PRACTICAL

HI-FI SOUND

the complete guide to perfect listening in the home

Roger Driscoll

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Hi-Fi: an Introduction

The term 'hi-fi' is one which everybody knows, maybe conjuring up the thought of gleaming, elaborate control boxes with dials and knobs, and enormous loudspeaker cabinets. One might be forgiven for thinking that hi-fi is for the enthusiast, one who knows all about the technical jargon and the equipment specifications. Although the word is often used to describe audio equipment and its perfection–a pursuit of the ultimate by a knowledgeable few–to most of us hi-fi has a simpler meaning, namely the reproduction of music for our listening pleasure. It is with this aim in mind, to give the reader a little interest and pleasure, that this book has been written. Rather than delve deeply into equipment technicality, I have explained the essential points and where technical facts are necessary, I have illustrated them by familiar examples. It has been my experience as a teacher that the special details of a subject will follow easily if the basic points are properly made. We see an ever widening range of audio equipment coming onto the market, as time goes on, and technical refinements are made and featured in new models. I hope that the following chapters will help remove the confusion and allow readers to make a better choice for themselves when buying new equipment, perhaps for the first time.

There will be something to interest every reader of the book, the newcomer, the musical person for whom hi-fi is a means to an end, and the audio enthusiast too. In Chapter 1, after taking a look at the history of sound reproduction, we find how the ear responds to sound, and the advantage a stereo sound system provides by letting us discriminate and locate different sounds. The interesting properties of musical instruments and the sounds they make are the subject of Chapter 2, whilst Chapter 3 explains more of home audio equipment and gives advice on its essential elements, the pick-up and arm, records and tapes. There are recommendations to be found here of complete audio systems, in every price range, which will go together into a good hi-fi system. The classic and ever popular subject of loudspeakers follows in Chapter 4, and here will be found details of a simple, high quality loudspeaker system designed and made specially for readers of this book.

The final link, the acoustics of the concert hall and listening room where we hear live and recorded music, is made in Chapter 5, and there are some useful pointers on how to plan and furnish your listening room. At the end is a music list of well-recorded examples by which you can judge your audio system while you are enjoying the music. I hope readers will indeed find interest and enjoyment on their audio journey.

HISTORY

When we put on our favourite record, we might stop to think that maybe millions of other listeners are enjoying it too and could have been doing so within a few months of it being made. How different from the way things were, back in the last century, when music could be heard only by audiences in a concert hall or at privileged social gatherings. It would take years for a new composition to filter through to just a few thousand people. The audio equipment we now have in our homes, from the simplest record player to the most elaborate hi-fi installation, is a very powerful medium for communicating music of every age and style, and because we can listen as often as we wish, our appreciation and knowledge of the art may grow even faster.

Of course, we take all this for granted but it was necessary for ideas to be tried and developed before sound recording became possible. The industry has developed greatly even in the last thirty years, indeed, the term 'hi-fi' did not exist until about 1935.

The first arrangement enabling sound waves to be recorded and observed was produced by the French physicist Leon Scott

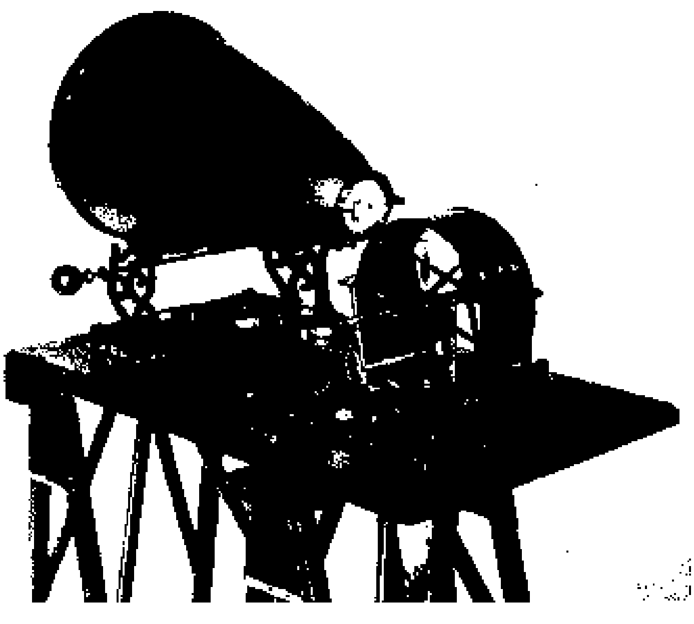


Fig. 1
Scott's Phonautograph of 1857.

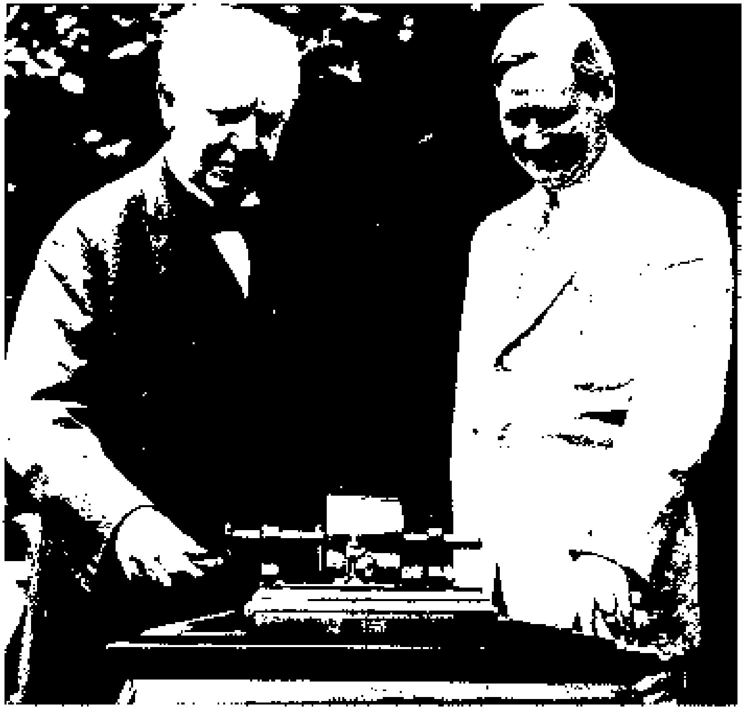


Fig. 2
Thomas Alva Edison (left) demonstrates his tinfoil phonograph in 1920.

de Martinville in 1857. It was called a phonautograph, and was made up of an acoustic horn and a piece of paper, coated with lampblack, which revolved on a cylinder driven by clockwork. Attached to the small end of the horn was a membrane and hog's bristle. When a person spoke into the horn the membrane vibrated and the hog's bristle traced a line on the black paper as the cylinder revolved. It was not possible for the delicate recording to be played back, but useful information was revealed about the path the line would take with different sound waveforms.

This idea of tracing out a groove on some smooth surface also inspired the American scientist Thomas Alva Edison when, two years later, he demonstrated a recording machine which could also 'reproduce automatically at any time the human voice perfectly'. His invention, the tinfoil phonograph, became the first commercial machine for storing and reproducing sounds. The recording was made on a tinfoil sheet wrapped around a cylinder which was manually rotated as the groove was traced out. A person would speak into a diaphragm (like that in a telephone mouthpiece) to which was fixed a sharp metal-cutting stylus, and this made the impression on the foil surface. The indentations on the metal surface were more permanent, and would stand up to being retraced. Sound reproduction was possible by use of a similar replay stylus and diaphragm, and although reproduction was harsh and grating, it was clearly audible.

At the time, many phonographs were manufactured and demonstrated. They were used for dictation recording and sold to the public in England, but interest faded because of the machine's limitations and rudimentary performance. It was well remembered by the American people, however, for they continued to use the word 'phonograph' long after the Edison machine was overtaken by the disc systems we know today.

Although there were refinements to the phonograph, it was to be another kind of reproducer which would eventually become established. Emil Berliner, a German *émigré* living in America, turned his attention to sound recording in 1887. In place of the cylinder he used a disc, supported by a moving platter or turntable. The disc, which was of zinc, was covered with an acid-resistant coating and the recording made by cutting a spiral groove into this coating with a stylus coupled to an acoustic horn. When the disc was immersed in acid, an etched record was formed, which could be played back or used to make other discs from plastic materials. This was a great advantage in terms of the commercial possibilities of the system and, as readers know, it was so easy to put on and play the disc. Its technical quality, also, was far superior to the cylinder records, because the motions of the groove were lateral, not vertical. Berliner's invention was called the Gramophone, and it was developed into a commercial product almost at once.

In 1895 Berliner established a Gramophone Company in America, and then the Deutsche Grammophon Gesellschaft (DGG) record company in Germany in 1898. It was not long before many thousands of records were being produced daily, in several countries of the world. Almost every owner of an audio system and records will have a picture of one of the first Berliner gramophones, and a little terrier named Nipper with his head cocked in concentration at the machine. This historic picture appears, of course, on all His Master's Voice records, and was painted in 1899 by Francis Barraud for The Gramophone Company in London, just after they marketed the machine.

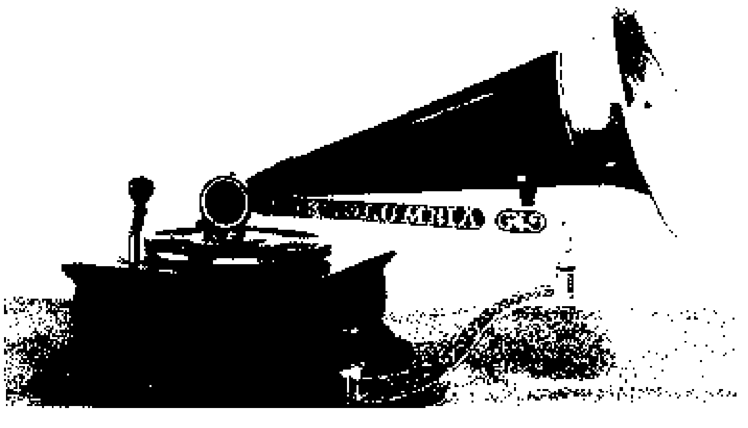


Fig. 3
The Columbia AJ Disc Gramophone of 1903.

(Later the company became Electric & Musical Industries, EMI). The picture has become the most famous trade-mark in the world.

During the first few years of the twentieth century, the gramophone became more refined. The stylus and diaphragm devices were more accurate, and were called 'sound boxes'. These had the replaceable metal stylus attached to the centre of a light, resonant membrane, which was coupled acoustically to the horn. The weight of the horn was balanced so that it did not apply too

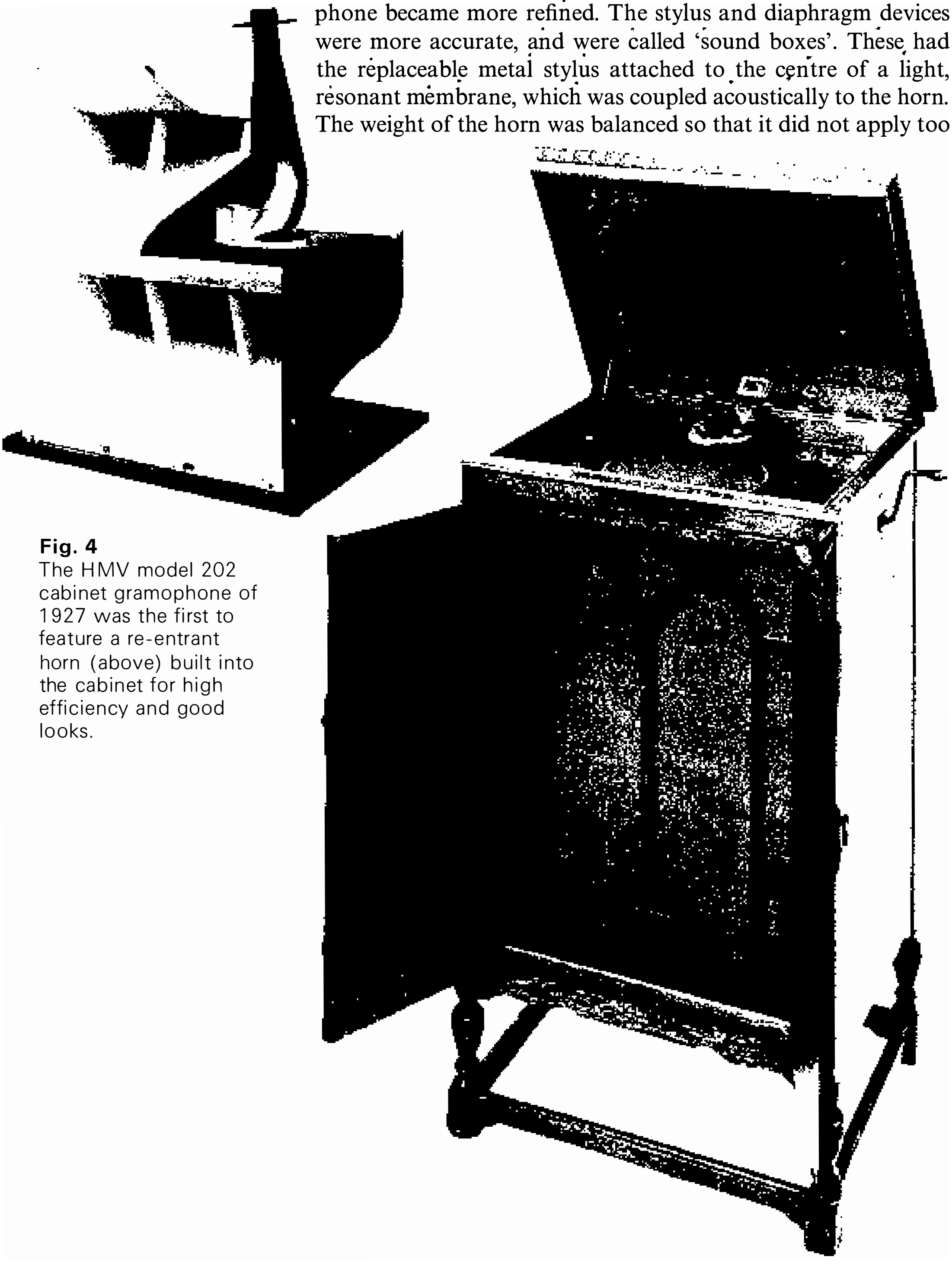


Fig. 4
The HMV model 202 cabinet gramophone of 1927 was the first to feature a re-entrant horn (above) built into the cabinet for high efficiency and good looks.

much force onto the stylus. In 1906 the Victrola was marketed, a cabinet gramophone in which the horn was made part of the internal construction of the cabinet, rather than being perched on top of it. By about 1910, it was clear that the disc record had won the battle; the record companies stopped manufacture of all cylinders as the choice of disc gramophones widened further. In the 1920s, the gramophone became the most popular form of home entertainment; in the year 1927, DGG produced more than 5 million discs, and HMV launched the first cabinet gramophone with an exponential re-entrant horn (Fig. 4).

Already problems could be foreseen in the further development of the gramophone, however. All the gramophones of the time were purely mechanical devices, requiring cumbersome acoustic horns for good efficiency. Important developments were taking place by then in the electronics and communications fields. Lee de Forest in America had invented the triode valve, which allowed signals to be amplified by electronic means. Advances in technology had produced microphones, delicate electro-acoustic transducers which could convert sound waves into electrical signals. It was natural for these important ideas to spread into the recording industry.

If the home gramophone were to become an electro-acoustic device, some way had to be found of replacing the acoustic horn with a transducer, i.e. one which could convert electrical signals fed to it into sound waves. This need was met by two American scientists, C. W. Rice and E. W. Kellogg, when they invented the moving-coil loudspeaker in 1925. This is described later in Chapter 4. Its great advantage was its simplicity of construction and relatively small size; moreover, if louder sounds were required, this was possible simply by driving the loudspeaker with more electrical power, rather than using even larger acoustic horns. As readers know, this device became established with almost no opposition and is universally used today. At the same

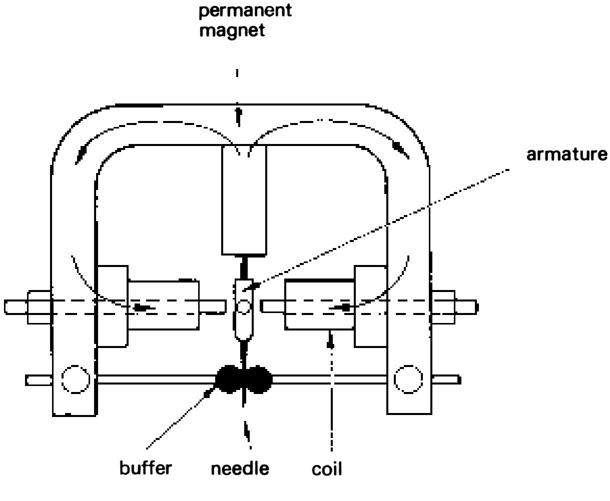
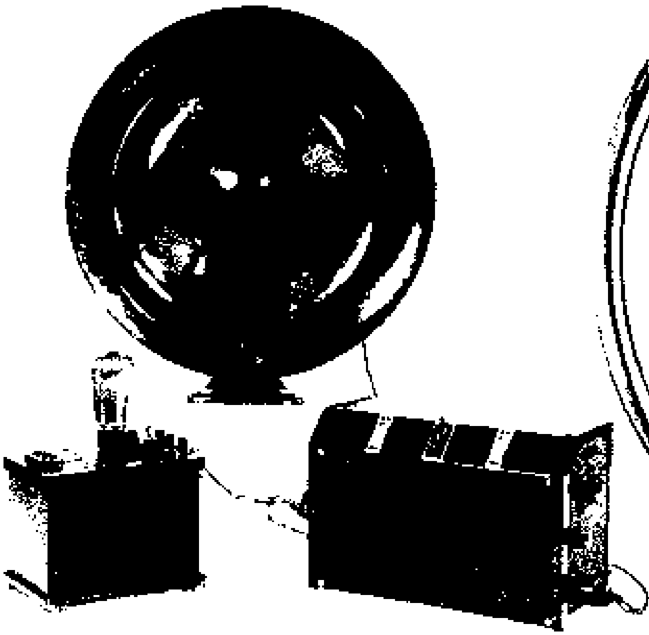


Fig. 5
'Gramophone microphone'. One of the first magnetic pick-ups made in 1926.



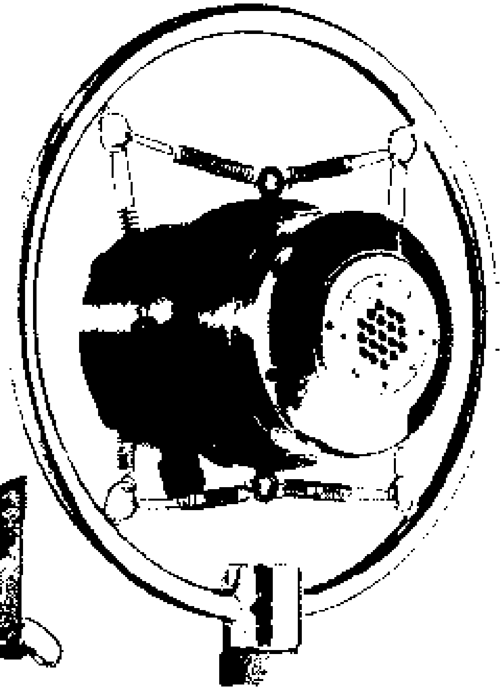


Fig. 6 (Left)
The first radio set marketed by Philips in 1927 to use a moving-coil loudspeaker. The power unit is on the left.

Fig. 7 (Right)
One of the first moving-coil microphones designed and made by Alan Dower Blumlein in the early 1930s.

time, electro-mechanical transducers were required which would do the job of the sound box in gramophones. Initially they were called 'gramophone microphones', and one of the first is shown in Fig. 5. Although relatively crude devices, they were lighter than the sound boxes, and they produced from the record an electrical signal which could be processed in an electronic amplifier. Hence by 1926 the principles of the gramophone were the same as they are in present day hi-fi equipment; the pick-up generated an electrical signal which was electronically amplified and fed at higher power to a loudspeaker which changed back the signal into radiated sounds. In that year Brunswick, in association with the Radio Corporation of America (RCA) announced the first electric record player console which it named the Panatrope.

These developments in loudspeakers and amplifiers also allowed radio programmes to be heard by a much wider audience. Hitherto, radio had been primarily a medium for the hobbyist, who would listen over his crystal set. The same developments were applied in the recording studio. Acoustic records, where the master disc was cut from a stylus energised directly from the sound source, were replaced by discs which were cut by electromagnetic cutting heads, energised from amplifers and microphones. These new records had far higher recorded levels and dynamic range than their predecessors, giving reproduction quality in the home much nearer the standard we are used to now.

One of the most important contributors in this period was the British scientist Alan Dower Blumlein, who developed some of the first high quality, moving-coil microphones, cutting heads and lathes on which the first 'electric' records were made. His contributions were many, in areas such as television, communications and electrical measurement as well as sound recording. It is remarkable to point out that Blumlein, back in the 1930s, proposed a method of stereo recording–the closely spaced microphone pair–which experience has since shown to be the best method of accurate two-channel recording, and one very widely adopted in present day recording technique. Certainly one of the great scientists of the time, he was sadly killed in an aircraft crash during World War II at the age of thirty-nine.

The recorded repertoire and public interest in record-playing equipment were now served by a specialist magazine, the *Gramophone*, founded in 1923 by Sir Compton Mackenzie. It featured regular reviews of new record releases and a section written by an 'expert committee' on the technical developments. There were many still to be made. The 1930s saw the new superhet radio receiver, in which station interference and 'whistles' were largely eliminated by using a fixed tuned frequency in the set's amplifying stages. In America, Juke Boxes began to appear, and with more than 25000 in operation by the mid 1930s, accounted for a further 20 million records sold. RCA Victor introduced the Duo Junior, a small turntable designed to be jacked into the radio set and offered for sale at just $9·50. It created thousands of new record fans; artists such as Bing Crosby and Louis Armstrong featured on 10-inch shellac discs selling for 35 cents each. Loudspeakers became more sophisticated; the first commercial electrostatic loudspeaker system was developed by the Automatic Musical Instrument Co. for use in their coin operated phonographs, and Bell Telephone Laboratories demonstrated the first two-unit loudspeaker with bass and treble sounds divided by filters between the two speakers. Separate high-frequency speakers or 'tweeters' became available in England, too, in 1931, and the bass reflex enclosure was introduced commercially by Jensen in 1937. Its details will be described later in the book (Chapter 4).

There were problems, of course, during the War; manufacturers resorted to grinding up old records and re-using them with a filler to stretch the supply of shellac. This had come mostly from Singapore and supplies had ended although demand for records had kept growing. But the hi-fi era had yet to come. One of the most significant post-war developments was the emergence of tape as a recording medium, for it gave far greater flexibility to the recording engineer. Then, in 1948, came the revolutionary Long-Playing record from Columbia, giving twenty-three minutes playing time on each of its 12-inch sides. sides. The 45-rpm 7-inch vinyl single followed in 1949, and there was something of a battle between the two speeds–$33\frac{1}{3}$rpm for the LP and 45 rpm for the single–but both had their features.

Long playing time was the main advantage of the LP whilst compactness was all important for the short musical numbers released in the 'pop' industry, and so both have remained.

The technical quality of the LP record was exceptionally high, although it did call for further big changes in the cutting and playback heads. All the previous domestic pick-ups were 'heavy-weights', with tracking weights of 3 or 4 ounces, i.e. around 100 grams, and even the first of the 'middle weights', introduced by Telefunken in 1939 for 78-rpm shellac discs, still tracked at 25 grams. The new LPs could not stand up to this, and less than half this figure was needed for reasonable record life. The answer was the pick-up with a cantilever; in this the stylus was decoupled from the heavy transducing element by being mounted on a long, thin bar, the cantilever, which made it free to move under far less downward force. The British engineer Stanley Kelly was one of the first to develop such a pick-up for Cosmocord, in the early 1950s, featuring a sapphire stylus and a playing weight of 10 grams. His later work included the ribbon loudspeaker, which is described in Chapter 4. Cosmocord and Garrard made millions of crystal cantilever pick-ups for record players and radiograms in the 1950s, which was really the start of the hi-fi age. During that decade world record sales trebled; the quality of the LP was so good that manufacturers sought hurriedly to improve the performance of their record decks, amplifiers and loudspeakers.

Hundreds of new audio equipment companies appeared, for example the famous British loudspeaker manufacturer KEF,

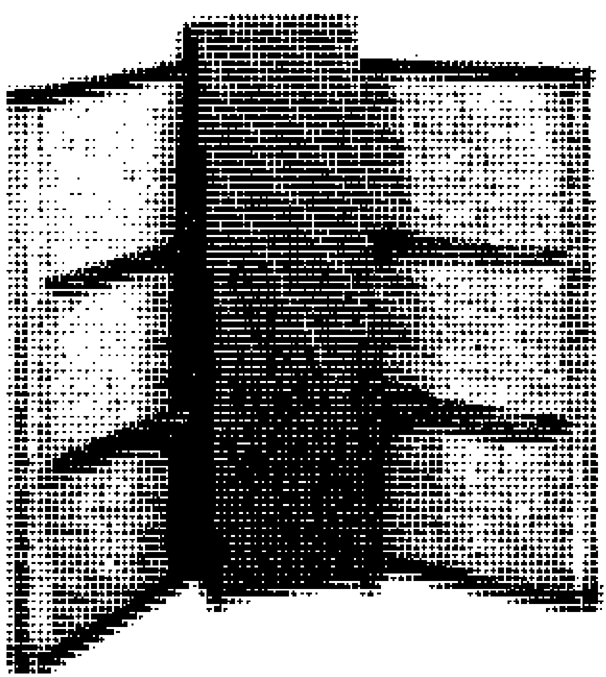


Fig. 8
A prototype full range electrostatic loudspeaker by Quad, made in the early 1950s. The system stood about 1.5 metres high, with a central treble unit and bass sections in the wings, but never reached production.

first started trading in the mid 1950s. Another British firm, Quad, announced the first full-range electrostatic loudspeaker, which is described in Chapter 4. Peter Walker, who designed it, had been working on the principle for some years before, and there were several prototypes which never reached the public eye. For interest one of them is shown in Fig. 8. The introduction of a standard system of two-channel stereo sound for LP records in 1958 called for even further specialised development from pick-up engineers. One of the famous designs of the 1960s was the Decca ffss integrated pick-up and arm, shown in Fig. 9. Stereo record grooves vary in depth as well as width, requiring the stylus to have freedom to move in all directions in a plane, so tracking forces of 2 or 3 grams or less became the order of the day. The public began to choose their audio equipment from specialists in each product area; an amplifier by Leak, a turntable from Garrard or Collaro and a loudspeaker from Kef, Quad or Wharfdale. A separate FM tuner in the mid 1950s and a cassette deck in the mid 1960s became additional elements as hi-fi developed. The compact cassette was introduced, firstly by Philips and then by other companies who agreed to standardise the format. In 1956, the first British magazine devoted mainly to hi-fi equipment, *Hi-Fi News & Record Review*, was launched. The *Gramophone* and *Hi-Fi News* were followed in the 1970s by many more specialist British audio manazines, as current readers know; for example, *Popular Hi-Fi*, *What Hi-Fi*, *Hi-Fi Answers* and the IPC magazine *Practical Hi-Fi*, with its lively readers' letters page. A similar variety of reading matter is available in many other countries of the world, confirming the wide-established interest in hi-fi and recorded music.

Today the listener can choose his records and tapes from a vast recorded repertoire and his equipment from hundreds of models from the world's manufacturers. Hi-fi is a major consumer market with an appeal for every person, from the outright audio enthusiast to the newcomer, and the many musical persons for whom audio equipment is a key to musical enjoyment. To know a little more would be helpful to all, and in the following chapters I have set out to explain the most important points, whilst always remembering the main purpose and truth of it all, a wider enjoyment of music.

First a few words on the basic properties of sound, and the way we hear it.

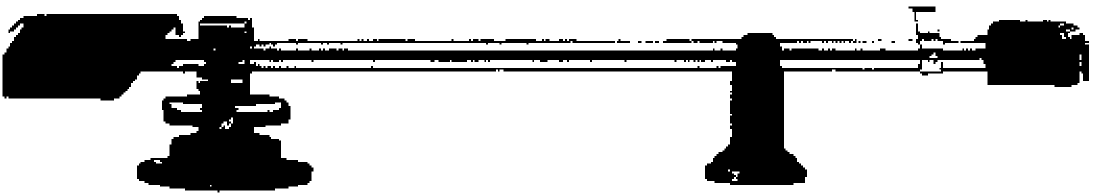


Fig. 9
The Decca Mk I pick-up and arm designed for stereo LP records.

Unlike radio waves or light waves, which can travel in 'empty' space, sound requires a material medium to travel in. This medium might be air, through which, for example, the sounds heard in a concert hall travel (or the 'noise' of traffic in a busy street), or a solid material, such as a wall or the wooden panels of a loudspeaker cabinet. When sound travels in air, it is said to be an 'airborne sound' or an 'airborne noise', whilst when it travels in a solid structure it is called a 'structure-borne vibration'. It is possible for vibrations in solid structures to cause airborne sounds, and vice-versa; for example a hi-fi system playing in one room can be heard in a neighbouring room due to small vibrations set up in the walls. Similarly, the vibrations of a musical instrument produce sounds in the air which travel out to the listener.

Sound travels in the form of waves which are produced by the vibration of the sound source. The forward vibration of, say, a violin string, pushes against the air which contacts it, and compresses the molecules forward. Because of the fluid or elastic nature of air, these compressions are passed on to layers further in front. When the vibrating string moves backward, the air layers are rarified instead of being compressed, and this state is passed on in exactly the same manner. By this means, a sound *pressure wave* is created which travels outwards from the source at a definite speed (Fig. 10). The speed with which this sequence of events takes place is the well known *speed of sound* or the *velocity of sound* denoted by the letter c. The speed of sound is 344 metres per second or about 1150 feet per second. It means that, in any one second, any particular feature of a sound wave travels a distance of 344 metres in air.

The rate at which the vibrations of the sound source (and the alternate compressions and rarefactions of the air) occur is called the *frequency* of the sound and is denoted by the letter f. Not all frequencies are audible to the human ear; a listener near to the source will hear the sound if this frequency is anywhere between about 20 times per second and 20 thousand times per

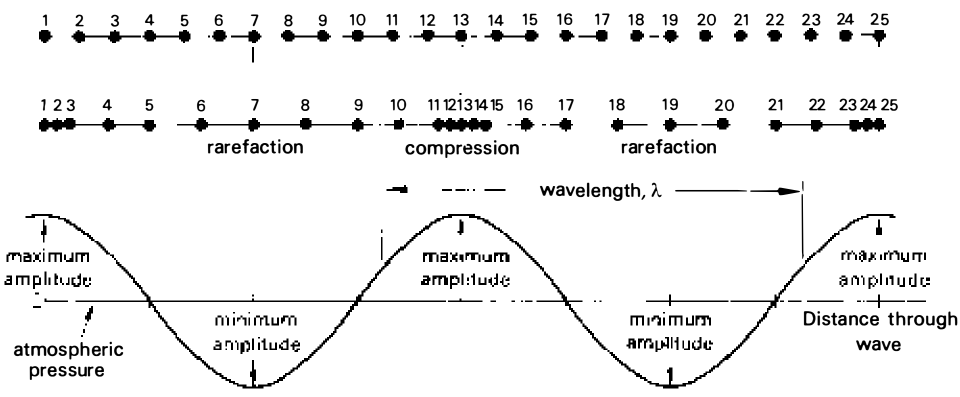


Fig. 10
Showing pressure changes in a simple sound wave. All music and speech sounds are made up of such simple waves, called sine waves.

second. The symbol Hertz is used for this measurement, abbreviated to Hz. Thus the audible frequency range extends from 20Hz to 20000Hz or 20kHz (1k=1000). Frequencies below 20Hz are called infrasonic whilst those extending above the audible range are called ultrasonic. (Analogies can be made here with light waves; for example compare these terms with infrared and ultraviolet. The colours red and violet are at the low and high extremes, respectively, of the visible spectrum).

The *wavelength* of a sound is the distance between any two points in the sound wave which are identical and is denoted by the symbol λ. The speed of sound is then simply related to its frequency and its wavelength by the formula $c=f\lambda$, where c is the speed of sound, f its frequency and λ its wavelength. For all practical purposes, the speed of sound, c, never changes, so there is always a fixed relationship between the frequency f and wavelength λ. Table 1 shows how the wavelength of sounds are related to frequency. Notice how widely the wavelength varies for different audible frequencies. Low, bass frequencies have wavelengths of several metres, whilst high frequency treble sounds have wavelengths of only a few centimetres or less. These results have very important implications for accurate sound reproduction in a listening room.

Table 1 Relationship between frequency and wavelength

Speed of sound c=344m/s

Frequency, f	**Wavelength, λ**		
Hz	metres	feet	inches
20	17·2	56	4
40	8·6	28	2
50	6·8	22	7
70	4·9	16	2
100	3·4	11	3
150	2·3	7	6
200	1·7	5	8
500	0·7	2	3
1 kHz	0·3	1	2
2	0·1		7
4	8 cm		3
8	4		$1\frac{1}{2}$
10	3·4		$1\frac{1}{3}$
14	2·4		1
17	2		$\frac{3}{4}$
20	1·7		$\frac{1}{2}$

Reflection and diffraction

Large objects such as chairs or bookcases in a listening room, cast sound shadows at medium and high frequencies, whereas at low frequencies the sound diffracts around these objects and is

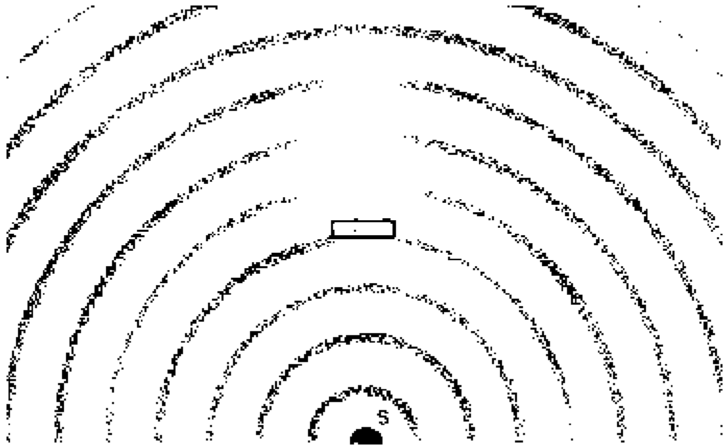


Fig. 11a
Sound waves diffract around an object when it is small compared with their wavelength.

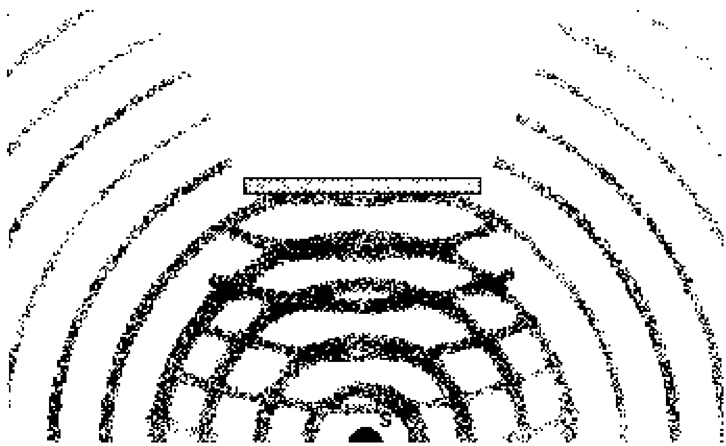


Fig. 11b
Large objects reflect the sound back and cast a 'sound shadow' where the sound cannot be heard.

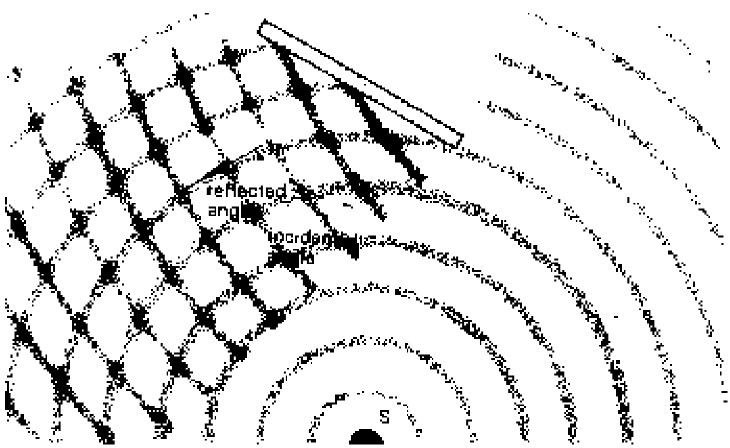


Fig. 11c
Large objects reflect sound at the same angle at which it arrives on their surface.

transmitted as though they were not there. This is because when sound waves encounter an object in their path, their behaviour will depend upon the size of the object in relation to their wavelength. Just like light waves, sound waves will *diffract* around very small objects, i.e. those which are very small in comparison with the wavelength of the sound, but large objects will *reflect* the sound, and cast a 'sound shadow' beyond the object, which dampens or muffles the sound, (Fig. 11). Since high-frequency sounds have a very small wavelength, however, almost any object of significant size in a listening room will interrupt the passage of a high-frequency sound. Hence the sound field in a listening room will not be the same at all frequencies in the audible spectrum, but will be affected by the contents of the room, depending upon the frequency. More will be said of listening room acoustics in Chapters 2 and 5.

Sound levels

When a musical instrument or loudspeaker is radiating sound, it must be losing energy to the air. The air is a medium which is transferring sound energy from the source to the listener. The sound source is losing energy at all points around it, and so to find the total amount of sound energy being radiated in any one moment, i.e. the *acoustic output power*, we add up all the powers flowing through each small region of air surrounding the source. In fact the *intensity* of a sound gives us just this information; it tells us the power per unit area flowing away from the source, at whatever point we measure it. A very loud sound therefore has a high intensity, whilst a very soft sound has a low value of intensity. If the sound source radiates equally in all directions, i.e. if it is omni-directional, we can easily find the total sound power it is radiating if we know the intensity at any point. Many practical sound radiators, however, do not radiate in this simple manner, as we shall see in Chapter 2.

As we move further away from the source of sound, the area over which the sound power flows will increase. Then the power per unit area will be smaller, so the intensity must fall and the sound will become less loud. The ear can accommodate an enormous range of intensities; at middle frequencies, say around 1 kHz, most listeners can just detect a sound with intensity 10^{-12} watts per square metre, i.e. 1 micromicro watt per square metre. At the same time a listener can appreciate the arresting sound of a bass drum, which has an intensity of 0·1 watt per square metre a metre or so away. Hence it is normal for sounds with intensities of many million times each other to be encountered in everyday life. With such large numbers involved, it is simpler to use a different scale to express intensities. In fact we use a decibel scale to express these ratios which we call the *intensity level* of a sound. Taking the quietest sound which most listeners can just hear, 10^{-12} watts per square metre, as reference I_o the intensity level of a sound of intensity I is simply given by:

intensity level $= 10 \log_{10}\left(\frac{I}{I_o}\right)$ dB, with the abbreviation dB for decibels.

Table 2 Intensity ratios using the decibel scale

Intensity I (watts/sq. m.)	**Intensity Level dB** (dB re 10^{-12} watts/sq. m.)
100 000	170
10 000	160
1 000	150
100	140
10	130
1	120*
0·1 (10^{-1})	110
0·01 (10^{-2})	100
0·001 (10^{-3})	90
0·0001 (10^{-4})	80
.	.
.	.
.	.
.	.
.	.
10^{-12}	0

**Normal maximum level encountered*

Table 2 shows how the intensity of a sound can be expressed more conveniently using the decibel scale. An intensity which is 1 million times higher than the reference or 'threshold' intensity then has an intensity level of 60 dB, a more manageable figure. An intensity level of 120 dB is the highest figure which most listeners are able to accept without the sensation becoming painful. This is known as the 'threshold of pain'. Needless to say, such high levels as this are scarcely ever encountered in the everyday sounds we hear. It is possible, however, for certain musical instruments to produce sound levels which easily exceed 100 dB, as we shall see in Chapter 2. The sound intensity levels of a range of sources known to us is shown in Table 3.

Table 3 Typical sound intensity levels and relative 'noisiness'

Sound Intensity Level *(dB)*	**Source**	**Subjective Loudness**
220	Nuclear weapon detonated at 500 metres	Destructive
200	Atlas guided missile launch at 100 metres	Destructive
160	0·303-inch rifle, peak sound level at user's ears	Dangerous to hearing
140	Jet aircraft launched from naval carrier	Dangerous to hearing
130	Early submarine engine room	Deafening–threshold of pain for most people
120	Hard rock band	Deafening in the long term but not dangerous in small doses

Sound Intensity Level *(dB)*	Source	Subjective Loudness
110	Accelerating motor cycle at 2 metres	Very loud and uncomfortable
100	Noisy aeroplane cabin or loud car horn at 3 metres	Very loud and distracting
90	Pneumatic road drill at 3 metres or noisy urban street	Very loud
80	Tube train in tunnel at 35 m.p.h. or school dining hall with untreated walls	Loud
70	Busy traffic at a few metres	Quite loud
60	Noisy office, or person speaking at normal voice 1 metre away	Fairly loud
50	In modest saloon car at 30 m.p.h.	Intrusive but not loud
40	In quiet saloon car or soft background music in well furnished room	Fairly quiet
30	Suburban street in evening or average domestic room noise	Quiet
20	Quiet suburb at night	Very quiet
10	Quiet whisper at 1 metre or rustle of leaves in tree in slight breeze	Poetically quiet
0	Threshold of hearing or audibility	Dead quiet

THE PERCEPTION OF SOUNDS

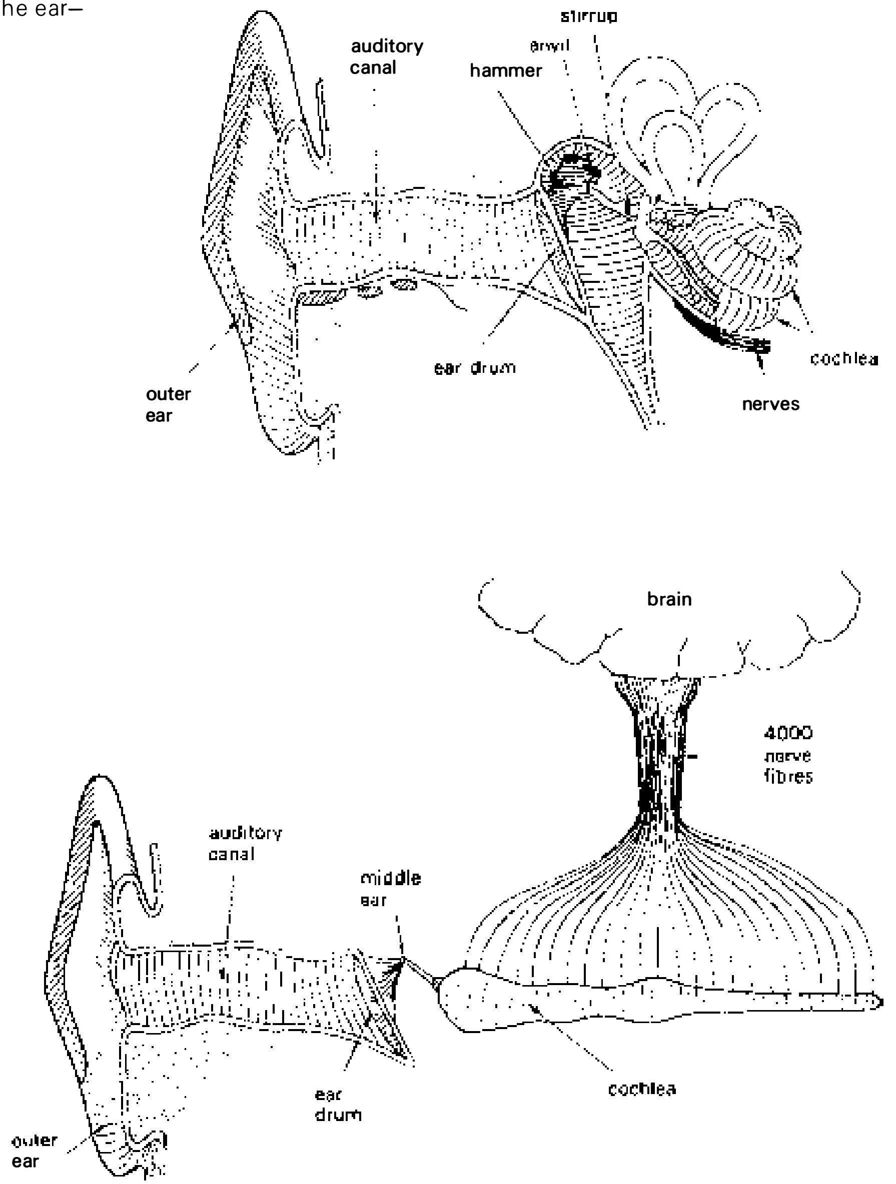
The ultimate destination of a sound is the hearing system of the listener. Our perception of sounds–the way the hearing system functions–is very complex and is influenced by many factors. Some of these are best explained in Chapter 2 when they can be illustrated in relation to music and musical instruments, but it is helpful to look at some of them now.

The ear

The ear divides into three main parts, the external or outer ear, the middle ear and the inner ear (Fig. 12). The outer ear comprises the pinna, which leads to the ear canal and terminates in the eardrum or tympanic membrane. The pinna is that part of the ear which we can all see, but it seems to serve little purpose in our hearing except to direct a little more sound energy into the ear at high frequencies. The ear canal is more important. About an inch (2·5 centimetres) in length, and with a volume of about 1 cubic centimetre, the ear canal provides an acoustic *resonance* (see p. 35) in the centre of the audible frequency range, i.e. at about 3 kHz or so, which means that it provides a useful magnification or amplification of sound pressures and increases our sensitivity to sounds. A consequence of this is that the ear is made less susceptible to damage by being more aware of high sound intensity levels.

The eardrum or membrane is a vibrating element which transfers the air movements into a mechanical motion. This

Fig. 12
Showing the essential components of the ear–brain system.



membrane weighs only about 15 milligrams, and its movement is no more than about 2 millimetres. The middle ear consists of an air-filled cavity of about 2 cubic centimetres volume, which contains three very small, hard bones connected to the eardrum. These bones, called the hammer, anvil and stirrup, act as a mechanical lever and further magnify the force provided by the eardrum. The lever and transformer action of the middle ear gives a remarkable 30-fold magnification of force acting at the eardrum, which is passed on to the inner ear.

The stirrup bone connects to a very important part of the ear, called the cochlea, which forms the principal part of the inner ear. The cochlea is a coil-shaped tissue, of length about 3·5 centimetres and formed into $2\frac{3}{4}$ spiral turns. This complex organ contains fluids which, when subject to pressure from the middle ear, produce nerve responses in some 4000 nerve fibres connecting the cochlea to the base of the brain. The exact behaviour of the cochlea and the way by which it describes the nature of a sound to the brain is still not well understood. Information about the intensity of a sound is believed to be determined by the ear system. Thus, the nerve responses to the brain do not occur as a continuous wave but as a series of short pulses. The greater the sound intensity, the greater the number of nerve pulses forwarded by the cochlea to the brain. The frequency or pitch of a sound, on the other hand, is now thought to be determined in the higher centres of the brain, and not by the ear as described above. Certainly the ear is remarkable in its ability to discriminate between sounds of different frequency. It has a very fast response so that only a few cycles are needed for any frequency to be identified and there are no less than 1 500 different frequencies within the audible spectrum which the ear can distinguish as separate. No known electrical system can discriminate as precisely and rapidly as this.

Space and depth: binaural listening

We have two ears, just as we have two eyes. Our visual judgment of the *relative* distance of objects away from us is greatly assisted by seeing them with two eyes. Readers can very easily verify this by holding up two similar objects, for example pencils, at different distances and then closing an eye. This is also true of our ability to judge the *relative* positions of different sound sources. In fact listening with two ears (binaural listening) provides us with the exact equivalent of binocular vision.

When we listen to sounds, either in a concert hall or a domestic listening room, the sound from any source reaches both ears. Sounds which come from directly in front will produce exactly the same sensation at each ear, but this will not be true of sounds which come from the extreme left or right. Sounds which come, say, from the extreme right of the listener will take a little longer to reach the left ear. This *time* difference is quite small–it will not exceed about 0·6 milliseconds–but it is important in terms of how the ears respond. Another difference between the sounds reaching the two ears results from the listeners head 'blocking' certain frequency components of the sound. Low-frequency sounds will diffract around the head but middle and higher frequencies will be reflected and will not directly reach the ear

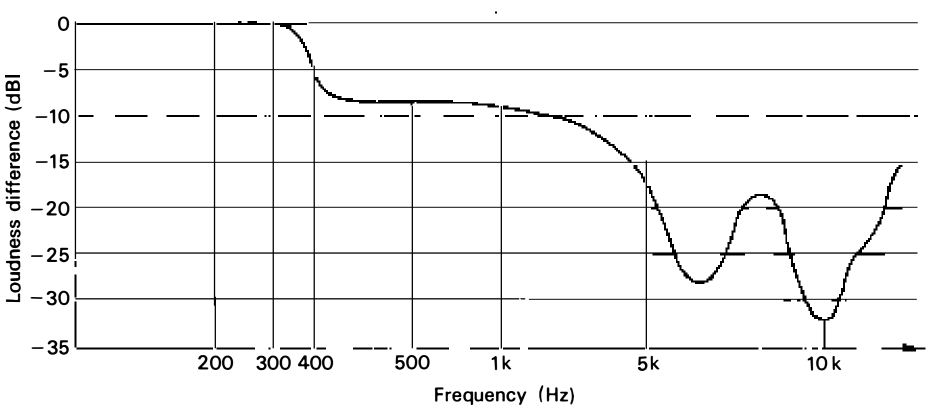


Fig. 13
Curve showing the change in loudness at each frequency when a sound source is moved horizontally from a position 45 degrees left to a position 45 degrees right of the listener.

which is farther from the source. Again, this *frequency spectrum* difference provides an important 'clue' to the location of the source. It can be seen that high-frequency sounds are greatly attenuated–in fact by more than 30 dB in intensity. Because certain frequencies are attenuated, there will also be an overall *intensity* or loudness difference between the sounds reaching the left and right ears, and this too is significant (Fig. 13).

