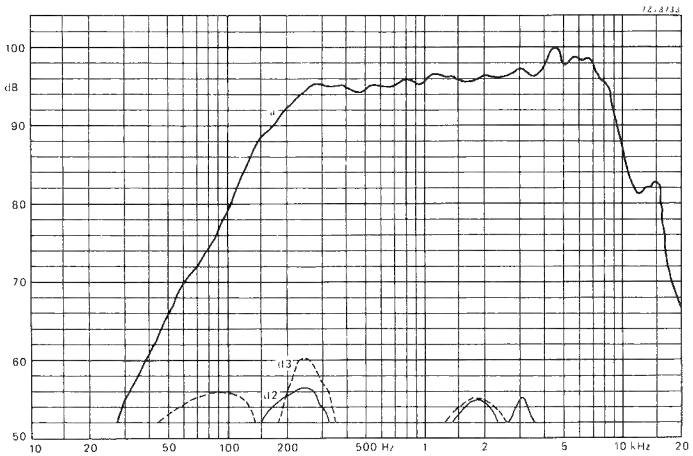




	Sq4	Sq8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,4	Ω
Resonance frequency	220		Hz
Power handling capacity, unmounted			
measured with filter: 72 μF, 2,1 mH	50		W
36 μF, 4,5 mH		50	W
Operating power	4		W
Weight	0,8		kg

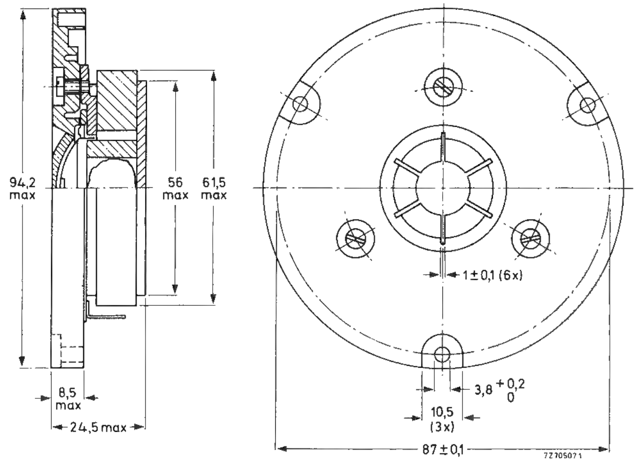


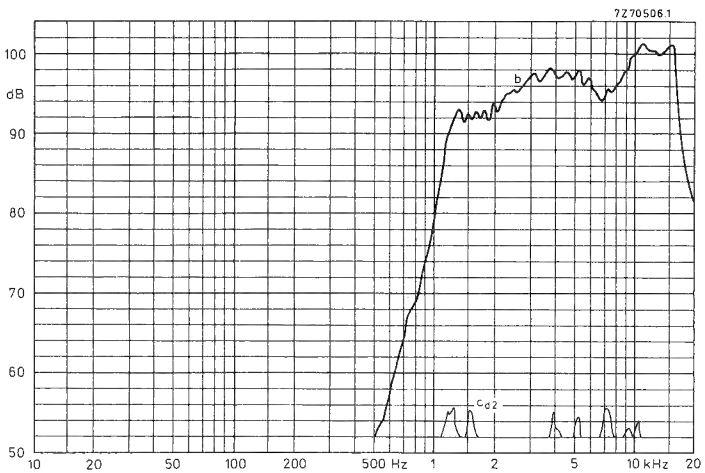




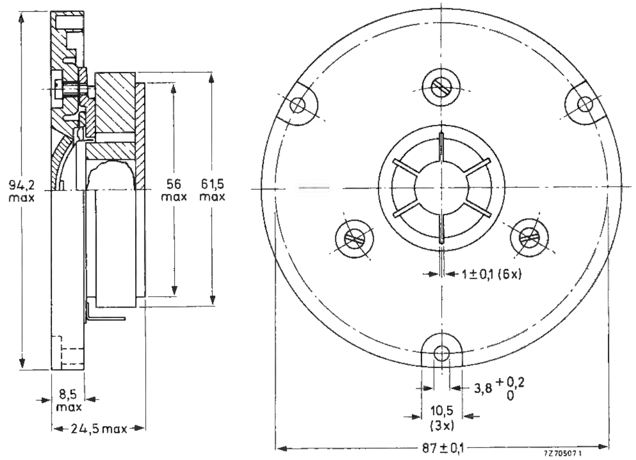
	T4	T8	
Rated impedance	4	8	Ω
Voice coil resistance	3,4	6,3	Ω
Resonance frequency	1200		Hz
Power handling capacity, unmounted			
measured with filter: 12 μF - 0,35 mH	20		W
at 2000 Hz 8 μF - 0,5 mH		20	W
measured with filter: 5 μF - 0,2 mH	40		W
at 4000 Hz 3,2 μF - 0,35 mH		40	W
Operating power	4		W
Weight	0,25		kg