When listening to an orchestra in a concert hall, the listener employs all three of the aural clues described above–time, spectrum and loudness differences (especially the time difference)–to locate each section of the orchestra as it plays. Small movements of the head, which are continually being made, allow precise location as well as judgment of the 'size' of the sound source. Instead of being confused together, different sounds are 'mentally separated'. There are also, of course, sounds reaching the listener indirectly, i.e. sounds which have been reflected from the walls of the hall and which arrive from every possible direction, front, sides, above and even behind. These too are included in the listener's binaural assessment of the sound, so he also gains information about the listening environment, i.e. the size, space and acoustical quality of the concert hall.

There is another important property which the hearing system possesses, and that is the ability to concentrate on one sound and exclude others. This is a subconscious activity on the part of the brain, and in many respects parallels the behaviour of the eyes. Thus the angle of visual perception extends over 150 degrees or so, though it is normal to see in detail over a much narrower field of view. It is possible, however, to bring into focus any area by sudden movement of the eyes.

When a person is listening to another in a room where many others are talking the 'steerable directivity' feature of the hearing system allows him to concentrate on the one voice and to exclude the others, thereby obtaining more information from the desired sound. The same will apply in a concert hall; the listener is able to separate, for example, reflected or reverberant sounds from direct sounds, and to further appreciate the nature and composition of the sounds.

All these points have important implications for an audio reproducing system if we are to try and recreate the original sound experience.

MONO AND STEREO

In the early days of hi-fi, and for many years before, it was usual to obtain a recording of music or speech from the signal derived from a *single* microphone placed at some central position near the sound sources. The recording was replayed through a single amplifier to a pair of headphones, each of which reproduced an identical sound signal at the listener's left and right ears. Alternatively, the sound signal was fed to a single-loudspeaker system placed at some convenient position in a listening room. Such a sound system was given the now familiar name, *monaural* or *monophonic*, meaning that it relied upon only *one* recording and reproduction sound channel. Single-channel sound systems are

Fig. 14
An early recording session.

still in wide use today; for example the household telephone system is a monaural system, whilst many radio broadcasts and cinema sound systems are monophonic. (The term monaural is preferred for systems which replay the sound over headphones and monophonic for those which employ loudspeakers. There are, of course, important differences between the two, for example reproduction over loudspeakers involves the listening room acoustics, whereas headphone listening does not).

A single-channel sound system cannot provide the listener with the same information he would derive if allowed to listen normally, with his two ears, at the same position as the recording microphone. The microphone picks up sound at only one point, whereas the listener detects the sound at two points. Furthermore, the microphone cannot discriminate between different sounds as the listener can. The situation is not improved if several microphones are employed at different points say, of a dance band or an orchestra, if their outputs are merely combined to provide a single sound channel. The result will be that the listener's auditory perspective, his judgment of spatial depth and width and ability to locate the source of a sound, is completely lost because the information he needs is not transmitted. He will be treated instead to a performance in which, although the frequencies and amplitudes of the component sounds are essentially preserved, all directional information is lost. It would be just as though he were placed at some central point near the original sound sources, but with one of his ears totally covered. A good example of this, which listeners can experiment with themselves, is the conventional hearing aid. If a person with normal hearing listens through it in, say, a crowded room, he will find that although the hearing aid makes sounds louder, it also makes them less intelligible. The reason is that the microphone is equally sensitive to every sound reaching it and cannot discriminate between them as the listener can. Hence the microphone, which has no ability to distinguish between different sounds, is imposed upon the listener's hearing system, in the same way as in a monaural sound recording chain.

One better method might be to make a model of the listener's head, and fix in this 'dummy' head two microphones, one at each ear position. If the two microphone signals, which would be different, were recorded and replayed separately, over two spaced loudspeakers or over two separate headphones, then this would recreate the sounds in a very similar way to those the listener would himself hear at the 'live' performance. Although a little obvious and simplistic, this method of recording forms the basis of the familiar binaural method of sound reproduction. It goes a long way to providing the listener with the realism of reproduced sound in which positional information, 'size' and spaciousness are present. A binaural sound system is a *two*-channel system, and the two channels must be preserved throughout the entire recording and reproduction chain.

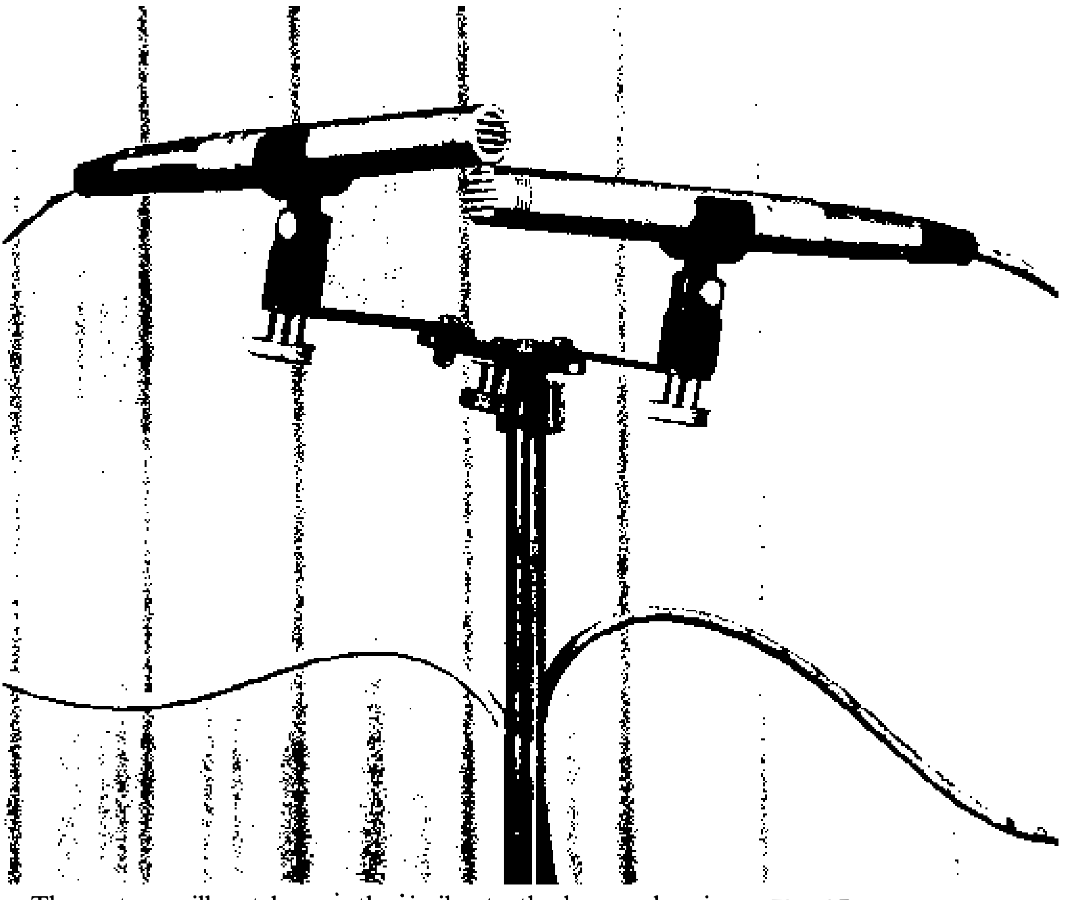


Fig. 15
The closely spaced stereo microphone pair first suggested by Blumlein and often used in present day recording technique.

The system will not be exactly similar to the human hearing system. The 'dummy' head will not move, as the listener's head is continually doing while assessing the sounds. The microphones must be carefully chosen to provide the same directional characteristics as the listener's ears, and even then they cannot discriminate between sounds in the same way the listener can. If reproduction takes place over headphones, it must be remembered that *both* ears should hear different versions of *both* signals, just as they would in the concert hall. It must therefore be arranged that appropriate time delay and amplitude spectrum differences are introduced into the 'crossed' signals.

Two-channel sound reproduction is more frequently replayed over a pair of spaced loudspeakers, when it is called a *stereophonic* sound system. This is the one familiar to every reader, to which he has been listening for some years. Assuming the loudspeakers are identical, and radiate appropriately into space, a stereophonic sound system provides the auditory perspective that would be gained at the 'live' performance. In a good system the illusion of sounds being correctly positioned, laterally across the sound stage as well as in depth, is well created. In hi-fi this is called *stereo image*.

It is not only directional information which a stereo sound system can convey. Because the listener is able to separate the sounds he hears, and to employ his own directional hearing quality, he is better able to appreciate the balance, frequency content and acoustical properties both of the musical instruments themselves and of the recorded listening environment. A common recording method for stereo sound is the closely spaced microphone technique (Fig. 15), as originally suggested by A. D. Blumlein, mentioned earlier in this chapter. Some recording engineers and producers employ a 'multi' microphone technique for stereo, in which a number of microphones are placed at key points around and within the sound source area. The microphone outputs are suitably mixed and directed to the left- or right-hand channels, or both, according to their relative positions, so as to produce the most accurate and pleasing perspective, musical balance and clarity.

MULTI-CHANNEL SOUND SYSTEMS

If a two-channel stereo system can so much help in recreating the original sound experience, it might be thought that with a greater number of independent channels, we could approach the original even more closely. Then we would have a multichannel stereo system which many will have heard in cinemas specially equiped for 'Cinemascope' or 'Cinerama' films. Four, five or even seven separate sound channels are employed, each one covering a definite part of the original sound stage. In such systems, which are very costly, each channel is arranged to be very similar to its neighbour and to overlap it slightly to avoid sound 'holes'. Then a smooth sound image is created, which can extend right across and to the *sides* of the listener. Thus the listener's sense of direction is enhanced, and he gains the impression of being transported *into* the sound field.

There would seem to be a clue here as to what would ideally be required for a sound system to *accurately* recreate the same sound field which exists for the listener at a 'live' performance. Sounds reach the listener from every possible direction, with each directed ray of sound adding a little information to his overall impression. If this situation is to be recreated in the listening room, it means that a very large number of separate sound sources will be needed, so arranged as to 'enclose the listener in sound'. It would, of course, be quite impracticable to consider this as a commercial possibility, but it has in fact been done experimentally. A system such as this is capable of convincing the listener that he is actually one of the audience at a theatre, rock concert, or any 'live' sound session that he may wish to hear. A very complex experiment, both to record and replay, but it does produce results which are exciting and remarkable.

Quadraphony

Perhaps we could go some way towards the ideal listening situation in the home by increasing, to some extent, the number of separate channels of our domestic reproducing system. Two-channel stereo imparts to the sound a degree of width and depth, but these impressions do not extend very far forward of the two loudspeakers. What is required is more of the ambience of the hall or studio in which the recording was made, which is more likely to be perceived from reflected sounds which arrive from the sides and back. This is where a quadraphonic system, which, as its name implies, features four separate channels, will help. With two further recording microphones free to monitor sounds travelling into the audience it becomes possible to capture directional information, derived from both direct and reverberant sound, from all angles in the horizontal plane surrounding the listener.

The number of channels is ideally suited to the domestic listening room, because it is normally rectangular and will thus easily accommodate four loudspeakers in a symmetrical layout. When this new technique was developing, in the early 1970s, there was no problem in making recordings since four-channel tape recorders had by then become established, even domestically. More recently, four-channel discs have become available, with the four separate information channels encoded onto a single groove, to satisfy the all important criterion of compatibility. Unfortunately, however, no single standard has evolved for cutting quadraphonic discs, and this has led to complications for the consumer and some loss of interest.

SUMMARY

From the very early days of sound recording, when it was remarkable enough even to identify the gritty sounds captured from carbon microphones onto shellac discs, the technique of sound recording and reproduction has been to enhance not only the accuracy or fidelity of sound, but also the listener's experience of being present at a 'live' performance. Achievement of this must be the ultimate aim, for there is no more authentic musical experience than being present in an audience to musicians who are playing for one's pleasure. The mechanism of our hearing involves a continuous assessment of sounds which reach us from every direction, in terms of time difference, frequency content (pitch) and loudness, and these clues are further refined by the ability of the brain to discriminate between them. If this situation is to be relived, then the same sound field must be created *by the sound sources* in the listening room, a near impossible task. Practical multi-channel sound systems can go some way to doing this, the stereophonic and quadraphonic reproducers that are in our homes today, of which more in Chapter 3.

2
Listening to Music

Music is the most human of the arts. Music expresses and communicates moods, feelings and human experiences which are too deep for words. This must be why it is possible to enjoy good music without any insight into its nature or composition. There may also be other reasons, which are too difficult to explain, for just as the mechanism of hearing is very complex, so is the relationship between our sensory capacities and the physical nature of musical sound, what might be called our taste in music. The importance of music to us is so much better expressed by a more famous writer–

> *Music is the most entirely human of the fine arts, and has the fewest analoga in nature. Its first delightfulness is simple accordance with the ears; but it is an associated thing, and recalls the deep emotions of the past with an intellectual sense of proportion. Every human feeling is greater and larger than the exciting cause–a proof, I think, that man is designed for a higher state of existence; and this is deeply implied in music, in which there is always something more and beyond the immediate expression.*
>
> (SAMUEL TAYLOR COLERIDGE.)

What he says is not confined to the rarified atmosphere of the concert hall. An audio system in the home can serve the same important cause–that of a wider appreciation of music–to people of all musical taste. Through repeated listening, as and when we wish, we can become acquainted with the most ambitious musical compositions, and be introduced to new music and to new artists. There are no better reasons for our pursuing an interest in high-quality sound reproduction. Whilst music is an art, sound reproduction is a science, so it is not surprising that music has been the subject of much scientific investigation. We are concerned with the musical sounds made by the human voice and by instruments played by musicians, with how these sounds are made and with the physical nature of the sounds themselves.

It is helpful (as well as interesting) to know more about the

Table 4 Objective and subjective terms

Musical term & meaning	Scientific term & meaning
Tone A sound sensation having pitch.	No corresponding term. Sometimes used to describe a single frequency sound (sine wave).
Pitch Subjective judgment of frequency on a musical scale.	**Frequency** Number of cycles in one second (Hz).
Loudness Subjective quality of sound intensity. Can refer to any kind of sound sensation.	**Intensity** Acoustic power per unit area. Difficult to compute in the case of complex musical sounds.
Timbre or Quality Character or distinguishing feature of a complex musical sound.	**Harmonic structure** or **Frequency spectrum** Relative amplitudes of frequency components of any sound.
Transience Suddenness, or rate of growth and decay of a sound. Short sound.	No corresponding term. Complex sound containing many frequency components. Discontinuous sound.
Partial tone Any of the single frequencies which make up a complex musical sound.	No corresponding term. Would have same meaning as musical term, i.e. any one component in frequency spectrum.
Overtone A partial tone of frequency higher than the fundamental in a complex musical sound.	No term used. Would have same meaning as musical term, i.e. any component in frequency spectrum higher than fundamental. Sometimes the term *harmonic* might be used since overtones include these.
Harmonic A partial tone whose frequency is a simple, whole number multiple of the fundamental.	Same as musical term; i.e. any frequency component in spectrum which is a multiple of the fundamental.

nature and main properties of music and speech sounds, in order to specify a home audio system which will reproduce them faithfully. A sound has a number of special characteristics; a musician would speak of these as pitch, loudness, quality or timbre and transience. These are subjective terms, and the only way to explain them reliably, i.e. consistently, is to use objective or scientific words. We have explained two of these scientific terms in Chapter 1, they were frequency and intensity. The corresponding subjective terms are pitch and loudness respectively. (See Table 4)

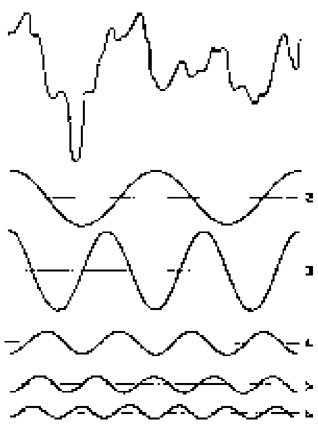


Fig. 16
A short interval of the sound of a violin when playing music and below it harmonic components.

Pitch The pitch of a musical note determines its place on the musical scale, and is the subjective quality of the frequency of the note. In music, it is very rare that a musical note will contain only one frequency component (Fig. 16) but the pitch of the note is always judged to be the pitch of the lowest frequency component the note contains, i.e. the fundamental. Hence the ear judges pitch by assessing the time it takes for a musical sound to repeat itself.

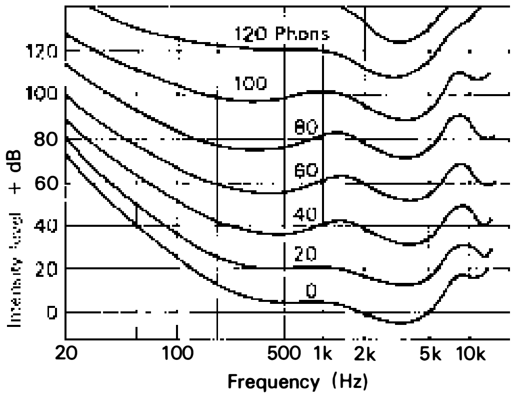


Fig. 17
The equal loudness curves. Every point on a given curve has the same loudness in phons. We judge a sound to be twice as loud when its loudness level is increased by 10 phons.

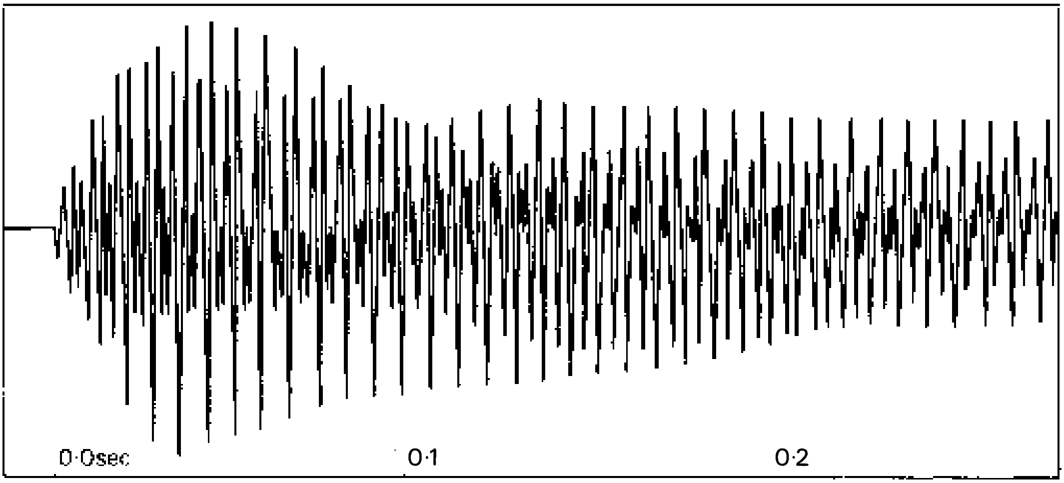
Table 5 Intensity, loudness, musical notation

	phons	dB
ppp	20	40
pp	40	50
p	55	60
mf	65	70
f	75	80
ff	85	90
fff	95	100

Intensity The intensity of a sound, is a measure of the power flow away from the source of sound, but the loudness of the sound is how the ear perceives this power flow. In fact our sensitivity to sound power depends upon the frequency we are hearing and upon the intensity at which we hear it (Fig. 17). We use a special unit to indicate how loud a sound can be, it is the Phon. It is well known that at low sound levels, say intensities of 40 dB or 50 dB, we are much less sensitive to extreme bass and treble notes, whilst at high levels, say 100 dB or more, corresponding to the peak sound levels in a concert hall, we are about equally sensitive to all sounds whatever their frequency. The relationship between intensity level, loudness level and musical notation are shown in Table 5. Notice that the difference between the quietest and the loudest sounds is about 60 dB, in scientific terms, and this is an important point when considering the range of sound levels an audio system will have to accommodate.

Quality or timbre It was said above that a musical note usually contains a number of different frequency components. Some of these may be simply related to the fundamental note, i.e. they may be simple multiples of the fundamental, when they are called *harmonics*. This is not always true, however and a more general term used is *overtone*, which describes any of the higher component frequencies present along with the fundamental. The overtones or frequency content of a musical sound describe what the musician calls the quality or timbre of the sound. Whether it is smooth, as for the piano, or incisive, as for the violin. If the harmonic or frequency content of a musical sound is sufficiently altered, the instrument which produced the sound becomes more difficult to identify. There are many factors which determine the quality of a sound, as we shall see later in this chapter.

Transience This is perhaps the most interesting feature of all and has much to do with the way a musical instrument is sounded. When the instrument is first sounded, the sound produced con-



tains two parts; a 'steady' part, which may persist for a few seconds, and a time dependent part, which 'rides' on the steady part but which decays away quite rapidly with time. Hence the *initial* sound is different from the steady sound produced, the latter often lasting much longer. (Fig. 18). Every musical instrument will exhibit a transient part to its characteristic sound, though in some cases this part will almost dominate in importance, even to the extent of identifying the instrument being played. A piano note, which is struck with a hammer, produces a transient which persists and contributes considerably to the overall subjective impression of the sound. The notes on the harpsichord, on the other hand, are plucked; the sounds are short lived with the initial transient forming their greater part. Even in the case of instruments where the transient has about the same importance in relation to the steady component, for example if we compare the piano with the cello, it becomes difficult to tell much difference between their sounds if the starting transients are removed.

Fig. 18
The build-up and decay of note C (512 Hz) played on a piano. The time-scale shows intervals of tenths of a second. Notice how higher harmonics within the envelope rise and fall even as the fundamental dies steadily away.

A hi-fi system must process transient sounds very accurately, preserving the shape of their waveform and any sudden changes in amplitude. Any limitation in the system which alters wave shapes will detract from accurate transient resolution or transient precision. If sudden amplitude changes are blurred, the system will not have what is termed *transient attack* and the sound will lose its edge. (A similar effect is produced by turning down the treble control on the amplifier.)

MUSICAL SCALES

As we have said there are an infinite number of possible frequencies or changes of pitch in the audible range, extending as it does from 20 Hz to 20 kHz. The point would seem to be, can we hear frequencies *continuously* within this range, or is there some limiting difference in frequency which we are unable to detect?

Fortunately for the construction of musical instruments, and, for that matter, audio equipment, there is a limiting value of pitch change which we cannot hear. At low frequencies, say around 100 Hz, a change in frequency of 3 per cent is barely detectable, whilst in the region above 500 Hz the threshold of pitch discrimination is as small as 0·2 per cent. The result is that there are a finite number, about 1 500 discrete frequencies within the audible range which most listeners, if tested, would be capable of identifying.

These figures are important to the audio equipment designer, when determining the maximum speed variations he can allow in record turntables or tape recorder drive mechanisms, but they are just as important to the design and development of musical instruments. It would hardly be possible for a musical instrument to provide 1 500 separate notes for the musician to play in accordance with what the composer and the listener could possibly hear. (An instrument which comes closest to this is the pipe organ, an instrument of vast size and cost of which more will be said later). To solve this problem *musical scales* were established early in the development of Western music.

A scale is a series of sounds or notes, arranged from low to high frequencies by definite frequency intervals. Each interval between two notes is determined by the *ratio* of their frequencies, not by the frequencies themselves. The way in which these ratios are arranged through the frequency range distinguishes one type of musical scale from another. The two most important basic musical scales are the 'just' or 'natural' scale and the 'equally tempered' scale or scale of equal temperament (Table 6). The natural scale is based on the fact that frequencies which have simple, whole number ratios are most pleasing to the ear. The most 'consonant' or pleasant sounding interval corresponds to a ratio of 2:1 i.e. an octave. Notice, on this scale, which is based on musical C, that the frequency ratios of notes C, E and G are in simple number ratios of 4:5:6. The whole scale is based upon the

Table 6

The natural or just scale

Name of note	C	D	E	F	G	A	B	C
Frequency in terms of C=f	f	$\frac{9}{8}$f	$\frac{5}{4}$f	$\frac{4}{3}$f	$\frac{3}{2}$f	$\frac{5}{3}$f	$\frac{15}{8}$f	2f

The equal temperament scale

Name of note	C	D	E	F	G	A	B	C
Frequency in terms of C=f	f	$2^{\frac{1}{6}}$f	$2^{\frac{1}{3}}$f	$2^{\frac{5}{12}}$f	$2^{\frac{7}{12}}$f	$2^{\frac{3}{4}}$f	$2^{\frac{11}{12}}$f	2f

fact that the most pleasing combination of two notes is one expressible by two whole numbers, neither of which is large. Many musical instruments can be tuned to this musical scale, or they can be 'tempered' to other scales. Some instruments, however, have notes which are normally fixed, for example the piano or the pipe organ. These instruments require a scale where the frequency interval between notes is the same at all points of the scale, which it is not for the natural scale.

This is necessary so that music can be played in a different key, where the keynote is the sound or note with which the musical scale or composition begins. This would be no problem for a violin for this instrument may easily be tuned and can play any frequency within a continuous range on each of its strings. The piano, organ, and most wind instruments would require new notes to accommodate a change in musical key. In the more widely used equally tempered scale, the octave is divided into 12 intervals in which the frequency ratios are the same at every point. In fact the ratio has the value $2^{\frac{1}{12}}$ for each semi-tone, (12 semi-tones equal one octave) so there is consistency between each note regardless of key. Although the above are the basic musical scales, composers of recent generations have been moved to extend the range of musical expression in their work by pioneering new forms of musical scale, for example the 12-note system, but these are beyond the scope of our discussion.

RESONANCE

Because musical scales consist of definite fundamental frequencies, musical instruments and composition can be based upon them. Musical instruments are therefore designed to produce a limited range of frequencies, and to discriminate against others. This is why virtually every musical instrument ever designed relies upon some kind of *selective* mechanical or acoustical system to produce its sound. A selective system of this kind implies a *resonant* behaviour which means that there is some frequency at which the system is most easily excited into motion (also called the *natural frequency* of the system). The resonant properties of almost every kind of object have been employed for this purpose in musical instruments. Strings, bars, plates, membranes, reeds, pipes, horns, the air itself, and more recently electronic circuits, are all featured in making musical sounds (Table 7). Almost any mechanical or acoustical system exhibits resonance.

At resonance, very little energy or force need be applied to the system in order to excite it into motion very vigorously, and this property can be turned into useful account. For example, we can design the 'body' of a violin to resonate acoustically at some frequency in the audible range, when it will radiate sound efficiently. At frequencies other than resonance this radiation is

Table 7 Instruments and resonant properties

Instruments	Properties
Whistle, Pipe organ, Flute, Piccolo	Air column (acoustical) *resonance* excited by blowing or pumping; some tuned with keys on pipe
Violin, Viola, Violincello, Double Bass, Piano, Harp, Guitar	Vibrating string *resonance* excited by bowing, plucking or striking; tuned by string length changes with further tonal colour and acoustic efficiency added by instrument's body
Clarinet, Saxophone, Oboe, Bassoon, Bagpipes, Harmonica, Accordion	Mechanical (reed) *resonance* and (in some) *resonant* pipe which gives further efficiency, tonal compass and tuning
Horn, Trumpet, Cornet, Tuba, Trombone	Lip reed *resonance* coupled to *resonant* pipe for efficiency and tuning
Xylophone, Chimes, Tambourine, Bells, Drums, Cymbals, Triangle	Vibrating bar, membrane or plate *resonance* in most cases untuned but in drums coupled to air volume *resonance* below membrane
Human voice	*Resonant* membrane (vocal chords) of variable tension with oral/nasal cavities and larynx forming complete mechanical/acoustical resonant system
Electronic musical instruments	Tuned (*resonant*) electrical networks and electronic oscillators/amplifiers

greatly reduced, so that the resonant system is also selective. Another example is the acoustical or air resonator in a loudspeaker cabiner or 'reflex' cabinet. We can arrange that the system resonates at some bass frequency in order to increase the efficiency of the loudspeaker as a low-frequency radiator. Resonance can sometimes be a bad thing. Under certain circumstances, when resonance gets out of control, it can lead to damage or destruction of the system being excited. An example of this is the infrasonic (very low frequency) destruction of the Tacoma bridge in America in 1949.

The elements of these resonant systems are shown in Fig. 20. Familiar to some readers will be the electrical resonant system where the resonant frequency is the one at which current flows most easily. The magnified electrical current obtained can be used, for example, to provide a note on an electric organ.

In the mechanical system the elements are mass, springyness or stiffness and friction, which corresponds to the resistance or loss element in the electrical system. An example of such a resonant system in a musical instrument is the string in a piano or violin. The mass and springy elements are distributed uniformly throughout the string, so the length of the string determines the resonant or natural frequency of vibration. Another example is the mechanical reed in, say, a clarinet or saxophone. The reed possesses

Fig. 19
Showing the destruction of the Tacoma Narrows bridge in 1949. It was set into motion by high winds, and its long slender girders were not stiff enough to prevent its eventual collapse.

the same basic mass and springy elements, together with frictional losses, and is excited into vibration by the musician blowing air into it.

In the acoustical resonator, the mass and springy elements are provided by the air in the system. A certain volume of air moves to and fro in the neck or vent of the vessel, and this constitutes the moving mass of air. This moving mass compresses the fixed

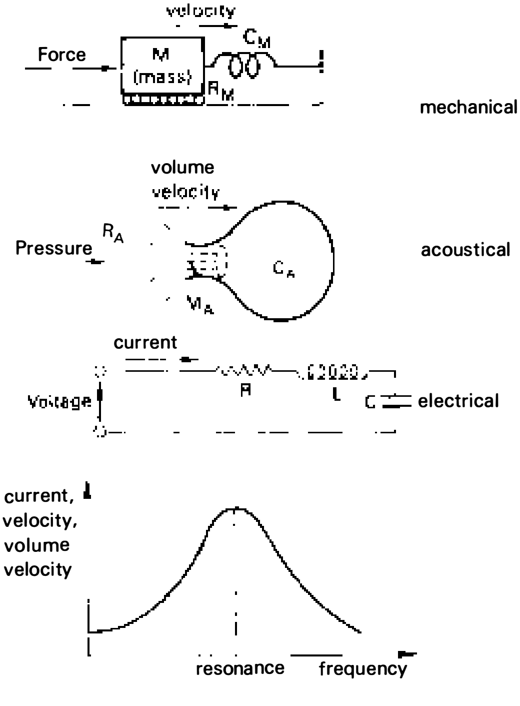


Fig. 20
The basic elements of different resonant systems and the curve of how each responds. At the resonant frequency each system responds most easily.

volume of air contained in the body of the vessel, and this has springyness or stiffness. There will be a frequency at which the air mass cannot move any faster, and is most easily excited. This is called Helmholtz resonance, after the scientist H. L. Helmholtz who first developed the theory of the system. The greatly magnified air velocities in the mouth of the vessel cause sound radiation to take place. A simple example of this is when we blow across the mouth of an empty bottle. If the blowing action is directed suitably, a sound of definite pitch can be heard clearly. This sound will be lower in pitch for larger bottles. Exactly the same principle applies to the reflex or 'tuned' loudspeaker cabinet (see Chapter 4), except that the vent resonance is excited by the loudspeaker unit from within the cabinet. It is worth noting that this system can also be employed as an acoustic absorber, by lining the neck of the vessel with a soft absorbent material. There are yet other applications of the Helmholtz resonator in room acoustic design, as we shall see in Chapter 5.

FEATURES AND SOUNDS OF MUSICAL INSTRUMENTS

Every musical instrument exhibits resonance. Sometimes more than one resonant system operates to give an instrument its characteristic sound. There are many ways, too, in which the resonant systems can be excited. In string instruments the strings are plucked, as in the harp or guitar, bowed, as in the violin or double bass, or even struck, as in the piano. These actions can be shown to 'shock excite' the strings, which means that the driving force contains many frequency components. For example, when a bow is drawn across a violin string, the string moves discontinuously, being dragged with the bow a small distance and then slipping back to a point beyond its rest position, due to its elasticity. The time taken for this sequence of events to repeat determines the fundamental frequency of vibration, i.e. the pitch of the instrument.

String instruments

String instruments are among the most traditional in music, and form a large section of the orchestra. They employ a number of tightly stretched strings, often of metal, coupled to a hollow body or sounding board which exhibits its own complex vibrational behaviour and serves to augment the sound output of the strings. The strings produce overtones which are all harmonics of the fundamental, and whose relative amplitudes are similar between instruments excited in the same way. The characteristic sound, of course, is quite different for each particular instrument, because the body modifies the relative amplitudes of certain overtones according to its shape and size, as well as adding its own transient characteristics. Hence the body imposes its own formant qualities on the sound of the primary vibrators, and contributes greatly to the final sound. The larger the body or sounding board, the more efficiently it will radiate sound.

The lute, zither, harp, guitar and harpsichord are all plucked-string instruments. The lute, first made more than a thousand years ago, is the forerunner of the modern guitar (Fig. 21). The six strings of the guitar are stretched between adjustable pegs and a tailpiece, which transmits their vibration to the hollow body. The instrument is finely tuned by the pegs, but the pitch is varied when playing by pressing the strings against the frets on the finger board. The other hand of the musician is used to pluck the strings, sometimes with a pick or plectrum, a small flat piece of metal or plastic held between the thumb and first finger. The intensity with which the strings are plucked affects the overtone structure, making the sound more incisive, but the early harmonics have the highest amplitudes (Fig. 21).

The harp, generally a classical instrument, has its strings

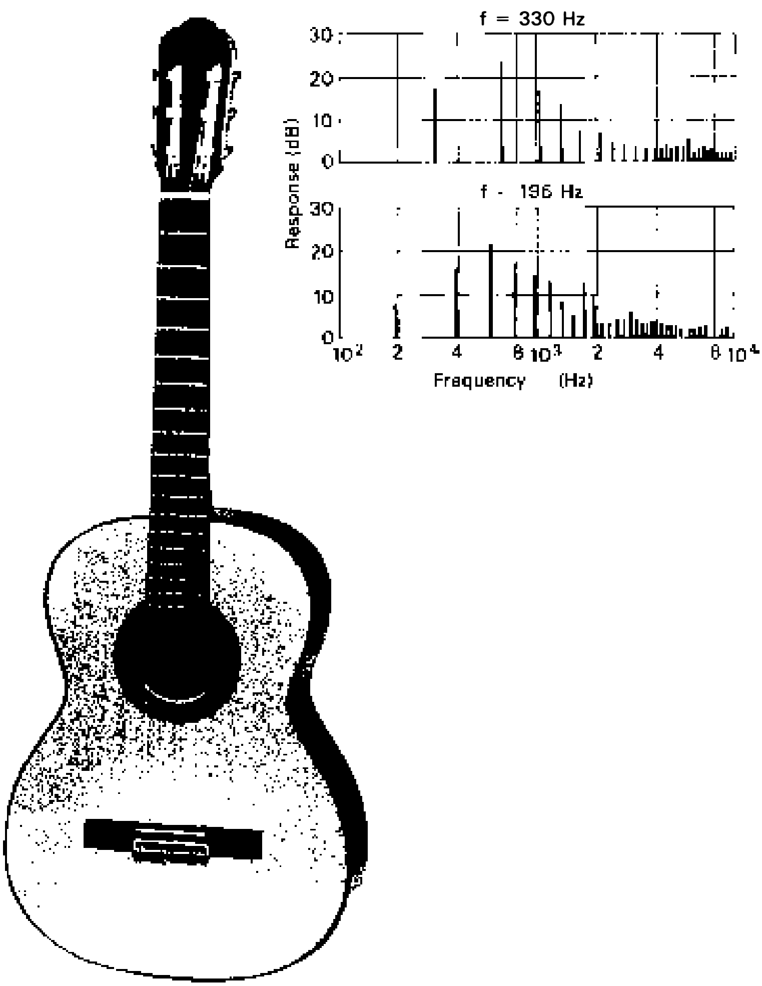


Fig. 21
The acoustic spectra of two notes of a guitar, fundamental frequencies shown.

stretched vertically within a triangular frame, (somewhat like the lute, its forerunner) one member of which forms the sounding board. Since the strings stand clear of the sounding board, their own transient behaviour dominates and persists for some time. The instrument is provided with several pedals which operate a disc-wrestpin mechanism at the top end of the strings to change their pitch when playing, each pedal relating to all strings tuned to the same key. Conventional pegs are also fitted for 'static' tuning. The strings are plucked with one finger (not with the nails or any sharp object) when definite pitch is required, or with all the fingers of both hands when a more reflective mood is required, as in a *glissando*. The characteristic sound is well known to be smooth and warm, rather than plosive and sharp, which makes this string instrument something of an original in music making.

The violin, viola, cello and double bass are in essence differently scaled versions of the same basic instrument. The principal differences are, therefore, in the range of frequencies they cover rather than in their musical quality. The construction and action within this family is almost identical, except for the gauge and length of the strings. The four strings are stretched between adjustable pegs and the tailpiece on the bodies of the instruments. The strings pass over, and firmly contact, the bridge, which couples them mechanically to the upper part of the body, called the 'belly'. The curved belly is connected to the 'back' of the body by a small soundpost. Resonance is provided by the vibration of these elements as well as by the vibration of the air in the two 'f' holes in the belly (Helmholtz resonance), all contributing to the instrument's timbre or quality. It can be seen from Fig. 22, which shows two examples of steady notes produced on a violin and double bass, that the overall response produces a sound rich in harmonics, with even the 7th and 8th harmonic levels being comparable with the fundamental. This fairly even distribution over many harmonics gives the violin its characteristic tone, especially at the higher reaches of its range, where it is incisive and arresting. Note that at lower fundamental frequencies, there are more harmonics but their amplitudes fall more rapidly, giving the violin a more feathery, resiny sound. The same spectral distribution is featured in the double bass (Fig. 22) where the contribution from the large body is more significant above about 100 Hz. Because of this it is important in reinforcing the bass portion of the orchestra. However, it is probably better known as a bass rhythm instrument in dance bands, when the strings are plucked.

The piano and harpsichord are both keyboard instruments, although in one case the strings are struck and in the other case plucked. It is this that accounts for the major difference in tonal quality between the two and not any important constructional difference. The harpsichord and grand piano are, in fact, quite similarly constructed. In the piano, a felt hammer strikes the

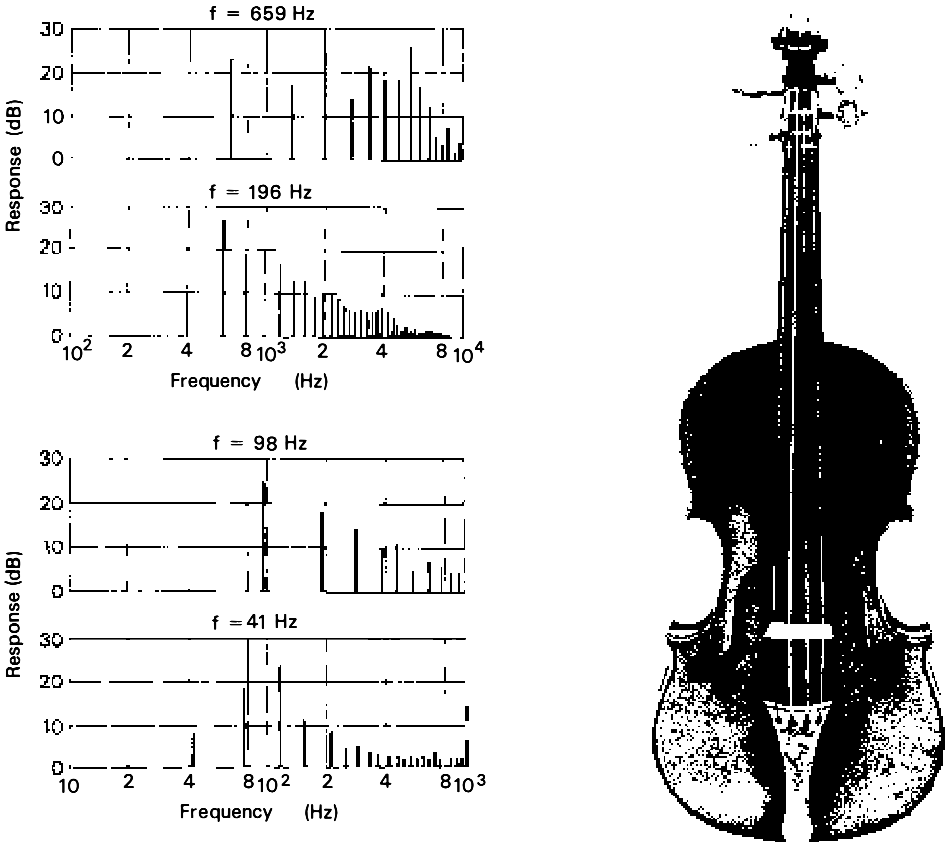


Fig. 22
The acoustic spectra of two notes played on a violin, above, and double bass, below, with photo of violin, fundamental frequencies shown. Notice the relatively high amplitudes of the harmonics in relation to the fundamental.

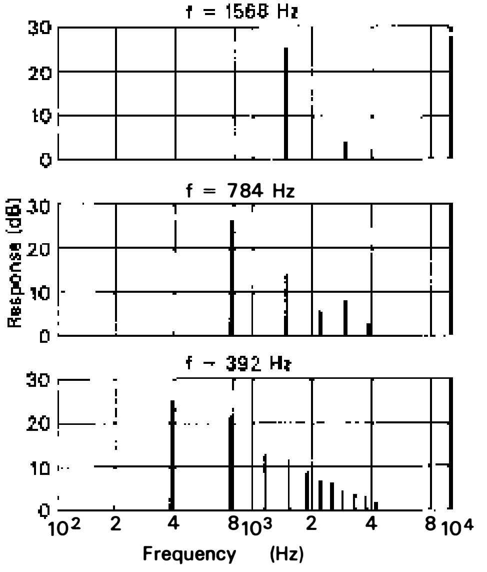
metal strings, which are closely coupled to a large soundboard, immediately behind them. The string/hammer mechanism is arranged vertically in the 'upright' piano or spinet, whilst in the larger grand piano it is horizontal. The grand piano is a more efficient radiator at low frequencies, and so its construction plays a more important part in determining low-frequency tonal quality. Some control of the instrument's tonality is effected by the intensity with which the notes are played. The actual shape and size of the soundboard and cabinet of the piano do not contribute equally to all notes, but provide a richer overtone structure in the lower register and a more resonant tone.

In the harpsichord the notes are plucked, so there can be little or no control of note quality by the musician. Sound energy is released from the strings more suddenly, so that the notes are shorter lived. Hence the initial, harmonically rich, sound dominates to give the harpsichord a sharper tonal quality.

Wind instruments

Another major section of the orchestra is composed of wind instruments. These instruments employ a combination of air and mechanical reed resonators as the primary vibrating elements, and are excited by blowing or pumping. The elementary air reed

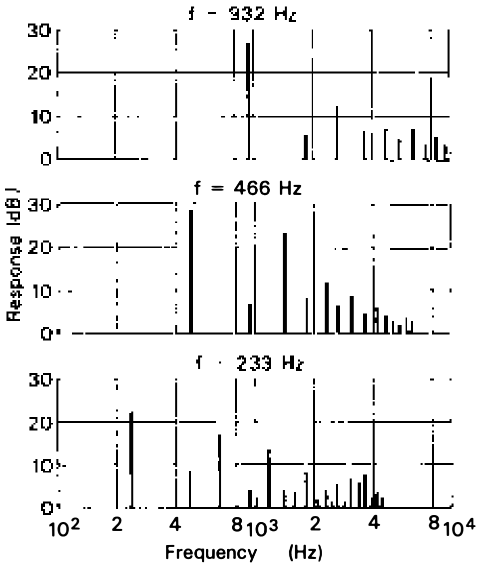
Fig. 23
The acoustic spectra of three notes of a flute, fundamental frequencies shown. At higher frequencies the note is fairly pure.



instrument (the recorder, for example,) is a simple resonant pipe whose frequency can be varied with the finger holes along it. When blown it produces an alternating air flow at the fipple hole in front of the mouthpiece which makes the pipe resonate. The flute is somewhat more refined, because it has keys to allow more rapid changes in pitch on the musical scale, however, it is still an 'air reed' instrument in that it is the movement of air in the mouthpiece that makes the pipe resonate. The flute produces a very pure tone in its higher register, though the pipe contributes more overtones to lower fundamentals (Fig. 23). The piccolo is very similar, except that its smaller pipe length puts it at a higher point of the musical spectrum. Sound radiation takes place at the pipe resonance frequency both from its open end and from the side holes.

More complicated wind instruments contain a mechanical reed system as the primary vibrator coupled either directly into the air, as in the mouth organ or accordion, or into an associated resonant pipe which adds further to the instrument's tonality, as in the clarinet or saxophone. The mechanical reed, usually made of bamboo, operates in a similar manner to the air reed and is excited into a sawtooth vibratory motion by the musician blowing. The reed radiates all harmonics of its fundamental, but these are further modified by the different size and construction of the clarinet and saxophone.

In the clarinet, the flared pipe is small and narrow, which means that it has a high acoustical impedance (a quality similar to resistance). The nature of this impedance is such that the higher harmonics produced by the reed do not excite the pipe, resulting in a smooth note (Fig. 24). In fact much of the energy of the clarinet note resides the fundamental, giving a clear, bright sound.



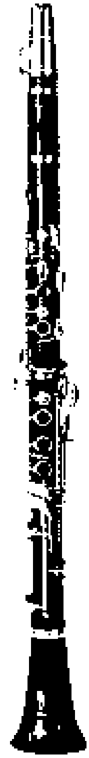


Fig. 24
The acoustic spectra of three notes played on a clarinet, fundamental frequencies shown. Notice the complex pattern of harmonic amplitudes at low fundamental frequencies.

In the saxophone, on the other hand, the pipe and flared mouth are much larger than in the clarinet, so that more of the higher overtones are transmitted to give the instrument a characteristically more 'reedy' tone, which is also full and rich owing to the greater sound output from its mouth. The frequency spectra of the saxophone do not show any definite pattern within the instrument's musical compass (Fig. 25).

The oboe, English horn and bassoon incorporate even more complicated primary vibrators, in the form of a double mechanical-reed system in the mouthpiece (Fig. 26). All these instruments combine with a resonant pipe which has keys to change its

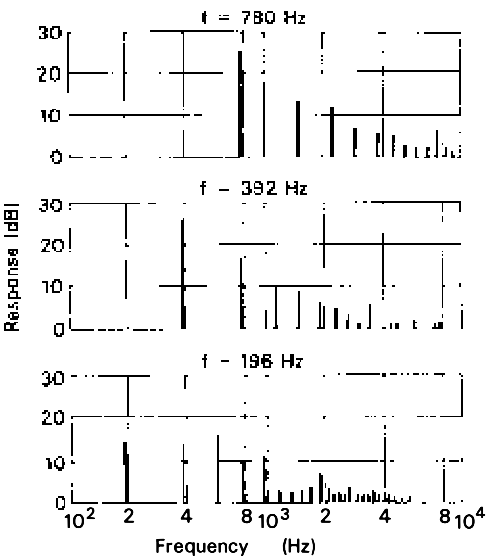


Fig. 25
The acoustic spectra of three notes of an alto saxophone, fundamental frequencies shown.

Fig. 26
The acoustic spectra of three notes played on a bassoon, fundamental frequencies shown. Notice how the harmonic pattern varies greatly with fundamental frequency.

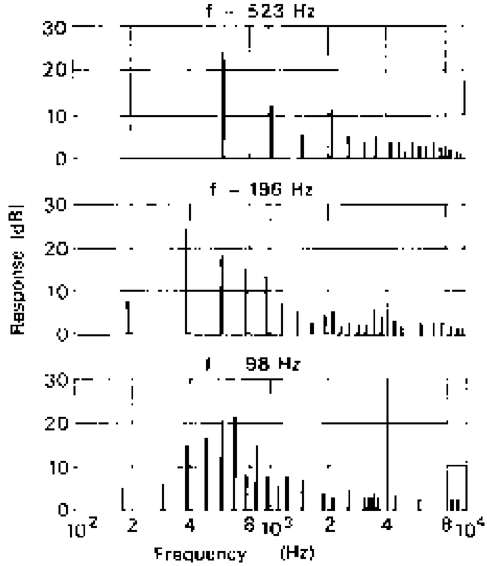
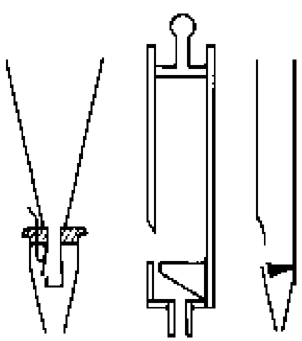


Fig. 27
Three different pipes used in the pipe organ: left, a metal mechanical reed pipe; centre, a rectangular, wooden air reed pipe; right, a round metal air reed pipe.



resonant frequencies in accordance with the musical scale. The tonal qualities of these instruments are more subtle and complex. For example, the oboe has the brightness of the flute but is also reedy, whilst the bassoon, which covers a lower register, is eerie or whimsical. All these sounds may be called upon by the composer to provide any shade of musical expression.

The organ: a special reed instrument

The pipe organ, the most magnificant of all musical instruments, has a history stretching back more than 2000 years. The pipe organ is almost a complete orchestra in one instrument with each note having its own separate generator. The sound generators are resonant pipes, either of the air-reed or mechanical-reed type, and there can typically be several hundred of them assembled in the instrument, many concealed from view. The instrument features several keyboards or manuals, each one controlling a particular row or rank of pipes (Fig. 27). Also there is a pedal keyboard, which controls pipes of lowest pitch.

The air-reed or 'flue' pipes may be of round, rectangular or triangular cross section, and of uniform or tapering shape. They are constructed of either wood or metal and are fed with an air supply from motorised compressors. The pipe proportions, materials and air supply pressures all have an effect upon pitch or tonal quality. Also, the organ employs metal reed pipes, which add further to tonal character by emphasising pitch and melody. By a combination of 'stops' and 'mixtures', particular pipes in a rank can be silenced or combined together, upon depression of a single keynote, to enhance tone colour even further.