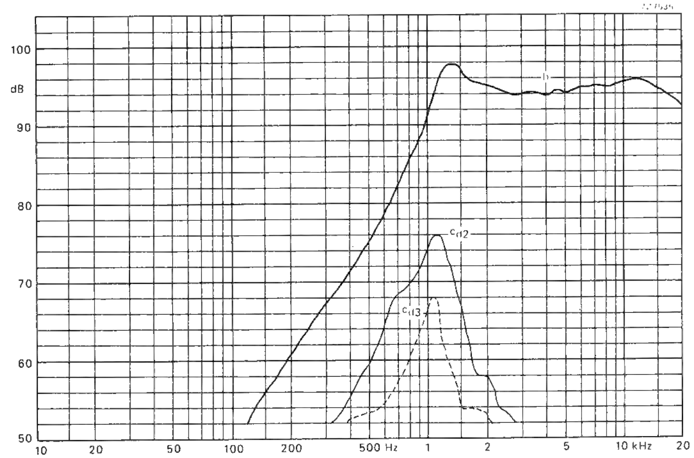
The loudspeaker has a polycarbonate dome and an aluminium-silver, copper clad voice coil.





		T4	T8	
Rated impedance		4	8	Ω
Voice coil resistance		3,4	6,3	Ω
Resonance frequency		1450		Hz
*Power handling capacity, unmounted				
measured with filter at 2000 Hz	12 μF, 0,35 mH	20/4		W
	8 μF, 0,5 mH		20/4	W
measured with filter at 4000 Hz	5 μF, 0,2 mH	50/6		W
	3,2 μF, 0,35 mH		50/6	W
Operating power		6		W
Weight		0,25		kg



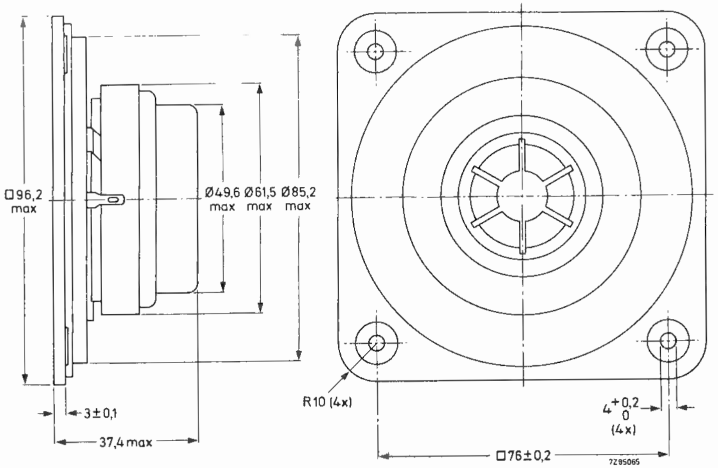


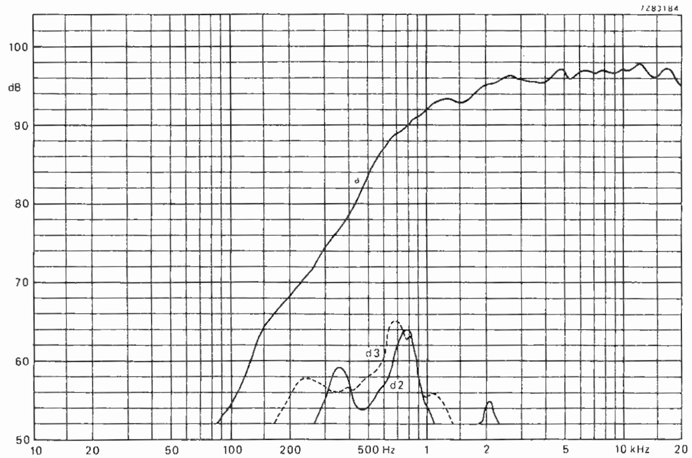
* First figure = system power, second figure = loudspeaker power handling capacity.

AD01420/T

1 inch high power tweeter

	T4	T8	T15	
Rated impedance	4	8	15	Ω
Voice coil resistance	3,4	6,3	12,5	Ω
Resonance frequency		950		Hz
*Power handling capacity, on IEC baffle		3,5		W
in series with capacitor 12 µF	20/3,5			W
6,8 µF	50/4,5	20/3,5		W
4,7 µF		50/4,5		W
3,3 µF			20/3,5	W
2,2 µF			50/4,5	W
Operating power		4		W
Weight		0,26		kg



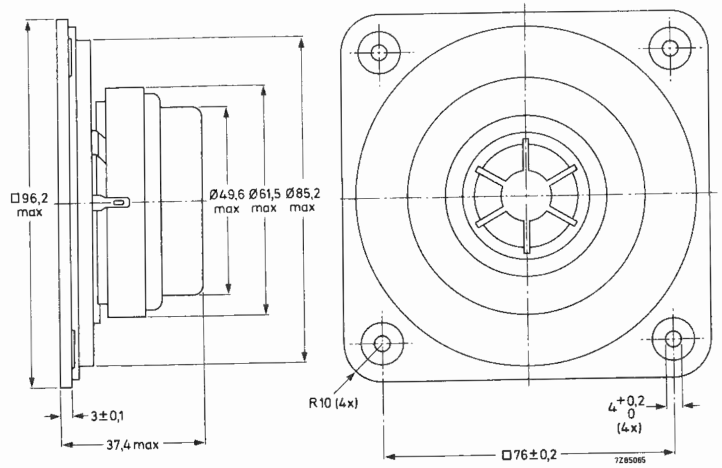


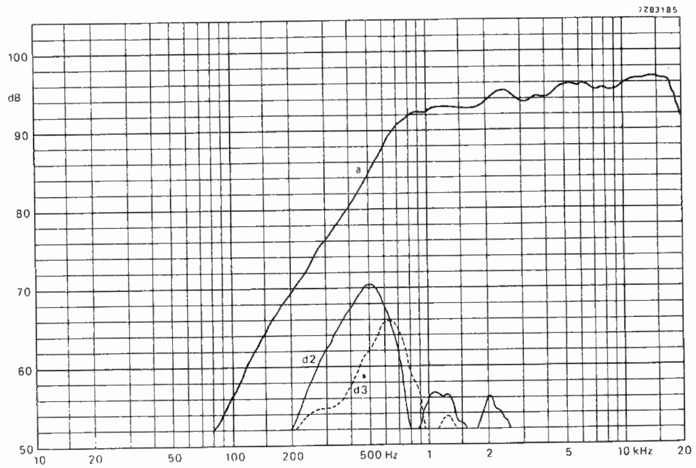
* First figure = system power, second figure = loudspeaker power handling capacity.

AD01430/T

1 inch high power dome tweeter

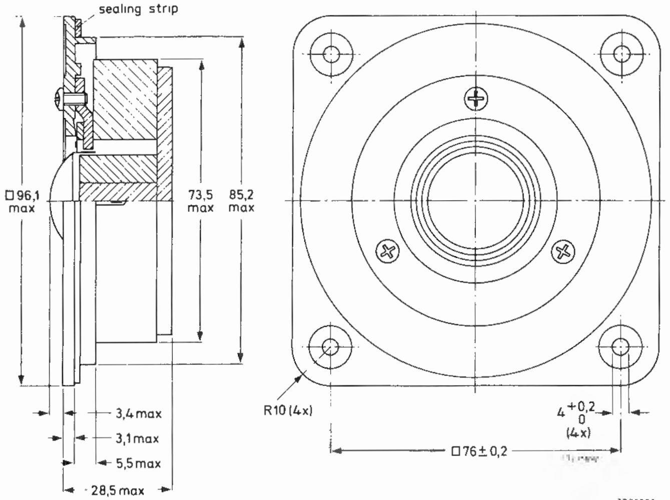
	T4	T8	T15	
Rated impedance	4	8	15	Ω
Voice coil resistance	3,4	6,3	12,5	Ω
Resonance frequency		1200		Hz
*Power handling capacity, on IEC baffle				
in series with capacitor 12 µF	20/3,5			W
6,8 µF	50/4,5	20/3,5		W
4,7 µF		50/4,5		W
3,3 µF			20/3,5	W
2,2 µF			50/4,5	W
Operating power		6		W
Weight		0,26		kg