The central keyboard or manual controls what is called the

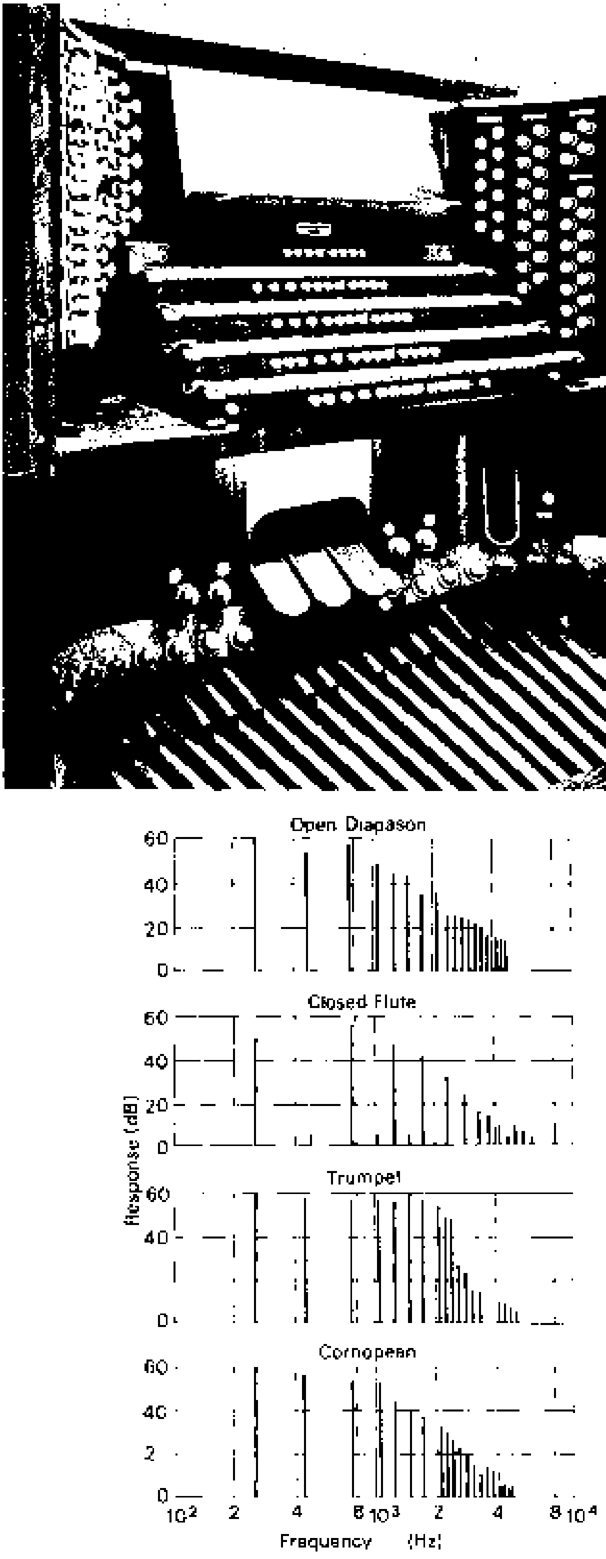


Fig. 28
Acoustic spectra of four different organ pipes of the same pitch and photograph of modern console.

'Great' organ, which consititutes the body of the instrument's sound, actuating the principal flue choruses which express the main theme of any music being played. The keyboard above the 'Great' is called the 'Swell' organ, whose pipes are enclosed within a chamber or swell box. A pedal control allows the player to change or swell the loudness of this group of pipes over a substantial range, adding further expressional effects. The Swell organ contains a variety of flue and reed pipes. The keyboard positioned below the Great is called the 'Choir'. This organ controls flue and reed pipes of highest pitch, and offers softer notes and mixtures, used for accompanying voices. In some organs a fourth keyboard is provided, above the Swell, called the Solo organ. This controls higher pressure pipes, used for highlighting particular themes or providing high power for musical climaxes. These four manuals, together with the pedal manual, combine to give the pipe organ tremendous tonal possibilities and sound power. Unique among the instruments of the orchestra, the pipe organ produces by far the highest power at the lowest audible frequencies – the largest pipes have a 'speaking' length of 32 feet (9·75 metres) and can create steady sound fields of 115 dB at 16 Hz over much of the auditorium. The spectrum of an organ pipe (Fig. 28), of either the air-reed or mechanical-reed type, is very rich in overtones, a great many of which have amplitudes comparable with the fundamental.

Brass and percussion instruments

The brass section of the orchestra is composed of wind instruments which differ from the woodwind group in that the pulsating action of the air is produced by the player's lips, stretched over the mouthpiece. The precise lip action and air movement is affected by the shape of the mouthpiece, which varies between instruments.

Lip reed instruments include the simple bugle, the trumpet, trombone, tuba and French horn. In instruments provided with keys, the length of the brass tube can be changed which alters the tube resonance in accordance with the musical scale. The coiled tube of which the French horn is formed, for example, is about 12 feet in length. Since the keys serve only to vary the tube length, sound radiation from brass instruments does not take place from side holes along the tube, only from the relatively large mouth or 'bell'. This is another difference between brass and woodwind instruments.

The trombone features a cylindrical u-shaped tube with a telescopic 'slide' allowing pitch to be varied in the same way that the violin family can vary pitch, by a continuous glide through the fundamental frequency range. Only these instruments can do this. The trumpet, trombone and French horn produce sounds which are rich in harmonics throughout their frequency ranges. The tone of the trumpet, for example, has brilliance and clarity.

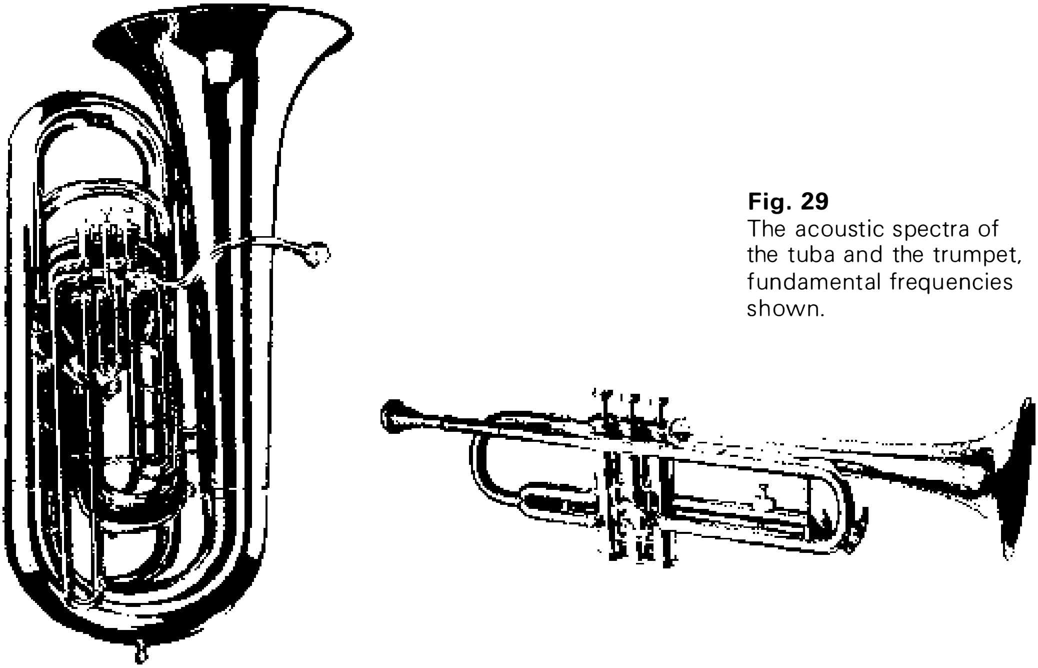
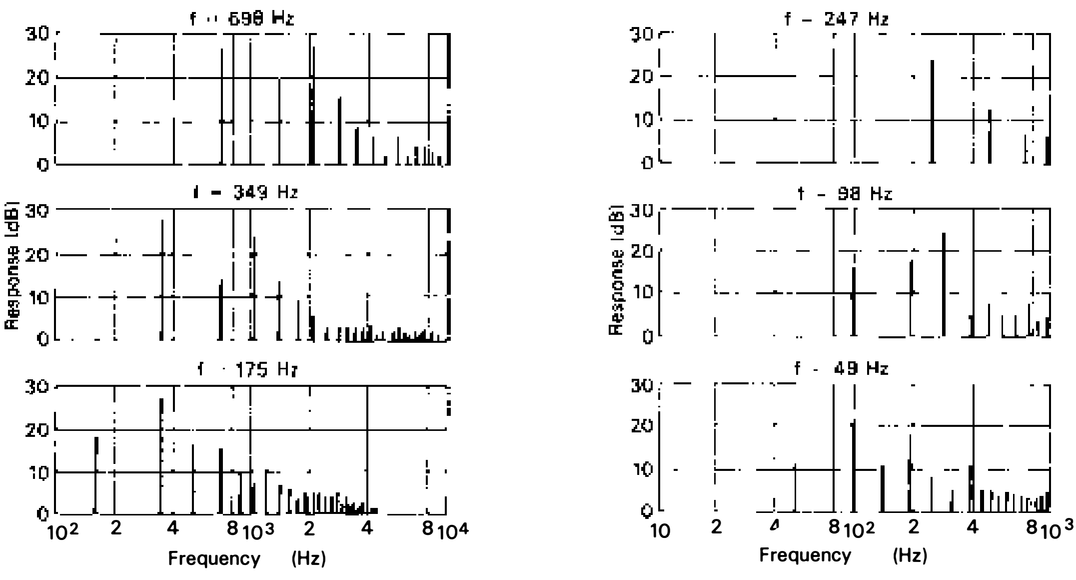


Fig. 29
The acoustic spectra of the tuba and the trumpet, fundamental frequencies shown.



It can be strident and also soft with some richness.

The tuba serves as the bass of the brass group. Since the principal output of the tuba lies in the narrow range from about 40 Hz to 300 Hz, its acoustic spectrum will contain only a few harmonics, especially for fundamentals above 200 Hz (Fig. 29). Its tone is therefore deep and smooth. The role of the brass in music varies with the type of composition. In classical music, the lip reed instruments, with the exception perhaps of the French horn, are not called upon to express subtleties in tonal quality. In modern jazz, on the other hand, the trumpeter may be a musician of exceptional brilliance and celebrity.

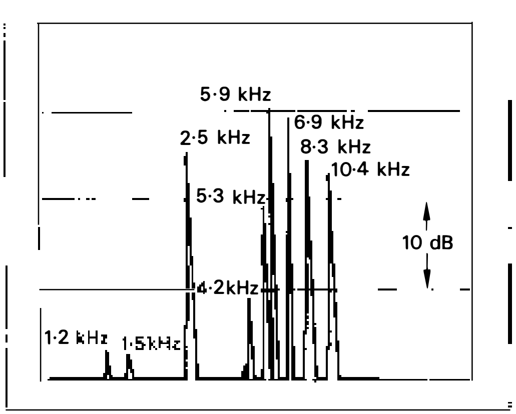
Percussive instruments are excited at a great many frequencies by some form of hammer blow. Some of them, for example the xylophone, chimes or tubular bells and kettledrum, have definite pitch whilst others such as cymbals, tambourine, bass drum and triangle have no definite pitch. In drums and cymbals the resonant properties of vibrating membranes and plates are employed. The frequencies at which membranes and metal plates respond are not harmonically related, and the energy in these overtones shows no definite trend. The spectra change quite markedly with time, and there is no fundamental note which persists long enough for the ear to latch onto. The explanation is that the energy at some natural frequencies is diverted into others, so that frequencies appear and subside randomly as time progresses (Fig. 30). The spectrum of a cymbal, for example, extends right across the audible range, and well above it.

The xylophone consists of a series of resonant bars whose length determines the pitch, whilst the kettledrum or tympani has a leather skin (membrane) stretched over a large hemispherical bowl. The air within the bowl forms a broadly tuned Helmholtz resonator which assigns a clearly audible pitch to the sound of the drum. Various types of kettledrum sticks may be used, for example felt, rubber or wood, each giving a different tonal quality. When a bell is sounded, its initial output has an ill-defined pitch, but almost immediately the higher partial tones die away and a note of clear pitch can be heard.

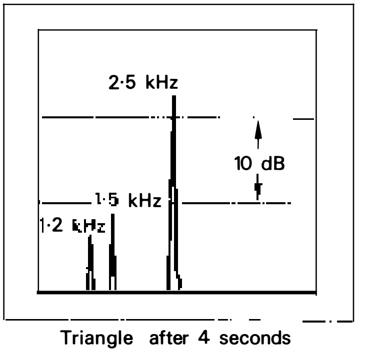
Fig. 30
The acoustic spectra of a triangle at the moment of being struck and four seconds later. Notice that the overtones are not harmonic and change with time.

Electrical musical instruments

Just as musical sounds can be produced by the traditional



Triangle



Triangle after 4 seconds

instruments of the orchestra–instruments which convert mechanical vibrations, initiated by musicians, directly into acoustic waves–so it is possible for musical sounds to be created by electrical means. Perhaps the most familiar instruments are the electric guitar and the electronic organ.

In the electric guitar, the body is not designed to radiate sounds directly, but couples its vibrations into an electromagnetic transducer which converts the sound waves into an electrical signal. The transducer is fitted behind the bridge or strings and the signal is amplified and reproduced over a loudspeaker. Since it is not required to radiate sounds efficiently itself, the body of the electric guitar is made more massive than an acoustic guitar, and is finished more ornately to give visual appeal. Modern electric guitars often feature electronic circuitry within the body allowing the artist to modify the form of the signal leaving the transducer. These include pitch change and tone adjustment, effected by control knobs on the top of the body (Fig. 31).

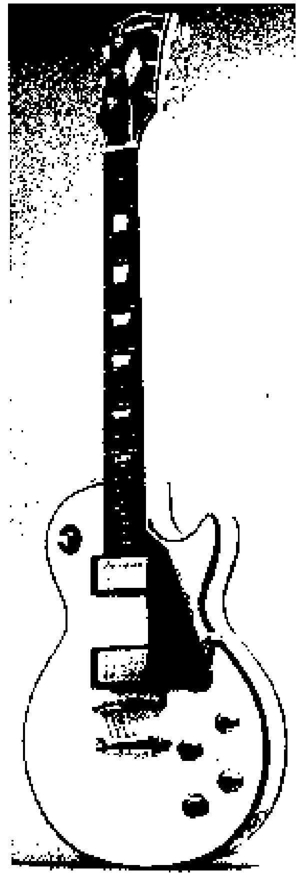


Fig. 31
An electric guitar showing controls which allow the player to make tone adjustments to an inbuilt amplifier.

Thanks to the revolution in electronic systems design due to digital techniques, developments in electronic organs in recent years have been remarkable. Basically, an electronic organ aims to recreate some of the sounds of traditional musical instruments by generating their sounds electronically and playing them over loudspeakers contained within the instrument. Some of the early electronic organs, produced about twenty years ago, gave only an approximation to the sounds of a pipe or reed organ, or did not even attempt to emulate its sound. Nowadays, however, the scope of the instrument is incomparably wider and it can easily stretch to recreating almost all of the instruments of the orchestra. Much more than this, all manner of signal processing is available, with transient attack and decay, vibrato, swell and tremelo effects, pitch transposition and automatic rhythm all under the musician's control. The fundamental notes are derived from digital frequency generators or 'clocks' operated by the keyboard (Fig. 32). The pulses from these generators are then fed to 'waveshaping' circuits to give the effects. These are operated by banks of special tabs or buttons near the keyboards.

One such instrument is the Aurora Classic, made by the Hammond company. This highly sophisticated instrument features two manuals, one relating to the 'upper orchestra' and allowing such instruments as the flute, clarinet, saxophone, oboe, violin, piano and harp to be simulated. The lower manual produces fundamentals of lower pitch and includes the sounds of some of these instruments and others like the cello. There is also a pedal keyboard giving such sounds as bowed and string bass, the bass clarinet and tuba, as well as organ pipe notes of lowest pitch. The instrument features more than twenty automatic rhythm sections for which the tempo, volume and tone balance can all be separately adjusted. Rhythms such as the waltz, swing, rock and

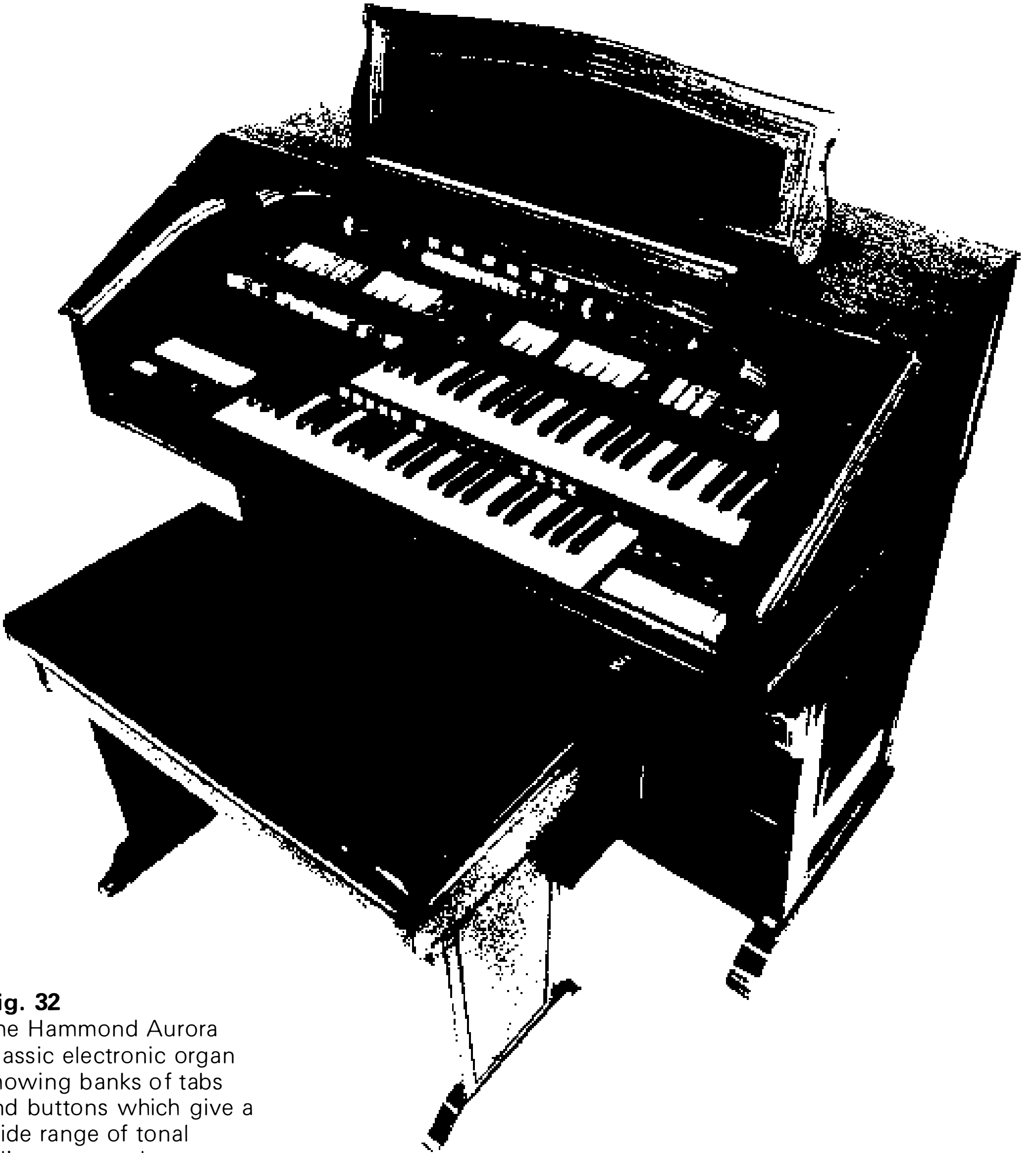


Fig. 32
The Hammond Aurora Classic electronic organ showing banks of tabs and buttons which give a wide range of tonal adjustment and expressional effects.

jazz rock can be used as a basis for composition. All these features are backed up by a vast range of effects designed to give the greatest flexibility in making music with the instrument. The electronic organ is not a toy, but an original and flexible musical instrument in its own right. The Hammond Aurora Classic, for example, costs more than £4000 (Fig. 32).

The human voice

The flexibility of the human voice is immeasurably greater than that of any musical instrument. The amount by which an individual can control the tone quality of his voice, and the degree of information that he can thereby convey, are truly immense. This

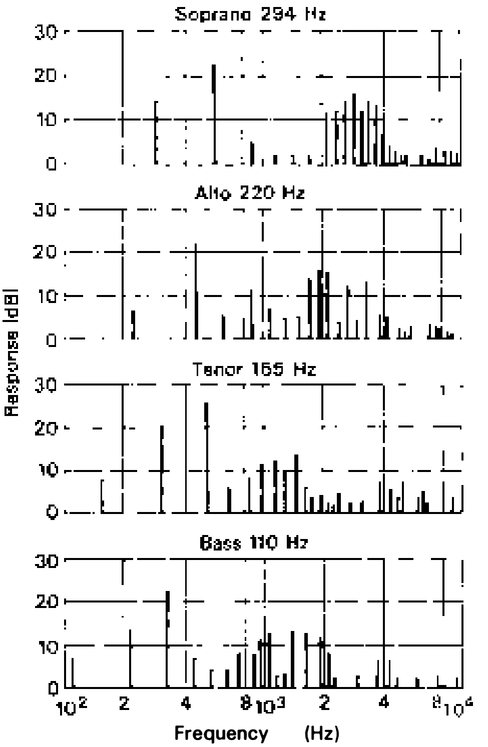
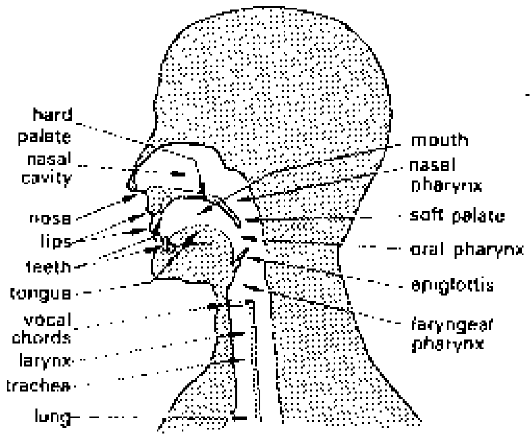


Fig. 33
The essential components which generate and modify the sound of the human voice and the acoustic spectra when the vowel 'ah' is sung by four singers at different points of the musical register.



might seem obvious for here we have not the sound of an instrument made by man, but the voice sounds of man himself!

The voice mechanism is shown in Fig. 33. It consists of three main parts; the lungs and associated muscles which provide a continuous flow of air between breaths; the vocal chords stretched across the larynx which convert this steady flow into a sawtooth vibratory motion and so constitute the primary vibrator, and a system of resonant cavities formed by the throat, tongue, mouth, lips and nasal cavity which impose formant qualities upon the basic voice sound. The action of any one can be varied at will to modify the loudness, pitch and tonal quality of words spoken or sung. The basic rate at which the vocal chords vibrate will vary from person to person. For a gruff voice it might be as low as 60 Hz, whilst for a shrill voice it could be 350 Hz. This is the basic pitch at which a person normally voices words, but it may be varied over about two octaves if desired, as might happen when singing.

Depending upon the particular sound being made, different parts of the speech mechanism are called into operation. For example, the vowel sounds a, e, i, o and u are determined principally by the tongue and lip positions, acting on the sound of the vocal chords, i.e. voiced sounds. Other sounds, for example ph, t, ch, require the tongue, teeth and lips for their formation, but not the vocal chords. Hence these sounds are said to be unvoiced. Another example of unvoiced sounds is whispering; although the resonant cavities of the nose and mouth are formed exactly as for normal speaking, there is no vibratory supply of air to them from the vocal chords, only a steady breathing noise. The nasal cavity is employed when sounds such as m, n or ng are made. In the English language, speech information is carried principally by the consonants, not the vowels. It has been shown that only one vowel sound need be retained for speech to remain quite understandable.

The acoustic spectra of a bass, tenor, alto and soprano singing voice when producing the vowel sound 'ah' are shown in Fig. 33. It is interesting to see that the form of the spectra are similar, except that they are shifted in frequency. These spectra were taken, not during the build-up or decay of the sound, but during a short interval of the 'steady', held sound. It would not be wise to conclude that these four persons have indistinguishable voices, except for pitch, because this information is very limited and the spectra shown take no account of the transient characteristics of the voices, which would certainly be different. The many subtleties of voice control, in terms of frequency, timbre and output, allow the human voice to express almost every shade of emotion, and permit the listener to distinguish when a particular person is speaking.

SOUND LEVELS AND AUDIO

To the discriminating listener the range of frequencies covered by an audio system is important. The limits of audibility are from 20 Hz to 20 kHz so one might conclude that any high-quality audio system must cover this range. Some would argue that this is an unnecessarily wide range, whilst others would say that musical sounds extend beyond this range. Before discussing these views, we might investigate the limits of frequency coverage for musical instruments.

Frequency range

When a musician speaks of the range of a voice or musical instrument, he means the frequency range of the *fundamental* frequencies which the voice or instrument can encompass. The range of human voices and musical instruments is shown in Fig. 34. These are shown both as a comparative line drawing and as a list of frequency limits for easy reference. By far the widest range is covered by the pipe organ, whilst the range of the kettle drum is only a few tens of Hz. If we exclude the pipe organ, we see that every possible fundamental note of music and speech can be captured if a frequency response range extending just a little above 4 kHz is maintained. An audio system with an upper frequency response limit of about 4 kHz would ensure that not a note is missed when reproducing most kinds of music and speech. What it would not do, however, is to convey the appropriate tonal

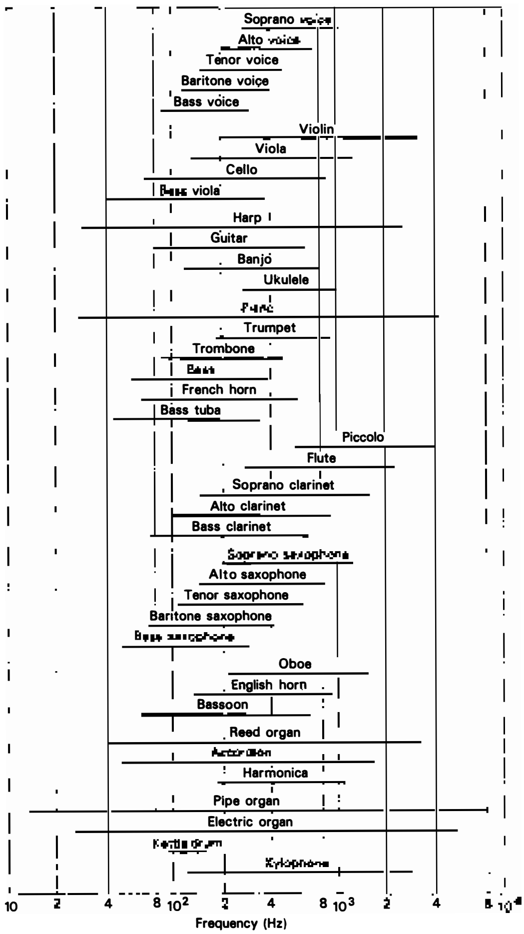


Fig. 34
The frequency range of fundamentals covered by different musical instruments and voices.

Lower Limits

	Hz
Pipe Organ	16·351
Piano	27·5
Contrabassoon	30·868
Harp	32·703
Reed Organ	36·708
Double bass	41·203
Tuba	43·654
Chimes	48·999
Bass Saxophone	51·913
Bass Trombone	55
French Horn	61·7
Cello	65·14
Kettle Drum	87·307
Bass Voice	87·3
Clarinet	110
Trumpet	146
Violin	196

Upper Limits

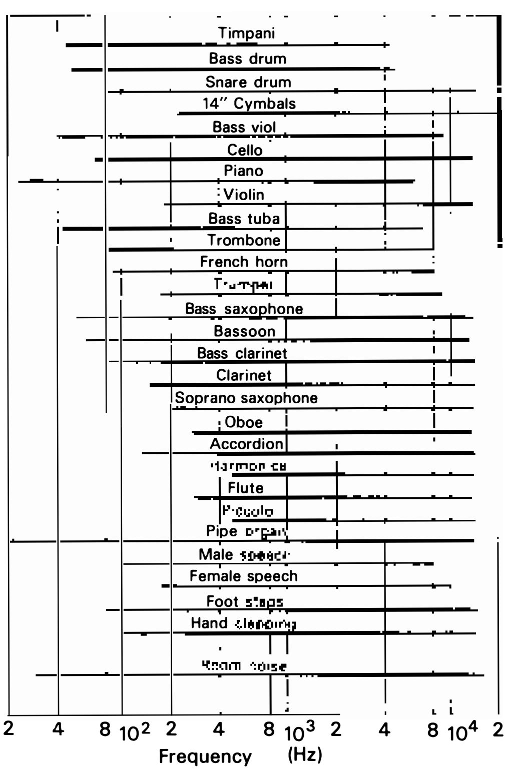
	Hz
Kettle drum	7,000
Bass Tuba	8,000
Singing voices	9,400
Harp	11,000
Organ	13,000
Xylophone	13,000
Saxophone	14,000
Cello	16,000
Violin	16,000
Cymbals	20,000

quality of the sounds, because it would suppress most of the overtones.

What would be helpful would be a list of the extreme limits of frequency which these voices and instruments may produce, which would include any sub-harmonics of fundamentals and every overtone. This information is shown in Fig. 35. Whilst the fundamental ranges of Fig. 34 place each instrument at some definite point of the musical scale, the total frequency spectra of Fig. 35 are very similar for most of them. Note how many instruments require an upper frequency limit of more than 10 kHz for faithful reproduction. In other words, it is quite common when listening to music 'live' for the ear to receive sounds of frequency well above 10 kHz. The pipe organ produces overtones which extend beyond 16 kHz, whilst the spectrum of cymbals extends beyond 20 kHz. The low frequency limits of Fig. 35 can be seen in some cases to extend below the minimum fundamental frequency limits of Fig. 34, and this is because these instruments generate sub-harmonics (i.e. fractions of the fundamental frequency) as well as mechanical soundings which would be necessary to their faithful reproduction.

This suggests that an audio system of the highest quality requires a frequency response range extending from 16 Hz to 20 kHz. There are very good practical arguments against design-

Fig. 35
The total frequency range required for perfect reproduction of various instruments, voices and other sound sources.



ing an audio system with such a low bass limit as this, and many high-quality systems do not extend this far down in frequency. It is not a difficult matter, however, to design a system with an upper frequency limit of 20 kHz, and many systems reach this limit, or exceed it. More is said about this in Chapter 3.

Power and dynamic range

Just as the total frequency span of music and speech is important, so is its maximum acoustic power. This information will indicate what will be required of the power amplifier in an audio system to faithfully recreate the musical dynamics. Table 8 lists the maximum acoustic power of a selection of the instruments in an orchestra or band, including those which contribute most to loud climaxes. The figures were obtained by asking the musician to play a note or chord loudly on the instrument, at some point in the lower part of its fundamental frequency range, where most of the energy in music resides. Using a reference power of 10^{-12} watts, and the same formula as that explained in Chapter 1, we

Table 8 The maximum acoustic power of a selection of instruments and speakers

Instrument/ Voice	Peak Acoustic Power *Watts*	Sound Power Level *dB*
Baritone voice	·02	103
Choir of 75 singers	2	123
Triangle	·05	107
French horn	·05	107
Flute	·05	107
Piccolo	·08	109
Double bass	·16	112
Bass tuba	·21	113
Bass saxophone	·3	115
Trumpet	·33	115
Grand piano	·45	116
Bass drum (36″ × 15″)	10	130
Trombone	6·5	127
Cymbals (15″)	9·5	129
Pipe organ	12·6	131
Orchestra (15 piece)	9·0	129
Orchestra (75 piece)	66	137
Small bookshelf loudspeaker	·1	110
Freestanding high power loudspeaker	4·0	125
Large open air 'pop' concert (over loudspeakers)	250	144

can give these powers a decibel rating, too. Note that these are the absolute maximum powers the instruments, either singly or collectively as an orchestra, can generate. These powers will vary, moment by moment, as the mood of the music changes, with a typical norm being about one-twentieth of the values listed (i.e. about 12–15dB lower). Hence during a piece of music an orchestra will generate an acoustic power which averages about 3watts, with occasional peaks of over 60watts and with softer moments of a few tenths of a watt or less.

The same is true of the peak acoustic powers possible from the loudspeakers. These powers cannot be achieved at every frequency, for example in the bass or treble ranges. Hence on wide-frequency music the peak powers fed to loudspeakers must be far more limited, especially for small speakers. The best loudspeakers have peak output powers approaching that of a full orchestra, and all are helped by the fact that when in use at home, they radiate into a far smaller room volume than a 'live' orchestra in a concert hall. Hence the sound energy per unit volume of the room, what is called the sound 'energy density', can be the same as in a concert hall with a far smaller power radiated. The maximum sound intensity in the home listening room can therefore be almost as high as it would be at a 'live' performance.

It is the *dynamic range* of music which most loudspeakers are unable to recreate. This was explained on p. 32 as the difference between the quietest and loudest sounds in music, and its figure is about 60dB. Because of the technical power handling limitations which are necessary for loudspeakers–but which do not exist for musical instruments–it becomes impossible for most speakers to recreate the highest power levels in music and at the same time render the quietest sounds at clearly audible levels, for example above background noise in the home. More is said about this in Chapter 4.

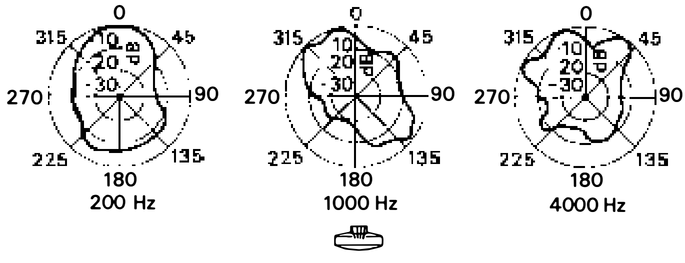
Directionality

As for any system which radiates sound, the intensity of the sound radiated from musical instruments and from a person's lips, varies with the direction of observation. This is an important point, for it can mean that the quality of the notes heard will change as the orientation of the instrument changes with respect to the listener. To see how sound is emitted from musical instruments in a horizontal direction with angle round them we refer to a circle diagram or polar diagram as it is called. Some reference point is chosen, as 0 degrees, and the relative acoustic output at all other angles is plotted with respect to this. Some examples are shown in Fig. 36. In these diagrams, the further out the solid line is from the centre of the circle, the greater the output of the instrument in that particular direction. The particular path taken round the instrument is indicated from the drawings shown, which are plan views, so that the polar diagrams are taken in a horizontal plane when the instruments are positioned as shown.

String instruments in the violin family have very complex sound radiation characteristics. At lower frequencies the belly and back of the body vibrate in unison, for example at 200 Hz the two halves of the polar diagram are similar. At higher frequencies, however, the body is a very complex vibrator, as shown by the irregular shape of the diagram. This irregularity implies that the two halves of the body move quite independently. The same is true for larger instruments in the violin family, except that they exhibit complex behaviour at lower frequencies.

Violin

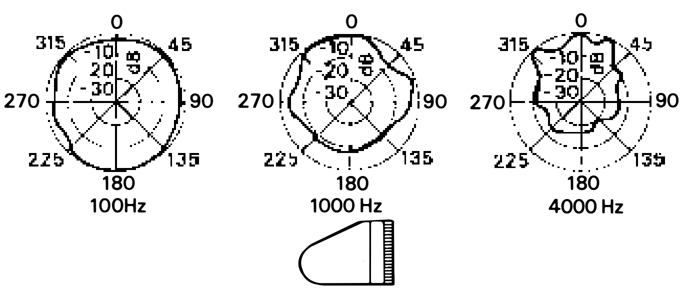
(a)



The open lid of the grand piano affects its high-frequency polar response. The reference axis here is in the direction of the open lid, which is always pointed towards the audience. This is not surprising, for it can be seen that above 4 kHz almost all of the acoustic output of the instrument is confined within about 45 degrees either side of the open lid. The polar diagrams of the grand piano show fairly good symmetry, even though the instrument is not of symmetrical shape. Hence the sounding board and cabinet vibrate 'all of a piece'.

Piano

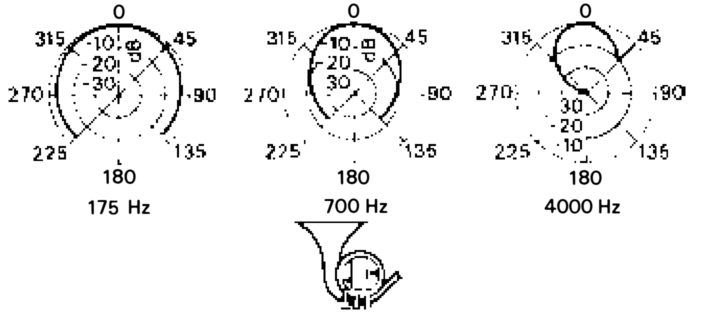
(b)



Wind instruments such as the clarinet, saxophone, trumpet and tuba have quite smooth polar diagrams over their entire frequency ranges. This follows from the fact that radiation from the mouths of these instruments dominates, which of course are simple circular shapes. By the familiar principle explained in Chapter 1,

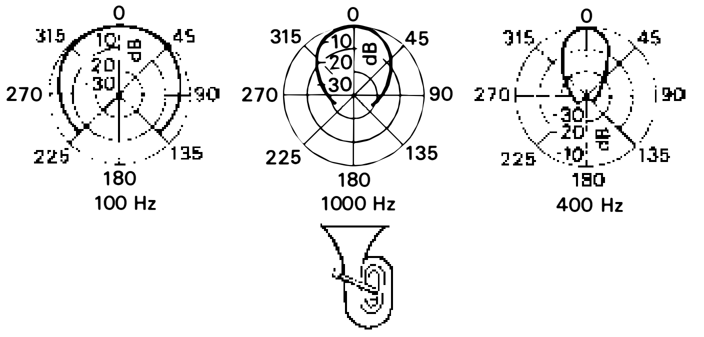
French Horn

(c)



Tuba

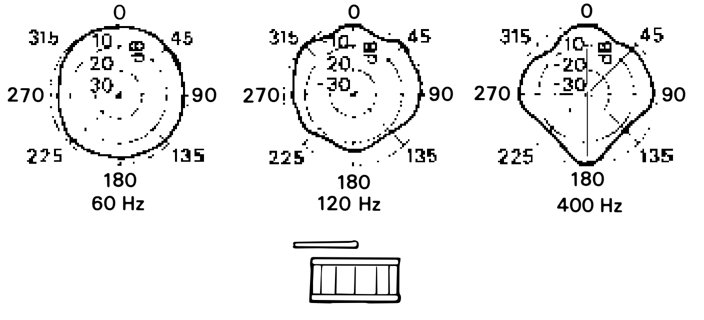
(d)



these instruments will exhibit more confined polar characteristics (i.e. will become more directional radiators) when the size of the open mouth becomes comparable with the wavelength of the sound it radiates. Examples are the clarinet above 4 kHz, the alto saxophone above about 2 kHz and the tuba, whose large 'bell' makes it directional above 400 Hz.

Drum

(e)



The directional characteristics of the human voice are of special interest, because of how they affect an intelligible reception of words. At low frequencies, the sound is almost omnidirectional, and even at the uppermost frequency limits of speech, there is very little discrimination over an angle of 90 degrees to either side, i.e. between extreme left and right. The loss of high-frequency

Human Voice

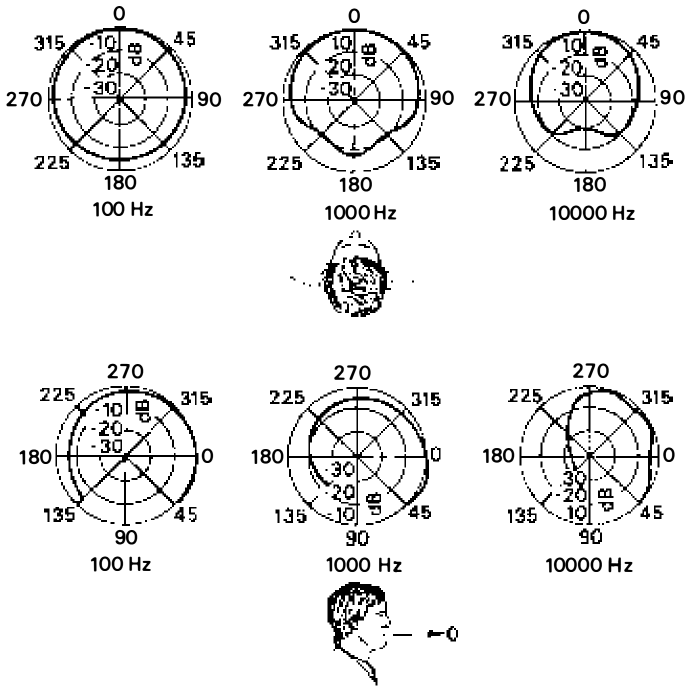


Fig. 36
The directional characteristics of some musical instruments and the human voice. The polar diagrams are horizontal and taken clockwise around the instruments when positioned as shown, except for the last diagram for the voice which is taken clockwise around the head in a vertical plane.

detail beyond this range is considerable. The vertical polar diagrams show how important it is for a speaker not to turn his head downwards when speaking to a large audience. These results are in accordance with experience. If a person moves his head away by more than about 50 degrees his voice becomes less intelligible.

Now vowel sounds are voiced at the basic speaking pitch, i.e. at low frequencies, which diffract around the head and indicate more of the character of the voice regardless of direction. The consonants, on the other hand, contain more high-frequency information, so that these are directed principally forward and carry most of the intelligence of speech.

These results show that for a clear reception of the subtle overtones of music and the intelligence detail of speech, the listener needs to have a clear, direct path between his ears and the sound sources. When listening to reproduced sound over loudspeakers, the listener needs to be within an angle of 30 degrees or so of the axes of the loudspeakers, especially at high frequencies, where both the reproduced sounds and the loudspeakers themselves become more directional. More will be said of this in Chapter 5, but first let us consider the hi-fi equipment itself.

3
Audio Systems

Chapters 1 and 2 outlined the basic principles of sound and music that are necessary to any study of sound reproduction in the home. Our purpose in this and the following chapter is to look more closely at the individual components of a home audio system.

SOUND REPRODUCTION SYSTEMS

There are basically two types of sound reproduction system: direct transmission and storage or recording types. The simplest form of direct sound system requires only a microphone to convert the original sound into an electrical signal, an amplifier to raise the level of this signal and a loudspeaker to change the amplified electrical signal back into sound waves. The whole system is connected by wires, so that the listener is confined in time and proximity to the original sound source. Examples of such a system are the public address installation in a theatre or the sound reinforcement equipment used by musicians on stage. The use of a radio link, in place of wires, removes the proximity limitation, i.e. radio transmission allows the audience to be of any size and located at any place with respect to the original sound source. Hence communication on a massive scale is possible if we use a radio system. This ability to communicate on a massive scale is what hi-fi is all about (quality considerations apart).

A hi-fi system goes one step further and removes the time restriction because it places an intermediate recording-storage medium between the original sound and the listener. The media most widely adopted today are tape (open reel and cassette) and vinyl disc, i.e. the LP record. High-quality sound recording on standard disc or tape material means that music can be heard at any time, in any place in the world, as often as the listener desires. Hence a modern hi-fi system is a perfect communication medium.

Whilst new technologies are beginning to appear in the recording studio, the type of audio equipment available today for domestic use will remain essentially as it is for many years to come. Indeed, some of the links in the sound recording and reproducing chain, for example the microphone and the loudspeaker, are never likely to change from their present form.

Transducers Transducer systems (loudspeakers and microphones are examples) convert sound waves into an electrical signal, or vice versa. There are similarities between microphones and loudspeakers, but one important difference is the much larger sound power which the loudspeaker is required to handle. Hence great accuracy is required of a microphone, whilst both accuracy and efficiency are required of the loudspeaker.

Microphones, however, are not normally regarded as part of a hi-fi system because they are encountered in the first stages of the sound recording process, i.e. in the recording studio or the concert hall. Microphone performance quality is a vitally important part of good sound recording, however, as any recording engineer will tell you.

Another transducer of special interest to the home listener is the record pick-up, and this is explained later in the present chapter. Advice is also included on how to match different pick-up arms to the pick-ups most generally available today. We shall look, also, at the recording media, the disc record itself and the tape material and cassettes. Finally, I shall give examples of audio equipment in different price categories which will match well into a complete home system. Since loudspeakers are such an interesting part of hi-fi, a separate chapter, Chapter 4, is devoted to them.

Before considering the audio amplifier we should discuss the distinction made in recent years between analogue and digital hi-fi systems.

Analogue and digital

An analogue system is one where every link in the sound reproduction chain – the record itself, the pick-up, the amplifier and the loudspeaker – aims to preserve exactly the sound information as it was recorded. In other words, the audio reproduction system processes an exact replica or analogy, of the original sound (loudspeakers and microphones, therefore, must be analogue devices). Exactly the same is true in the recording process. Every link from the studio microphone through to the special cutter head which cuts the groove on the LP record, aims to preserve the original sound precisely. We say that an analogue recording and reproduction system is a system of high resolving power, because it is designed to preserve every tiny detail of the original sound. Such a system is shown in Fig. 37. Because the familiar analogue sound recording and reproduction system has such fine resolving power, it is sensitive to degradations which can occur along the path between the original sound and the listener.

These can arise from inaccuracies in the studio microphone

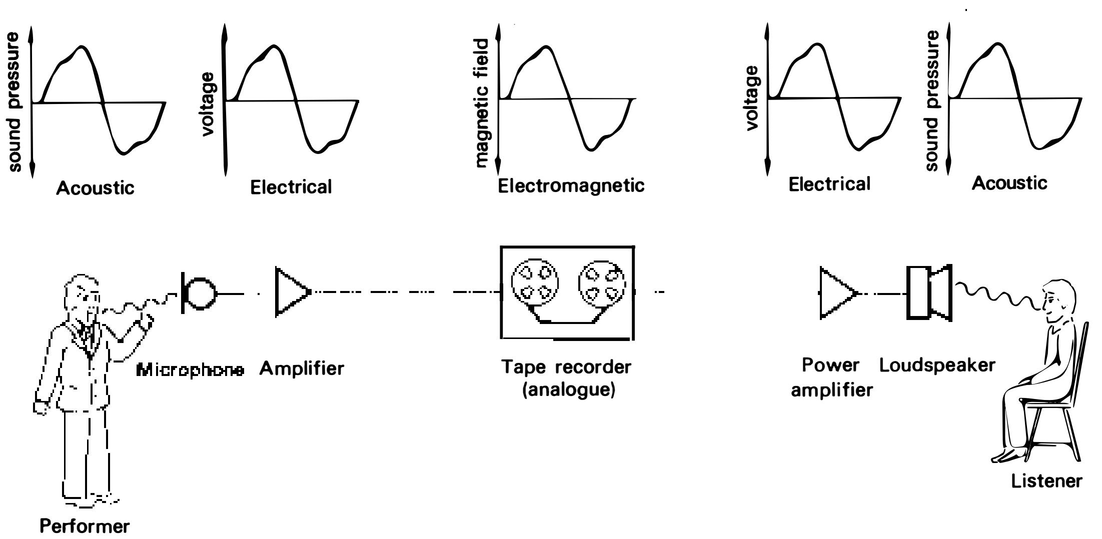


Fig. 37
The traditional analogue sound recording and playback system. The original signal is preserved at every point in the system.

or in the operation of the disc cutter head, and in 'noise' from the recording amplifiers or the master tape recorder, and these will be passed on to the final recording. Similarly, in the reproducing chain noise and distortion will be contributed by the audio amplifier, the pick-up and the loudspeaker. Small speed variations in the turntable, too, will degrade what is finally heard, because the analogue system must be capable of detecting such small deviations. More will be said of these problems later. Over the years, improvements in electronics design, in the fine mechanics of transducers such as pick-ups and loudspeakers, and in the engineering tolerances of record turntables and tape recorder mechanisms, have led to performance standards in traditional analogue audio systems which are of the very finest. The very best domestic audio systems of today are extremely accurate and can satisfy the most critical listener, although they can be very expensive.

Since some of the problems in sound reproduction can never be overcome completely–noise and distortion can never be reduced to zero–it is natural to wonder whether we are not now approaching the limits that can be achieved, cost effectively, with present technology and traditional analogue methods. Readers will be aware that words such as 'digital' or 'digital technique' have been bandied about in the past few years, though there will be many who are not aware of the fundamental differences between the familiar analogue technique and the new digital one. With the recent advances which have been made by the electronics industry, particularly in the field of digital electronic devices, it has become possible to process an electrical signal far more accurately, however complex the signal may be. Also, the costs of digital processing have, as the technique has developed, become steadily lower. It is true to say that digital

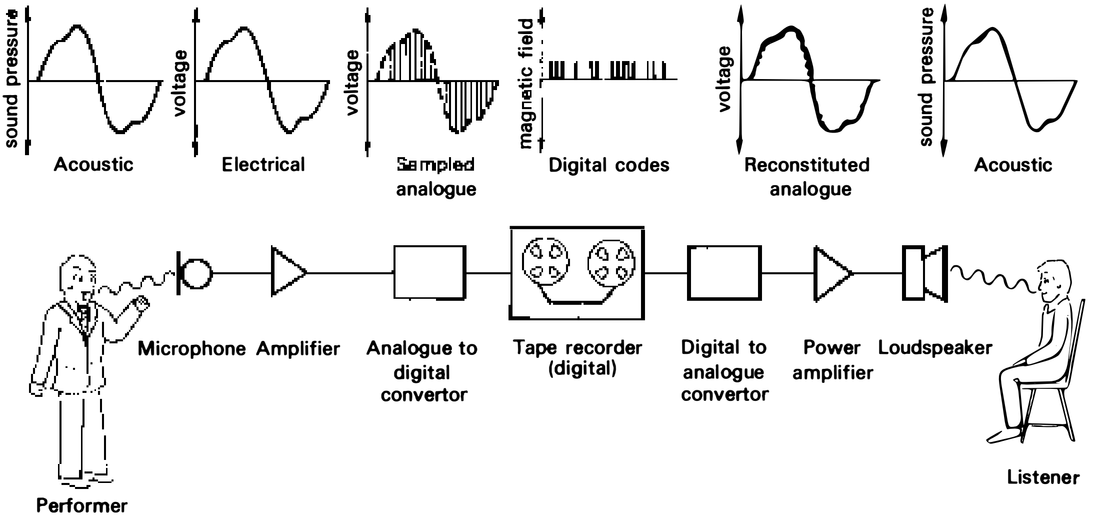


Fig. 38
The new digital sound recording and playback system. The original signal information is changed into a digital code at the recording stage, which makes it less susceptible to noise and distortion.

electronics has enormously expanded the applications to which electronics may be put, both industrially and in the home.

Digital techniques have begun to be applied in the sound recording process. Some of the LP records currently available (and mentioned in the music list in Chapter 5) were made with the use of a digital tape recorder in the recording studio. Briefly, a digital system of sound recording converts an audio signal from its exact, analogue form, into a special code, called a binary code. The binary code is quite a simple one although it can still accommodate all the necessary information contained in the audio signal. A diagram of a digital recording chain is shown in Fig. 38. The signal begins and ends in analogue form, as it must for our ears to recognise it, but in the tape recorder it is converted into digital form (analogue to digital conversion), in fact into a binary code of discrete numbers or digits. Hence the signal which is recorded onto tape is not continuous, as an analogue signal, but takes the form of particular, discrete values. In this form, the signal is far less susceptible to noise, distortion and speed variations occurring in the tape recorder. In fact a digital recording system is more immune from these problems because it does not attempt to preserve the exact, continuous form of an audio signal.

It is necessary, of course, for the digital system to carry *sufficient* signal information so that after it is converted back into analogue form (digital to analogue conversion) the result is indistinguishable from the original sound, but the limitations lie only in the electronics of the system, not in the tape medium or in the mechanics of the tape recorder. There is no problem here, however, since digital electronics is an area where there is still enormous opportunity for technological improvement. In fact experience has shown that even the first generation of digital

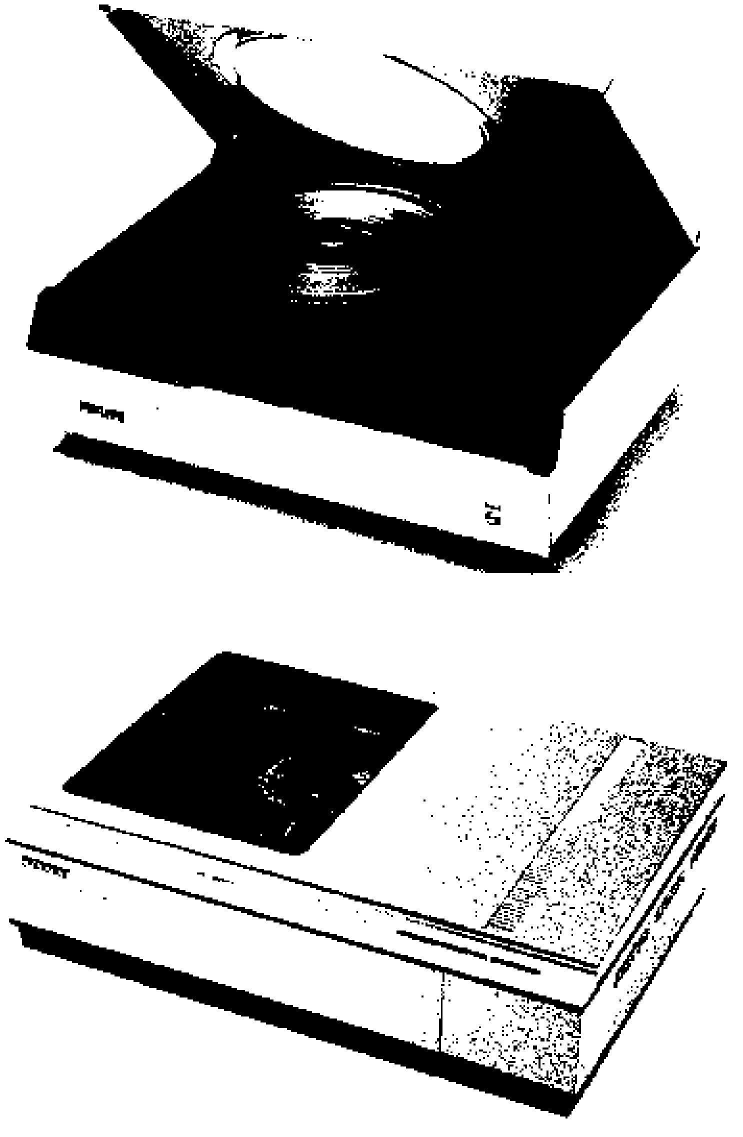


Fig. 39a
Two examples of digital disc record players of the future by RCA and Philips.

systems are as good as some of the best analogue systems which have been evolved over many years. Digital tape recorders, of which several have been developed by the broadcasting organisations and record companies, have performance qualities which are far superior to the best analogue studio recorders.

It would, of course, be logical to extend digital recording techniques to the reproduction chain, so that the signal remains in digital form until just before it reaches the loudspeaker. Whilst a great deal of research is currently being directed towards this end, there are, as yet, no domestic record players or tape recorders available which operate digitally. A digital home audio system

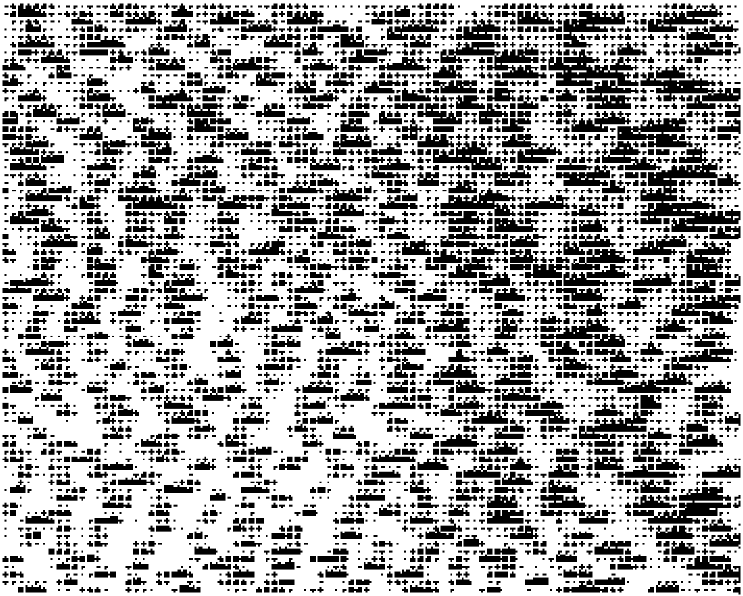


Fig. 39b
The pulse code modulated surface of a digital audio disc.

must be carefully considered before it is introduced for general use. For example, the system must be standardised throughout the recording industry, just as analogue LP records, turntables and pick-ups have become over many years past. Unless this is done, public interest will not be sustained and the technique will not become established–as was the case with four-channel disc-recording methods. There is little doubt that one day, the standard technique for sound recording and reproduction will be a digital technique. Its precise form is as yet undecided, and it will almost certainly be ten years before we see a generally agreed system become adopted. It could even be longer, that will be up

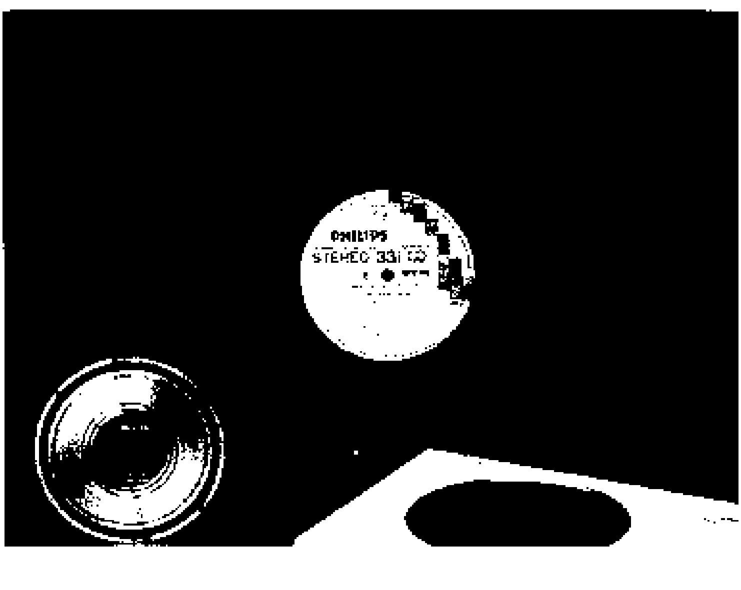


Fig. 39c
Comparison of a familiar LP record with a digital audio disc of the future

to the recording industry, meanwhile the analogue LP record has a history extending back more than thirty years. It could comfortably have just as long and healthy a future if the new technology is not properly managed. As a matter of interest, certain of the major companies have shown examples of what digital record players and discs for home use could be like. Two of these are shown in Fig. 39.

AUDIO AMPLIFIERS

An amplifier is a device for increasing the level or amplitude of a signal. Readers will be familiar with the term amplifier applied to power amplification necessary to drive loudspeakers but in fact amplification is required in many parts of a sound reproduction chain and all pieces of electronic equipment–tape recorders, cassette machines, tuners, etc.–have several stages of signal amplification. Quite often a signal is amplified then reduced (attenuated) many times before reaching its final destination. This is done so as to minimise noise or distortion problems at points where they are more intrusive.

An audio amplifier in the home first raises tiny signal voltages obtained from the pick-up, cassette deck or tuner, to levels where they can be adjusted, say by the tone controls or filters. Then these signals are turned into high electrical power for feeding the loudspeakers. The two tasks are sometimes done by separate pre-amplifiers and power amplifiers, or the two may be combined in what is called an integrated amplifier. The output power varies greatly between different amplifiers; for a small budget audio system the output power will be about 20 watts per channel, whilst a medium output power would be about 50 watts per channel. In audio systems of the highest quality which can deliver true dynamic range and musical peaks to the loudspeakers, an output power in excess of 80 watts per channel would be required. An example of a modern audio amplifier which would suit most domestic requirements is shown in Fig. 40.

The audio amplifier is the strongest link in the reproduction chain being, as it is, a wholly electronic device. However, there are a number of problems that can occur with an audio signal even when it is in electrical form, as in the amplifier, tape recorder or tuner.

What are the misfortunes that can befall a perfect signal to degrade its quality? Sadly they are many, and often inter-related in such a way that only a compromise can be achieved.

Noise

Noise is a serious offender. This rather loose term covers a great many different phenomena but basically noise is any signal which

is not related in time to the original signal but occurs in addition to that signal. Noise can be random, such as the characteristic noise of recording tape or the inter-station noise of an FM tuner. This is commonly referred to as 'white' noise, because of the analogy with white light which, like noise, has a wide frequency spectrum. Some forms of noise are cyclic or repetitive such as mains frequency hum or impulse noise from a car engine. Noise can be generated within a sound reproduction chain, or can be picked up from an external source and amplified along with the wanted signal.

Fig. 40
A modern audio amplifier in the middle price range with medium output power and typical facilities.

It is very difficult to design equipment which can differentiate between noise and wanted signal and even more difficult to separate the noise from the signal once it is there. There is no system at present which can remove noise from a signal without altering the signal in some way although for certain specific types of noise there are noise reduction systems available which can give a definite improvement to a signal in relation to noise; the Dolby system discussed later in this chapter is an example. Filters on the amplifier–scratch, rumble, etc., can give a subjective improvement to a noisy signal but only at the expense of frequency response and often only make the situation worse.

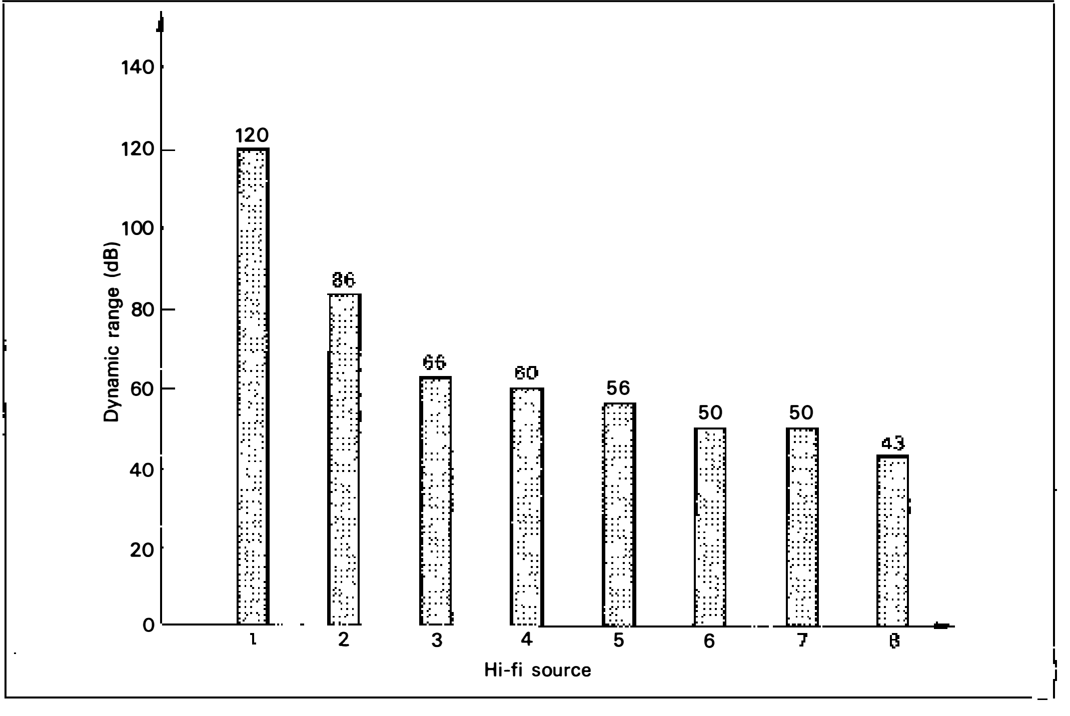
The only effective way of dealing with noise at present is to try to prevent it getting into the signal in the first place. Some types such as tape noise are determined by physical properties of the medium and no amount of careful design will reduce it below a certain level. Also as far as amplifiers are concerned, the ultimate noise performance is limited by the physical phenomenon of thermal noise produced by its resisters at room temperature. However, noise produced by external sources can often be reduced or eliminated by careful circuit design and construction and the use of electomagnetic and electrostatic shielding.

There are certain types of noise which occur only in the presence of a signal and in some cases it is difficult to say whether the effect should be called noise or distortion. A good example of this is what is known as modulation noise on tape recording. Whereas normal tape noise (also called zero-modulation noise) is constant regardless of input signal, modulation noise *only* occurs when a signal is present and is therefore partially masked by the signal itself. However modulation noise is quite distinctive, giving the signal a rather granular sound and is subjectively more disturbing than the constant hiss of zero-modulation tape noise.

In hi-fi, noise is generally specified as a −dB figure below a stated signal level or as a ratio in dB. The ratio between the noise level of a system and the maximum level that it can handle is called the dynamic range, and is also quoted in dB. This is illustrated in Fig. 41 for some of the components of a home audio system.

The measurement of noise is often quite complex due to the random nature of the quantity but generally broadband noise such as tape hiss is measured by integrating the noise signals over the entire audio spectrum and is usually given an rms (root mean square) value. (This quantity is used to indicate the total energy in the noise signal.) Also weighting curves are commonly used to give a more favourable noise reading. Unfortunately there is little correlation between the various weighted measurements and the so-called 'flat' measurement. It is therefore difficult to compare specifications of equipment when different measuring techniques are involved. However, as a rule of thumb, weighted measurements tend to be about 6 to 10dB better than

Fig. 41
The dynamic range that can be obtained from each link in the audio chain.
1 Audio power amplifier
2 Audio pre-amplifier
3 High-quality loudspeaker
4 High quality tape recorder
5 High quality disc record
6 Dolby cassette recorder
7 F.M. tuner
8 Cassette recorder



unweighted. It is generally accepted that 60dB unweighted S/N (signal/noise) ratio is a good figure to aim at for a hi-fi system.

Distortion

Distortion is another widely used and misused term which covers a multitude of sins. Unlike noise, distortion is invariably related in time to the input signal and because of this well-defined relationship distortion can often be reduced to insignificant levels using various circuit techniques, the most important being the use of negative feedback. Reduction of distortion in transducers is far more difficult and usually the distortion generated by a loudspeaker is at least 100 times worse than amplifier distortion.

The most commonly specified form of distortion is Harmonic Distortion. This can consist of several harmonically related distortion products which are often specified collectively as total harmonic distortion (THD). If a frequency which is the 2nd harmonic of the fundamental is added to the original signal, then the shape of the waveform will be changed. The exact nature of the change depends upon the amplitude and phase of the harmonic in relation to the fundamental, as is the case in music. This also applies to any other harmonic although in natural sounds generally only the lower-order harmonics (2nd, 3rd, 4th, etc) have sufficient amplitude to seriously affect the appearance of the basic waveform. It is this change in the shape of the waveform that causes the sound to be distorted.

Because hi-fi systems are usually reproducing more than one frequency at any time, another type of distortion is encountered. This is called intermodulation distortion and is most severe when two or more signals are being passed through the same channel although intermodulation of harmonic products also occurs. Intermodulation distortion is not harmonically related to the fundamental signal and so tends to be more disturbing to the ear.

Other types of distortion have gained prominence recently and are of a more subtle nature because they only occur under certain transient conditions and therefore do not show up with the more traditional 'steady state' measuring techniques. These phenomena have been given various names such as transient intermodulation distortion (TIM) and slewing induced distortion (SID).

Distortion in general is caused by some form of non-linearity in a system. In other words the output does not faithfully follow the input. This is illustrated in (Fig. 42) with the case of an audio amplifier. Some forms of distortion are proportioned to the input signal level whilst others are more or less constant and therefore appear worse at low signal levels. An example of this effect is found in crossover distortion which is due to mismatch in the output devices of certain types of audio amplifier.

There are many ways of measuring and specifying distortion each of which produces very different results. Fortunately

Fig. 42
Showing how a signal is distorted when the 'transfer curve' for an amplifier is not straight.

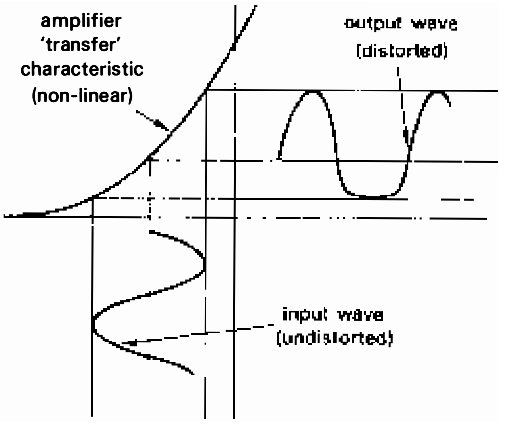
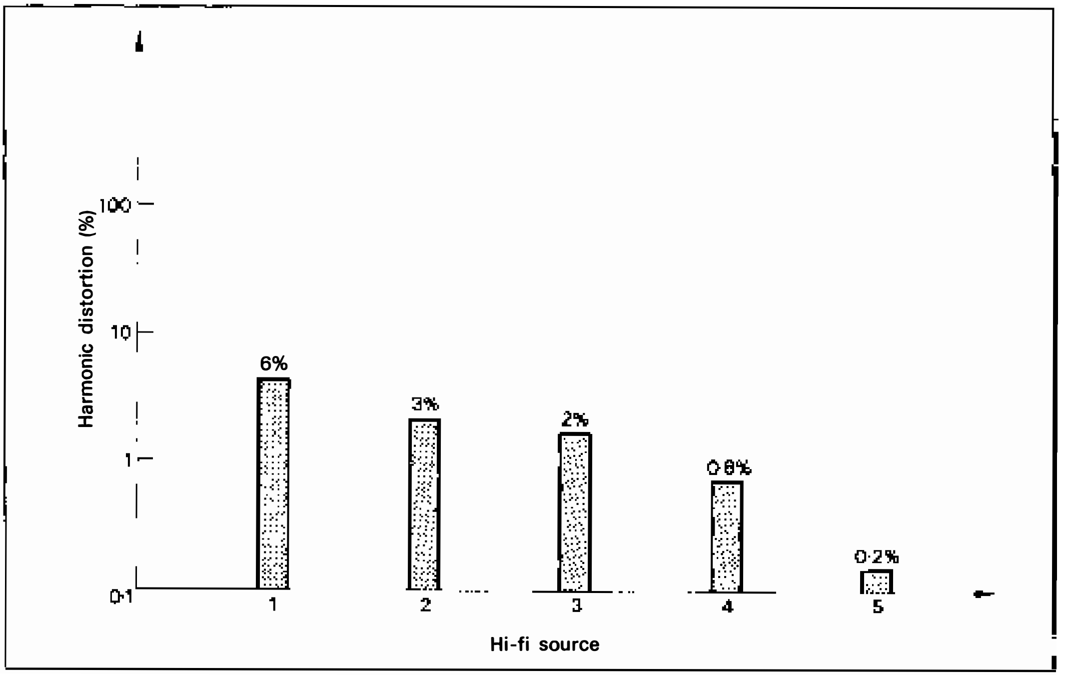


Fig. 43
Harmonic distortion produced by each link in the audio chain under normal listening conditions.

1 High quality loudspeaker
2 Pick-up cartridge
3 Domestic tape recorder
4 Studio tape recorder
5 Audio power amplifier

common sense prevails and two or three types of measurement are being standardised by most manufacturers. (There are indeed international standards relating to most aspects of hi-fi measurement.) Total harmonic distortion (THD) is the most common and this is usually specified as a percentage related to a given maximum output value at a given frequency. Intermodulation distortion (IM) is also often given and there are at least two different standards in common use. A figure of 0·1 per cent is considered good for an audio amplifier.

It is difficult to quote an acceptable level for distortion as some types are far more disturbing to the ear then others. Also, as mentioned earlier there are vast differences in the levels of



distortion generated by different parts of a system. Figure 43 gives a rough idea of the range of levels that can be encountered. It is interesting to note that, whilst in sound reproduction certain harmonic distortion components of an audio signal are more noticeable than others, there is no such distinction in music. Thus, we are not more sensitive to these particular harmonics when they are present in a musical note heard 'live' in a concert hall.

Frequency response

Frequency response is the ability of a system to reproduce all frequencies at a constant level (assuming the input is constant). In fact it is not necessary or desirable to have a frequency response or bandwidth that is many times greater than the response of the human ear and wide bandwidth is invariably obtained at the expense of some other parameter such as noise or distortion; 20 Hz to 20000 Hz is considered to be the normal audio bandwidth although very few adult persons can hear up to 20 kHz. It is interesting to note that many manufacturers put undue emphasis on frequency response, a typical example being domestic open-reel tape machines which are often specified to 20 kHz or higher at a speed of 19 cm/s (0·19 m/s in SI units) whereas professional studio recorders may only be specified to 15 kHz at the same tape speed! The important thing to consider is the design compromises that have had to be made in order to achieve an extended frequency response and the ability of the equipment to maintain this specified response during normal use.

In most cases the extent of frequency response is not so important as the overall flatness or uniformity of response. Sudden dips and peaks in a response are particularly to be avoided as these are accompanied by severe phase changes which have a clearly audible effect on the sound quality. This is especially true of loudspeakers, as we shall see in the next chapter. Frequency response is specified as the range of frequencies which are reproduced within stated limits, i.e. 20 Hz – 18 kHz ±, 3 dB under specified operating conditions and referred to a given frequency, usually 1 kHz. Some examples of frequency response are illustrated in Fig. 44.

Crosstalk

Crosstalk or channel separation is another problem which can occur in any part of a stereophonic or multi-channel system and is caused by part of a signal from one channel 'leaking' through into the next. If this leakage were 100 per cent in a stereo system then the stereo effect would cease and the system would become in effect monophonic. However, in most cases separation is greater than 20 dB (10 per cent leakage) and this is sufficient to maintain an adequate stereophonic effect. Again transducers tend to be the worst offenders, gramophone pick-ups in particular,

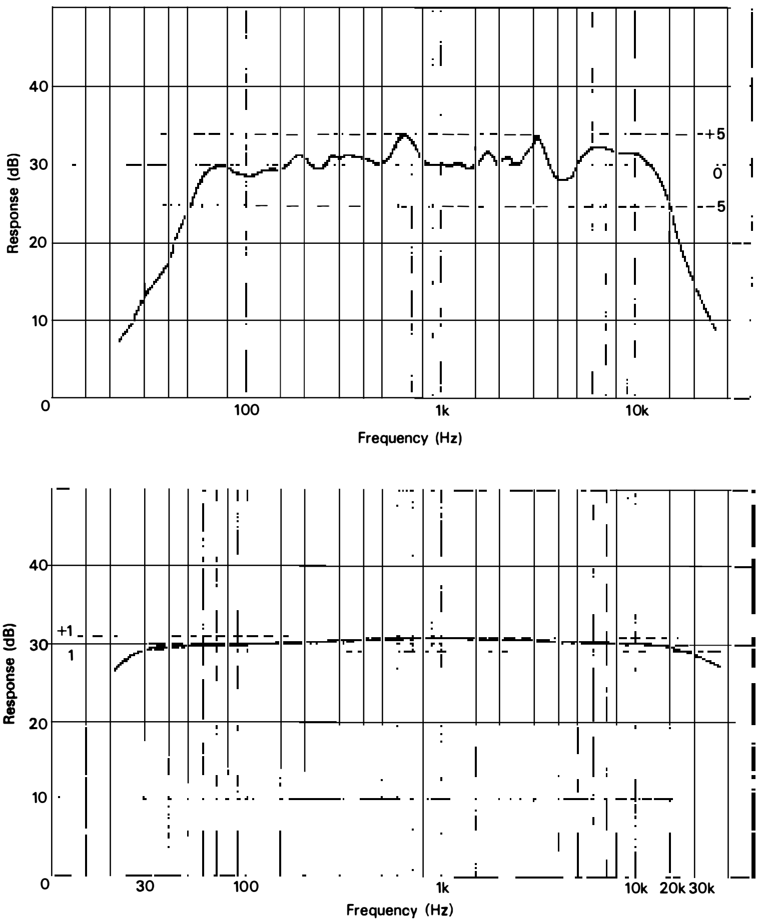


Fig. 44
Typical response curve for a loudspeaker, (above), and for an audio amplifier, (below). Wholly electronic systems such as amplifiers always have very smooth responses, but this is never the case with transducers such as loudspeakers.

also gramophone records and tape machines often have poor crosstalk figures. As always amplifiers tend to be well behaved with crosstalk figures of around 40–60dB. Like most other imperfections in the hi-fi system, crosstalk is usually frequency dependent being worse at high and low extremes of frequency so any valid specification should state the frequency of measurement.

THE LP RECORD

Having looked briefly at the audio amplifier and made a few general points about the performance of audio equipment, let us look more specifically at some of the components of the hi-fi system. There is no better place to start than the record medium itself. The LP record is the principal source of all the music we will ever hear on our home audio system. It has been with us for more than thirty years, and even the introduction of compact cassette tapes about fifteen years ago has not greatly diminished its importance. There are many points in its favour; it is very convenient to use, even the totally untrained user can easily find the music he or she wants on one of its sides. There is an enormous recorded repertoire on disc, thousands of music lovers have large and valuable record collections of music built up over many years. The LP provides the better part of an hour's playing time and at a cost which, in real terms, is still well below what it was many years ago. Another important point is the size of an LP disc; it is large enough for important notes about the music and the recording artist to be conveniently presented on the sleeve. Last, but by no means least, is the quality of reproduction which a typical LP record can provide. In recent years, this has become more variable, although at best the LP does give exceptionally good sound quality. Its principal drawback is its very fine surface, which is easily damaged, so that records will always benefit from careful handling.

It was suggested to me about ten years ago that by now there would be few or no LP records around on the market, because other media would have displaced them. I will leave readers to judge for themselves how accurate a prediction this has turned out to be. The LP record is still the most popular choice for home entertainment, so let us go into the stages in its production. Figure 45 shows, in a much simplified form, the many processes that take place before the recorded performance finds its way into the grooves of the record we put on in our listening rooms.

The very first stage involves actually recording the original performance and producing a 'master' tape. The original performance may take place in a concert hall complete with audience, or it may be a studio production. Studio performances are usually slightly superior, technically, because in the studio there can be complete control of the recording conditions. Concert hall performances may place constraints on the recording engineers and production team due to the acoustics of the auditorium and the need to maintain the correct environmental conditions for both performers and audience.

Microphones–at least two but generally many more–are placed to achieve the desired balance between instruments and performers. Depending upon the recording techniques used, the entire balance can be set at the start of a session and recorded directly onto the master tape, in which case the balance cannot subsequently be changed. Alternatively, individuals or small groups of performers can be recorded onto separate tracks of a multitrack tape, either simultaneously or independently, as is frequently done in popular music recording. This allows the balancing to be carried out at leisure away from the heat of the recording session, and it allows various 'mixes' to be produced for different purposes. In such cases the so called mixdown produces, for example, a stereophonic (two-channel) master for disc cutting, from an original 12- or 24-track tape. The direct

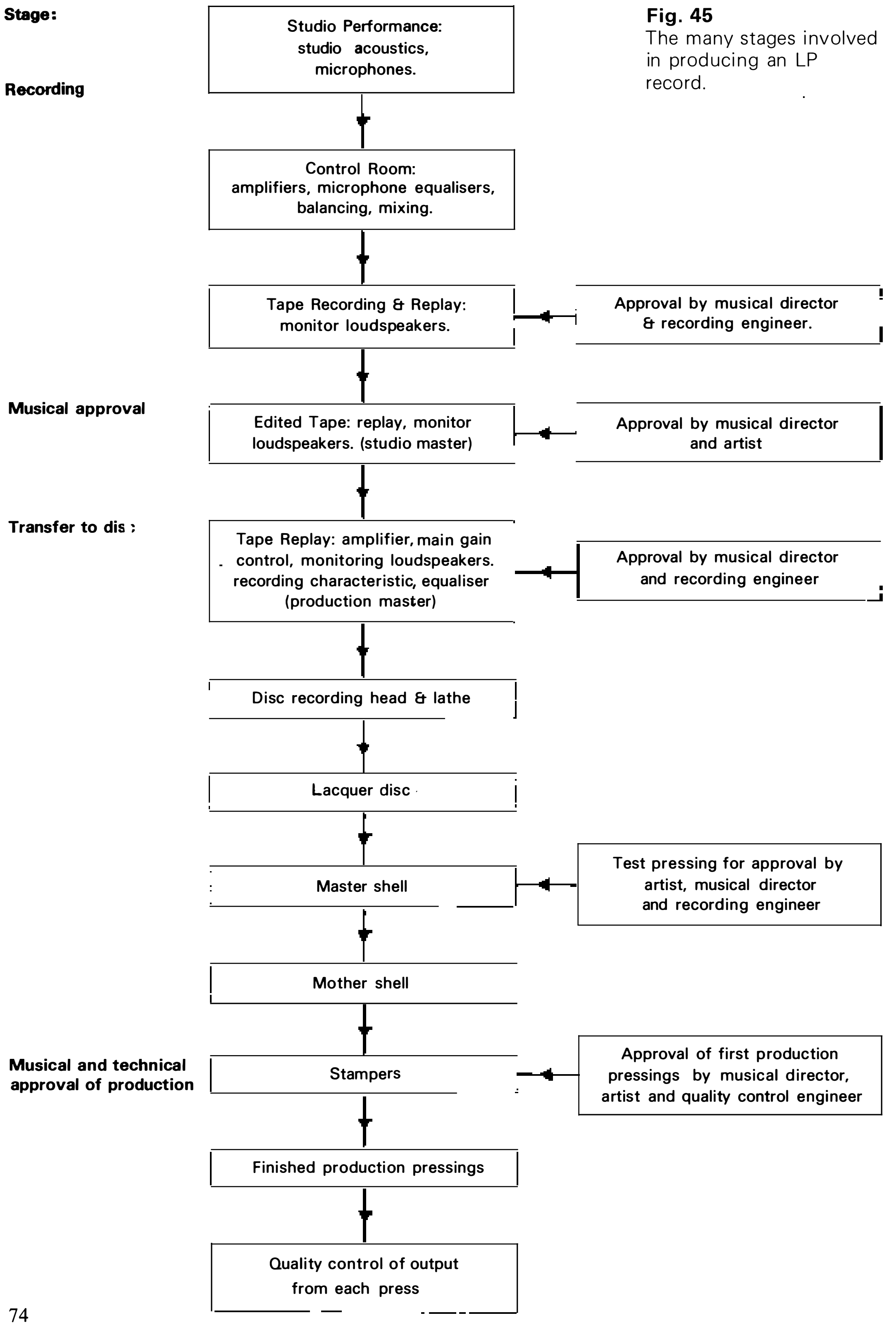


Fig. 45
The many stages involved in producing an LP record.

recording which contains all original programme material is called the studio master. The final version after all mixing, balancing and adjustments is called the production master. Special attention is then given to the technical requirements of the disc cutting stage, which comes next. It is possible, for example, to produce a production master tape which is perfect on studio replay but which cannot be used for cutting a disc. The reason is that the disc will eventually be played under varying conditions quite different from those maintained in the recording studio.

Transfer

In the transfer from tape to disc, the production master is replayed (after necessary adjustments by the cutting engineer) on a special tape-machine which feeds the disc-cutting lathe. These machines usually feature a pre-read head which produces a signal giving automatic control of the cutting lathe to obtain the best compromise between recording level and playing time.

Fig. 46
The cutting engineer examines the groove quality cut by the recording head on a lacquer disc.

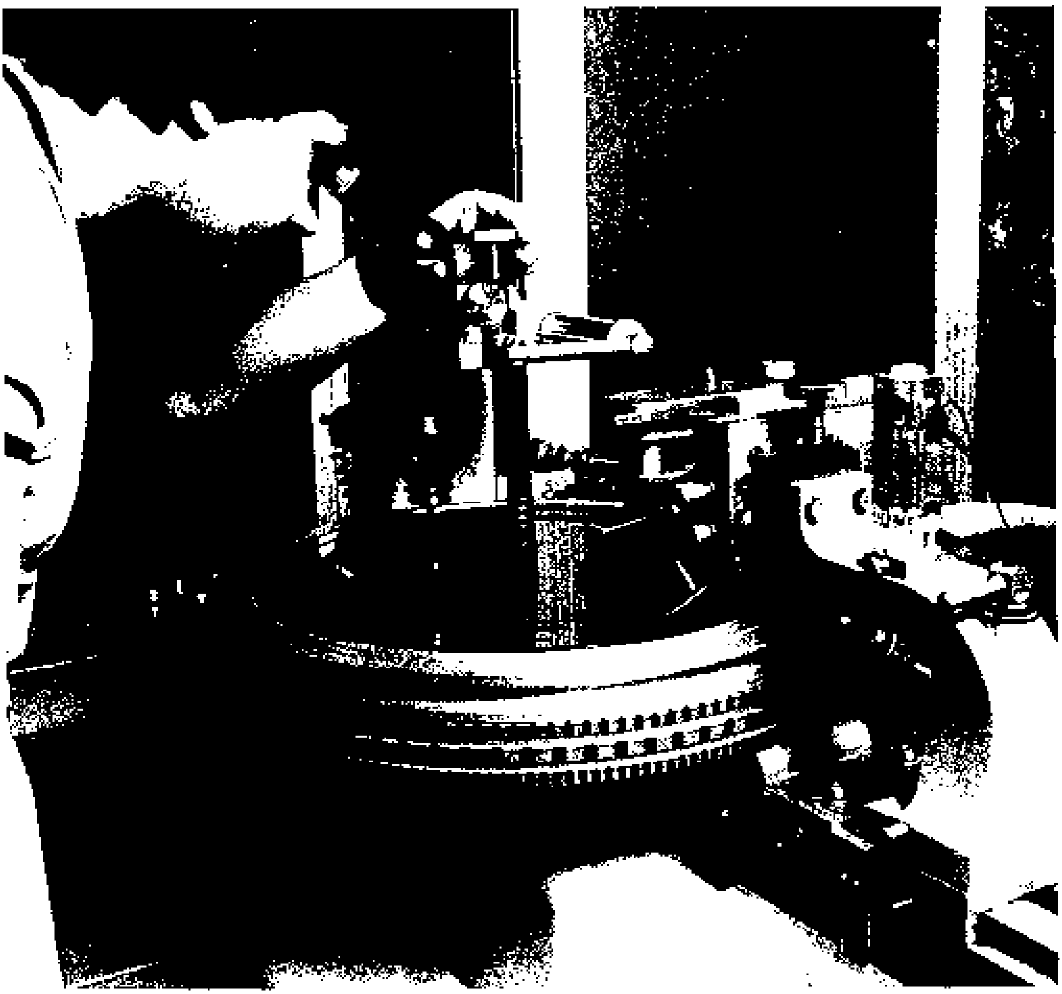
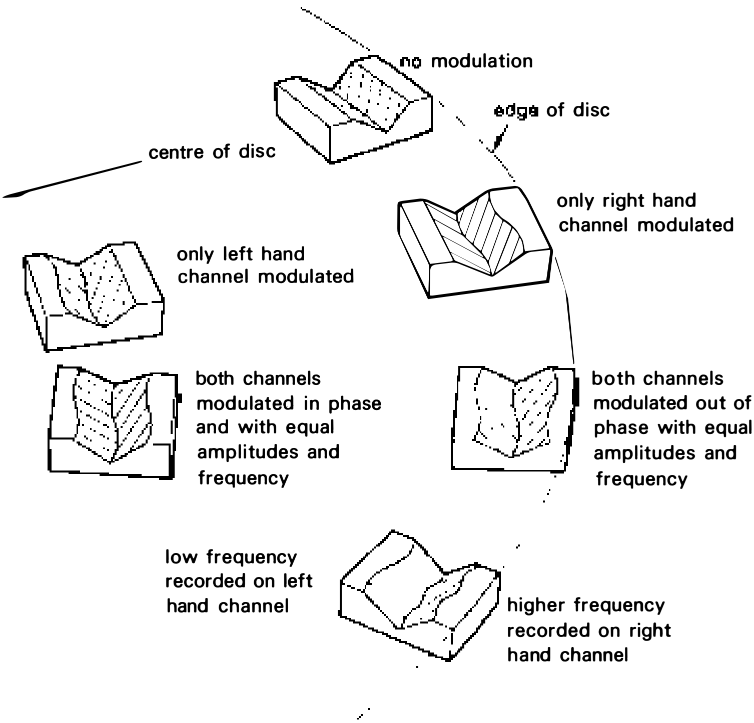


Fig. 47
Examples of the shape the groove wall takes when modulated onto each wall, but the groove has a much more complicated shape with music or speech waveforms.



There are many complex problems associated with disc cutting and the cutting engineer must be very skilled at obtaining the optimum results on disc from a given master tape. The disc on the cutting lathe is usually known as a 'Laquer' and is a flat aluminium disc coated with a soft plastic laquer–originally cellulose acetate but now cellulose nitrate is used (this medium for disc cutting was developed by the British pioneer Cecil E. Watts in 1938). The surface of this blank laquer disc must be exceedingly flat because of the very small dimensions of the groove that is cut into the surface. The lathe (Fig. 46) consists of a precisely engineered turntable on which the blank disc rests and the cutterhead which incorporates the chisel-shaped cutting stylus. The cutterhead is driven radially across the surface of the disc as the turntable rotates (at $33\frac{1}{3}$rpm), thus producing the familiar helical groove which is the modulated programme signal. In a stereophonic disc the signal modulation takes the form of variation in depth and lateral displacement of the groove. In a mono recording the depth is constant and only lateral displacement occurs, examples are shown in Fig. 47. The cut surface of the laquer is so soft that even a few playings with a modern lightweight pick-up would damage the grooves. Obviously our familiar vinyl pressings cannot be manufactured directly from such a delicate original, therefore several intermediate processes are involved.

The laquer is metalised with silver to make it electrically conductive. Then it is electroplated with nickel to form a thin metal mirror image of the original surface. This plating, after reinforcement with a solid metal plate, is called the 'master'. The electroplating process is repeated on the master to form one or several 'mothers'. The mothers in turn are electroplated and the separated platings are further reinforced to produce 'stampers'. These stampers are by now very hard mirror images of the original laquer, and often several stampers are produced from each mother.

The stampers actually form part of the mould that is used to 'press' each disc. Two different stampers are produced, one for each side of the disc. The stampers are fitted into a hydraulic press which closes onto a charge of PVC material called 'vinylite', and applies heat and pressure. After a cycle time of about thirty seconds the finished pressing is ejected and trimmed ready for packaging and distribution. Many pressings can be obtained from one set of stampers.

Because of the many intricate stages in the disc production chain the quality of commercial pressings may fall short of the original master quality. This is unfortunate because exceptionally fine results are obtainable from the disc production process. However, there are other factors which also affect the reproduction from disc records and these will become clear in what follows. One final word about what are known as 'direct-cut' discs. These are records where all tape-recording operations in the production stage are eliminated and the lacquer is cut directly, as the musicians play, following the studio mixing and balancing operations of the recording engineer. Hence the number of links in the recording chain is greatly reduced, which gives even greater

Fig. 48
The lacquer being silvered by a chemical spray.

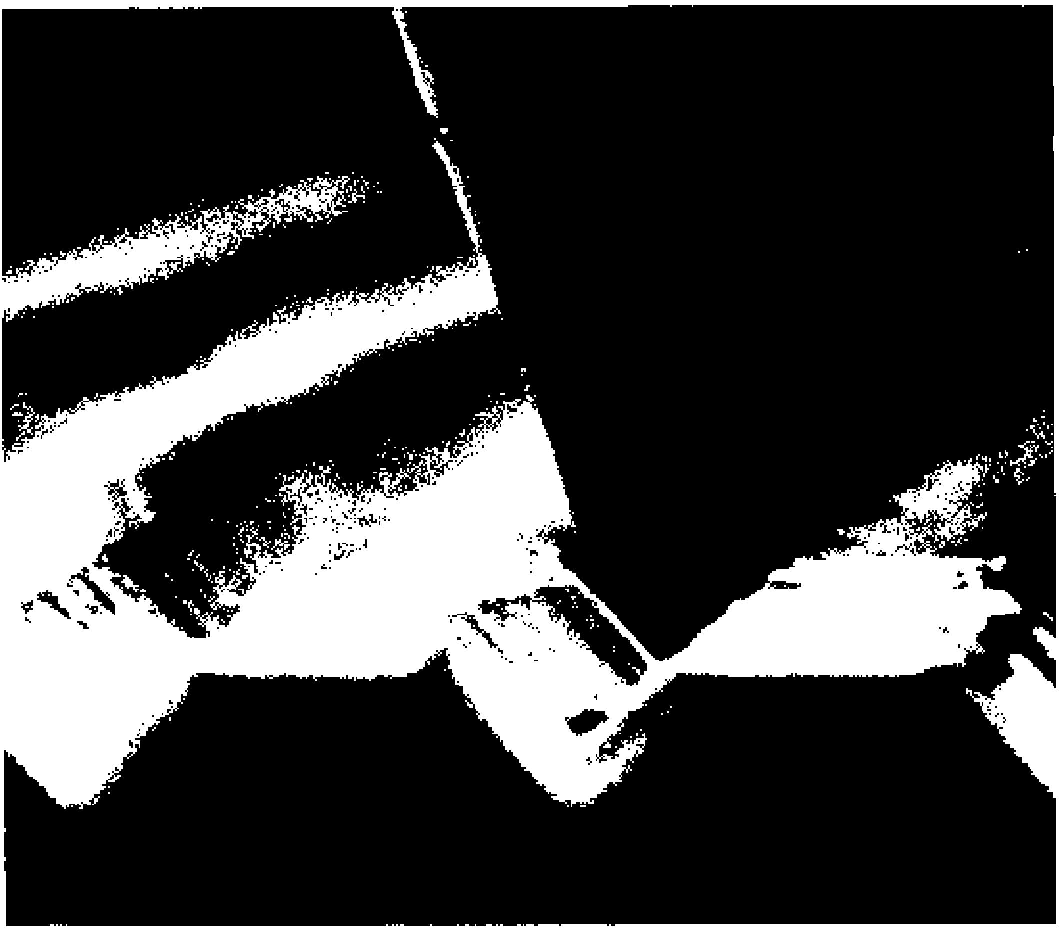
Fig. 49
(Left.) Separating the master from the lacquer.

Fig. 50
(Below.) A charge of PVC, sandwiched between the labels, is placed onto the stamper for forming into an LP disc.

fidelity, especially in terms of an extremely quiet, noiseless surface on the record, and improved dynamics. Naturally, the complete success of a direct-cut disc depends upon the musicians, producer, engineer and lathe operater giving of their best, all at the same time. There is no possibility of editing, and the whole side of a direct-cut disc must be cut in one continuous session. This is to be contrasted with conventional tape-master methods, where edits may occur, and often do, every minute or two during the final recording.

Quite a variety of direct-to-disc recordings have become available in recent years, from several countries in the world and even from some of the well-known record companies. A few have been included in the music list in Chapter 5. Generally, their technical quality is remarkably good, and much better than conventionally recorded discs. However, direct-cut discs are rather more costly than usual, because their high technical quality is matched by very high production standards and, of course, because mistakes in direct-to-disc recording are more expensive to rectify.

Fig. 51
A greatly magnified photo of a stylus resting properly in a stereo record groove.



As we sit back and enjoy our favourite music, we never think of the very violent forces at work on the surface of the disc. The modulations on the disc are measured as velocities (in fact recorded velocities) with units of metres per second. Alternatively, we may wish to consider the accelerations of the groove (velocity $\times$ frequency) and the stylus which travels in it. It is not uncommon for accelerations to reach 16000 m/s^2, which is more than 1600 times the acceleration due to gravity! The tiny area of stylus contact on the groove walls is such that it exerts a pressure of 50000 lb per square inch (3×10^8 Pascal)! It is almost surprising that either record or stylus survive even a single playing. Nevertheless, discs and stylii do, as we know, last through many hours of playing time, assuming that the arm and cartridge are properly set up and that the stylus remains in perfect condition.

Figure 51 is a greatly magnified photograph of a conventional spherical stylus resting properly in the groove of a stereo record. Notice the perfect profile of the stylus and its symmetrical position with respect to the groove walls. Although the depth, width and spacing between adjacent grooves will vary with the signal amplitude the width of the groove at the top will typically be only one two-thousandth of an inch and the depth about one-thousandth of an inch. In fact the stylus is in tiny, two point contact with the groove, one contact point on each wall. Provided this situation remains throughout the playing of the disc, the pick-up will make a perfect transcription of the recording. Figure 52

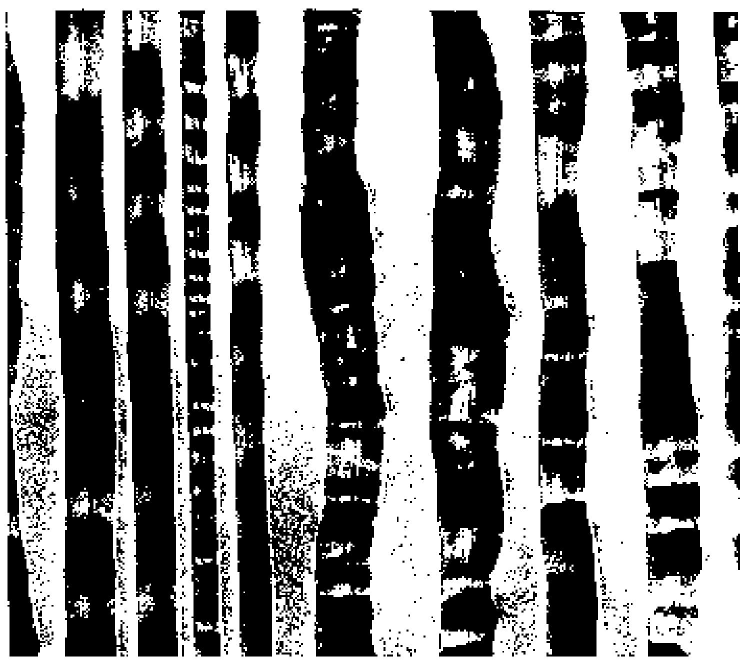


Fig. 52
Path of grooves on a stereo record during an interval of orchestral music. Note low-amplitude high frequencies at left and high-amplitude low frequencies in centre grooves.

shows what route the stylus takes during an interval of music. If, for any reason, the stylus leaves its proper seating in the groove, mistracking occurs which may be audible and even damaging to both stylus and record. The energy which moves the stylus comes from the record, as the groove moves past the stylus at fixed angular speed i.e. 33⅓rpm. The situation can be likened to a game of tennis, with the part of the stylus taken by the ball and with the racket representing the record. As the ball reaches the racket, it is brought to a complete stop and is accelerated again in the opposite direction, almost instantaneously. The energy possessed by the ball due to its motion (i.e. kinetic energy) must be given up to the racket as it is redirected. So it is with the record groove and stylus. Although the stylus has very low mass, it still has energy of motion, and this is imparted to the groove each time the stylus is forced into a different direction. It follows that some wear must take place, even under conditions where the system is performing exactly as designed, although with pick-ups becoming ever lighter and more faithful transducers, the problem is not a serious one. In practice, however, the stylus may become significantly worn with age, or damaged, and this will lead to more rapid deterioration in the condition of your records.

The first of the so-called 'hi-fi' generation of pick-ups were of the crystal type, employing the well known piezo-electric effect, i.e. they produced an electrical signal in response to being deflected by the motion of the stylus. These types were gradually replaced by ceramic pick-ups, which employed similar principles but were more rugged and a little more accurate. As techniques developed further, it became possible to make magnetic pick-ups at reasonable cost, and these types, of the moving magnet (mm) or moving iron (mi) variety, now dominate the audio market. More recently we have seen the introduction of moving-coil pick-ups, and these feature extremely low moving mass and very accurate tracking ability. They produce very low signal voltage levels, however, and so special pre-preamplifiers are required with them before connection to the amplifier. Moving-coil cartridges are now more popular and their cost, although still quite high, is becoming more realistic. An example is shown in Fig. 53. It is still true to say, though, that the very best of the moving-magnet cartridges can equal some of the moving-coil types. The sensitivity of a pick-up cartridge is a measure of the output voltage it will generate as a specific frequency and recorded velocity. Usually it is quoted in millivolts per centimetre/second recorded velocity, and for all magnetic pick-ups its value will lie in the range 0·5 to 2·5mv/cm/s. Its value gives no indication of the performance quality of the cartridge, but it is advisable to make sure that the amplifier you are using does suit the sensitivity figure of your cartridge.