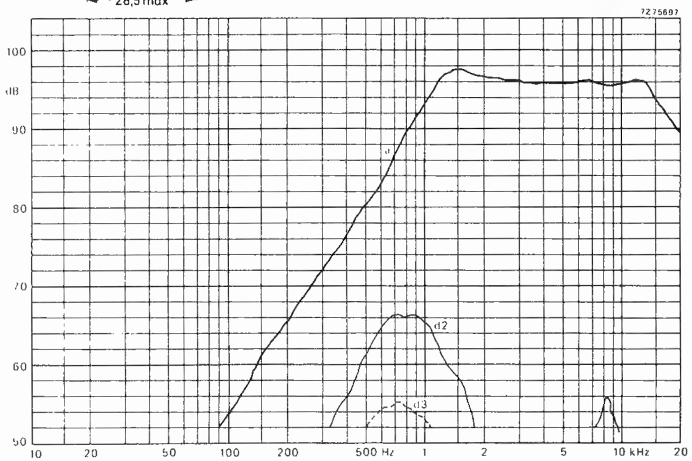




** First figure = system power, second figure = loudspeaker power handling capacity.*

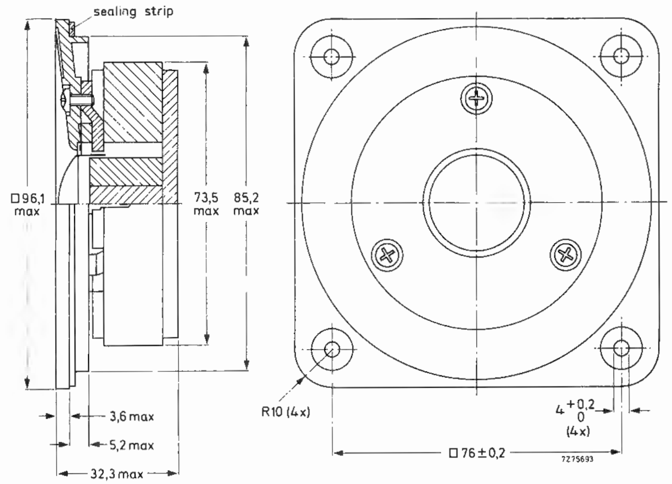
		T4	T8	T15	
Rated impedance		4	8	15	Ω
Voice coil resistance		3,4	6,3	12,5	Ω
Resonance frequency			1250		Hz
*Power handling capacity, unmounted					
measured with filter at 2000 Hz	12 μF, 0,35 mH	20/4			W
	8 μF, 0,5 mH		20/4		W
	3,3 μF, 1 mH			20/4	W
measured with filter at 4000 Hz	5 μF, 0,2 mH	50/6			W
	3,2 μF, 0,35 mH		50/6		W
	1,5 μF, 0,8 mH			50/6	W
Operating power			5		W
Weight			0,5		kg

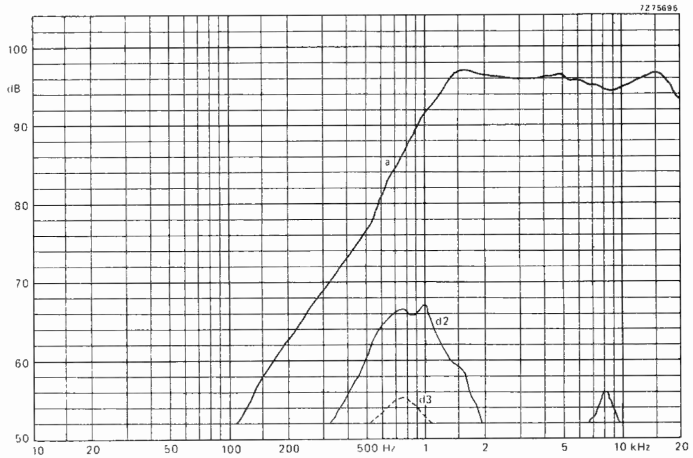




* First figure = system power, second figure = loudspeaker power handling capacity.