The record, arm, pick-up cartridge and record deck or turn-

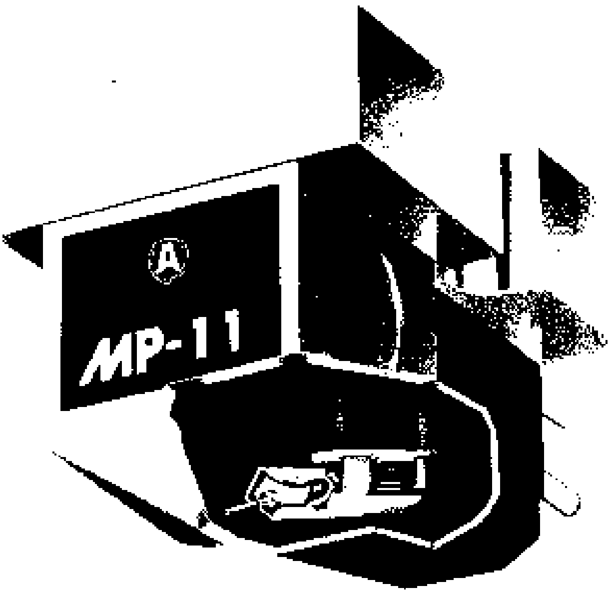


Fig. 53
A modern magnetic pick-up, the Osawa MP11 moving-coil cartridge.

table are all parts of a single mechanical system and they interact with each other to affect the overall sound quality of the audio system. The turntable itself has been found to have an important influence on the fidelity of reproduced sound. I mentioned earlier how small cyclic speed variations of the turntable are one of the problems of high-fidelity sound reproduction. They can be a source of interference in a studio tape recorder as well as a domestic record deck. These slow variations in turntable speed give rise to what are known as 'wow' and 'flutter' being characterised by changes in the pitch, especially of fairly pure tones which decay slowly, such as those of the piano. A high quality turntable will have wow and flutter figures of 0·05 per cent or less, which refers to the percentage speed variation from its accurate value. For interest, Table 9 lists the percentage wow which most listeners are capable of detecting for different sound sources.

Another problem with the turntable is the phenomenon known as 'rumble'. This again is a cyclical disturbance, but it does not influence the pitch of reproduced sounds as much as it causes an intrusive low frequency 'rumbling' noise to be added to the music, being due to imperfect motor bearings. This quantity is often quoted as a dB figure below a specified output level from the pick-up. It should be at least 60dB below the normally specified signal level in a good turntable.

Many modern turntables feature an adjustment on the pick-up arm called bias or sidethrust compensation, and this provides a side force, acting towards the outer edge of the turntable, to counteract the force to which the pick-up is subject owing to

Table 9 The percentage wow detectable on music

	frequency swing (%)	
instrument	peak-to-peak	'r.m.s.'
Piano	0·6%	0·22%
Oboe/horn	0·7	0·25
Violin	0·8	0·3
Cello	0·8	0·3
Symphony orchestra	0·8	0·3
Dance band with piano	0·8	0·3
Singing (solo)	0·9	0·32
Harp	0·9	0·32
Singing (choir)	1	0·36
Brass instruments	1	0·36
Dance band without piano	1	0·36
Choir	1·2	0·45
Jazz	1·5	0·56

the spiral groove on the disc pulling the pick-up inwards, towards the record centre. This adjustment will always be fully explained in notes accompanying the turntable or arm (sometimes turntables are supplied without arms, so the user can select his own particular arm of which more will be said later) but remember that correct bias adjustment is important for correct tracking of your cartridge.

Another important setting is the downforce or tracking force applied to the pick-up to keep it comfortably seated in the record groove. Nowadays, tracking forces range from about 1 gram to 3 grams or so, depending upon the quality and characteristics of the pick-up. Information as to the optimum tracking force for your pick-up will be readily available from the pick-up manufacturer, and is easily set. Again many modern turntables which come complete with arms have calibrated tracking force gauges mounted onto the arm, so you can easily set the correct figure. It is worthwhile checking this now and again, for it is most important to the proper performance of the pick-up. Too low a setting is often worse than a slightly high setting, for it will lead to groove jumping and damage to your records, even after only one or two playings. These features are shown in Fig. 54 which is a typical example of a modern turntable and arm combination in the middle price range.

Stylus and record care

In normal use, when playing records on your hi-fi system, you will not need to give the equipment any special attention, except for the pick-up stylus and the records themselves, which are by far the most delicate parts of the whole installation. The comments above about bias compensation and tracking force are thus important, and the pick-up arm should always be handled with care. You may wish to change the pick-up 'head' to an alternative

one, perhaps to play old LP or 78 rpm records, and when doing so always ensure that the head is fitted properly, so that the stylus is exactly vertical. If this is not done, uneven record and stylus wear can result, and 'crosstalk' between the two stereo channels will be impaired. Always take a little care with the pick-up arm and cartridge.

Stylus wear may be too difficult to assess from a simple examination with the naked eye, and if you suspect it has become damaged – through a careless drop onto a record or the turntable – you can take it to your dealer who will arrange for it to be examined under a microscope or shadowgraph. This should in any event be done periodically, say every year; the cost and inconvenience are small in comparison with the value of your record collection. Consider that an assortment of 100 LP records will probably have cost you about £400, and some may be irreplaceable.

You may need advice when it is necessary to replace your stylus, it may have some special tip shape such as elliptical or line contact. These profiles have been developed to optimise the stylus-record contact under difficult modulation conditions, and it is important that the correct one is fitted to preserve the performance of your cartridge. Always exercise the same care when handling the records. Do not touch them on the modulated surface, even with clean hands, since small salt or grease deposits will detract from perfect performance. Store the sleeved records

Fig. 54
Showing the adjustments provided for arm and cartridge in a present day turntable.

A counterweight
B tracking-force ring
C anti-skating compensator

vertically, such that it is *just* possible to remove the one you wish to play without disturbing its neighbours. Be careful when removing a record from its sleeve, not to score its surface with the edge of the paper container. Make sure that the inner paper container does not become folded at the corners, or it will exert uneven pressure on the disc when it is stored, which in time leads to 'warping'. The same applies if leaflets or booklets accompany the record; avoid folded edges or creases across the leaflet for these will cause ridges to bear against the flat record surface.

Warping has become a more frequent problem with LP discs in recent years. It means that the disc has become twisted and unflat, normally over some segment of its surface, which is easily seen at the edge of the disc as it revolves. In some cases the pick-up arm can be seen to rise and fall as it encounters the warped section. The speed of the stylus past the groove is actually changing at these moments, which can cause wow or swishing sounds to be heard over the speakers. A more serious form of warping is when the record becomes dished, so that it makes contact with the turntable only at its centre. It could then happen that the record will slip and become unplayable. These problems have arisen partly because modern LPs are made thinner than in the past for economic reasons but remember record flatness can be strongly affected by storage and handling conditions. Slight warping can be remedied in time by storing vertically as explained above.

There are many cleaning devices available today which are claimed to be effective, but few of them are in my experience, the reclamation procedure can sometimes do more harm than good. There will be little or no need for cleaning if the record is always handled carefully and stored away from dust. If you consider it necessary, clean the record with a lint free cloth, very lightly moistened with an anti-static cleaning fluid. Avoid excessive use of these fluids. Alternatively use a little distilled water instead. It is even better not to apply fluids of any kind yourself, but instead use a proprietary cleaning cloth which is already impregnated with a controlled amount of fluid.

Because atmospheric conditions vary, and synthetic materials such as nylon are used in domestic listening rooms, your records may become electrostatically charged as you handle them, i.e. they may acquire an electric potential. When charged in this way, the record may attract dust from the atmosphere onto its surface. This problem is not an easy one to solve, because the environmental conditions which influence it can change. It is even more difficult to make the record material immune from ever acquiring a charge, without affecting its fidelity. Several proprietary devices are now available which attack this problem; some create a charge designed to neutralise the charge on the record (anti-static guns) whilst some apply a fine deposit on the disc surface, designed to prevent any build-up of charge (anti-static preservatives). In many cases 'static' problems will not be serious and will be minimised by the cleaning procedure described above – sparing use of an anti-static cleaning fluid. Also, to minimise any build up of static electricity, always withdraw the record slowly from its inner container, as far as possible avoiding friction. This is best achieved by resting the finger tips on the centre of the disc while bringing the base of the thumb against its outer edge. In cases where static charge becomes a problem, however, one of the anti-static preservative fluids might be used. The preservative will ensure an essentially permanent (or at least semi-permanent!) neutral record surface. My own experience is that treatment is quite effective, and I have found its use does not, significantly, impair the fidelity of the recording. Under very critical listening conditions, a slight loss of extreme treble clarity might be noticed, so remember that this fluid should be applied once only, and that it is not an alternative to cleaning. An LP record is a highly complex precision product so personal care and common sense will be very important factors in preserving your collection.

Handling a record correctly

Matching pick-ups and arms

Many turntables come complete with their own arms and fitted cartridges, ready for use after simple setting up and connection of the amplifier and loudspeakers. For those who wish to be more particular, and make the selection of the arm and the cartridge for themselves, it has become common for manufacturers to supply the turntable on its own, with provision for the user to fit his own choice of arm. The question arises, what is the best combination of arm and cartridge in such cases? There is such a great variety of possible combinations that I am giving for readers who are considering this option some further explanation and information on the point. First, a word of explanation. We spoke earlier in the book about the phenomenon of resonance, and how every mechanical system will exhibit it. This is true of the arm and cartridge combination as well. The arm and cartridge housing are rigid bodies, and the stylus is connected to them via springy, compliant supports. These compliance elements are, of course, part of the internal design of the cartridge, and constitute what is known as the cartridge or pick-up compliance. They ensure that the stylus is quite free to move in the record groove. The arm and cartridge fixed mass, pivoted on the delicate bearing at the base of the arm, provide the mass element of the system which has a particular resonant frequency. Obviously, this resonant frequency must not occur within the audible frequency range, or it will gravely upset the operation of the pick-up. This is avoided by making the pick-up compliance relatively high, but not so high that the stylus is not well enough supported under the influence of the downward tracking force. It is now generally agreed that a fundamental arm-cartridge resonance frequency somewhere in the range 10 Hz to 17 Hz is optimum; if it is higher than this it may interfere with bass frequency information contained on the record, while if it is much lower the arm will be too sensitive to vibrations transmitted through the turntable e.g. from the listener moving about in the room.

The mass of different arms, and the compliance figures for different cartridges available today, will be found to vary within quite wide limits. Since the low frequency resonance is determined by the combination, it is important that the right choice be made so as to give a resonance frequency within the range mentioned above. Table 10 gives a listing of the majority of pick-up arms and cartridges currently available, with their mass and compliance figures. Ideally, a selection within the 'optimum' range should be made, although it can be seen that this still leaves the user with a wide choice. Arms shown with a higher than usual mass should be combined with cartridges of slightly lower compliance. The trend these days is to make pick-up arms of

Table 10 Arm/Pick-up matching chart

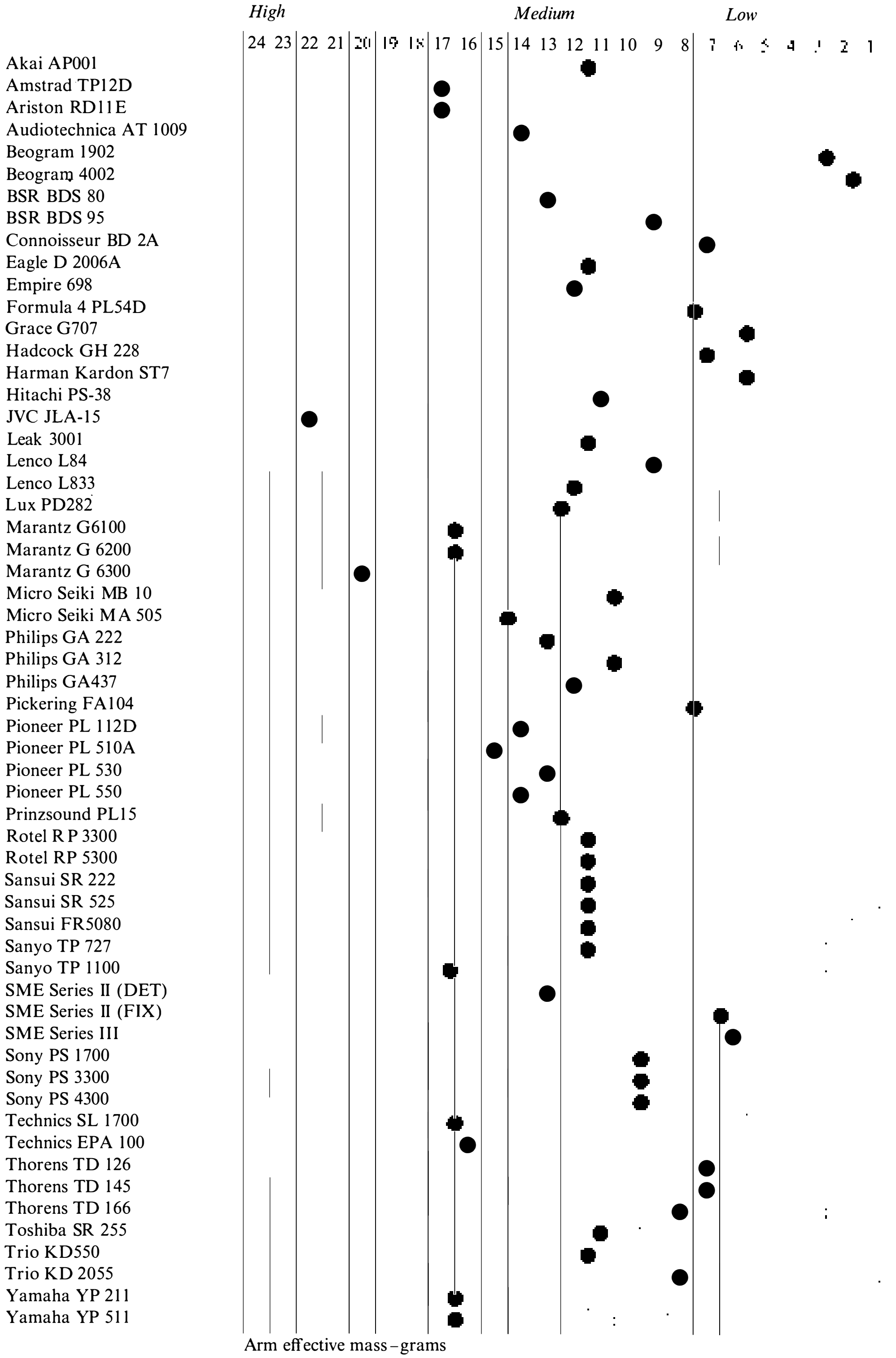
	Effective mass* of arm without cartridge *g*	Maximum Compliance (f=7 Hz) MM/N**	Optimum Compliance (f=10 Hz) MM/N**	Minimum Compliance (f=17 Hz) MM/N**
Low Mass	0	86	42	15
High	1	74	36	13
Compliance	2	65	32	11
	3	57	28	10
	4	52	25	9
	5	47	23	8
	6	43	21	7
Medium Mass	7	40	19	7
Medium	8	37	18	6
Compliance	9	34	17	6
	10	32	16	5
	11	30	15	5
	12	29	14	5
	13	27	13	5
	14	26	13	4
High Mass	15	25	12	4
Low	16	23	12	4
Compliance	17	22	11	4
	18	22	11	4
	19	21	10	4

* based on the average mass of a pick-up cartridge = *6 grams*–most cartridges are between 5–8 g.

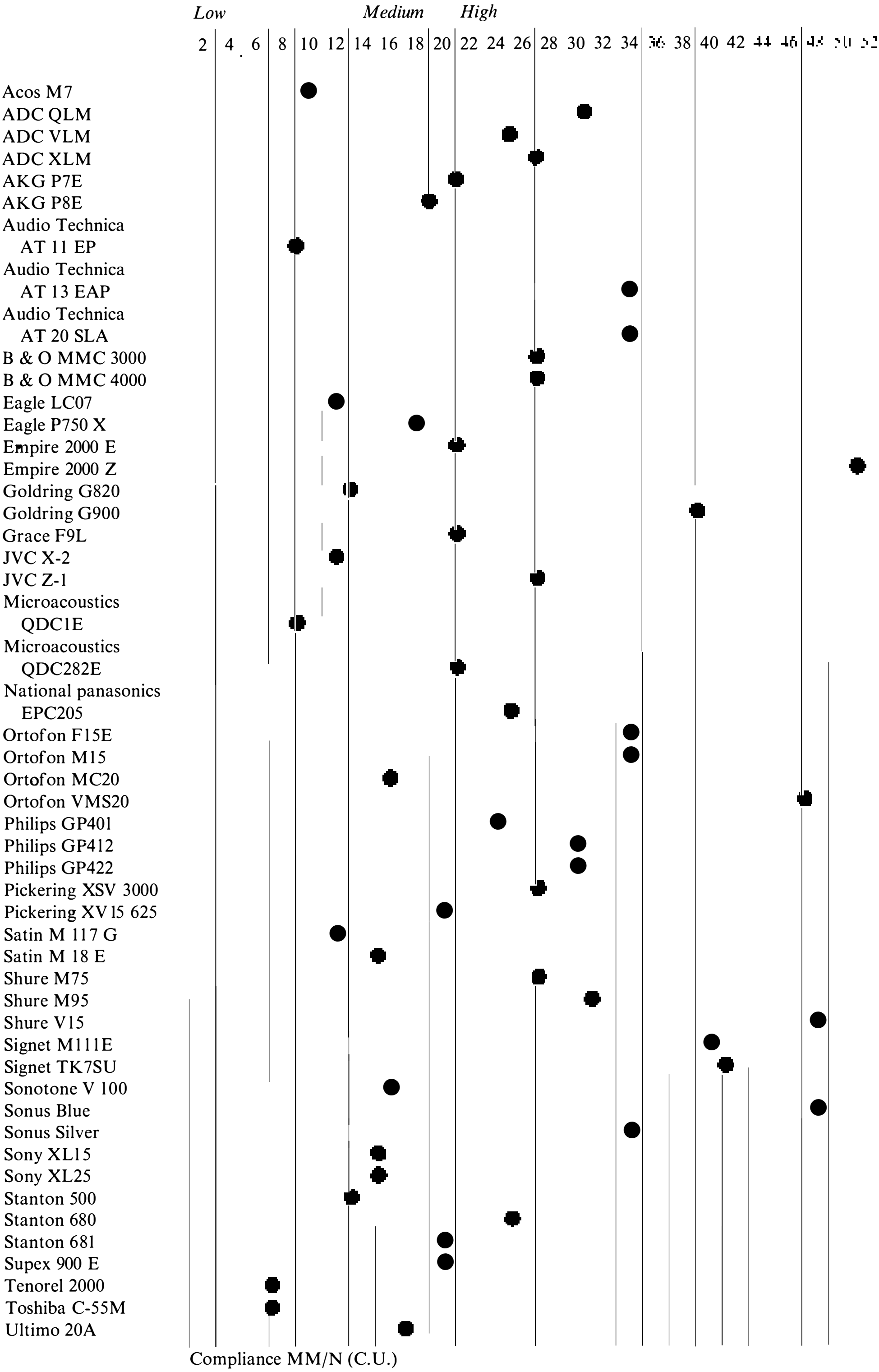
** *or* compliance units (C.U.) *or* cm/dyne × 10^{-6}.

Optimum values of compliance–matching with an arm of given effective mass are shown in column 3, but any compliance value within the maximum and minimum limits will give satisfactory results.

Pick-up arm effective mass chart



Pick-up compliance chart



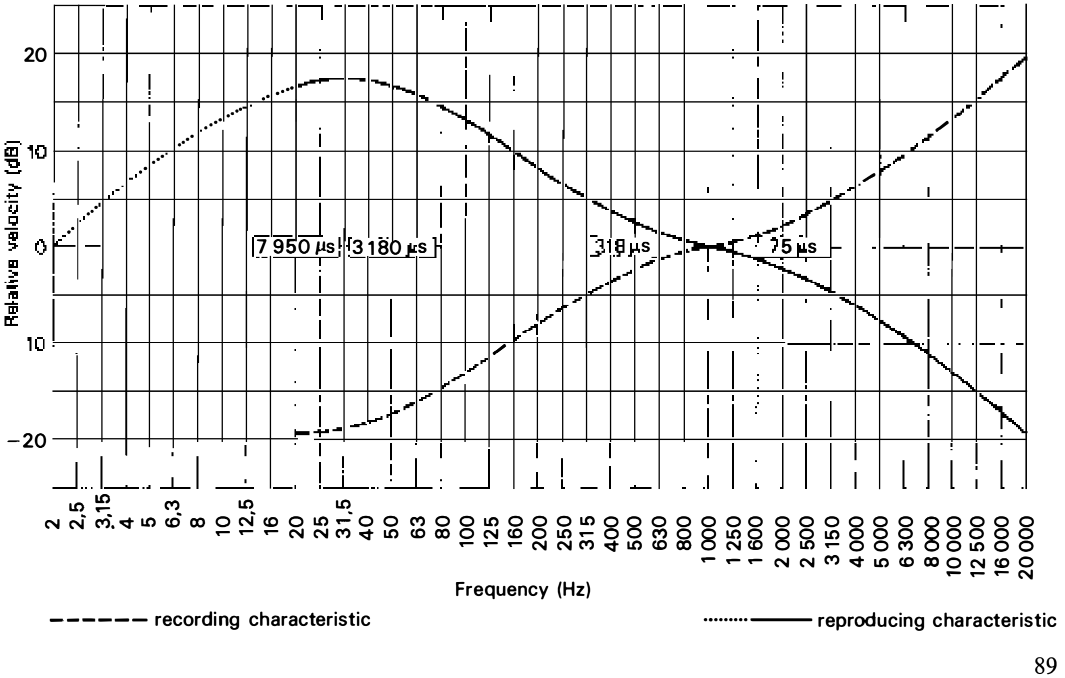
low mass, suiting cartridges of higher compliance, of which more have become available. Moving-coil cartridges tend to have fairly low compliance values, and this may necessitate additional mass fixed into the headshell, which must be properly counterbalanced when the arm is set up. Out of interest, Fig. 55 shows the latest equalisation characteristic for LP records, which features the recently approved low frequency replay modification (shown dotted), designed to minimise the audible effects of very low frequency problems occurring with records and turntables.

TAPE SYSTEMS

The alternative medium for sound recording and reproduction in the home is tape which, as we have seen, forms an important part of disc recording as well. This is because tape is a very convenient recording medium and can easily be edited, which is not possible with disc. The concept of magnetic tape recording is quite simple. The electro-magnetic recording head on a tape recorder contains coils wound on a 'soft' magnetic material. When an audio signal is applied to the coils, a varying magnetic field is set up in the material. If another suitable magnetic material –the recording tape–is passed close to the head, it will be permanently magnetised in accordance with the audio signal variations. The tape material is said to be magnetically 'hard' because it retains, as it must to be useful, the information magnetically impressed upon it. The recording head, on the other hand, only becomes magnetised when a current is passed into its coils, hence the term magnetically 'soft' material.

When the recorded tape is played back, the reverse happens and

Fig. 55
The latest equalisation curve used in making an LP record. The extended part of the upper curve, shown dotted, has been introduced to minimise the audible effects of very low-frequency problems in turntables or records.



a principle called electro-magnetic induction induces an electric current into the coils of the playback head as the tape passes by it. Hence a signal is produced just like the one originally used to make the recording.

The tape medium consists of a plastic, polyester backing or base material, onto which a magnetic lacquer or paint is coated, mixed with a binding agent so it will adher to the backing. Still most widely used is a lacquer derived from ferric oxide–common rust–although other compositions have been used in more recent years. The tape manufacturing process has become very complex and refined, as more additives such as lubricants, stabilisers and adhesion promoters have been included with the magnetic lacquer in order to improve the quality of the medium. The polyester backing has great strength even though it is very thin, and resists stretching, heat and moisture under normal operating conditions. Various thicknesses are employed depending upon the application; for example the tape thickness will be about 35 microns (millionths of a metre) for professional use, and about 20 microns for standard applications. Different lengths and reel or spool sizes are available to give some choice of maximum playing time, which can be further extended by use of different playing speeds, standardised at $3\frac{3}{4}$, $7\frac{1}{2}$ & 15ips (inches per second) for all domestic machines.

Similarly, the thickness of the lacquer will vary from about 7 microns for very high frequency e.g. video work to 18 microns for high-quality audio applications. The width of the tape has been standardised for general applications, normally it is $\frac{1}{4}$inch, and sometimes $\frac{1}{2}$inch, 1 inch or 2 inches for studio use. The tape is made in 12-inch or 18-inch 'jumbo' rolls, which go through a very complex series of chemical coating, polishing and finishing processes in which the very tightest tolerances must be held. As tape has become a more popular medium, there has been a movement towards more recording channels or tracks on the tape, slower speeds and higher frequency performance, which has made the manufacturing process even more a precision task.

Those faithful to 'open-reel' tape recorders–the same format as is used in the recording industry–will know that a similar degree of precision engineering goes into a modern domestic tape recorder. The first domestic open-reel machines became available in the early 1950s, and this type of machine has, over the years, sustained a good following among the advanced amateur and semi-professional recordist. The fact that this group has never widened very much is probably due to the application required of the user to operate a good machine and make recordings for himself, the high cost of the machines and the lack of any significant repertoire of pre-recorded tapes. This situation was transformed, however, shortly after the introduction, in 1963, of a totally new, compact format for recording tape. This was the compact cassette, launched by the famous Dutch firm Philips at the Berlin Radio Fair in the autumn of that year. Fortunately for all those who use it, and there are many outside the audio industry, the cassette has become standardised. This came about for two main reasons; firstly, the advantages of the cassette, its compactness, ruggedness and simplicity of use, were clearly seen (even in comparison with other self-contained formats, e.g. tape cartridges which were competing with it) and secondly, Philips decided to offer the idea to every other manufacturer without patent protection so it could become a generally available, compatible system. Agreement was reached and, as readers know, the market for cassette tapes and recorders increased enormously. There is now a single standard for the construction of the cassette (See Fig. 56) and the speed at which it runs, $1\frac{7}{8}$ ips, and its use is firmly established in instrumentation, business administration, language laboratories and the audio world.

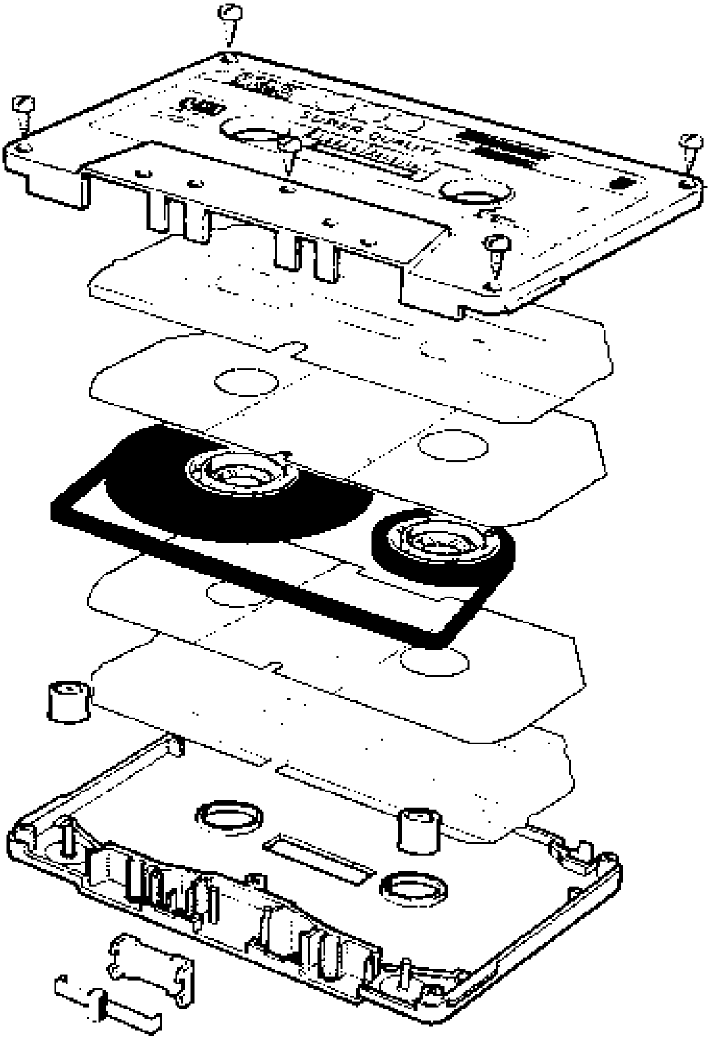


Fig. 56
The components of a modern tape cassette.

The convenience and standard format of the compact cassette and its consequent wide acceptance, has meant that it can be sold, not only as a basic 'blank' tape medium for the amateur recordist, but also in pre-recorded form, as an alternative to disc record. Hence the cassette recorder is now about as common a component of a home audio system as the record deck. Cassette recorders are far less delicate than the record deck and arm, and the same is true of the cassette compared with the record. Cassettes, of course, should be kept away from magnetic fields and excessive heat. Almost all new record releases these days are also available on cassette, and already a substantial catalogue of pre-recorded cassettes exists for the home listener. In the early years cassette quality was rather poor in comparison with the average LP, but the gap has narrowed, as new tape formulations and the Dolby tape noise reduction system (described below) have become available for cassettes. In general, I think that the quality of LP recordings is more consistently good than that of the cassette versions, while 'special' or direct-cut disc releases are far superior in technical quality.

Making pre-recorded cassettes

Of course cassette tape systems also have a major advantage over disc in that they can be used to record as well as replay, giving the user a potentially wider choice of material. On the other hand, pre-recorded cassettes are costlier to produce than disc because each cassette must be copied from a master tape which is a longer process than the pressing techniques used for discs.

Going back to the master tape stage of the disc process, the studio master is copied to produce a high speed production master (running master). This master is played on a special high speed machine running at 32 times the original master speed (609·6 cm/s, 240 ips). Slave cassette recorders are linked to the master machine, and also run at 32 times the normal cassette running speed. The master tape is in the form of a continuous loop with a burst of 6 Hz cue signal signifying the start position. The slave recorders carry the 3·2-millimetre tape on large platters so that several copies of the master are obtained sequentially on each reel.

The recorded reels of cassette tape are placed on an automatic tailoring and winding machine which cuts the copies to the appropriate length, splices leader tape and leads the tapes into empty cassette packages. The cassettes are then labelled and packaged ready for distribution.

Different cassette tapes and uses

It is almost ironic to note that, in the days when tape was available to the hi-fi user only in open-reel format, there were never any problems as to which tape to use. It was usually a question of trying a few spools from different manufacturers, and settling on one or two which you found to your liking. Now, with cassettes in use by a far wider public, many of whom have no particular interest in the technicalities of the medium, there is a bewildering choice of tape formulations, not all of them compatible with every cassette machine.

Because the tape width and operating speed are standardised in cassettes, for compatibility reasons, the only area where changes can be made to improve performance quality, is in the magnetic coating, i.e. the chemical formulation on the tape. The first formulations were of ferric oxide, and then came chromium oxide, which gave more extended high-frequency response and better noise performance. It was not simply a question of buying a new cassette for your machine, however, for the new chromium oxide tape caused greater recording head wear, and it needed different electrical bias and equalisation on the home machine.

Bias concerns the high-frequency signal (normally about five times the highest audio frequency) on which the audio signal 'rides' during the recording process in order to get the best magnetic impression on the tape for a given magnetic field created by the recording head. It is fundamental to the process of magnetic tape recording, but it was found that the amount of this bias frequency was different for chromium oxide tape formulation than had hitherto been needed for the traditional ferric oxide. The other quantity, equalisation, concerns the amount of 'boosting' of high audio frequencies as they are recorded onto the tape, in order to improve noise performance. Again, because the new chrome tape gave inherently better high-frequency performance, it needed different equalisation. Hence it was not long before new cassette recorders were needed, which featured special heads and switchable bias and equalisation controls. In recent years, there have been other formulations, all designed to get the best from cassette tapes. We have

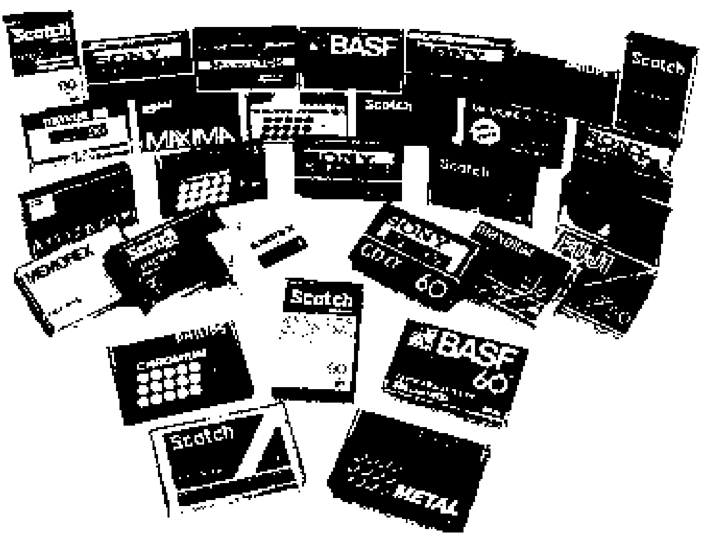


Fig. 57
A range of commercial 'blank' cassette tapes available today.

seen ferro-chrome formulations, which have some of the advantages of both ferric and chrome materials, and very recently, pure metal coatings, using pure iron particles instead of oxides. A range of them is illustrated in Fig. 57. Whilst some of these new materials can be used on most cassette recorders, some of them can only be used on modern machines. To assist readers with this point, I have got together most of the current cassette tape brands and cassette recorder specifications, and tried to match them up. Table 11 lists the great majority of all cassette tapes available and gives their formulation, whilst Table 12 lists many of the cassette machines now on the market and indicates the tape formulations for which they are designed. As can be seen, many machines are adjustable for more than one type of cassette. One manufacturer, in anticipation of the home listener's confusion, has developed a cassette recorder which features electronic selection of bias and equalisation of any cassette tape put into it, fully automatically. This machine is the JVC Model KD-A8, and no doubt there will be other electronic aids of this kind in the future, see Fig. 58.

Fig. 58
The JVC KD-A8, which automatically sets bias and equalisation for whichever tape you use.

Table 11 Cassette tapes

Product	Type[1]	Bias[2]	Equalisation *us*	Output[3]
Agfa L.N.S	Fe	N	120	N
Agfa Color	Fe	N	120	H
Agfa Superferro	Fe	H	120	H
Agfa Carat Ferrum+ Chrom	Fe Cr	($FeCrCro_2$)H	120	H
Agfa Stereochrom	Cr	(Cro_2)H	70	N
Agfa Superchrom	Cr	(Cro_2)H	70	H
Ampex 20 20	Cr	(Cro_2)H	70	H
Ampex		*No Data*		
Audio Magnetics XHEI	Fe	N	120	H
Audio Magnetics Super	Fe	N	120	H
Audio Magnetics Extra	Fe	N	120	H
Audio Magnetics Plus	Fe	N	120	H
BASF LH	Fe	N	120	N
BASF Ferro Super LH	Fe	N	120	H

BASF Ferro Super LHI	Fe	H	120	H
BASF Chromdioxis	Cr	(Cro_2)H	70	H
BASF Ferrochrom	FeCr	(FeCr)H	70	H
EMI Standard	Fe	N	120	H
EMI Super	Fe	N	120	H
EMI Hi-fidelity	Fe	N	120	H
Fuji FX-I	Fe	N	120	H
Fuji Fx-II	Fe	(Cro_2)H	70	H
Grundig Studio 'Fe'	Fe	N	120	N
Grundig Hi-fi CrO_2	Cr	(CrO_2)H	70	H
Grundig Profi 'FeCr'	FeCr	(CrO_2FeCr)H	70	H
Grundig Super Hi-fi 'CrO_2'	Cr	(CrO_2)H	70	H
Hitachi UD-ER	Fe	N	120	H
Hitachi UD-EX	Cr	(CrO_2)H	70	H
Maxell UD-XLII	Fe cobalt	H	70	H
Maxell UD-XLI	Fe cobalt	N	120	H
Maxell UD	Fe	N	120	H
Maxell UL	Fe	N	120	N
Memorex MRX_3	Fe	N	120	N
Memorex High Bias	Cr	(CrO_2)H	70	H
Philips Ferro	Fe	N	120	N
Philips Superferro	Fe	N	320	H
Philips Superferro I	Fe	N	120	H
Philips Chromium	Cr	(CrO_2)H	70	H
Philips Ferrochromium	FeCr	(FeCr)H	70	H
Philips Metal	metal	(metal)H	70	H
Pyral Maxima	Fe cobalt	N	120	H
Racal-Zonal 'Professional'	Fe	N	120	H
Racal-Zonal XL Gold	Fe	N	120	H
Racal-Zonal XL Silver	Fe	N	120	H
Ross RX1	Fe	N	120	N
Ross RX2	Fe	N	120	N
Scotch Dynarange	Fe	N	120	N
Scotch High Energy	Fe	N	120	H
Scotch Chrome	Cr	(CrO_2)H	70	H
Scotch classic Ferri-chrome	FeCr	(FeCr)H	70	H
Scotch Master I	Fe	N	120	H
Scotch Master II	Fe	(CrO_2)H	70	H
Scotch Master III	FeCr	(FeCr)H	70	H
Scotch Metafine	metal	(metal)H	70	H
Sony FeCr	FeCr	$N(CrO_2)$	70	H
Sony CDX	Fe	(FeCr)H	70	H
Sony AHF	Fe	N	120	H
Sony BHF	Fe	N	120	N
Sony CHF	Fe	N	120	N
Sony Metallic 46	metal	(metal)H	70	H
TDK -D	Fe	N	120	N
TDK -AD	Fe	N	120	H
TDK -SA	Fe	N	70	H

1 Fe = Ferric Oxide tape (LH), Cr = Chromium dioxide (CrO_2), FeCr = Ferrochrome, metal = pure metal coating.

2 N = normal bias (Fe), H = high bias (CrO_2 or FeCr or metal)

3 N = Normal, H = high output.

Table 12 Cassette machines

Product	Power[1]	Bias[2]	Equalisation[3]	Level Indicator[4]	Noise Reduction[5]
Aiwa AD6500	M	LH, Cr, FeCr	70/120	T.V.U., PkL	D
Aiwa AD1250	M	LH, Cr, FeCr	70/120	T.V.U.	D
Aiwa AD6300	M	LH, Cr	120	T.V.U. PkL	D
Aiwa AD6800	M	LH, Cr, FeCr,	70/120	T.V.U., PkM	D
Aiwa AD 6400	M	LH, Cr, FeCr,	70/120	T.V.U., PkL	D
Akai CS 34D	M	LN, Cr	120	—	D
Akai GXC 740D	M		—	—	D
Akai CS 7020	M	LN, Cr		T.V.U.	D
Akai CS 7040	M	LH, LN, FeCr	—	T.V.U. PkL	D
Akai GXC 709D	M	LH, LN, Cr, FeCr		T.V.U., PkL	D
Akai GXC 7300	M	LN, Cr, FeCr		T.V.U., PkL	D
Akai EXC 5700	M	LN, Cr, FeCr		T.V.U., PkL	D
Alpha CD 101	M	LH, Cr, FeCr	—	T.V.U.	
Alpha CD 601	M	LH, Cr, FeCr	—	T.V.U.	D
Amstrad 7050	M	LH, Cr, FeCr	—	T.V.U.	D
Amstrad 7060	M	LH, Cr, FeCr		T.V.U.	D
Amstrad 7070	M	LH, Cr, FeCr		T.V.U.	D
BASF 8200 CrO_2	M	LH, Cr	70/120	T.V.U.	D, DNL
Brocord 5000	M	Auto, LH, Cr		PkM	D
Brocord 1900	M	LH, Cr, FeCr	—	V.U.	D
Crown CTR 375	M	Auto, LH, Cr		ALC	—
Crown CTR 300	B, M	LH			—
Decca DC 2000	M	LH, Cr		T.V.U.	D
Decca DC 1100	M	LH, Cr	—	T.V.U.	—
Dual C919	M	LH, Cr, FeCr		T.V.U.	D
Dual C939	M	LH, Cr, FeCr		LED	D
Eumig Metropolitan Concert	M	LH, FeCr, Cr	—	LED	D
Ferguson 3280	M	LH, Cr		T.V.U.	D
Garrard GC300	M	LH, Cr		T.V.U.	D
Garrard GC350	M	LH, Cr	—	T.V.U.	D
Goodmans SC 4000	M	LH, Cr, FeG		T.V.U., PkL	D
Grundig CN 830	M	LH, Cr, FeCr		T.V.U.	D
Grundig CN 730	M	LH, Cr (Auto)		T.V.U.	D, DNL
Grundig CN 930	M	LH, Cr, FeCr		T.V.U.	D
Grundig CN 1000	M	LH, Cr, FeCr	—	T.V.U.	D
Grundig CNF 300	M	LH, Cr, FeCr	70/120	T.V.U.	D
Hitachi D220	M	LH, Cr	70/120	T.V.U.	D
Hitachi D550	M	LH, Cr		T.V.U.	
Hitachi D555	M	LH, Cr, FeCr		T.V.U.	
ITT ST 720	M	LH, Cr		T.V.U.	
ITT 8021	M	LH, Cr, FeCr	70/120	T.V.U. PkL	D
JVC KD 35		—			D
JVC KD 720	M	LH, Cr, FeCr			D
JVC KDS 200 II		—			—
JVC KDA8	M	automatic	variable	T.V.U.	D
Lenco C 2003	M	LH, Cr, FeCr		PkM	D
Lenco C 1102	M	LH, Cr		T.V.U., PkL	D, DNL
Marantz 5010	M	LH, Cr, FeCr		T.V.U.	D
Marantz 5020	M	LH, Cr, FeCr		T.V.U.	D

Product	Power[1]	Bias[2]	Equalisation[3]	Level Indicator[4]	Noise Reduction[5]
Marantz 5025	M	LH, Cr, FeCr	—	—	D
Marantz 5030	M	LH, Cr, FeCr	—	T.V.U., PkL	D
National Panasonic RS 612 US	M	LH, Cr	70/120	T.V.U.	D
Neal-Ferrograph 102	M	LH, Cr	70/120	T PkM	D
Neal-Ferrograph 103	M	LH, Cr,	70/120	T PkM	D
Philips N 2521	M	LH, Cr, FeCr		T.V.U.	D, DNL
Philips N 2534	M	LH, Cr		—	D
Philips N 2538	M	LH, Cr, FeCr		—	D
Pioneer CTF 9191	M	LH, Cr	70/120	T.V.U., PkL	D
Pioneer CTF 1000	M	LH, Cr, FeCr	70/120	T.V.U., PkL	D
Pioneer CTF 4040	M	LH, Cr, FeCr	70/120	T.V.U.	D
Rotel RD 10F	M	LH, Cr	70/120	T.V.U.	D
Rotel RD 30F	M	LH, Cr, FeCr	70/120	T.V.U., PkL	D
Sansui		—			
Sharp Optonica RT1155	M	LH, Cr, FeCr		T.V.U., PkL	D
Sharp Optonica RT2050	M	LH, Cr, FeCr	70/120		D
Sharp Optonica RT3535	M	LH, Cr, FeCr	70/120		D
Sharp Optonica RT5838	M	LH, Cr, FeCr		T.V.U.	D
Sony TC 118 SD	M	LH, Cr, FeCr	70/120	T.V.U.	D
Sony TC 177 SD	M	LH, Cr, FeCr		T.V.U., PkL	D
Sony TC 525	B, M	LH, Cr	—		
Sony EL 5	M	LH, Cr, FeCr		T.V.U.	D
Sony TC 158 SD	B, M	LH, Cr, FeCr		T.V.U.	D
Sony TC 229 SD	M	LH, Cr, FeCr		T.V.U.	
Tandberg TCD 330	M	LH, Cr		PkM	D
Tandberg TCD 310 II	M	LH, Cr		PkM	D
Tandy			—		
Teac A 103	M	LH, Cr		T.V.U.	D
Teac A 150	M	LH, Cr		T.V.U., PkL	D
Teac A 303	M	LH, Cr		PkM	D
Teac A 480	M	LH, Cr, FeCr		T.V.U., PkL	D
Technics RS-676 USD	M	LH	—	—	D
Technics RS-615 US	M	LH, Cr	—	T.V.U.	D
Technics RS-630 TUS	M	LH, Cr		—	D
Technics RS-640 USD	M	LH	—	T.V.U.	D
Technics RS-9900	M			—	D
Telefunken C 3300 Hi-fi	M	LH, Cr		PkM	D
Telefunken MC 2400	M			T.V.U.	D
Toshiba PC 230	M	LH, Cr		T.V.U.	D
Toshiba PC 3060	M	LH, Cr	70/120	T.V.U.	D
Toshiba PC 4020	M	LH		T.V.U.	D
Toshiba PC 4030	M	LH, Cr	70/120	T.V.U.	D
Toshiba PC 6030	M	LH, Cr	70/120	T.V.U.	D
Toshiba PC 5460	M	LH, Cr, FeCr	70/120	PkM, VU/Pk	D
Trio KX 520	M	LH, Cr, FeCr		T.V.U.	D
Trio KX 620	M	LH, Cr, FeCr	70/120	T.V.U.	D
Trio KX 720	M	LH, Cr, FeCr		T.V.U., PkL	D
Trio KX 830	M	LH, Cr		T.V.U., PkL	D
Trio KX 1030	M	LH, Cr, FeCr,		T.V.U., PkL	D
Uher CG 362	M	LH, Cr, FeCr		PkM	D

Product	Power[1]	Bias[2]	Equalisation[3]	Level Indicator[4]	Noise Reduction[5]
Uher CR 210	M	LH, Cr		PkM	—
Uher CR 240	B, M	LH, Cr, FeCr		PkM	D
Uher CG 330	M	LH, Cr	—	T.V.U.	D
Uher CG 350	M	LH, Cr		T.V.U.	D

1 M=mains, B=battery

2 LH=normal Fe, Cr=chrome, FeCr=Ferrochrome, auto=adjustable, LN=low noise Fe

3 70=High (CrO_2), 120=normal (Fe)

4 T.V.U.=twin V.U. meter, PkM=peak reading meter, PkL=peak LED indicator, ALC=automatic level control

5 D=Dolby system, DNL=DNL system

note.–some machines have combined bias/equalisation switches.

One final point on the Dolby noise reduction system. This is a system specially designed to reduce the noise contributed by tape recorders and was first designed for studio tape recorders, about fifteen years ago. It is called a differential system, in that it operates only when the music is quiet. During these passages, it senses the low signal level and electronically raises the level of this signal as it is being recorded. On the louder passages of music, when noise would be unnoticeable, the system drops out and allows the proper dynamics of the music to be rendered. During the playback process, the tape recorder has inbuilt circuitry which provides the inverse, de-emphasis of these low level signals. Hence signals which are most susceptible to the effects of noise become clearer, which gives an important improvement in the overall recording system. Most of the LP records we hear today were made on studio tape recorders incorporating the Dolby noise reduction system. About ten years ago, a simplified Dolby system was introduced into cassette tape recorders, when their technical quality was in need of improvement. The system has been widely adopted (under licence) by audio equipment manufacturers and many, though not all, present day cassette recorders have a Dolby facility which may be used when making recordings. Similarly, the recording industry has adopted the same, simplified system, in making pre-recorded cassettes, and most are now recorded with it, as is shown clearly on the cassette.

Since cassettes recorded with the Dolby system have low level signals artificially raised, it follows that they should be replayed only on cassette machines which can carry out the inverse operation of restoring these signals to their proper relative level during the playback process. In practice, however, the only difference between 'Dolby' and 'non-Dolby' cassettes will be in the treble balance, the former having a greater treble emphasis than the latter. This difference is not normally too much of a distraction, however, as listeners to pre-recorded (Dolby) cassettes in their car will have found. Car radio-cassette machines do not incorporate the Dolby system.

CHOICE OF EQUIPMENT

Having described the main components and the alternative media, disc and tape, it is time to consider matching them together into a complete audio system. One link, perhaps the most interesting in the audio chain, the loudspeaker, has the whole of the following chapter devoted to it, and readers will wish to read about it further before making their final choice. For completeness, however, I have included some examples of loudspeakers in this section of the book, where I have selected equipment which can be assembled into a complete home audio system. These are arranged into three separate price categories, so as to meet every requirement. Before making these recommendations, a word on how your choice should be made. Firstly, readers should decide on whether they want to make a choice from separate audio units, turntable, amplifier, loudspeakers, or whether they prefer to buy a complete, all-in-one unit, called a music centre, which includes everything except the loudspeakers. In fact even the speakers will be easy to select if you are choosing a music centre, since many are made for use with specific music centre models. Music centres are quite compact units, and the market in them is wide, including models at budget prices, so readers who want a compact hi-fi system, without the bother of building it up from separate units, have a very wide choice. Generally, a choice should be made from speakers of smaller size, since the power amplifiers in most music centres are not designed to deliver very high power levels. Again there are many speakers of compact size designed for use with your music centre. An example is shown in Fig. 59. Apart from being designed to fit available space in the listening

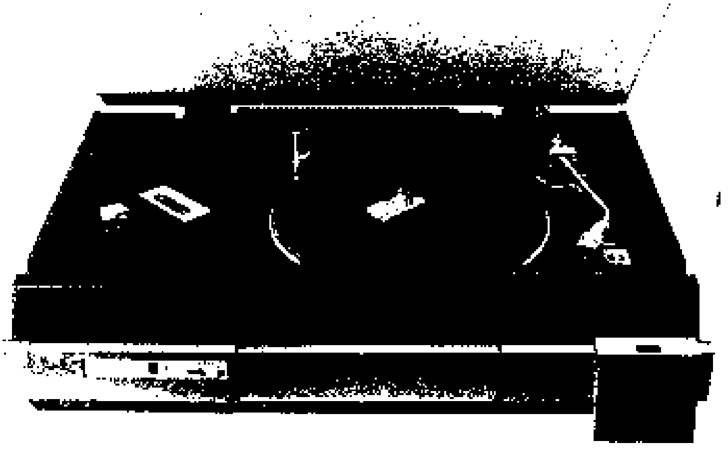


Fig. 59
One of the latest 'high specification' music centres featuring full-function remote control. Sony HMK 9000.

Table 13 Audio equipment in three price ranges

Item	Top price range
Pick-up cartridge, arm and turntable. (Abbreviated pu, arm & deck).	Linn LP 12 deck, Grace G707 arm and Supex SD 900E, m/coil pu. Linn ASAK m/coil pu. Micro Seiki DQX 1000 deck. Ortofon MC30, m/coil pu. Syrinx PU2 arm. Entre 1, m/coil pu. Mission 774 arm. ADC ZLM Mk II pu. SME Series III arm. Shure V15 IV pu. Micro Seiki LC80 m/coil pu. JBE deck. Head amplifiers for m/coil pu's: Lentek head amp. Ortofon T30 head amp. Fidelity FRT 5 head amp.
Amplifier, Tuner and Cassette deck. (Abbreviated amp., tun, & cass.) Where separate pre-amplifiers and power amplifiers are recommended these are abbreviated pre-amp. and power amp. respectively.	Naim Audio pre-amplifiers and power amplifiers. Quad 44 pre-amp. and 405 power amp. Rappaport PRE-1E pre-amp. Mission 772 pre-amp. and 775 power amp. Meridian 101 pre-amp. and 105 power amps. TVP-1 pre-amp. and TVA-10 power amp. (These are valve amplifiers). Yamaha T2 tun. Optonica ST9100 tun. Aiwa AD6900 Mk II cass. Nakamichi N582 cass. Sony TCK 8B cass. Philips AH 280 pre-amp. AH 380 power amp. and AH 180 digital tuner. Philips N 2552 cass. Revox B77 open reel tape recorder. Philips N 4520 open reel tape recorder.
Loudspeakers, and Headphones.	KEF Model 105 Mk II. Spendor BC3. Quad Electrostatic-Headphones by Stax, Sennheiser & Koss, e.g. Stax SR5, or SRX3; Koss PRO 4 AAA; SennHeiser HD430.

room, rather than for the best performance quality, a possible problem with a music centre is that the whole unit must be sent for maintenance if just one of its elements–amplifier, tuner, disc player, cassette deck–gives trouble. Even so, a music centre represents a functional general purpose audio system, giving good results from both tape and disc recordings, and these features will be well received by many listeners.

Perhaps, however, you wish to make your choice from separate, self-contained units, so you can include in your system those units whose performance and appearance you like best. Maybe you already have such a system, but wish to upgrade one of its components, such is the flexibility of separate unit audio. Of course, separate units, from different manufacturers, are not all universally compatible; indeed, they are not designed to be. For example, one does not try to match a modest amplifier in the budget price range to a pair of very costly, well extended loudspeakers designed for much more powerful amplifiers. Similarly, one must not invest in a specialist pick-up cartridge, only to mount it into a basic arm for which it was not designed, and will not track properly. Whilst the enthusiast will have checked all the relevant maker's specifications, he will still wisely ask advice before making his purchase. I would suggest the same to any purchaser; seek advice from an audio dealer who has the equip-

Middle price range	Budget price range
Thorens TD160 deck with Formula IV Mk III arm. DUAL 506 deck with own arm and Ultimo 10X pu. Rega 3 deck with Rega arm or SME 3009 Series 3S arm. EEI 500 pu. Grado F1 + pu. Ortofon M20FL pu. Hadcock 228S arm. Micro Seiki DQX 500 deck with own arm.	Connoiseur BD2 deck with own arm and ADC QLM34 Mk III pu. Sansui SR222 Mk II deck with own arm. Ortofon FF15E Mk II pu. Shure M95ED pu. Pioneer PL512 deck with own arm.
A&R A60E amp. A&R T21 tun. Yamaha CR620 integrated tun./amp. Sansui G5500 integrated tun./amp. Aiwa ADL40 cass. Tecnics RSM22 cass.	JVC AS3 amp. Optonica SM5100 amp. NAD 3020 amp. and 4020 tun. Nytech CTA252XD Mk II integrated tun./amp. Aiwa ADM100 cass. Sony TCK6B cass.
Lentek S4. Philips AH 585 Motional feedback. Rogers Compact Nonitor. Celef Mini Pro HE. The 'Hi-Fi Sound' Loudspeaker design, see p. 141 Headphones by Sennheiser & Koss, e.g. Koss K 145; Sennheiser HD 424X.	Videotone GB1. KEF Celeste Mk III. NAD Model 8030. The 'Hi-Fi Sound' Loudspeaker design, see p. 141. Headphones by Sennheiser & Koss, e.g. Koss K 125; Sennheiser HD 414X.

ment of your choice or who is prepared to obtain it for you. He will arrange for the equipment to be demonstrated, perhaps in your own home listening room.

As to the amount you should spend on your new audio equipment, that must be a desicion for you to make. Generally, though not in every case, the more money you spend, wisely, the more closely you will eventually come to hearing the very high quality results that are possible from the best home audio systems of today. It is implicit in this statement that the listener who spends a large sum *wishes* to appreciate the highest quality sound reproduction, or has that ultimate aim.

There are many reasons why a person buys a particular item of audio equipment, or any other item. His budget, his other priorities and his personal preferences. I cannot advise on this, but I can assemble some examples of audio equipment in different price categories, all of which will match well and give good performance (see Table 13). There are no doubt other items which readers know from their own experience combine into a good system, but space limitations prevent me from including them all. In the next chapter we take a look at loudspeakers in more detail, and after that the listening room, in Chapter 5. We shall then be ready to judge and enjoy music on our audio systems, and to this end I have given a music list at the end of Chapter 5.

4
Loudspeakers

During the fifty or so years since the loudspeaker was first invented, it has become totally established in sound reproduction. The relatively simple device that Rice and Kellog first described, has remained essentially unchanged to the present day. The experience of the past fifty years has shown that it would be difficult to find an alternative to this system which would give the same performance quality together with simplicity of construction and flexibility of use.

There are, however, a number of different loudspeaker systems on the market today, many of them containing several speaker units covering different parts of the audible range. This chapter aims to give an account of these different systems as well as to explain the principles and advantages of each. The final section gives details and specifications of the Hi-Fi Sound speaker listed in Table 14.

MOVING-COIL LOUDSPEAKERS

The speaker familiar to most readers is actually called a moving-coil speaker. It consists simply of a light circular (or elliptical) diaphragm or cone, freely suspended from a metal frame or chassis by springy suspensions around its edge and near its centre. Attached firmly to the centre of the cone is a cylindrical former, and wound on this is a coil of wire called the voice-coil. The cone and former are often made of stiffened paper, and the voice-coil is of copper wire. (Fig. 60)

The coil and former are positioned between the poles of a magnet. The early speakers used electro-magnets (i.e. magnets which had to be energised from a direct current) but nowadays permanent magnets of soft iron or ceramic materials are used. When a signal is applied to the voice-coil, a force will be exerted on the coil, according to the elementary theories of electromagnetism. Since the coil is rigidly attached to the cone, it causes the cone to move. If the signal comes from a power amplifier which is amplifying the sounds of music, then the speaker cone will be made to reproduce music. This is the complete description of the operation of a speaker. It is the more remarkable that such an elementary device should be the subject of so much interest and controversy among hi-fi enthusiasts.

In fairness, this simple scientific theory is not an adequate explanation of how a loudspeaker behaves since, as we shall see, there are a number of practical limitations of a speaker which detract from good sound reproduction.

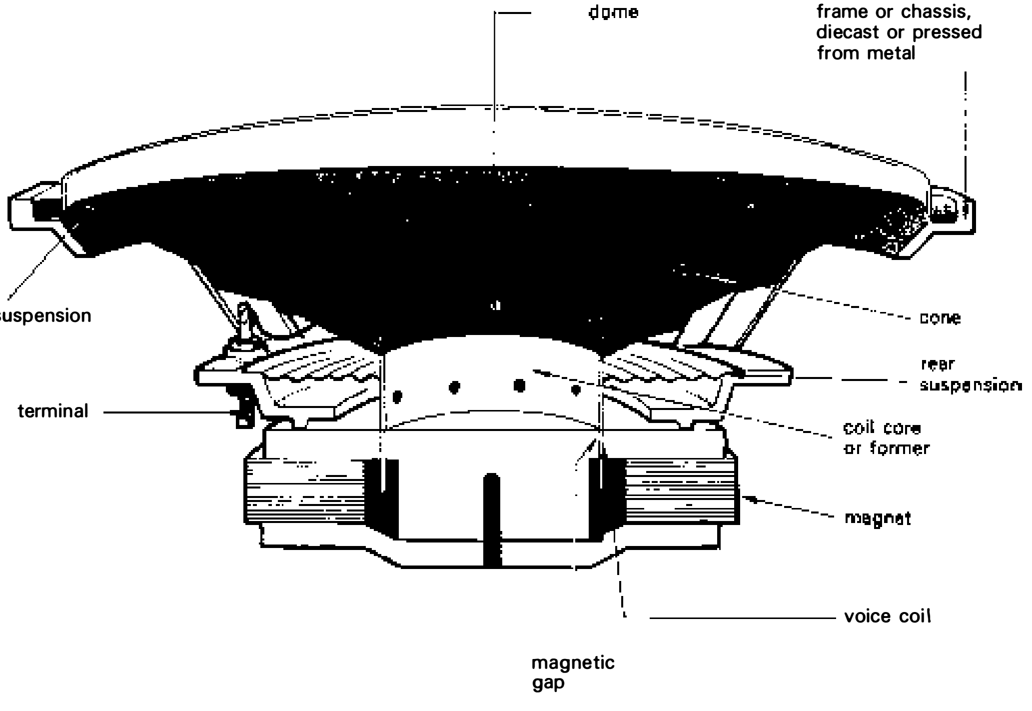
Bass enclosures

Just as for other sound sources, sounds will diffract around a loudspeaker cone, as they leave it, only if their wavelength is large compared with the diameter of the cone. This will always be true at bass frequencies, i.e. those below 200 Hz or so.

When the cone is moving, however, the front surface moves forward and compresses the air at the same time as the rear surface moves inward and rarifies the air. This means that at low frequencies the diffraction of these two disturbances around the cone would cancel each other and so reduce the efficiency of the speaker as a sound radiator. This is an important point for, as we said in Chapter 2, most of the energy in music and speech resides in the low frequency range.

The solution to this problem of 'back to front cancellation' is to mount the speaker unit into an enclosure. The enclosure may take many forms, and there are several different types in commercial use today, all of them designed to optimise the bass efficiency

Fig. 60
The construction of a modern moving-coil loudspeaker unit.



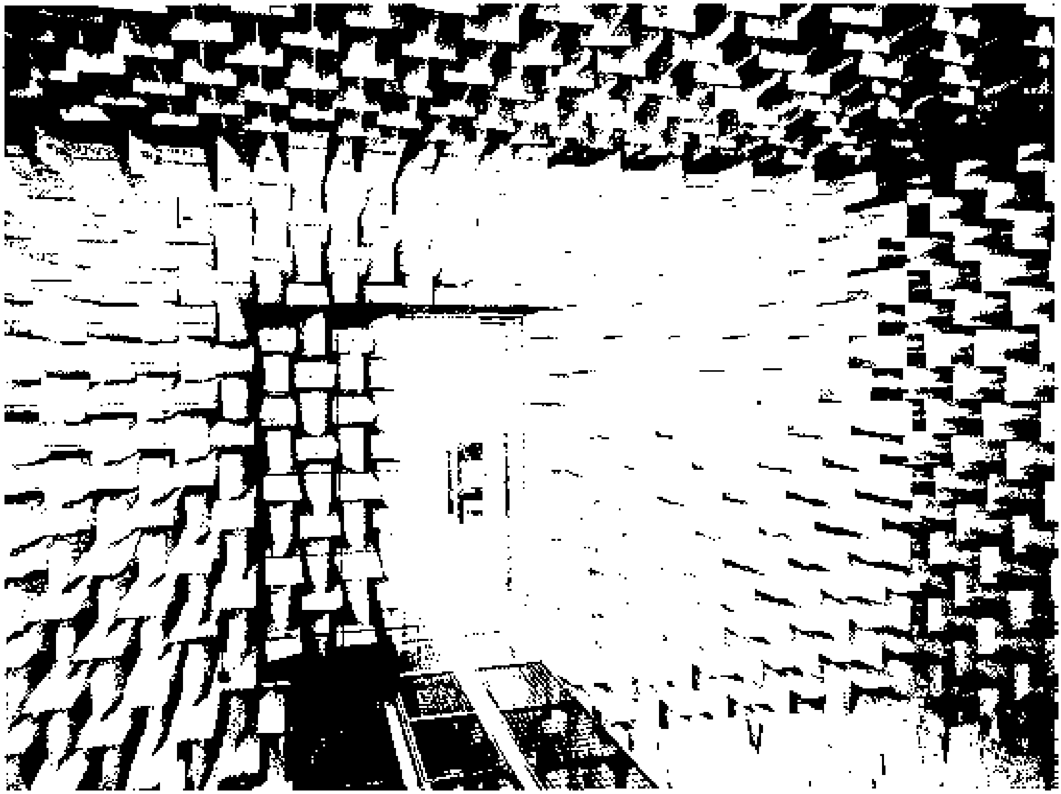


Fig. 61
A large anechoic (reflection free) room for audio measurements.

of the speaker. A loudspeaker enclosure is thus a housing for the speaker; it may be simply a flat, rectangular piece of wood, or 'baffle-board', with the speaker mounted in the centre, or it may be a cabinet which has an open back, or a completely sealed box of some kind. All these arrangements are enclosures, for they all serve to enclose or isolate the two radiating surfaces of the speaker so as to improve radiation efficiency. It follows that a loudspeaker enclosure serves a useful acoustic purpose only at low frequencies, where diffraction round the speaker unit is significant. When the enclosure is properly designed, low-frequency sounds will leave the front of the speaker only, and diffract around the *enclosure* without being cancelled. Then the speaker system will radiate usefully, equally in all directions.

Testing enclosures The most widely accepted way of observing how different types of enclosure affect the behaviour of a loudspeaker is to set the system up in a room called an *anechoic room* or acoustically 'dead' room. This room has special acoustic absorbing material, arranged in wedges, covering all of its six surfaces. The material absorbs almost all the sound energy that comes in contact with it, whatever its frequency, so that only direct sounds from the loudspeaker are picked up by the measuring microphone. From this we can detect the different output pressure levels from the loudspeaker at any frequency in the audio range, and get what is called the *pressure-response* curve or *frequency-response* curve of the loudspeaker, a result well known to hi-fi enthusiasts. Normally we take the response with the microphone on the axis of the speaker, and at some fixed distance from it (e.g. 1 metre), for it is on or near this axis that we shall listen to music.

Open-back cabinet

The most basic enclosure, widely used in radiograms, radio and television sets is the open-back enclosure. The response of a speaker in a variety of open back cabinets is shown in Fig. 62. Notice that if the flat baffle-board in which the speaker is mounted is made larger, the system responds usefully further down into the bass frequency range. Increasing the depth of the cabinet does not alter the frequency at which the output of the system begins to fall, but it does make the response more 'peaky' at this point. This is because the cabinet behaves something like an organ pipe, and exhibits an acoustic resonance at a frequency where the depth of the cabinet is one-quarter wavelength long. For a cabinet 61 centimetres deep, this frequency is therefore at about 120 Hz. This phenomenon occurs also at all odd multiples of this frequency, as can be seen from the ripples in the response shown in Fig. 62. Hence the next 'peak' occurs at about 360 Hz and so on.

Although this kind of enclosure will serve, it is obviously far from ideal since the peaky response will add a subjective 'boom' to sounds as the output from the back of the cabinet adds to that from the front, and will introduce its own transient 'ring' or 'overhang' to bass frequencies. As readers will know, the effect can often be heard from radiogram cabinets of this type, and sadly it colours reproduction.

Sealed enclosure

It would be better if sound radiation from the rear of the speaker could be totally enclosed, for then it would not interfere with the front radiation. This is the philosophy of the totally enclosed cabinet or sealed enclosure, (Fig. 63). This is a very convenient enclosure and is in very wide use today. It is sometimes called an 'infinite' baffle, in that, like a flat baffle-board of very large dimensions, it completely removes the rear radiation from the loudspeaker like mounting the unit in a wall. The sealed-cabinet enclosure is rather different, in fact, from an infinite flat baffle, because it seals behind the speaker unit a fixed volume of air, which reacts against the movement of the cone. The smaller the cabinet volume, the greater the stiffness of the enclosed air, and this affects the low-frequency limit of the speaker's response.

The basic loudspeaker drive unit, like any other mechanical system, has its own resonance frequency. It is called the speaker's 'mechanical resonance' or 'bass resonance', and for all loud-

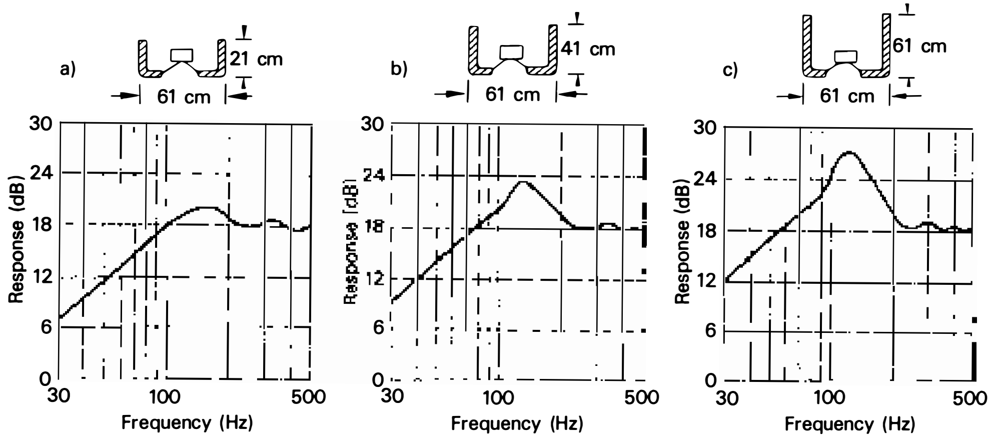
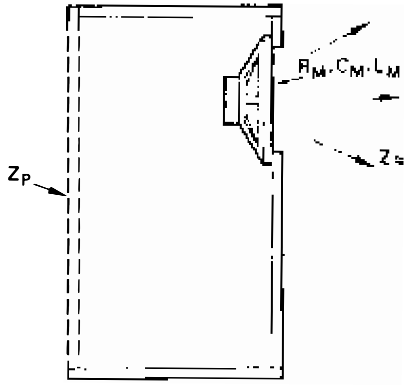
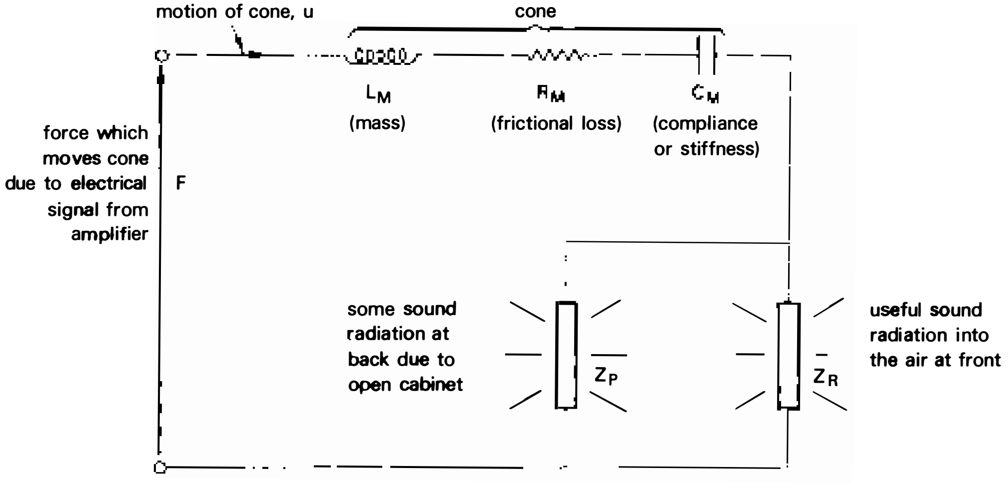


Fig. 62
The open-back loudspeaker cabinet used in radiograms and radio sets. The greater cabinet depth extends the bass response but makes it more peaky. The analogous circuit shows the basic elements of the cone, and sound radiation from front and back of the cabinet.





speakers it occurs at the *lowest* point of the speaker's useful operating range. For a bass loudspeaker, therefore, this resonance frequency will fall at around 40 Hz or so. (For the great majority of loudspeaker systems in use today, the bass resonance occurs somewhere in the range 25–100 Hz.)

The added air stiffness provided by this type of enclosure raises the bass resonance frequency of the speaker. Below this frequency the acoustic output of the speaker falls sharply, so that care must be taken in design or the speaker will lack fullness and body. Because smaller speakers have become more popular in recent years, to fit the available space in home living rooms, new design techniques have been evolved, with the aim of maintaining the efficiency of small speaker enclosures. For example, very small bass drive units radiate less power than larger cones, so 'long throw' units have been designed whose small cones can move over wider limits and handle more electrical power. Hence the loss of output caused by the tiny enclosure can be compensated by driving the speaker with more power at low

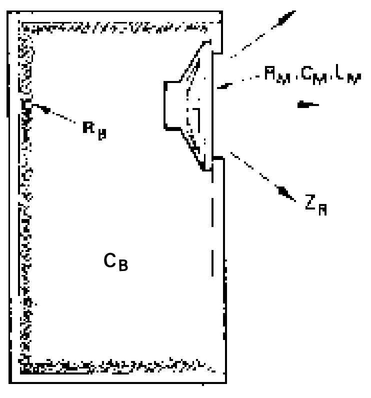


Fig. 63
The sealed box loudspeaker system and its analogous circuit. The enclosed air compliance raises the bass resonance frequency of the speaker whilst the cabinet lining makes the response smoother. Sound is radiated only at the front.

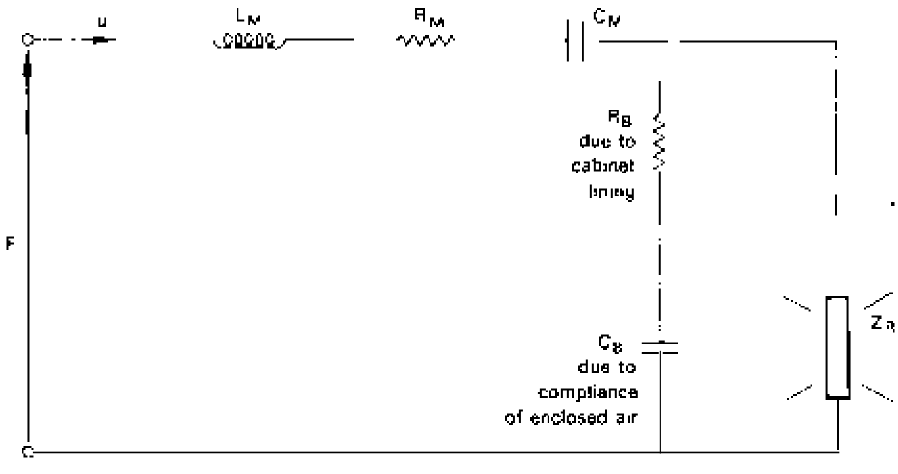
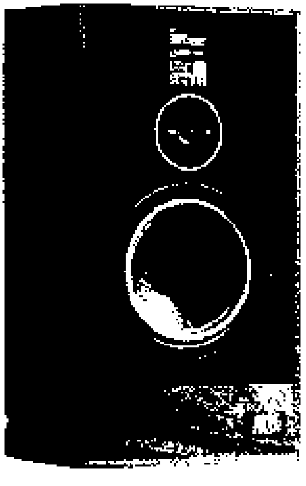


Fig. 64
A cross sectional view of a recent sealed cabinet loudspeaker system, the B&W DM5.

Fig. 65
The Acoustic Research model AR 25 acoustic suspension loudspeaker, which relies upon the stiffness of the enclosed air to provide bass driver suspension.



frequencies. This is known as 'equalising' the loudspeaker. These design techniques, which add to the cost of the system, solve the problem of inefficiency in small enclosures of the sealed type. An example of such a system is shown in Fig. 64.

The acoustic suspension loudspeaker

An effective solution to the problem of using a sealed cabinet to enclose a speaker, is to reduce the stiffness of the cone itself and use the enclosed air to provide most of the stiffness of the system. This has certain advantages. Firstly, the supporting stiffness of the cone is no longer provided by suspensions around its edge and centre, but by the air which acts uniformly over the entire cone area. Secondly, air is a more linear medium than normal stiffner materials and therefore introduces less distortion at high dynamics (i.e. when the cone movements are large and more easily distorted). This means that the speaker can reproduce fundamental sounds more faithfully, without introducing undesired harmonics of its own.

The acoustic suspension loudspeaker was first introduced commercially in America about twenty-five years ago. Its inventor was Edgar Villchur, who founded the famous American Company, Acoustic Research. This company adopts the acoustic suspension principle in all their speaker designs, of which some new ones have recently become available (see Fig. 65.).

The acoustic labyrinth and transmission line loudspeakers

Whilst it might seem expedient to remove the radiation of sound from the back of a loudspeaker, so as to prevent interference with the normal front radiation, it would be a better engineering solution if the energy in the rear sound wave could be turned into some useful account, since it represents about half the effective output from the speaker at bass frequencies.

One of the earliest methods by which this could be done was first proposed in the 1930s, as the Acoustic Labyrinth. This was a speaker in which a 'labyrinth' of internal baffles was used to guide the rear sound wave to the front of the cabinet, where it could add to the front wave (Fig. 66). The labyrinth consists of a folded

Fig. 66
Two different forms of acoustic labyrinth and their analogous circuits. The analogous circuit shows that the speaker resonance is quite heavily damped by the acoustically treated labyrinth.

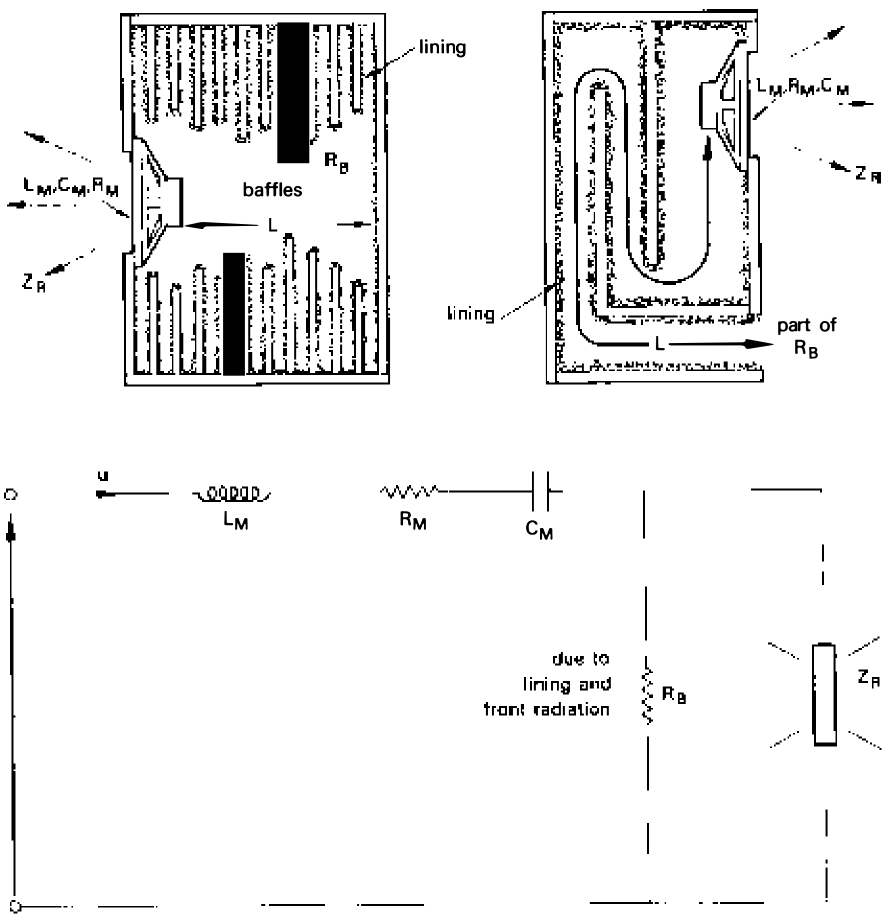
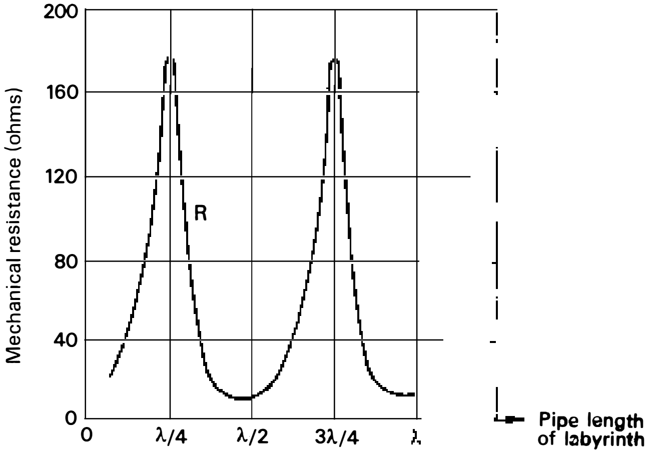


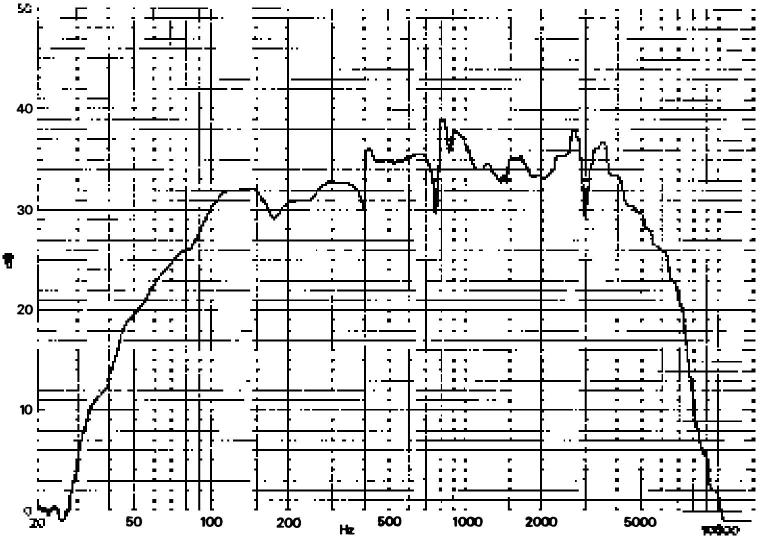
Fig. 67
Curve showing how the acoustic labyrinth loudspeaker is best matched to the drive unit when the line is one-quarter wavelength long. This is also true at higher frequencies.



resonant pipe which, when it is one-quarter wave in length, couples onto the radiated sound most efficiently. For example, if the length of the folded pipe is 3 metres, it will be one-quarter wave in length at 30 Hz. Hence the folded pipe serves to present to the rear of the speaker cone a relatively high acoustic-resistance component, into which sound radiation is coupled. The rear of the cone 'sees' a high resistance, and gives up energy to it efficiently. Figure 67 shows how this resistance varies with the wavelength of sound. It can be seen that at all odd multiples of one-quarter wavelength, the right coupling conditions are produced. For all even multiples of one-quarter wavelength, i.e. for all multiples of one-half wavelength, the cone sees a very much lower resistance, in fact at these points it is as though 'free air' is behind the speaker cone.

Sound energy which is coupled into the pipe travels along it and eventually reaches an open port at the front of the speaker

Fig. 68a
The response of a typical bass loudspeaker unit when fitted in a medium size cabinet 25 centimetres deep, sealed and unlined. Notice sharp changes in response at 400 Hz, 800 Hz and 3 kHz, i.e. whenever cabinet depth is a multiple of one-half wavelength.



baffle. The delay incurred in the path along the pipe serves to bring into phase the two waves from the speaker, that from the front and rear, so these add together and make the system more efficient overall. Unfortunately, because of the wide variations in resistance shown in Fig. 67, the output of the acoustic labyrinth loudspeaker would not be smooth, but would show peaks and dips as the pipe coupling changed with wavelength. Also, sound would be reflected within the pipe and this would upset smooth bass radiation. These problems are solved by lining the inside of the labyrinth with an acoustic absorbent material, such as fibre wool, cotton wool or long hair sheepswool. This serves to dampen the peaks caused by the pipe and reduces the undesirable effects of reflections occuring at the surfaces of the wooden panels inside. Absorbent linings of this kind are used in other types of loudspeaker enclosure, as some readers will know, generally to reduce reflected sounds within the cabinet. It can also be shown that (Fig. 68) the speed at which sound travels in these absorbents is lower than it is in free air, so the effect of the linings in a given cabinet is to make it behave the same as a *larger* enclosure would without lining.

This means that in the acoustic labyrinth, the lining effectively lengthens the folded pipe, and so extends the useful bass response of the system even further. In fact, if one goes a step further, and so lines the enclosure that the damped pipe presents a fixed, resistive impedance to the rear of the speaker cone, a different kind of enclosure system is produced.

This is known as the Transmission Line loudspeaker, whose basic operating principle is similar to the labyrinth, except that the pipe formed by the labyrinth of internal baffles is *filled* with low density acoustic absorbent material. Then the rear of the speaker cone should 'see' a fixed acoustic resistance of quite high

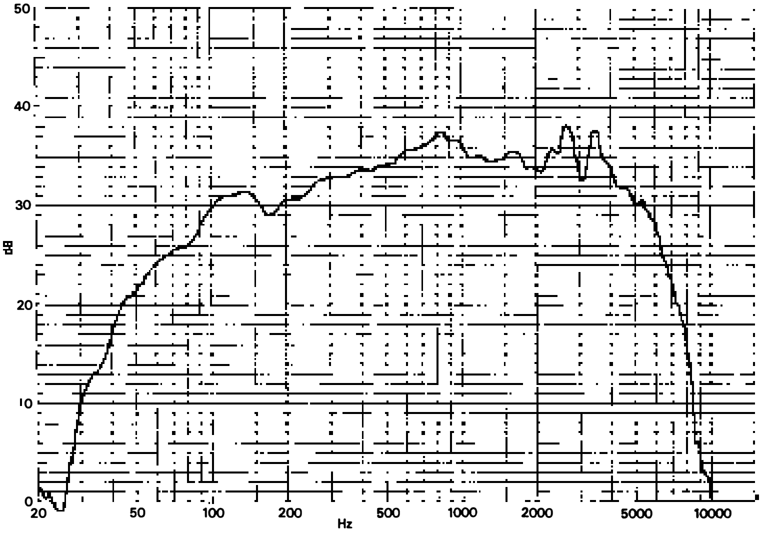


Fig. 68b
Same loudspeaker as in 68a but lined and filled with acoustic absorbent material. Reflections are heavily dampened and the speaker has a much smoother response at all frequencies in its range.

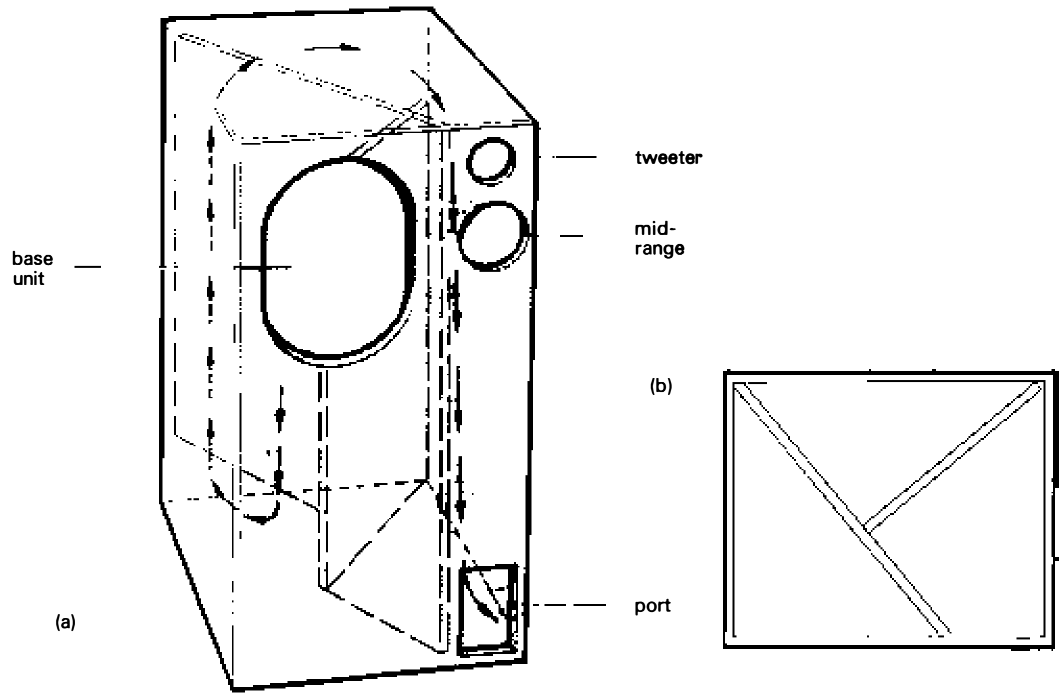


Fig. 69
The transmission line loudspeaker design by Dr A Bailey.

value, which will absorb most of the energy from the cone. In theory, the length of the pipe or line is not important to the operation of the system, because it correctly terminates or couples the speaker at all bass frequencies. In practice, it is not easy to absorb very low frequency sounds by the line unless it is of sufficient length, and it has been found that the line must be of at least one-quarter wave in length at the lowest frequency of operation, just as for a labyrinth. In fact the distinction between the labyrinth and transmission line loudspeakers becomes quite a subtle one.

Practical examples of the damped labyrinth or transmission line loudspeaker are the system designed by Dr A. Bailey (Fig.69), also those by I.M.F. Loudspeakers, and the design by the late B. J. Webb (Fig. 70), which has also been made available commercially as the Lentek Monitor Loudspeaker. A curve of the acoustic response of the Lentek Monitor is also shown in Fig. 70. The uppermost curve is the fundamental response. It was taken in an anechoic chamber, and the fall in response below 200Hz is inherent in its design. Sound radiation from the port increases at very low frequencies but this is not registered in the anechoic room because the port is near the bottom of the enclosure. However, when the speaker is placed on the floor in a normal listening room, sounds cannot diffract downwards and are reflected up to aid the overall output and give a flat response, right down to 30Hz. Interesting also are the lower two curves in

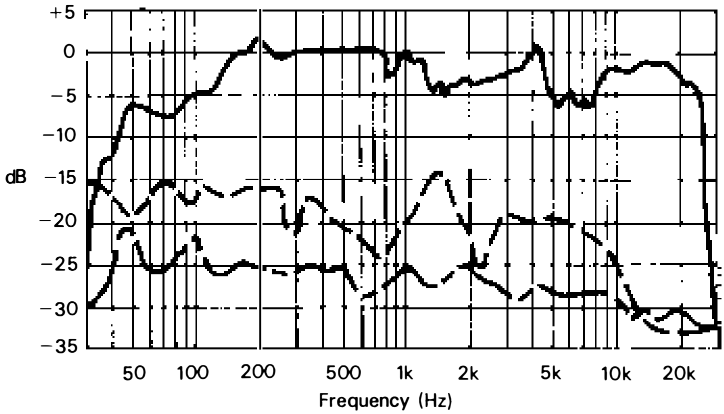


Fig. 70
The Lentek monitor transmission line loudspeaker, with acoustic response shown above, left. Notice how the harmonic distortion at low frequencies shows peaks at 30 Hz and 90 Hz.

this figure; they show the 2nd and 3rd harmonic generation of the speaker. They are quite low as they should be and do not detract from the performance of the loudspeaker, but the operation of the line is evident from their peaks and dips. At the fundamental, quarter wavelength frequency of 30 Hz, the 2nd harmonic component rises to a peak as the line comes into matching. The same is true at about 90 Hz, the three-quarter wavelength frequency. The trend in 3rd harmonic, however, is in the reverse sense. This component is mainly contributed by the driver unit, but it is reduced when the line comes in to matching the unit and remove sound energy from it. This loudspeaker system gives remarkable accuracy of reproduction in the upper bass reaches of voice sounds, and captures the low frequency ambient sounds of the concert hall with equal fidelity.

The reflex or 'tuned' loudspeaker

Another way of recovering the sound radiated from the rear of the cone in an enclosure is to use it to cause an air or acoustic (Helmholtz) resonance to take place in a vent or port at the front of the cabinet. This arrangement, and its circuit analogy, are shown in Fig. 71. The system is called a 'reflex' or 'phase-inverter' enclosure; it was first described in 1940, and is in very wide use today. The three essential resonance elements, mass, springyness or stiffness and resistance loss, are provided by the air, just as they are in the Helmholtz resonator found in some musical instruments. The volume of air within the enclosure provides the stiffness element, whilst the volume of air in the vent or port moves as a whole and so constitutes a moving mass of air. Finally, movement of the air in the vent causes sound radiation to take place at the mouth of the vent, so there must be some resistance element at this point which improves the efficiency of the speaker.

Like the sealed-box enclosure, this system will have a bass resonance frequency, determined by the speaker cone and the enclosure air stiffness. It will also have a second resonance frequency, that of the Helmholtz resonator, and the speaker is so

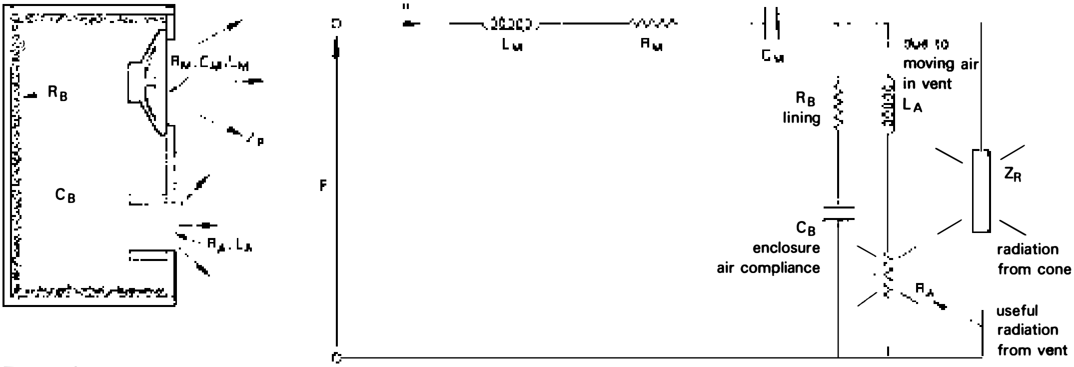
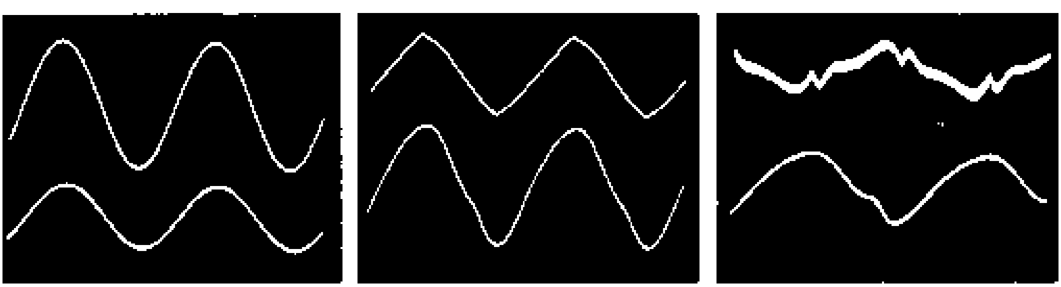


Fig. 71
The reflex loudspeaker cabinet and its circuit analogy. The acoustic resonator in the vent has the same elements, L, C and R as the cone, and sound radiation takes place more efficiently from the vent than the cone at frequencies below the cone resonance.

designed that this frequency occurs at a slightly lower point than the bass resonance. This means that, as the loudspeaker unit itself is beginning to loose efficiency, the air resonance comes in and boosts the output of the system. At this point (i.e. Helmholtz resonance) the movement of air in the vent is at its maximum, and is in phase with the motion of the front of the speaker cone. This continues as far up as the bass resonance frequency, and then the output of the cone starts to rise, whilst that from the vent begins to fall. At higher frequencies still, the output from the vent becomes negligible, and the cone takes over to radiate almost all of the useful sound output of the system.

This is shown in Fig. 72a), b), c). These are actual photographs taken from two microphones, one placed at the vent and the other very close to the centre of the cone of a reflex loudspeaker. Notice, in c), how usefully the vent is contributing to the output of the speaker, even though radiation from the *front* of the speaker cone is negligible. This is because the small sound pressure due to the rear is still being magnified by the vent. This can be thought of as coupling more effectively the relatively high mass of the speaker cone to the small mass of air outside, and so improving radiation efficiency. Those interested in the reflex loudspeaker will perhaps know that a third system resonance occurs, above that of bass resonance. This is that due to the cone and the air mass in the vent, and it helps in augmenting the cone output as it begins to rise to its final value in the upper bass frequency range.

Fig. 72



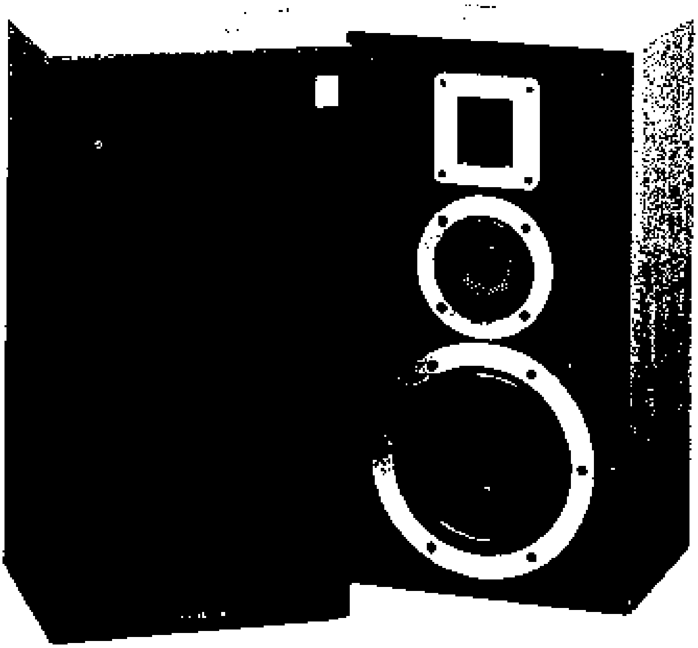


Fig. 73
Reflex enclosure.

Fig. 72
The sound output from the cone (above) and vent of a typical reflex loudspeaker system at three important frequencies:
(a) 114 Hz, well above speaker resonance. Cone output is large, vent output small.
(b) 56 Hz, below speaker resonance but near vent resonance. Now cone output is low and distorted while vent output is high and smooth due to resonance of air vent.
(c) 34 Hz, well below both cone and vent resonances. Cone output is very low and distorted but vent is still giving some useful output. (Relative output levels can be directly compared in (a) and (b), but in (c) the cone output has been greatly magnified).

There are numerous commercial examples of reflex bass enclosures in current use; it is likely that the reader's own loudspeakers are of the reflex type. A familiar modern example is shown in Fig. 73.

The auxiliary bass radiator (ABR) or drone cone loudspeaker

A possible problem with the reflex loudspeaker is that, because the air motion in the vent at bass frequencies is quite high, the system can produce harmonic distortion which gives a roughness to bass notes. Although it is usual to line the cabinet with acoustic material, both to broaden the air resonance and reduce internal cabinet reflections, there is a limit to the help these measures can provide. (Those who have studied the subject will know that the true damping of the reflex loudspeaker must be provided by the bass driver, not by cabinet lining. Otherwise the bass character would be 'flabby' and soft. Hence the high air velocities in the vent remain a problem.)

An elegant and economic solution to this problem is provided by what is called the ABR or Drone Cone loudspeaker enclosure. Like the reflex system, it is designed to improve efficiency by turning the rear cone radiation into useful account in a resonant device on the front of the cabinet, but it does so by employing another loudspeaker cone as a mechanical resonator. This cone is not driven from the amplifier like the main loudspeaker

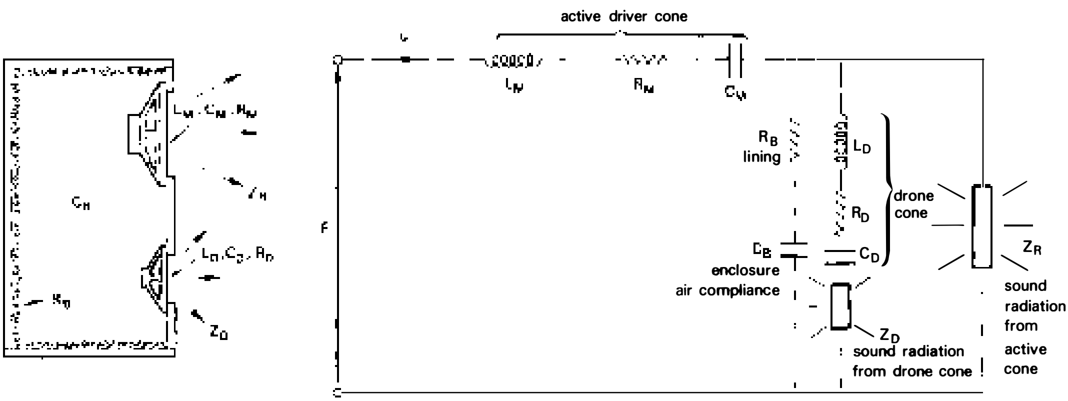


Fig. 74
The ABR loudspeaker replaces the air resonance of the reflex speaker by a mechanical resonance of the ABR cone. The active and 'drone' cone have similar resonant elements.

unit, but acts as a 'drone' by relying on the main unit to drive it into motion (Fig. 74). The ABR or drone cone can be mechanically similar to the main, driven unit, except that it is arranged to assist radiation below the bass resonance frequency of the main driver. Since the two cones will be of similar mass, the lower resonance of the drone cone is produced by making its suspensions less stiff, i.e. more compliant. Hence the principle of the ABR system is identical with the reflex speaker, but the performance of the ABR is more under the designer's control.

For example, the drone cone has a large area, so it can radiate sound as effectively as a small reflex vent but with a much smaller velocity. The drone cone will also move more uniformly, unlike the disturbed air in a reflex vent. Its resonance frequency can be adjusted conveniently, by small changes in the mechanical stiffness of its suspensions, whereas in the reflex loudspeaker such changes involve altering the area or depth of the vent.