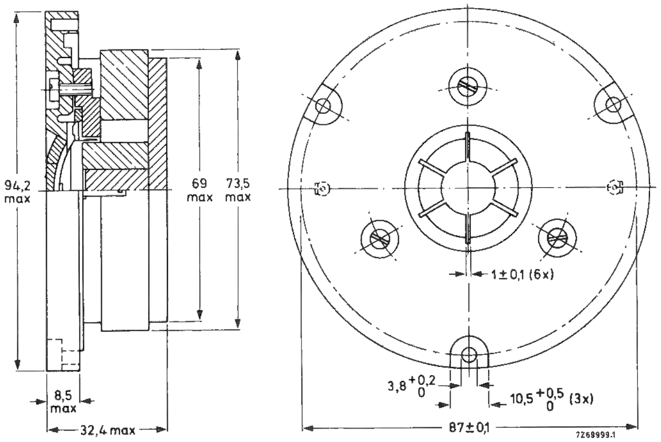
		T4	T8	T15	
Rated impedance		4	8	15	Ω
Voice coil resistance		3,4	6,3	12,5	Ω
Resonance frequency		1250	1250	1250	Hz
*Power handling capacity, unmounted					
measured with filter at 2000 Hz	12 μF, 0,35 mH	20/4			W
	8 μF, 0,5 mH		20/4		W
	3,3 μF, 1 mH			20/4	W
measured with filter at 4000 Hz	5 μF, 0,2 mH	50/6			W
	3,2 μF, 0,35 mH		50/6		W
	1,5 μF, 0,8 mH			50/6	W
Operating power		4	4	4	W
Weight		0,5	0,5	0,5	kg

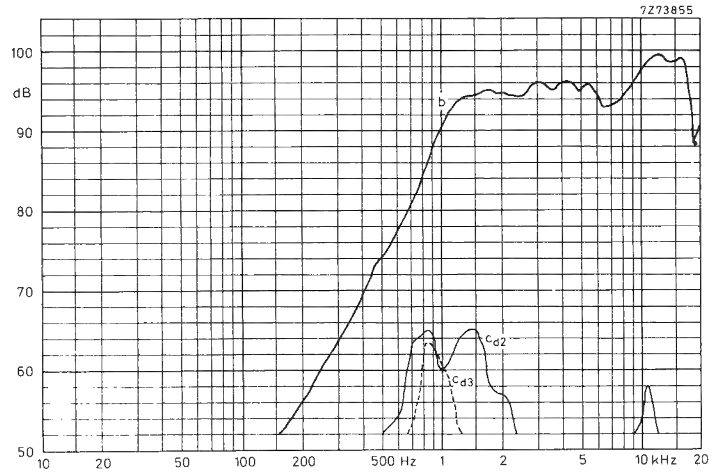




* First figure = system power, second figure = loudspeaker power handling capacity.

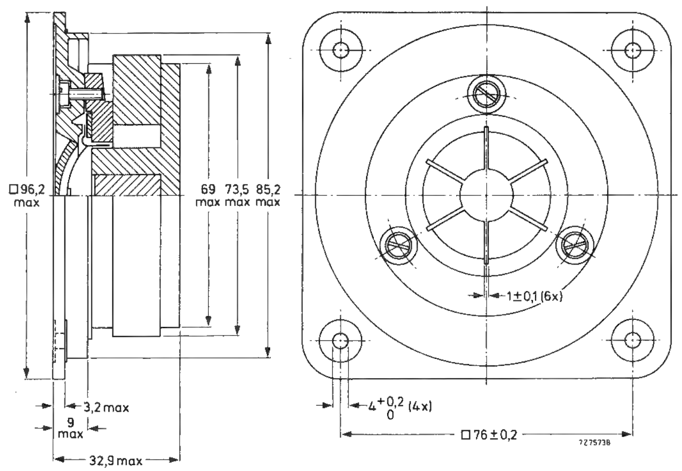
		T8	T15	
Rated impedance		8	15	Ω
Voice coil resistance		6,3	12,5	Ω
Resonance frequency		1000		Hz
*Power handling capacity, unmounted				
measured with filter	8 μF, 0,5 mH	20/4		W
at 2000 Hz	3,3 μF, 1 mH		50/6	W
measured with filter	3,2 μF, 0,35 mH	20/4		W
at 4000 Hz	1,5 μF, 0,8 mH		50/6	W
Operating power		2		W
Weight		0,5		kg