Interestingly, the Drone Cone speaker was first described in the mid 1930s, and offered in America by JBL in the 1950s. In England, Rola Celestion were the first Company to produce the system, though now there are several other commercial designs (Fig. 75). JBL first offered the system with small metal weights, which could be attached by the user to the centre of the drone cone to change its resonant frequency. This allowed the experimenter to tailor the extreme bass performance of his loudspeaker to his own requirements. Another point worth mentioning is that because the drone cone has a relatively low mechanical stiffness, most of the stiffness of the system is provided by the air in the sealed enclosure of the speaker. Hence the ABR loudspeaker behaves just like an acoustic suspension loudspeaker at low frequencies, i.e. in the range where the drone cone operates. A system adopted more by designers in recent years, the ABR speaker is said to have a 'warmer' bass character than a typical reflex design, due no doubt to its more uniform, controlled motion.

Fig. 75
The KEF model 104 'drone cone' loudspeaker. This has been a very popular loudspeaker; the drone cone is at the bottom.

Horn loading a loudspeaker
The few systems described above cover the methods used in almost every loudspeaker for obtaining efficient bass. Another method of improving efficiency, which can be applied at any audio frequency, bass, mid-range and treble, is to 'horn load' the loudspeaker. The theory and principles of horn loading are very complex, and it is no simple matter to design a system which will both improve efficiency and preserve accurate tonal reproduction. However, the history of the subject is a long one– some of the first experiments go back more than sixty years–and there are some very high quality designs which have become established.

We have described how, when an object is small compared with the wavelength of sound, the sound waves diffract around it and progress as though the object were not there. Similarly, when the wavelength of sound is much longer than the size, say, of a loudspeaker cabinet, the speaker will radiate sound equally in all directions. At low frequencies, for example, sound waves

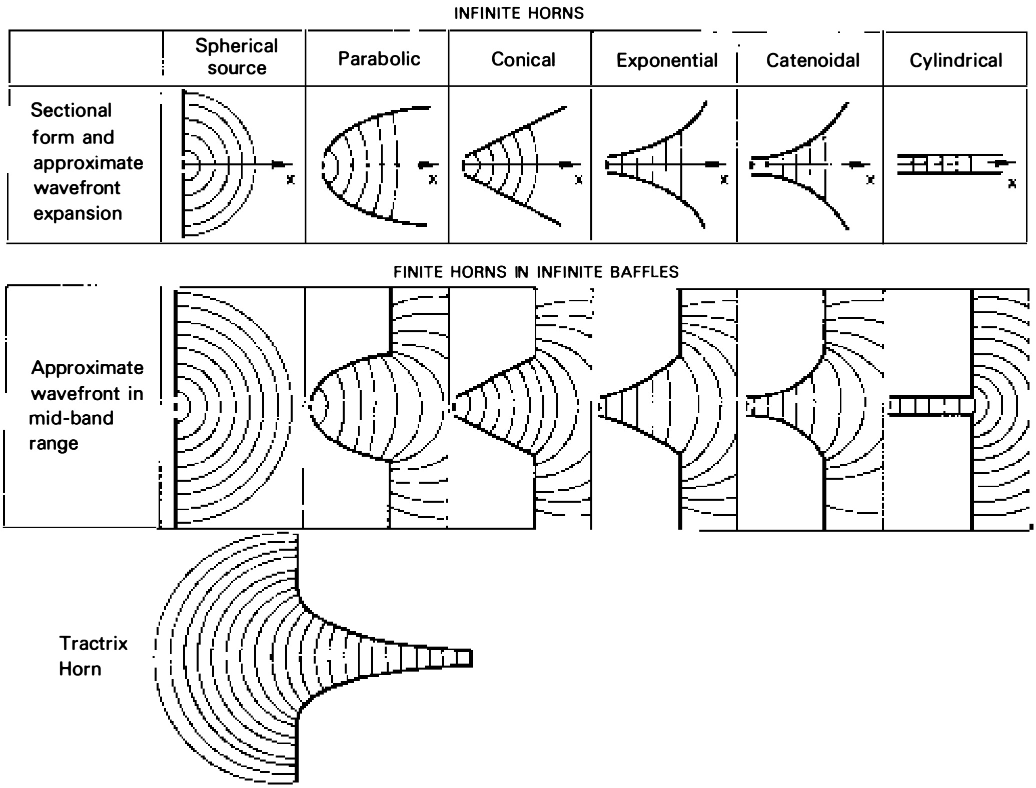


Fig. 76
Showing the types of wavefronts travelling in acoustic horns of different profile. The wavefronts are more curved in practical horns, because these have to be of a finite length.

always diffract around a speaker and so omnidirectional radiation takes place. This means that the sound waves advance as *spherical* surfaces or wavefronts, and as they get further away from the speaker, the wavefronts become less curved, i.e. more *plane*. The change in the area over which the wave advances, i.e. the expansion of the wavefront, becomes smaller as the wave moves further away from the source.

Although the theory is too complex to be described here it can be shown, scientifically, that when sound waves are *plane*, more useful energy is carried by them. This is why it is difficult for very small loudspeaker units to radiate bass frequencies efficiently. The wavefronts near the source are highly curved, and as such remove very little sound energy from the source. A larger source 'sees' a more resistive air medium, and imparts more energy to it. If some means could be found which would hold back the rate of expansion of the sound wavefront from a speaker, the speaker could impart more useful energy to the air. This is exactly what an acoustic horn is designed to do. It is based entirely upon the principle of preventing a rapid expansion of wavefront area, especially when very near to the sound source.

The two main factors which determine the efficiency of the horn over a particular frequency range are its 'mouth' area and the shape or 'profile' of the horn between its throat, where the speaker feeds the horn, and the mouth, where sound radiation takes place. For well-extended low-frequency performance, the mouth area must be large and the horn must not slope too steeply between mouth and throat (i.e. must have a low rate of taper). A variety of horn profiles have been tried over the years. One of the early profiles was the conical type, which has straight sides, and the parabolic horn, whose area grows proportionally with distance from the throat. Comparison of these with other types, however, show them to offer lower and less uniform throat resistances to the speaker unit, which makes them less efficient. These are illustrated in Fig. 76 and show the kind of sound wavefront expansions to be expected.

Notice how the wavefronts are more straight or plane near to the throats. A very widely-adopted horn shape is the exponential type, for which the rate of change of cross-sectional area is independent of distance from the throat. This horn is found to offer a very uniform trend in throat resistance, and within its efficient range of operation the wavefronts are almost plane. The catenoidal horn has similar advantages, and has only a very slightly different profile from the exponential one. If the mouth in these horns is adequately large, the throat resistance presented to the speaker has a smooth trend, but as the mouth is made smaller the curve becomes less regular and shows sharp peaks and dips (Fig. 77). The cut-off frequency of the horn, below which little or no useful sound radiation can take place, is determined by the 'flare' or 'taper' constant, i.e. the rate at which the horn changes area along its length. If sufficiently small taper rates are used, it is found that the lower limit of useful operation of the horn occurs when the mouth diameter is equal to about one-third of the acoustic wavelength. Thus a mouth diameter of 1 metre (40 inches) is useful down to about 100 Hz, and operation of the horn above this point will be satisfactorily smooth.

Efficient horn loading at bass frequencies requires horns which have considerable length and large mouth areas, so it is

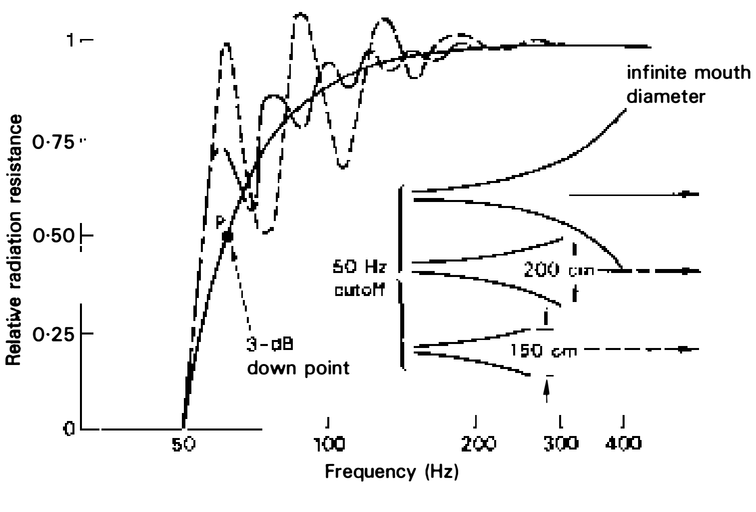


Fig. 77
Showing how the acoustic resistance presented by the air to an acoustic horn is made smoother by making the horn longer and wider at the mouth.

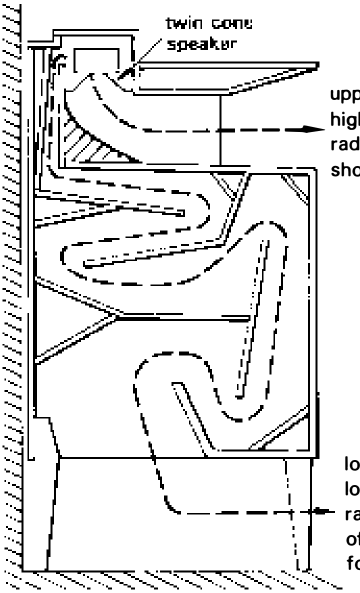


Fig. 78
The acoustic baffles in the classic Lowther horn loudspeaker system.

usual to 'fold' the horn in order to keep its external dimensions more convenient. In some cases the loudspeaker employs a combination of direct front radiation and rear horn loading. One of the classic systems of the 1950s was the Lowther horn (Fig. 78), employing front and rear horn loading for the entire frequency range. The rear of the drive unit is loaded by a long, folded horn of approximately exponential cross-section, whilst upper-middle and treble frequencies are radiated from a short horn, with only a single bend, near the top of the enclosure. Another system which is horn loaded is the Tannoy G.R.F.

A very successful range of horn-loaded loudspeakers for home use is that designed by P. W. Klipsch, who lives in America. Paul Klipsch has been known for his work in acoustic horn design for many years. It is said that he is a man who, before giving you the time of day, will first tell you how the clock works! Very great experience has gone into the design of the Klipschorn (Fig. 79). The system uses a folded three-sided (trihedral) corner horn, about 122 centimetres in length and of exponential cross-section. The walls of the room act as extensions of the horn, which is why the speaker must be positioned in the room corner. There are five or six different systems in the current range, with different horn sizes and efficiencies. All have linear horn loading in the treble register as well. Acoustic efficiencies of the order of 10 per cent or more are claimed.

The directional characteristics of a horn loudspeaker are similar to those of the drive unit which feeds it. At low frequencies it will be omnidirectional, but for large mouth areas the system becomes more directional at higher frequencies. An efficient bass horn needs a large mouth area which would make the system focus sounds

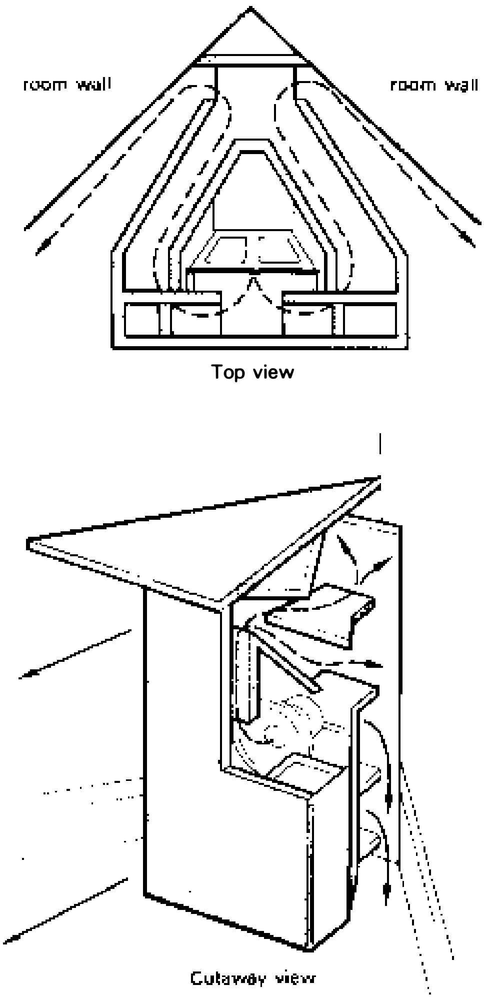


Fig. 79
The acoustic design of the Klipschorn.

into a narrow beam in the upper frequency range. In domestic systems such as the Klipschorn, the problem is overcome by employing special, smaller horns for the middle and treble registers. In some applications, such as in reproduction in large halls or theatres where horn efficiency is a great advantage, the problem is solved by using multicellular horns, i.e. horns containing a number of smaller horns, fed in parallel from a single drive unit (Fig. 80). At low frequencies the system behaves as a single radiator, but in the treble range the individual cells act independently, each one radiating in the direction it faces. The smaller dimensions of each mouth in relation to the acoustic wavelength make the system far less directional and this helps in projecting high-frequency sounds over a wide area of the audience.

Horn loudspeakers can be very efficient–perhaps five to ten times more than a direct radiator of similar dimensions–but

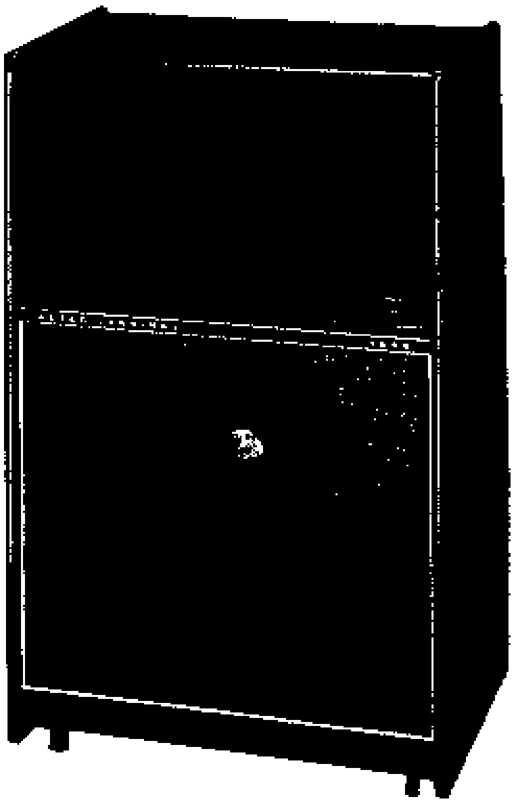
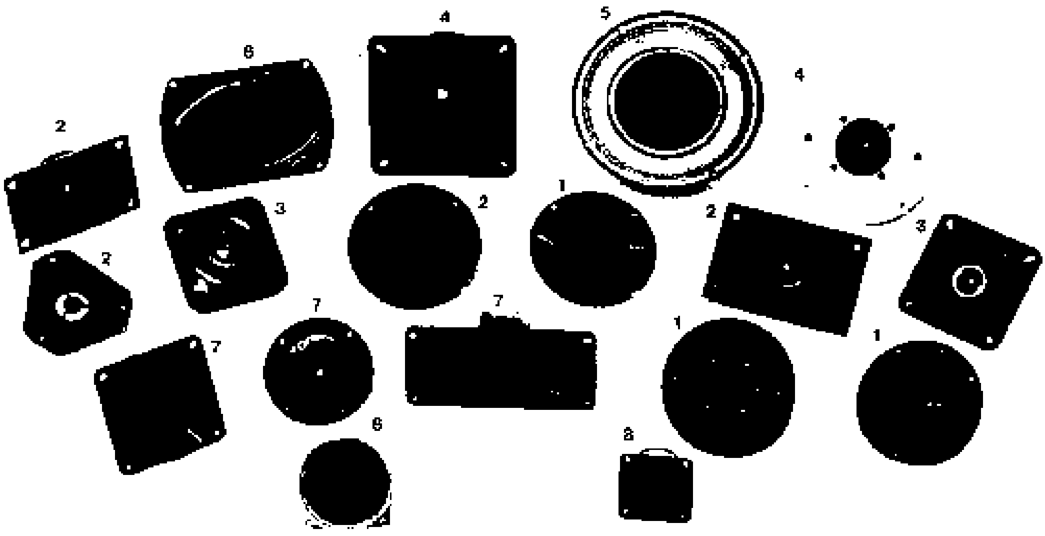


Fig. 80
A high-power stage sound loudspeaker system using a multicellular horn for treble radiation.

they must be very carefully designed if they are to be free from reflections and resonant irregularities in response. In the domestic listening room, horn-loaded speakers come closest to reproducing the true dynamics of extreme bass notes without distortion. One devotee in England, R. N. Baldock, has a 490-centimetre (16-foot) horn-loaded bass reproducer mounted in the sub-floor structure of his listening room! I have never heard any other speaker radiate the note of a 16 Hz organ pipe at its true level and tonal quality.

Middle and high frequency speakers

Loudspeaker enclosures, with their different forms of bass 'loading', look after the low musical register, but other units are required for the mid-range and treble frequencies. A loudspeaker system of the highest quality will contain two, three or even four separate drive units, each one designed specifically to cover a limited part of the audio spectrum. Depending upon the particular design, the bass driver will operate only up to a few hundred Hz, beyond which a mid-range or mid-frequency unit will be used. This unit will have been designed to give its best performance over about three octaves, so it will extend the response of the



system up to about 3 or 4 kHz. Beyond this point, a special treble frequency unit or 'tweeter' will operate, and usually cover the remaining part of the audio range, up to 20kHz. Occasionally, a 'super tweeter' or 'top top' unit will be used, which will faithfully reproduce extreme treble frequencies, say from 10 or 12kHz up to and beyond 20kHz.

Drive units designed for the higher frequency range are much smaller than bass drive units so that the shorter, high-frequency waves can be radiated over a wider angle. A mid-range unit will have a diameter of about 100–125 millimetres, or less, whilst a tweeter diaphragm may be of 25-millimetre diameter or less. These reductions in size bring other advantages. Firstly, a small-diameter cone will move more uniformly and so reproduce the rapidly changing high-frequency notes more faithfully. Secondly, the energy contained in the upper reaches of music is, on average, much lower than in the bass and lower middle register, so that mid-range and tweeter units need not have such heavy cones. Since in all moving-coil loudspeakers the cone movements decrease as the frequency rises, it follows that middle and high frequency drive units can be designed to give maximum efficiency within their confined working frequency ranges.

A range of middle and treble frequency drive units in current use is illustrated in Fig. 81. They range from simple paper cone units (of exactly similar basic construction to larger-diameter cone drivers) for use in inexpensive system design to specially formed 'dome'-shaped units coupled to complex Helmholtz air resonant cavities to maintain a smooth performance right across the range. One of these units, the Celestion model HF 2000, was chosen for the special loudspeaker design example given at the end of this chapter. Readers will find more information

Fig. 81
Examples of middle and treble speaker units in wide use.
1 High quality dome or pressure tweeters.
2 Medium quality dome or pressure tweeters.
3 Inexpensive dome or pressure tweeters.
4 High quality dome mid-range drivers.
5 High quality plastic cone mid-range driver.
6 Inexpensive cone mid-range driver.
7 Piezo-electric tweeters.
8 Inexpensive cone tweeters.

about this tweeter there. Dome-shaped diaphragms in tweeters have become very wide in application today.

Dome-shaped units The material used for the dome might be a fabric, impregnated with a suitable viscous damping substance, which will help improve their fundamental response and absorb sound energy which may travel in the dome itself. Alternatively, plastics such as melinex have been tried. The diameter of the dome will vary from about 37 millimetres to about 75 millimetres for mid-range speaker units whilst for tweeters it will be smaller, ranging from about 19 millimetres to 32 millimetres in diameter. In tweeters, metals such as titanium or beryllium are also used, which possess very high rigidity coupled with low mass, an ideal combination for the task of radiating sounds faithfully. The velocity with which sounds travel in these materials is very high, so that the energy stored in them will be very low, allowing the sound to be radiated more accurately.

Piezo-electric units Another kind of tweeter unit which has come into commercial use is the piezo-electric unit. This operates on rather different principles than the moving-coil speaker. The unit features a piezo-electric element which, when a signal voltage is applied to it, moves mechanically and so produces acoustic waves. The orders of movement of the material are quite small, so that the application is limited only to tweeter units. In some examples, an acoustic horn is coupled to the vibrating element to further improve its efficiency. One advantage of the piezo-electric tweeter is that it has a very high electrical impedance, which makes it almost self-protecting and allows crossover filters of the very simplest kind to be used to feed it. The performance of this type of unit can be of high quality.

In a modern loudspeaker system of high quality, there might be a bass driver of 25-centimetres diameter, with a paper or plastic cone and suspensions of rubber or PVC plastic, covering the range from 30 Hz to about 700 Hz; a mid-range unit of 10–13 centimetres diameter with a specially treated cone or a smaller diameter soft fabric dome, covering the range from 700 Hz to about 4 or 5 kHz, followed by a dome tweeter about 3 centimetres diameter for the treble range, i.e. extending smoothly up to around 18–20 kHz.

Crossovers

Since the different drive units in a complete loudspeaker system are designed only to operate over particular parts of the audio frequency spectrum, they must not be supplied with any other frequencies. The individual units must therefore be integrated together, and this is done with an arrangement of filters which constitutes what is known as the crossover network of the loudspeaker. The crossover is a very important part of the design of the speaker, for it must ensure that an integrated sound is produced, rather than a collection of separate sound sources.

The crossover circuit or network contains a number of capacitors, inductances and resistors, combined to form *filters*, which discriminate between different frequencies within the audio range. The bass driver will be fed from what is called a low pass filter, i.e. one which passes low frequencies but attenuates all others above a certain frequency, called the crossover frequency or crossover point. If a separate mid-range driver is employed, it will be fed from a band pass filter, i.e. one which passes frequencies within a certain band, but rejects all those above and below it, namely the upper and lower crossover points respectively. Finally, the tweeter will be fed from a high pass filter, which begins to pass frequencies above a certain point (the upper crossover point) while rejecting all those below it. The types of filter formed from the basic elements are called Butterworth filters, after the scientist who first suggested them. The reason this class of filters are most widely used in loudspeaker crossover-designs is that they give the maximum flatness within their frequency range together with the smoothest curve of attenuation beyond this range.

The Butterworth filters divide the audio spectrum between the different drive units such that the *power* fed to them adds up to equal the power fed to the filters from the amplifier. Hence these filters are designed so as to match the speaker units perfectly and to incur minimum power loss.

A very elegant engineering solution to the problem of loudspeaker design is the one adopted by the British company KEF Electronics in recent years. They see the crossover network, not just as an electrical filter, but as an instrument by which all the many factors affecting the behaviour of the drive unit are taken into account. The filter is therefore designed or 'synthesised' so as to provide the optimum acoustic response of the drive unit, and may differ considerably from the theoretical design value. There are some cases, of course, where this approach leads to results which are little different from the basic values of the theoretical filter, but equally there are others where it is significant in refining performance. These filters, and variations of them, are considered in more detail at the end of the chapter, where they will be applied to the loudspeaker design example produced specially for readers of this book.

In certain, more specialised applications, a loudspeaker system will be designed to have *active* filters. These are filters which form part of an amplifier which is built into the speaker enclosure. Each drive unit has its own associated amplifier and active filter circuit, and this allows the designer almost complete freedom to optimise the performance of the loudspeaker. Any differences in sensitivity or response uniformity between the

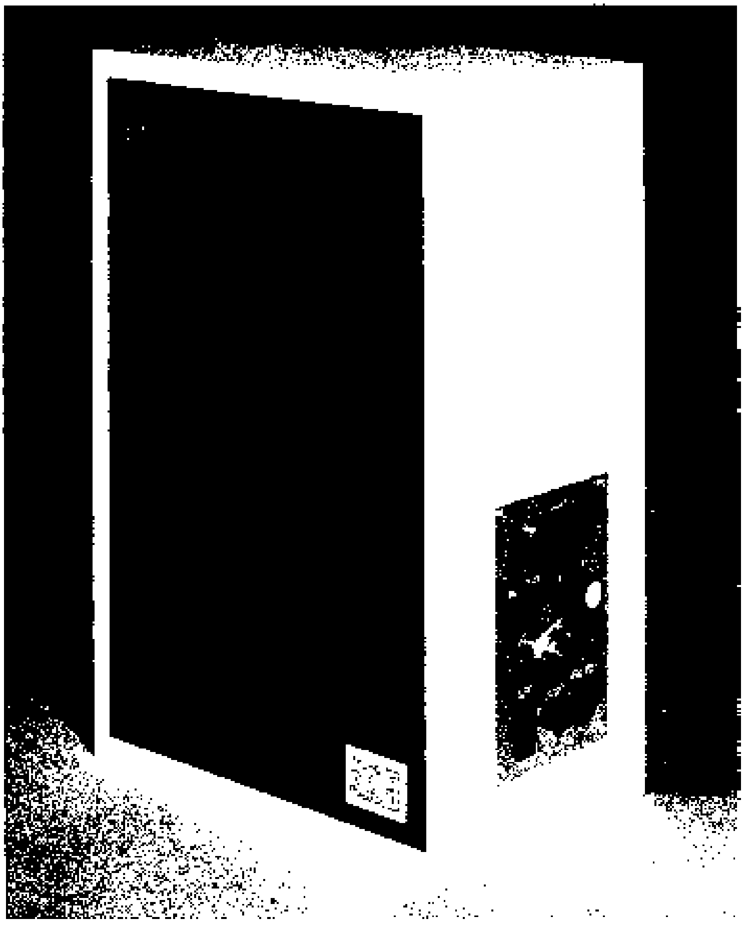


Fig. 82
An active loudspeaker system which incorporates the power amplifiers built into the cabinet. The Tycho SA5.

drivers can be compensated through the appropriate amplifier and active filter circuit. There is no problem about matching the loudspeaker to an external power amplifier, because these are inbuilt and the speaker is simply supplied from an ordinary pre-amplifier. Loudspeakers of this kind are not widely available on the domestic market, perhaps because they do not give the user the option of choosing his own power amplifier but instead add appreciable cost to the loudspeaker system. On the other hand, active speaker systems can be very convenient in small broadcast studios, where very high-quality performance and compactness are required (Fig. 82).

SPECIAL LOUDSPEAKERS

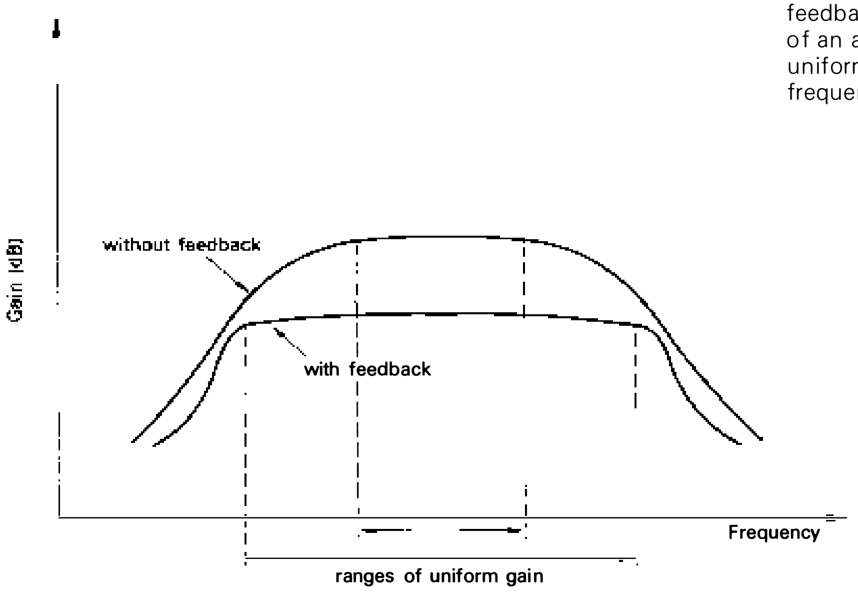
The moving-coil loudspeaker, and its variants in size, shape and profile, has been and is universally employed in speaker system design. However a few other special types were first developed in the 1950s, when home hi-fi was beginning to grow, and have since become firmly established in use. They will be known to many readers and so will be of special interest to us now. These three special types are the motional feedback loudspeaker, the ribbon loudspeaker and last but by no means least, the electrostatic loudspeaker.

The motional feedback loudspeaker

The motional feedback loudspeaker employs the principle of *negative feedback* featured in domestic amplifiers over the past forty years. Negative feedback is used to cancel the distortion that can occur in a system. In the amplifier, a small fraction of the output signal is returned or fed back to the input to be combined with the input signal. The combined signal is passed through the amplifier in the usual way, except that if the amplifier produces any distortions the signal fed back will reflect this when it is combined with the incoming signal. If the signal fed back is negative, i.e. if it is arranged to oppose the incoming signal, then the negative-feedback signal can be used to cancel or greatly reduce this distortion. This process is said to be causal, i.e. the correction cannot come before the error has occured, but in practice it takes place almost instantaneously so that the distortion is not heard. The application of negative feedback in an amplifier must reduce its overall 'gain', but this can be restored at some other point where distortion is less likely to occur.

Another advantage of negative feedback is that it can extend the range of uniform amplification of an amplifier (Fig. 83) and makes its gain more stable and consistent. These advantages are obviously tempting to the loudspeaker designer, especially at low frequencies where so much energy in music and speech resides and

Fig. 83
Showing how negative feedback makes the gain of an amplifier more uniform over a wider frequency range.



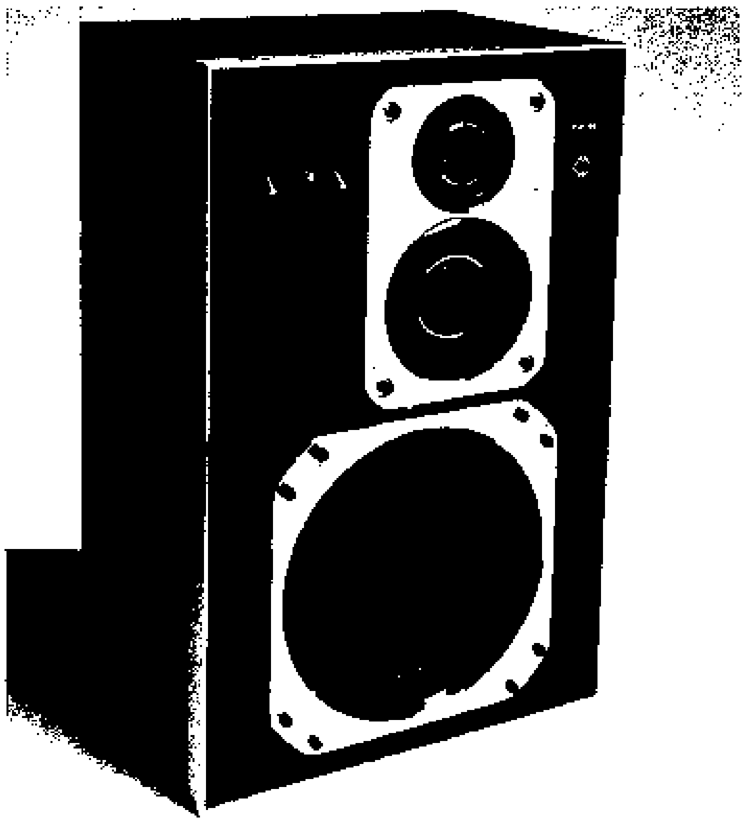


Fig. 84
Philips model AH 587 motional feedback loudspeaker. The system incorporates a motion sensing transducer mounted at the centre of its bass driver. There are separate power amplifiers for the mid-range and treble drivers but these do not feature motional feedback.

causes a speaker to produce quite high distortion. Whilst negative feedback is fairly easy to apply to a wholly electronic device like an amplifier, it is not quite so simple to apply to a speaker. What would be needed would be some device which monitors the *motion* of the speaker cone, translates it into an electrical signal, and feeds it back to an amplifier which could correct the electrical signal supplying the speaker. This is the principle of the motional-feedback loudspeaker.

There have been two or three commercial designs in recent years; one was sold under the name of 'Servo-Sound' and another well-known system is the Philips Motional Feedback Loudspeaker, or MFB speaker (Fig. 84). The Philips MFB loudspeaker has some interesting features which are worth describing. This small speaker contains two (in some models three) separate drive units, each one fed from its own, inbuilt amplifier. In all models the bass driver is a high-quality cone unit but, mounted at the centre of the cone is a very tiny, light transducer and associated amplifier stage.

The transducer senses the acceleration of the cone and, if on high bass or lower-middle music peaks its motion becomes non-linear, the transducer senses this and feeds a correction signal back to the power amplifier connected to the voice-coil of the bass unit. By this means the motion of the cone is controlled to within linear limits and harmonic distortion is greatly reduced. The motional feedback system operates throughout the bass and lower-middle frequency range, in fact up to about 500 Hz. Throughout this range, it can be assumed that the centre of the cone moves along with the entire cone area as a whole, i.e. the cone is in 'isophase' motion.

The crossover circuitry in the bass range is active (i.e. amplifiers are associated with the crossover circuit), so that the overall response in this range can be accurately tailored to give optimum performance. The result is that the MFB speaker gives a bass response which is exceptionally well extended, especially for an enclosure of such small size, typically this is 30 litres or so, although one model in the range is much larger and handles more power. Since the upper-range drive units each have their own inbuilt power amplifiers, the speaker can be used with any external amplifier, even a pre-amplifier will drive it. It also has an input level control so that an existing power amplifier may be used to drive it. It should be noted that the MFB speaker has very tiny signal power requirements; all it needs is a signal voltage, at least one volt, but higher voltage levels may be applied.

The MFB loudspeaker permits the listener to produce high dynamics from a small enclosure with far less distortion than would be possible from a normal system of similar size. The fact that the electronics are inbuilt may also prove an advantage where space is a problem. Alternatively, those with a rather modest amplifier and speaker combination, say a music centre, can upgrade their system by using a more powerful pair of MFB loudspeakers, which of course the music centre can adequately drive. On the other hand, the cost of the MFB system is higher and the listener does not have any choice of the amplifier he uses. The MFB speaker has become a successful alternative to conventional speakers, and it is an interesting development. It is well suited to those who want good quality sound reproduction without the problem of having to match up their equipment, or to those who may wish to improve on it.

The ribbon loudspeaker

A most interesting loudspeaker system, which took the audio world by surprise when it was first launched and is still in use today, is the Ribbon loudspeaker. It was developed in the early 1950s by Stanley Kelly, and it was fascinating to hear him talk about it recently, when I was researching for this book. The ribbon speaker was originally developed to provide a smooth, accurate response right up to the extreme treble range, not for the audiophile but for laboratory measurement purposes in calibrating microphones and earpieces for hearing aids. Remarkably, in the late 1940s, there were no wide-range sound sources available for such work, and it could take a whole day to calibrate a microphone by a method proposed by Lord Rayleigh almost a century before, the Rayleigh Disc method!

The ribbon loudspeaker design is based on the principle of the moving-coil speaker, i.e. the electrodynamic principle, except that the moving coil now becomes a very light metal ribbon which also acts as the sound radiator. Hence in principle the ribbon speaker is very simple; it consists of a light corrugated metal ribbon, which is made in fact from an aluminium-magnesium alloy, positioned between the poles of a powerful magnet. The electrical audio signal is applied between the two ends of the ribbon, and the force it experiences causes it to move and radiate sounds directly.

Fig. 85
An early Kelly ribbon loudspeaker unit with horn loading.

Production of the delicate ribbon radiator is difficult because of its tiny weight and special shape. The dimensions of the ribbon are 56 × 8 millimetres, whilst the weight is less than 4 milligrams. The ribbon is corrugated because, when it flexes appreciably or becomes warm due to high signal powers being fed to it, the corrugations help keep it rigid and prevent sagging. Its electrical resistance, too, is extremely small, only ·024 ohms, so the ribbon loudspeaker is a very low impedance device, both mechanically and electrically.

Because of its small size, the ribbon would not be sufficiently effective in radiating sounds directly, so an acoustic horn was designed for it. The length was made about equal to one-quarter wavelength at cut-off frequency of 1 kHz., about 10 centimetres, with a taper of something between exponential and catenoidal form. The mouth was made rectangular for directionality reasons, measuring 20 × 10 centimetres, with the shorter dimension horizontal, whilst the throat was of slightly larger size than the ribbon itself. The commercial product was first marketed in 1955, and offered a then unheard of smooth response between 3 kHz and 20 kHz. Built into the case was a high slope crossover circuit with a crossover point at 3 kHz, and this supplied a transformer, also built in, which matched into the low-impedance ribbon. When Decca bought the company and its design from Stanley Kelly in the early 1960s, they still continued to make it, and feature it today as part of their 'Decca London' range of speaker products (Fig. 85.) The Kelly ribbon speaker is an established alternative to moving-coil diaphragm units for smooth high-frequency reproduction. Its low mass enables it to recreate the subtle transients in the treble register with great accuracy. Whilst dome tweeter units have now caught up with it, there was a time when there was nothing to touch the performance of the Kelly ribbon speaker.

The electrostatic loudspeaker

In all the cases described earlier of enclosures for the moving-coil loudspeaker, the 'reflex' cabinet, the ABR system, and horn loading, the aim was to improve the efficiency of the speaker by

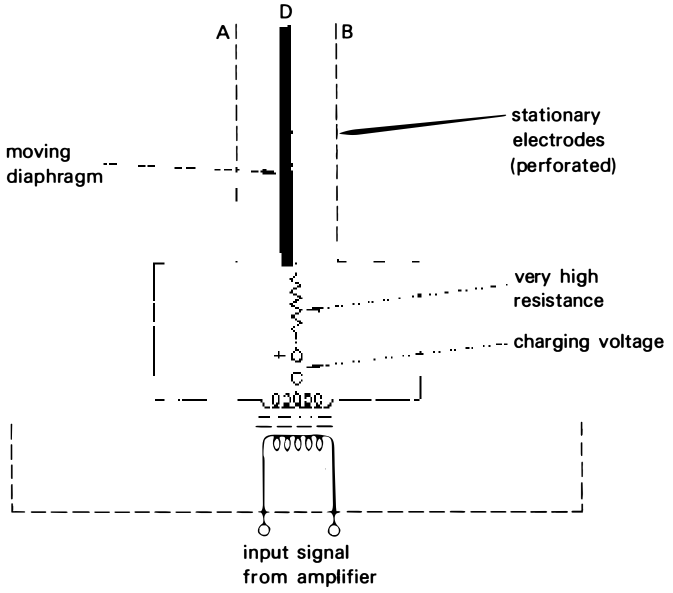


Fig. 86
The principle components of an electrostatic loudspeaker.

arranging that the radiation resistance (i.e. the air resistance 'seen' by the speaker and into which it imparts sound energy) is increased as much as possible. These systems work effectively but, to anyone looking at loudspeaker design afresh, and considering a totally different kind of loudspeaker, certain questions might be asked. For example, one might ask, why is it necessary to improve the efficiency of the moving-coil loudspeaker? Is it because the basic speaker is itself an inefficient device? In fact the answer to this question must be yes. A moving-coil speaker system is at best only a few per cent efficient; a typical value for a small bookshelf design is rather less than 1 per cent. Most of the energy supplied from the amplifier to a moving-coil speaker is wasted as heat in the voice-coil. The reason is that so much energy must be supplied to this coil in order to get the cone to move, because it is so heavy compared with the amount of air it disturbs. In other words, the relatively heavy cone is poorly matched to the very light air load. Thus the moving-coil speaker is bound to be inefficient.

If the weight of the radiating element in a speaker could be reduced, it would not only be more efficient but would operate far more faithfully under transient conditions, i.e. less energy would be stored in it and it would move almost exactly in conformity with the signal applied to it. This is the major advantage of the electrostatic loudspeaker. It operates quite differently from the moving-coil unit. Readers will know that it has become an established alternative, as a full-range system, to the more traditional moving-coil speaker, after extensive research in England and America over the last twenty-five years. The electrostatic loudspeaker has a light, tightly stretched plastic diaphragm, which is placed mid-way between two, fixed, flat plates, as shown in Fig. 86. The two outer plates are perforated so as to allow sound radiation to take place through them. Internal to the speaker is a source of high D.C. voltage, which is applied between the central diaphragm and the two plates, with the diaphragm positive. Then, through a special transformer, also fitted internally to the speaker, the signal voltage from the audio amplifier is applied between the fixed plates and the diaphragm.

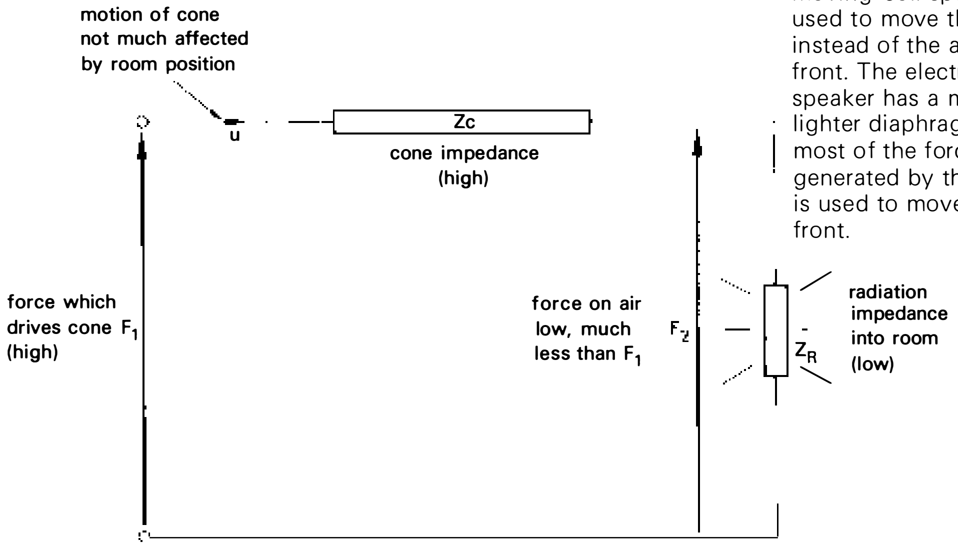
Even in the absence of any signal from the amplifier, there is a mechanical force between the plates and diaphragm, resulting from the high D.C. voltage which exists between them. As soon as a signal appears, this force varies in such a way as to attract the diaphragm towards one of the plates, depending on the polarity of the signal. The diaphragm will always return to its rest position when the signal disappears, because of its high stiffness. A very high resistance is connected in series with the diaphragm and its D.C. polarising voltage, so as to ensure that the electrical charge on the system does not change. If this condition of 'constant charge' operation is maintained, then the movement of the diaphragm will be linear, i.e. it will not produce any harmonic distortion. Also it will ensure that, as the diaphragm moves closer to one of the fixed plates, under conditions of high signal level, the voltage between the two will fall and reduce the probability of electrical damage which would follow them coming into contact.

The diaphragm of the electrostatic loudspeaker can be made any convenient size; the larger it is made, the more power the speaker will radiate. It has very low mass, and so the force which is generated to move it is the same force which acts on the air in front of it. It is driven electrostatically, and uniformly over its entire area, which implies that the system moves as a single body, with predictable accuracy. Figure 87 contrasts these properties with the operation of the moving-coil speaker. The heavy cone of a moving-coil speaker unit is driven at a single point at the centre, and the force available to disturb the air is far smaller than the driving force. Anything that can be done to increase the 'useful' power radiating force, i.e. in enclosure design, will help the speaker to become more efficient.

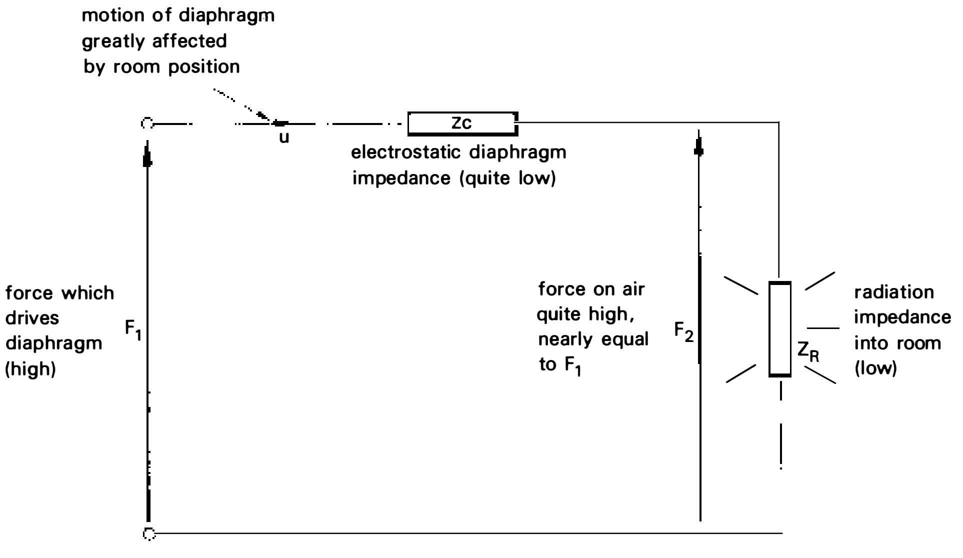
Readers may know that when a moving-coil speaker enclosure is moved closer to the corners of a room, its bass performance seems to improve. More will be said of room acoustics in Chapter 5, but the reason is that the corner of a room presents the speaker with a higher air resistance into which it couples more efficiently. This is not the case for an electrostatic speaker, however. The light diaphragm of this speaker is more profoundly influenced by any external load, and will actually become *less* effective as a sound radiator by placement in the corner of a room. Hence the speaker is placed a few feet from the room boundaries, where it 'sees' an almost purely resistive air load over most of its frequency range.

These are ideal and very attractive principles to the loudspeaker designer, but there are many practical considerations also, which must be reflected in a commercial system of the electrostatic type. Low-frequency limitations, reliability of the electrical systems and spatial directivity are just a few of them. Even so,

Fig. 87
A comparison, using analogous circuits, of the electrostatic loudspeaker with the moving-coil loudspeaker. Most of the force generated by the moving-coil speaker is used to move the cone, instead of the air in front. The electrostatic speaker has a much lighter diaphragm so most of the force generated by the system is used to move the air in front.



Moving-coil Loudspeaker



Electrostatic Loudspeaker

there are two or three commercial, full-range electrostatic loudspeakers available today.

From America has come the KLH Model 9, a full-range electrostatic system arranged as a tall panel, with separate smaller treble radiating elements to improve directionality. Another American system is the Dayton Wright array, arranged as a multicellular radiator of wide frequency range. In this system directionality is good, but my own experience was that smooth wavefront propagation was difficult to ensure owing to interference between neighbouring elements, which detracted from stereo image quality. Perhaps the classic full range electrostatic loudspeaker is the British one developing from the research of P. J. Walker and D. T. N. Williamson. Peter Walker is founder of the famous British audio firm Acoustical Manufacturing Company, with the name 'Quad' known to many audiophiles and musical persons alike. The Quad electrostatic speaker first became available in 1957 (Fig. 88a). The system covers usefully the range from 40 Hz to 20 kHz, and includes the necessary D.C. voltage source and network necessary to match the intrinsic impedance of the special diaphragm to the normal 8-ohm amplifier source. The diaphragm is divided into three sections, a

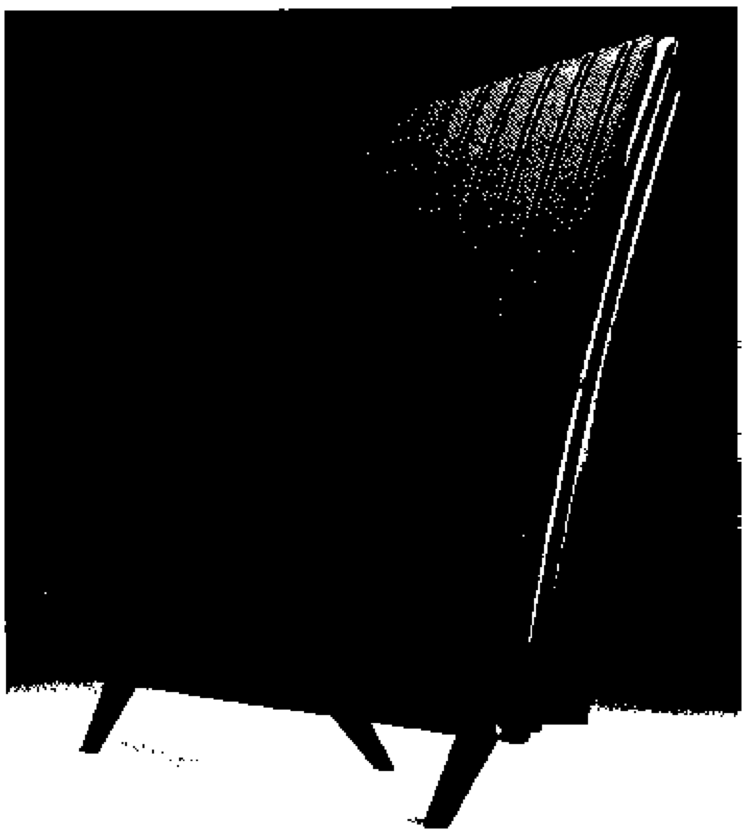


Fig. 88a
The Quad full range electrostatic loudspeaker has been a successful high-quality loudspeaker system for more than 20 years.

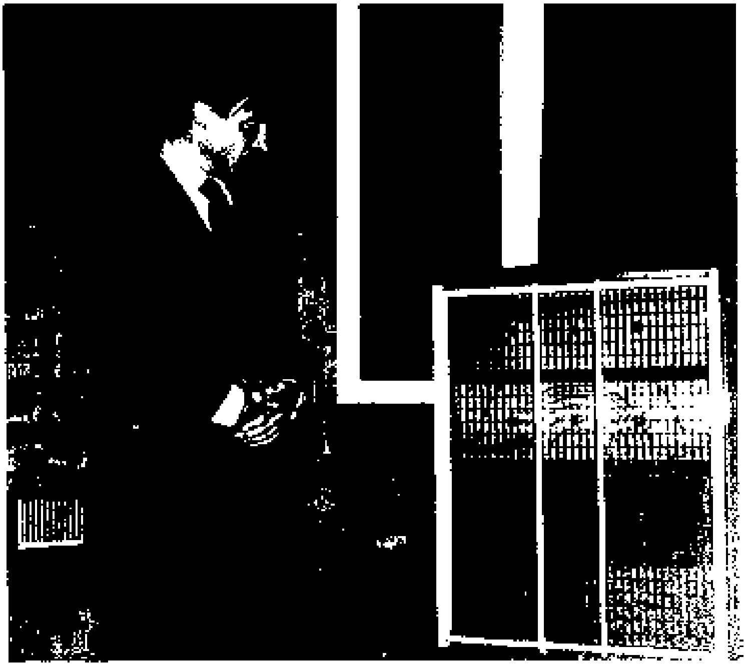


Fig. 88b
The prototype of possible future design of full-range electrostatic speaker from Quad. Notice the annular rings on the diaphragm, which are driven through filters over different parts of the audible range to ensure a smooth, wide angle of radiation at all frequencies.

central treble section about 15 centimetres wide and two adjacent 36-centimetre sections for the lower range, the overall height being about 61 centimetres and having a screen-like appearance.