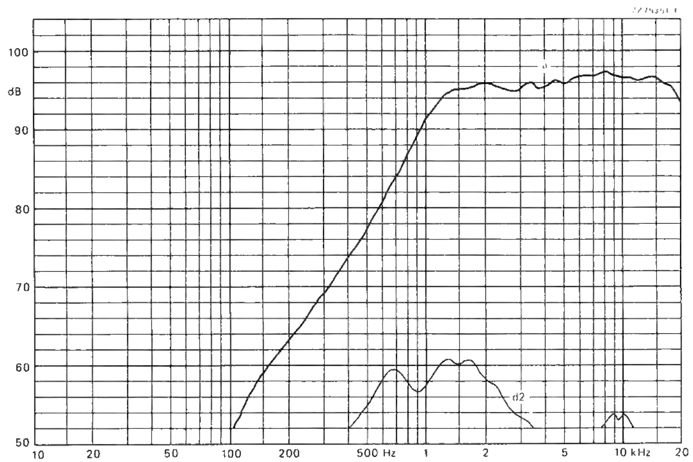




* First figure = system power, second figure = loudspeaker power handling capacity.

		T8	T15	
Rated impedance		8	15	Ω
Voice coil resistance		6,3	12,5	Ω
Resonance frequency		1300		Hz
*Power handling capacity, unmounted				
measured with filter	8 μF, 0,5 mH	20/4		W
at 2000 Hz	3,3 μF, 1 mH		20/4	W
measured with filter	3,2 μF, 0,35 mH	50/6		W
at 4000 Hz	1,5 μF, 0,8 mH		50/6	W
Operating power		3		W
Weight		0,5		kg

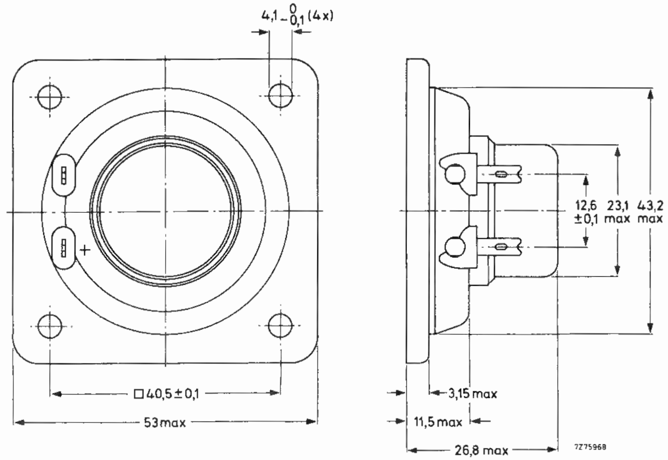


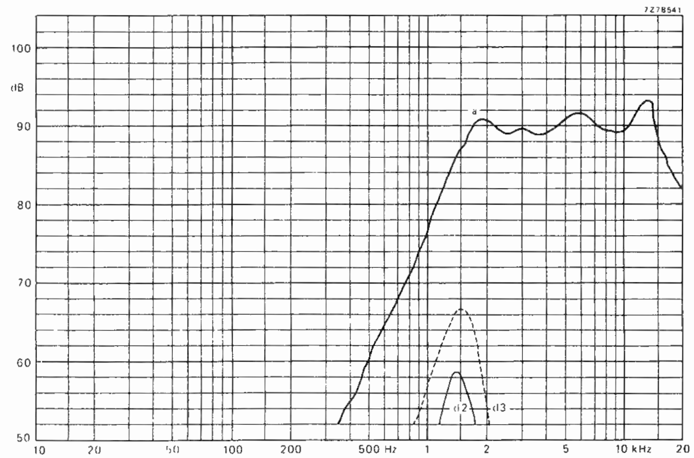


* First figure = system power, second figure = loudspeaker power handling capacity.

AD2296/T 2 inch high power tweeter

	T4	T8	T15	
Rated impedance	4	8	15	Ω
Voice coil resistance	3,4	6,6	13,5	Ω
Resonance frequency		1300		Hz
Power handling capacity		3		W
PHC system, cross-over 3000 Hz		6		W
in series with capacitor	12	5	2,7	μF
Operating power		6		W
Weight		0,05		kg





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