Its performance has commended it to many listeners, especially its fine transient resolution and openness in the middle register. Perhaps a little directional and limited in dynamic range, it still represents a truly remarkable technical achievement. This is especially so when one considers that in quite recent years, new measurement techniques allowing the time or phase response of this loudspeaker to be assessed, show its twenty-year-old design to have the accuracy of performance which Peter Walker predicted and which few modern speakers could achieve.

A recent design by Quad, though still in the experimental stage, is illustrated in Fig. 88b. Its essential principles are as described above, but it naturally reflects the designer's more recent experience. Its single-diaphragm radiator, which is taller than it is wide, is divided up into elements which are driven by a series of electrical attenuation and delay networks, internal of course to the speaker, which ensure that the radiator behaves as a simple, point source at all frequencies. Again we have a system which is well-behaved mechanically, most of whose motional force acts on the air load of the listening room. Its accurate motion can actually be predicted by the current which flows into it from the amplifier.

The development of this special loudspeaker will continue to hold the interest of the audio world. The moving-coil loudspeaker has the advantage of great simplicity, which would explain why so many of these systems have been created over the years. The design and electrical circuitry of the electrostatic speaker, on the other hand is, relatively complicated. I would think that neither system has a clear overall lead over the other. The moving-coil speaker, at its best, has the power handling capacity and dynamic range required for the most faithful recreation of sound in the home, whilst the electrostatic leads in transient precision and accuracy of operating principle. Since it would seem that the advantages of one cannot be featured in the other, the two will continue, side by side.

THE PERFORMANCE OF LOUDSPEAKERS

Modern trends

Fashions come and go even among scientists and loudspeaker designs. In the early days, smooth response was difficult to ensure because of the lack of uniformity in production of the cone material, which was almost always paper. Progress was gradual, and still is, with a few important advances, notably from those who attempted alternatives to the moving-coil diaphragm. Stanley Kelly's ribbon speaker and Peter Walker's full-range electrostatic design are the established examples. Over the years, new plastic materials for bass driver cones and suspensions and dome structures for high-range units illustrate how attempts have been made to improve the accuracy with which the loudspeaker can recreate the complex waveforms of music and speech.

During the early 1970s, it seemed fashionable to concentrate almost solely upon fidelity of reproduction at the expense, for example, of loudspeaker efficiency. A number of speaker designs have become available in recent years which, while claiming a natural, uncoloured sound, fall short of being able to reproduce the arresting dynamics and transient attack of much music material. It is not sufficient for a loudspeaker to have a rather smooth, soft sound if it does not have the peak power handling capacity that the music requires.

Yet another development of the 1970s was the linear-phase loudspeaker. Readers will know of a number of commercial designs, from such companies as Bang & Olufsen, Bowers & Wilkins, KEF Electronics, the American Company Dahlquist and Tecnics. These speakers aim to reduce the relative path length differences between the drive units of the speaker and the listener's ears, so as to preserve as far as possible the exact time relationship of all the frequency components which make up a sound. In general, linear phase systems have not been well received, not because phase-compensation cannot bring advantages for the listener, but because it is only one of a number of critical factors, all of which must be optimised in design. Aspects such as frequency balance between units and cabinet diffraction effects are every bit as important as correction of linear phase errors. A phase-compensated loudspeaker which succeeds in refining all the most important features of design is the KEF model 105. It is a very costly loudspeaker, but every attention is paid to positioning the drive units and contouring their separate enclosures so that stereo image stability and 'depth' effects are maintained.

Another feature of the KEF model 105, which I think represents what might be called the design trend of the present period, is the attention paid to ensure adequate power handling capability. To this end the speaker employs a filter in its bass section designed to improve the matching between the bass driver and the amplifier. This increases efficiency and allows a sealed enclosure to be used, which contains cone displacements and so gives better power handling capacity. In fact enclosures of the sealed type are more often seen nowadays, particularly in larger loudspeakers. The current trend, then, would seem to be to take advantage of the developments in cone materials, designed to make a loudspeaker a smoother, more accurate reproducer, but at the same time to combine this with adequate dynamic range and sensitivity. Perhaps we shall see other points emphasised in design in future years, although it is doubtful that progress will be other than it has been over many years past, continuous but gradual.

Even with the overall improvement in the performance quality of loudspeakers, their role is still perhaps the most interesting in sound reproduction. They are still the subject of much discussion and controversy. There are almost as many opinions about how a loudspeaker should sound as there are different models on the market today. There is as much argument in favour of a scientific approach to evaluating the qualities of a loudspeaker, which has become highly developed, as there is in favour of simply listening to it, subjectively, a practice which is even less reliable. I suppose these arguments arise because the loudspeakers we hear all sound so different. Although the musical balance of different speakers is closer than it was some years ago, the differences are still quite marked, even to the untrained ear.

Differences between speakers

Why should so many similarly designed speakers sound so clearly different? The reason mostly lies in the subtle upsets which a speaker makes to the precise *timing* of the different frequency components in a music signal. Although different designers will have different intentions, there are some difficulties which all of them have to accept. When a sound signal is first applied to a loudspeaker, it takes a little time for the energy to build up in the voice-coil to cause a movement of the cone. Secondly, the cone always has a small mass, and so it takes a further time before this mass can be made to move. When the cone finally moves, it does so by 'flexing' both along its radius and along its

Fig. 89a
Showing how a speaker cone flexes due to sound waves travelling around its circumference at: a_1 a low frequency, a_2 a high frequency.

a_1 a_2

circumference (Fig. 89). Sound waves are then travelling in the cone material itself and reach the listener after those radiated directly into the air. Finally sound energy which travels in the cone material can become stored in particular areas of the cone because of its non-uniform composition, and this means further time delays at particular resonant frequencies.

This is one of the most important causes of colouration in the performance of a loudspeaker since, at these frequencies, the response of the speaker can last longer than the duration of a brief music or speech transient sound. Yet another source of time delays is the storage of signal energy in the crossover circuit of the speaker. The overall result is that, particularly in the critical middle and treble register of sound, a speaker introduces small time delays in its response, and causes sounds to persist for far longer than they did originally in the music, with the result that the pressure response of the speaker shows many complex variations. With such complex patterns, there is no limit to the number of different sounds speakers may produce.

The effects of these complex delays on music and speech can sometimes be subtle, and sometimes very obvious. At higher frequencies they can cause a speaker to sound excessively bright and harsh. They can also cause an instability of image or confuse precise sound source location on stereo, because certain frequency components are delayed or persist and so dilute information contained in the desired signal. In the important middle range of music and speech, the time delays resulting from resonances in the speaker give a dominant 'hole in the wall' character to perceived sounds. The effect is well known, and is like that produced when a person is speaking with his hands cupped round his mouth.

An example is given in Fig. 90 showing a violin wave coming into and out of a poor loudspeaker. The upper waveform is the electrical signal of the open D string (fundamental frequency 294 Hz) of a violin, applied to an expensive commercial loudspeaker, and the lower trace is its acoustic response, taken in the anechoic chamber. Notice in the lower trace, how certain higher partial tones are strongly emphasised and shifted in time with respect to those in the upper wave. This speaker produces a very bright sound, which detracts from the 'feathery' quality normally expected from a violin. Hence this loudspeaker, like many others,

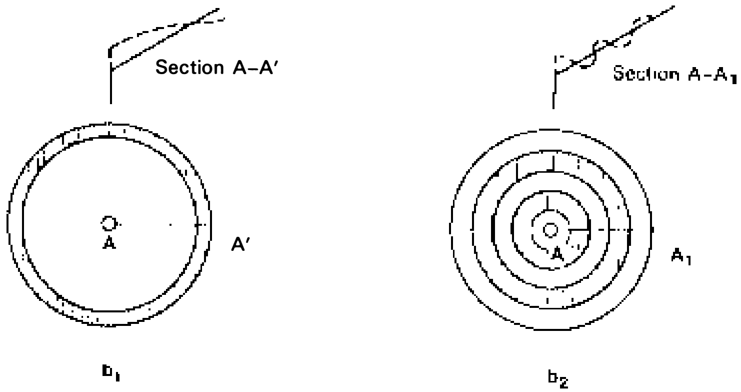


Fig. 89b
Showing cone flexure, due to sound waves travelling along the radius of a speaker cone at b1 a low frequency and b2 a high frequency.

detracts from the true tonal qualities of music and speech because it upsets the precise timing, or phase relations, of their constituent frequencies.

A loudspeaker would be ideal if it introduced no time delays in its acoustic response, but it would be perfectly acceptable if it introduced a delay which was the *same* for all audio frequencies. The trouble is that the speaker introduces differential delays which are different throughout the frequency range. One way of obtaining information about this problem would be to excite the speaker at each frequency in turn and then analyse, after removing the signal, its response with time, instant by instant. A simpler way would be to apply all frequencies at once, and then observe the complete response of the speaker with time after removing the test signal. Fortunately, a test signal of this kind is also one of very short duration, and it is called an impulse.

The impulse test signal, together with its frequency spectrum, is shown in Fig. 91. The impulse signal excites the speaker uniformly over its entire frequency range. The lowest frequency present, i.e. the fundamental, can be seen to be determined by the period of the pulse, i.e. the time interval between pulses, whilst the highest frequency present depends upon the pulse duration. Notice that the harmonics are very closely spaced–there will be

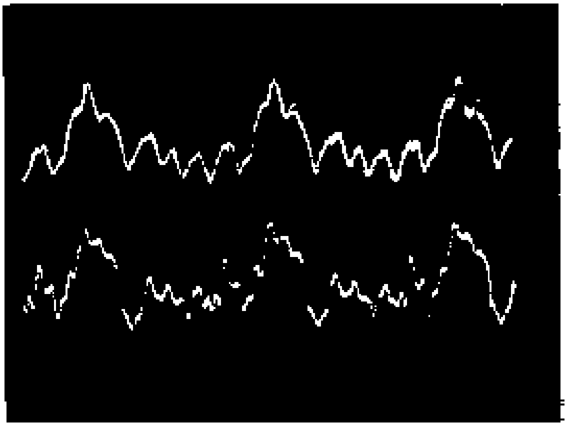


Fig. 90
The waveform of the open D string (294 Hz) of a violin fed to an expensive commercial loudspeaker (top) and the speaker's response (bottom). Notice how the higher frequency components are strongly emphasised and shifted in time due to poor transient response of the speaker.

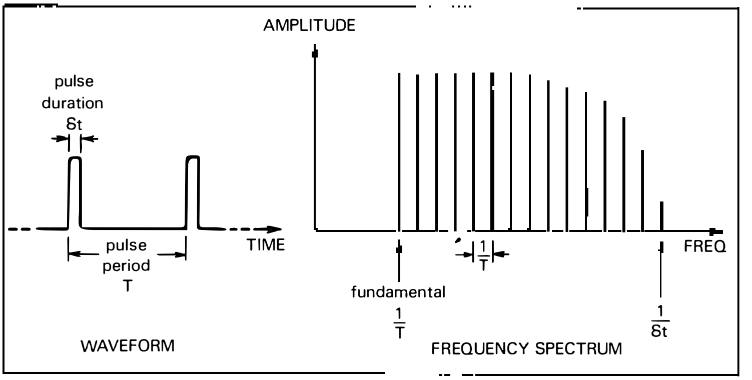


Fig. 91
Impulse test signal and its frequency spectrum.

more than 100 in a typical test pulse, so if the interval between pulses is changed, the response of the speaker will not change significantly. The height of the pulse, a few volts, must be so arranged that the speaker is never overloaded.

Use of the impulse test allows us to build up a series of time-delayed responses for the speaker, into a three-dimensional model, and this has been done by KEF Electronics, as well as other companies, in their research into the behaviour of loud-speakers. The test need not be done in an anechoic chamber (Fig. 61), as long as all the important time-delayed information from the speaker reaches the microphone directly, before the first reflection arrives from the walls of the room. If the test is repeated periodically, however, an anechoic room is required. This is the procedure generally used by loudspeaker reviewers. Some results are shown in Fig. 92. In each case the pulse duration was a very brief 80 microseconds (i.e. 80 millionths of a second), but the period of the pulse can be varied, depending upon the time it takes for the speaker's response to subside. The shorter the period, the better the speaker.

In Fig. 92a, we see the impulse response of the speaker used to reproduce the violin tone mentioned above. Notice how significant and persistent the residual response of the speaker is, even after 12 milliseconds, which was the pulse period. These residual components occur mainly in the treble range, and last many times longer than the test pulse, so giving the speaker its harsh sounding quality.

Fig. 92a
The impulse response of the poor quality speaker which reproduced the violin waveform show in Fig. 91 above. Notice the persistent response at high frequencies.

Fig. 92b
The impulse response of a speaker which has a resonance at 500 Hz. The residue in response at this frequency can be clearly seen.

Fig. 92c
The impulse response of the Quad electrostatic loudspeaker. This result shows very good preservation of the impulse.

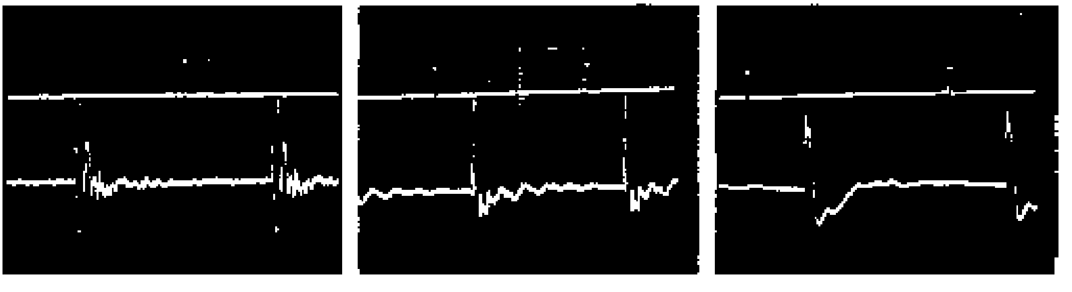


Figure 92b is the impulse response (period 12 milliseconds) of a speaker with a peak in its pressure response curve at around 500 Hz. The impulse test shows clearly that this is due to a resonance in the speaker, which persists for more than 10 milliseconds, i.e. more than 100 times the duration of the test pulse. When this speaker reproduces music or speech, the resonant phenomenon will dominate every sound, for it will last so long after it is excited by successive notes or words.

Figure 92c is the impulse response of the Quad electrostatic loudspeaker. The pulse period was again 12 milliseconds, so that the speaker was being excited down to about 80 Hz. Although the significant 'undershoot' in response, i.e. its fall well below the zero line, shows the speaker to have a limited low-frequency capability, the otherwise very accurate preservation of the pulse shape shows that this system processes many components in its range with near perfect time accuracy. In general, the impulse test is not the best pointer of the bass performance of a speaker.

The many different and highly complex responses obtained from loudspeakers when impulse tested gives us important clues as to why they sound different. Hence the test is a useful one, for it partly validates our experiences of speakers when they are auditioned on music in the listening room. But more of this in Chapter 5.

THE HI-FI SOUND SPEAKER

A final item, of particular interest and specially commissioned for readers of this book, is a high-quality loudspeaker design, the Hi-Fi Sound loudspeaker, offering very fine performance in an enclosure of modest size and cost. It features drive units of the very best quality, made by the famous British firm Rola Celestion, one of them specially developed for this design. The drive units will be supplied to another firm, Lever Audio Ltd., who, after consultation with the author have agreed to offer the design in various forms so as to meet every reader's requirements.

Firstly, it will be available in pairs, completely built and ready for use. Secondly, and at lower cost, it will be available in kit form, with every component provided for very easy assembly. Lastly, for those who wish to make the cabinet for themselves, details are given in this chapter and only the crossover and control board, together with the drive units will be required, available as a special pack from Lever Audio Ltd. The fullest details will be provided by them if you write to: Lever Audio Ltd., Hi-Fi Music Design, 29 Heathfield, Stacey Bushes, Milton Keynes, Buckinghamshire. Tel. Milton Keynes (0908) 316506.

Features of the loudspeaker

The Hi-Fi Sound loudspeaker is a system of modest size and cost, but designed by the author to be of high quality and using

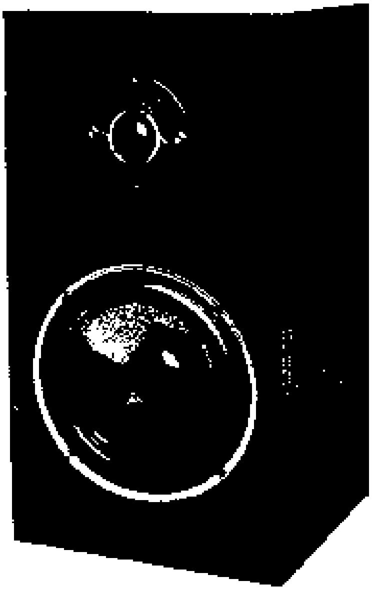
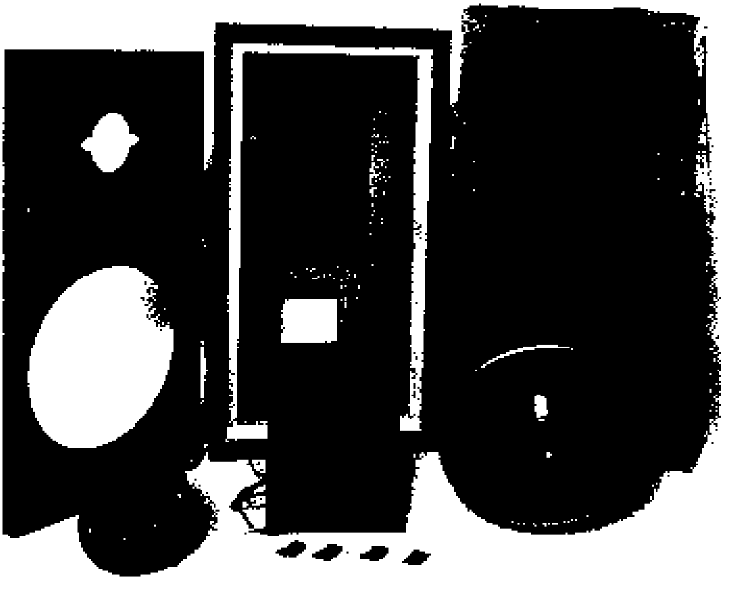


Fig. 93a
The Hi-Fi Sound loudspeaker design with grille removed.

very fine drive units. It measures 48 × 25·5 × 23 centimetres (19 × 10 × 9 inches) and can be used for cabinet top or stand mounting. It is best stood about 30 to 40 centimetres (12 to 15 inches) from the floor, but this is not too critical and the listener will wish to experiment in his own room. The enclosure is made from heavy chipboard panels, with bitumen lining on all three inner walls as well as absorbent filling to ensure smoothness and freedom from resonances in the lower range. The bass and mid-range drive unit is carefully matched to the enclosure and gives well extended and controlled bass performance. The tweeter is one of the best available, a dome unit with specially developed 'motor' system to give higher power handling capacity. Listeners will soon notice the treble accuracy and fine presence of this design. Even the fitting of the tweeter has been specially simplified, and this also serves to minimise diffraction errors in the extreme treble range. The finished design (with grille removed) is shown in Fig. 93a and in its component form in Fig. 93b.

The speaker has a treble energy control, fitted at the back, so the listener has some adjustment of treble balance to account for different rooms. The frequency response range of the speaker is from 70 Hz to 24 kHz, within 4 dB limits. It is of conventional 8 ohms impedance, and will present an easy load to any amplifier. Amplifiers with a continuous power rating of anywhere between 18 watts minimum to around 60 watts, can be safely used on normal music and speech programme. If your amplifier is of higher power rating, do not drive the speakers at maximum power level, but contain it to a more sensible level. In use you will find that the speaker has adequate sensitivity and power handling capability.

Fig. 93b
The components of the Hi-Fi Sound loudspeaker design.



Design and performance results

Earlier in Chapter 4 I mentioned the importance of the crossover network, which integrates the individual drive units together to form the complete system. This point is illustrated in the present design. Two Butterworth filters were used, of fairly simple design but arrived at after careful investigation of the behaviour of each drive unit. The final equalised network is shown in Fig. 94a, and its electrical response in Fig. 94b. Notice the very smooth trend which each curve takes, a very useful feature of the Butterworth filters as mentioned in Chapter 4. The two curves cross at 3 kHz, although acoustically they contribute equally at a frequency somewhat below this point, which is the crossover frequency of the system. The response of the speaker is shown in Fig. 95. The point by point response at each frequency in the audible range is shown in Fig. 95a, for both high and low settings of the treble energy control. It can be seen that there is a useful

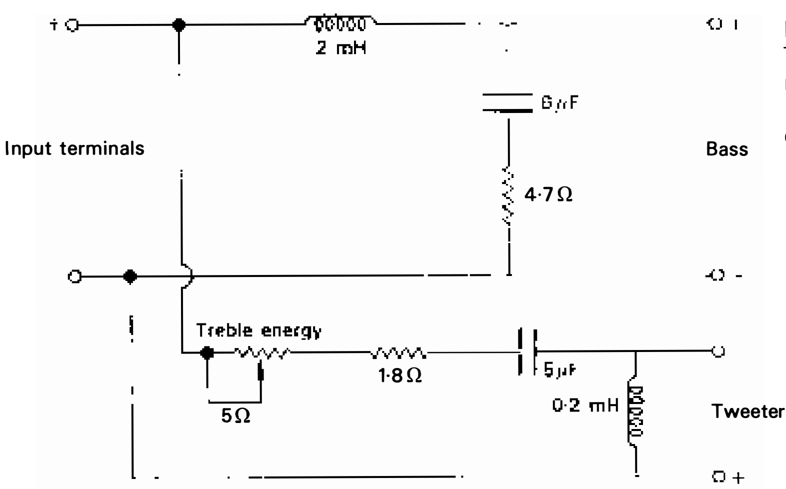


Fig. 94a
The final crossover network used for the Hi-Fi Sound loudspeaker design.

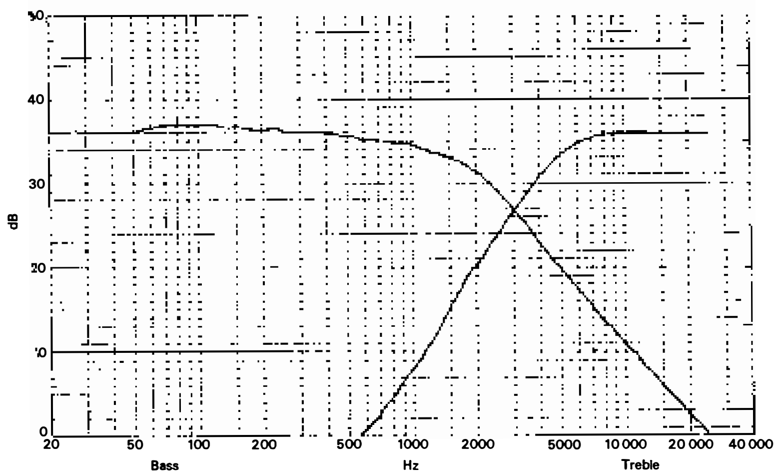


Fig. 94b
The smooth drive to each speaker unit provided by the network of Fig. 94a.

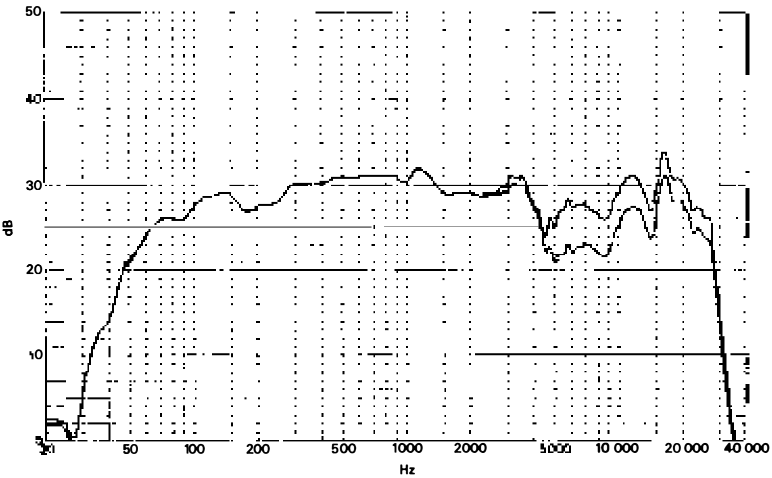


Fig. 95a
The response at each frequency of the Hi-Fi Sound loudspeaker.

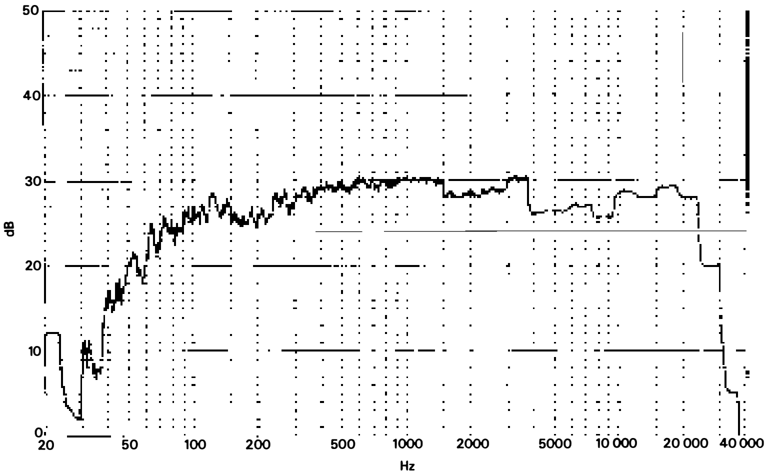


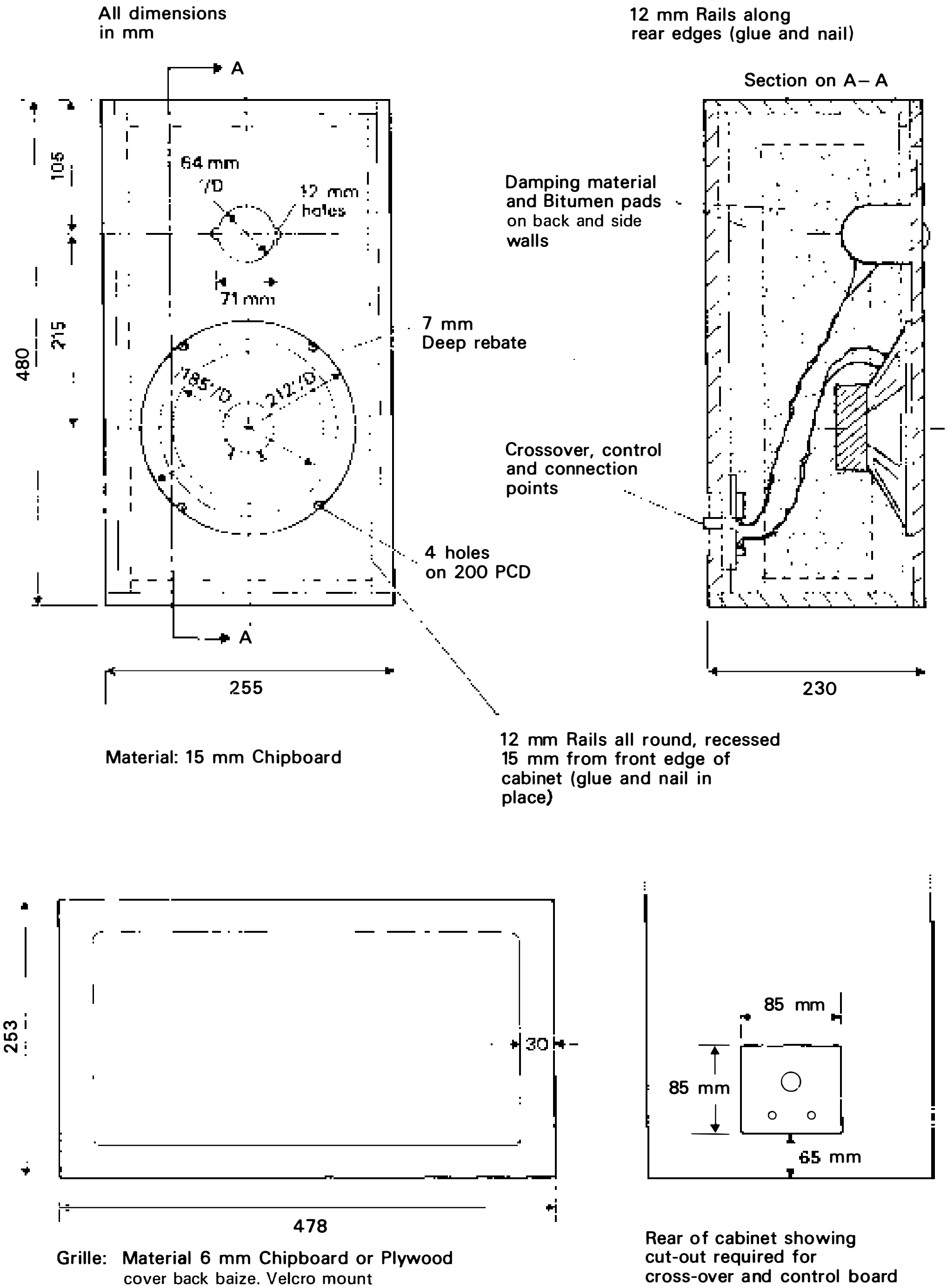
Fig. 95b
The response of the speaker when averaged in frequency bands.

range of adjustment provided by this control throughout the treble range above about 5 kHz. The response curve shows a level trend in the range between 200 Hz and 2 kHz, but the response level falls in the 2 kHz to 6 kHz region, which is a design feature arranged to give the best voice balance. The curve may not seem smooth in the extreme treble range, between 10 kHz and 20 kHz, but this results from the acoustic design of the tweeter and does not detract from good performance. This is confirmed in Fig. 95b, which is the response of the loudspeaker averaged in frequency bands. This result gives a better indication of the performance on music, where many frequency components will be radiated simultaneously. Experiment and extended listening tests have shown the final design to perform extremely well on a wide range of programme material, maintaining good dynamics, balance and treble smoothness. The spherical or polar response of the speaker is also smooth, giving precise stereo imaging, with good size and 'depth' to accompanied voice.

Notes for the home constructer

For those who intend making the cabinets for themselves, the details are given in Fig. 96. When you have made and assembled the enclosure, fit the bitumen sound deadening panels to both

Fig. 96
Showing the cabinet construction and grille of the Hi-Fi Sound loudspeaker.



internal side walls and cabinet back wall. Suitable panels are made by Bostik, and are easily cut to the appropriate size for the cabinet. Any suitable absorbent filling can be used, for example polyester fibre, glass fibre (which should be handled carefully) or even low density foam lining. The enclosure should be lightly filled with this material; enough should be used for the filling to remain in place when the cabinet is stood upright. Be careful not to allow the material to find its way behind the bass driver frame. The correct fixing screws must be used. For the bass driver four will be required and these are no. 8, $\frac{3}{4}$-inch pozi/supa raised counter-sunk type AB self tap. The colour is Dry Chemi black. For the tweeter, two screws are required and these are exactly as for the bass driver except they are flange head type instead of raised counter sunk. Use a 'pilot' 1.6 millimetre (1/16 inch) drill. Fit the drive units carefully so as to ensure an airtight enclosure. The drive units are of the highest quality and their details are as follows: The bass and mid-range driver is a 21-centimetre (8-inch) unit with plasticised fibre cone material and a neoprene roll front suspension. The voice-coil and magnetic pole diameter is 2·5 centimetres (1 inch), and a powerful barium ferrite magnet system is used, weighing 1·1 kilograms (2·3 pounds) and producing a flux density of 1·1 Tesla (11,000 Gauss).

The treble unit is a specially developed version of the familiar Rola Celestion type HF 2000. It has been designated the HF 2000HP, and features a polyethylene terephthalate diaphragm system driven by a 1·9 centimetres ($\frac{3}{4}$-inch) polyamide impregnated voice-coil with a high temperature adhesive and assembly technique specially developed for this design to give even better high frequency power handling capacity. The manufacturers have also simplified the fixing of the unit for the reader's convenience. The HF 2000HP employs an Alcomex magnet and motor system weighing 0·43 kilograms (1 pound) and producing a flux density of 1 Tesla (10,000 Gauss). When connecting the crossover leads to the drive units, note that the two are connected in opposing phase.

5
The Listening Room

SOUND WAVES IN ROOMS

I suppose that the subject of room acoustics appeals to many of us because it is such an elegant science. Both its mysteries and its cultural associations hold our interest; 'good acoustics' add the final touch to good music making. Also, of course, we can choose our music from the great recorded repertoire, so that good acoustics in the home listening room are important to a wider enjoyment of music. Indeed, the final link in the sound reproducing chain is not the loudspeaker, but the sound field between it and the listener. For the way in which sounds undergo change in a room and alter what we hear, is affected by many things; the size or volume of the room; its shape and curvature, i.e. whether the walls and other surfaces are plane or curved; the materials on these surfaces, and the other contents of the room including the audience.

When a sound wave in the air volume of a room reaches the surface of a wall it reacts in three ways. Some of the energy in the wave is used to set up compressional waves in the wall, so that the wall vibrates and transmits the sound into an adjacent room; some of the energy is absorbed by the wall (i.e. dissipated as heat energy in the wall material), but most is reflected back into the air and redirected across the room. Sound energy which continues to be reflected from one boundary of the room to another is called reverberation.

In 1895, at Harvard University, a young Professor of Physics named Wallace Clement Sabine was asked to do something about the severe acoustical faults in the Art Museum building. It seemed the room was very 'live', which means that it added too much reflected energy to direct sounds made in the room, and this confused their intelligibility. Sabine did not follow the previous practice of 'deadening' the room by hanging drapes and laying carpets, but chose a quantitative, scientific approach to the problem. The result of his work was a formula, known as *reverberation time*, which has been used to test rooms designed ever since.

Wallace Sabine, a pioneer in the field of acoustics.

Sabine looked very closely at the energy delivered to the room by the sound source. He saw the distribution of sound energy in the room as uniform and continuous, just like the distribution of the air itself. The word he gave to it was a 'diffuse' sound field i.e. one in which the rate of transfer of sound energy is the same in all directions, at each and every point in the room. This assumption implies that when a steady sound, e.g. from a speaker, is switched on the sound energy builds up smoothly, at any point in the room, as the number of reflections begins to increase. Although the build-up is smooth, its rate of rise changes with time. Initially, this rate is high because absorption at the walls has not become fully established. This absorption will increase, as more and more reflected waves encounter the room boundaries, and then the rate of increase of sound energy carried by the air must fall, because the steady sound source delivers a constant energy to the room. Finally, a situation is reached where the rate of supply of sound energy to the room equals that lost to the boundaries, and then the sound level becomes constant, i.e. 'steady-state' conditions are reached.

Exactly the reverse will apply when the loudspeaker is turned off. Initially, the rate of loss to the walls is high, so the sound level in the air falls quite rapidly, but as the number of reflections gets fewer, this rate of fall reduces. Quite obviously, if the walls of the room are given a different treatment, so that they absorb more energy, this sequence of events will occur in a shorter time. Another obvious factor is the size of the room, for this will affect the distances over which reflected waves travel. The surface area of the room is yet another influencing factor, because changing this will expose different absorbing areas to sound energy. Since the residual or reverberant sound energy has such a profound effect upon the intelligibility of speech or the fullness of music reproduced in the room, Sabine gave a quantitative meaning to this factor. The time taken for the reverberant energy to decay by 60 dB, i.e. to one millionth of its steady-state value, was called the reverberation time, T. The formula for this is given in Table 15 below. Table 14 lists the fraction of sound energy absorbed by different materials commonly used in buildings and for room furnishings. Table 15 lists a variety of halls, together with their uses and the desirable reverberation times. Since the absorption in a room will vary with frequency, so does the reverberation time. However, unless otherwise quoted the reverberation time is normally given at 500 Hz, since this is a frequency in the middle register where much of the energy in music and speech resides.

The reason that the definition for reverberation time concerns the time taken for a 60 dB fall in intensity is that such a fall will imply a decay to complete inaudibility for almost every sound encountered in normal listening situations. The reverberation time for a large concert hall is always greater than it is for a smaller room, as readers will have experienced from their visits to concert

Table 14 Sound-absorption coefficients of general building materials and furnishings

Materials	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Brick, unglazed	0·03	0·03	0·03	0·04	0·05	0·07
Brick, unglazed, painted	0·01	0·01	0·02	0·02	0·02	0·03
Carpet, heavy, on concrete	0·02	0·06	0·14	0·37	0·60	0·65
Same, on 1134 g hairfelt or foam rubber	0·03	0·24	0·57	0·69	0·71	0·73
Same, with impermeable latex backing on 1134 g hairfelt or foam rubber	0·08	0·27	0·39	0·34	0·48	0·63
Concrete block, coarse	0·36	0·44	0·31	0·29	0·39	0·25
Concrete block, painted	0·10	0·05	0·06	0·07	0·09	0·08
Fabrics						
Light velour, hung straight, in contact with wall	0·03	0·04	0·11	0·17	0·24	0·35
Medium velour, draped to half area	0·07	0·31	0·49	0·75	0·70	0·60
Heavy velour, draped to half area	0·14	0·35	0·55	0·72	0·70	0·65
Floors						
Concrete	0·01	0·01	0·015	0·02	0·03	0·02
Linoleum, asphalt, rubber, or cork tile on concrete	0·02	0·03	0·03	0·03	0·03	0·02
Wood	0·15	0·11	0·10	0·07	0·06	0·07
Wood parquet in asphalt on concrete	0·04	0·04	0·07	0·06	0·06	0·07
Glass						
Large panes of heavy plate glass	0·18	0·06	0·04	0·03	0·02	0·02
Ordinary window glass	0·35	0·25	0·18	0·12	0·07	0·04
Marble or glazed tile	0·01	0·01	0·01	0·01	0·02	0·02
Plaster, gypsum or lime, smooth finish on tile or brick	0·013	0·015	0·02	0·03	0·04	0·05
Plaster, gypsum or lime, rough finish on lath	0·02	0·03	0·04	0·05	0·04	0·03
Same, with smooth finish	0·02	0·02	0·03	0·04	0·04	0·03
Plywood paneling, 9·5 mm thick	0·28	0·22	0·17	0·09	0·10	0·11
Water surface, as in a swimming pool	0·008	0·008	0·013	0·015	0·020	0·025

halls. Its value, however, is not proportional to the room volume, and this is because a large hall also has very large absorbing surfaces which limit the duration of reverberation. The reverberation time is a most important index of the acoustical qualities of a room, and can be found quite accurately by the Sabine formula for almost any room. Its ideal value will depend upon the purpose to which the room is to be put.

In rooms for speech, some reverberation is desirable, because the reverberant sound energy adds to the direct sound and so helps to make a speaking voice more easily heard. Its value must not be too long, however, or else successive words will be confused together and intelligibility will suffer.

In very large concert halls, a longer reverberation time is, of course, inevitable. However, this will give the music the right fullness and blend as the right reverberant link is established between succeeding musical tones. Ideally, it should be possible to alter the reverberant qualities of a concert hall since different kinds of music require different acoustics. This has been done in modern halls, as we shall see later in this chapter.

Table 15 Halls and their uses, with desirable reverberation times

Type of hall	Typical use	Reverberation time *seconds*
Concert hall	Symphonies, romantic works, etc.	1·4 to 2·4
Concert hall	Opera, contemporary, etc.	1·5 to 1·8
Concert hall	Light music, operetta, using sound system	1·0 to 1·6
Churches	Organ, large choral	1·5 to 3·0
Cathedrals	Organ, large choral	above 3
Small theatres	Plays	1 to 1·5
Lecture theatres & conference rooms	Formal speech	0·8 to 1·2
Recording & broadcast speech studios	Clear speech	0·3 to 0·6
Domestic living room	General purpose	0·3 to 0·6

Note

$$T = \frac{\cdot 05V}{a_1S_1 + a_2S_2 + \ldots + a_nS_n}$$

V = room volume
a_n = the fraction of sound energy absorbed by S_n
S_n = the surface area absorbing a_n

In the small broadcast or music studio, a fairly short reverberation time is required, so that the broadcaster's voice can be conveyed clearly and neutrally to the home listener, or so that a music recording can be assessed with the minimum of intrusion of the studio acoustic. Exactly the same conditions should obtain in the home listening room as exist in a well-designed music studio. Again it is the aim to hear faithfully the recorded acoustic, not that of the listening room.

A survey of the reverberation times found in typical home listening rooms revealed the average curve shown in Fig. 97. It has been found that, with little variation, the reverberation time in most home listening rooms is about 0·4 seconds, falling slightly above 4kHz. This seems to be true for houses of all ages, for although rooms have become smaller in more modern homes, their furnishings are generally less absorbent. There is a greater tendency for wood and other acoustically reflecting material to be used in furniture, and décor generally.

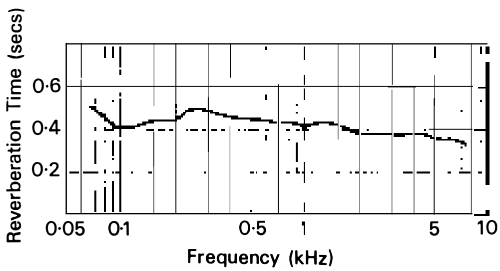


Fig. 97
The reverberation time at each frequency for a typical domestic listening room.

ROOM RESONANCES

Wallace Sabine took the view that a room 'fills up' with sound, just as a tank might fill up with water, and whilst his work was very successful, it overlooked some equally important properties of sound wave behaviour. Unfortunately, sound does not spread uniformly and continuously throughout a room as was first assumed, and this is because sound is by nature a wave motion. Its distribution in the air volume of a room is very much affected by the shape and curvature of the boundaries. Some surfaces may be convex, and will disperse the sound that comes in contact with them, i.e. redirect it over a wider area, whilst some surfaces may be concave, and will focus the sound into narrow beams, in exactly the same manner as light waves. This means that the distribution of sound energy in a room is never uniform as Sabine supposed, if only because of the irregular shape of the room. This uneven sound distribution will be further aggravated if the distribution of sound absorbents about the walls, floor and ceiling is uneven. But there is yet another reason why sound energy is unevenly spread in the room, particularly in the lower frequency range.

Just as in any mechanical system there are frequencies at which the system will most easily respond, i.e. resonant modes, so in the air volume of a room there are frequencies at which the air is most easily set into motion. These are known as the *normal modes* of air vibration, or *room modes*, and they profoundly change the acoustical response of the room at low frequencies. At each of these frequencies there will be points in the room where the sound pressure is high, called antinodes, and other points where the pressure is low, called nodes. Modes occur in a listening room when the distance between any two opposite surfaces, e.g. walls or floor and ceiling, equals one-half wavelength, or any multiple of one-half wavelength. A well-designed listening room will have its dimensions chosen so that the number of these modes is maximised and spread as uniformly as possible over the lower part of the audible frequency range. (See Fact Box on p. 153 for further details.)

The more, closely spaced, modes there are, the less easily they will be identified when listening to speech or music. To illustrate this point, take a look at Table 16 which shows the distribution of modes for three cubical rooms of different size, together with a room of less regular shape.

The important points to notice are that the mode frequencies get closer together at high frequencies, but at low frequencies they are relatively far apart. In the smallest room, the modes are furthest apart, and even the lowest of them fall clearly in the bass frequency range. In the smallest cubical room there are only four different modes between 55 Hz and 110 Hz, and these will be clearly audible on music reproduced in the room. The situation is a little better in the largest cubical room, though even here modes of different order occur at the same frequency, which again will strengthen their presence on the music. Now compare the modal distribution of the largest cubical room with the room of irregular shape, whose volume is similar to the cubical room. None of the modes coincide in this room, so there are many more of them

Table 16 The distribution of room modes

Mode order			Mode frequencies (Hz)			
n_1	n_2	n_3	3m cubical room	4·25m cubical room	6m cubical room	7·8 × 6·2 × 4·9m room
1	0	0	55	40	28	22
0	1	0	55	40	28	27
0	0	1	55	40	28	34
1	1	0	78	57	40	35
1	0	1	78	57	40	41
0	1	1	78	57	40	44
1	1	1	94	70	49	49
2	0	0	110	80	57	43
0	2	0	110	80	57	54
0	0	2	110	80	57	69
1	2	1	132	98	69	65
1	1	2	132	98	69	77
2	1	1	132	98	69	61
2	0	2	154	115	80	81
2	1	2	165	120	84	86
2	2	2	183	140	98	98

between, say, 28 Hz and 80 Hz. This room will impart some of its own character to reproduced music, but it will be far less distinct because of the higher mode density and smoother overlap between them.

Hence the low-frequency sound energy in small rooms tends to be concentrated at a few frequencies which are widely spaced and audible. The resulting low-frequency reverberation is known as 'bass boom' and is an undesirable feature of small untreated speech studios and, of course, home music rooms. In an established listening room the only solution is to deaden the boom effect by acoustic treatment, of which more will be said later. Those lucky enough to be planning a listening room should ensure that its dimensions are irregular (see Fact Box opposite) as this will minimise the chance of mode coincidence or degeneracy. Fortunately this problem is confined to low frequencies. Above about 300 Hz the number or density of modes becomes so high that it is safe to assume a smooth, diffuse sound field for almost any room. Figure

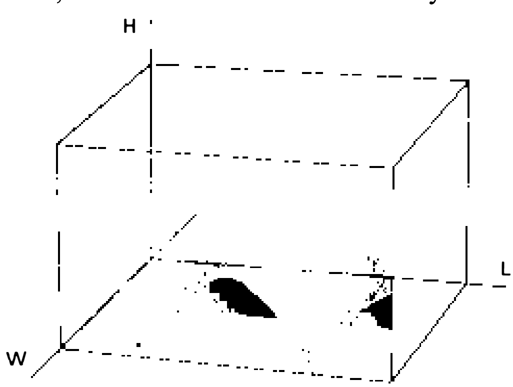


Fig. 98
The pressure variation throughout a listening room when a simple axial mode is being excited in the room. The height of the surface above the floor is an indication of the pressure amplitude, i.e. the audibility of the mode.

98 shows how pressure varies greatly with distance in a typical listening room when a simple mode is being excited. This will be the situation in most domestic rooms at frequencies below about 100 Hz.

Low-frequency sound production or reproduction in small rooms is presented with problems which are particularly serious in the low-frequency range. A uniform sound field cannot be assumed in smaller rooms because of the small number of discrete frequencies at which such rooms respond. This is not true of a large concert hall, however, where a diffuse sound field can obtain, right down to the lowest audible frequencies. Also, there are other important differences between sound production in small studios or home listening rooms as compared with large halls, and so we will consider them separately.

The formula to find the mode frequencies for a room of length L, width W, and height H, is

$$f_n = \frac{c}{2}\sqrt{\left(\frac{n_1}{L}\right)^2+\left(\frac{n_2}{W}\right)^2+\left(\frac{n_3}{H}\right)^2} \quad \text{Hz},$$

where c = the speed of sound, f_n = mode frequency and n = mode order; n_1, n_2, n_3 can each be any whole number, or zero. If n_2 and n_3 are each made zero, and n_1 is given successive numbers 1, 2, 3, 4 etc, we shall get what are called the *axial* modes of the room associated with its length. These are the simplest modes, and contribute mostly to the colouration of sound. If the length is the longest dimension then the lowest frequency mode is

$$f = \frac{c}{2} \times \frac{1}{L} \quad \text{Hz}.$$

(E.g. if L = 4 metres, then the lowest mode frequency will be $\frac{344}{2} \times \frac{1}{4} = 43$ Hz. The next mode would be at 86 Hz and the next 129 Hz. Simple modes are harmonically related.

Simple modes for other dimensions are found similarly. When the numbers n_1, n_2, n_3 all have values, the modes become much more complicated. It is soon found that the modes become numerous, and eventually a diffuse sound field can be assumed.

The total number of modes, dN, in a frequency range df, centred at a point in the audible range f, can be found approximately from the expression

$$dN = \frac{4\pi f^2 V\, df}{c^3},$$

where V = the volume of the room. For example the total number of individual mode frequencies within a 50 Hz range centred at 40 Hz for the 6 metre cubical room mentioned in the text, works out to be about 15, including coincident modes. Compare this result with that obtained for the Royal Albert Hall in London, whose volume is over 85 000 cubic feet. The answer we get is over 4000 separate modes. With so many in such a small frequency range, it would not be possible to hear any of them individually. In order to ensure that as many separate modes as possible occur in a room, we must arrange that its dimensions are in particular ratios. It has been found that dimensions which are in any of the following ratios (L:W:H) give best results:

1:0.62:0.39 or 1:0.69:0.43 or 1:0.78:0.62 or 1:0.82:0.72.

For example, the dimensions of the room of irregular shape referred to in the text are very nearly in the ratio 1 : 0.78 : 0.62.

A famous conductor once said, and not surprisingly, that the acoustics of a concert hall play a part in all stages of the musical process, in composition, in performance and in listening. Thus Bach reflected the difference between live and dry acoustics in his classic writings, and many musicians find their performances influenced by the acoustics of the space in which they are playing. The reverberation of a hall is claimed to assist even the playing of a violin. The blending effect on the small tones gives them strength and body.

The reverberation time of a concert hall can never be too long for an organist. Ample reverberation is part of organ music itself, and some of the pauses in Bach's organ works are designed specifically to exploit and expose the listening environment. A reverberation time of at least 2 seconds is required for such music, if it is less, the music falls apart into disconnected fragments. Pianists seem to prefer a dryer acoustic; where slow changes of chord are required, or where blending of notes played at fast tempo is desired, the sustaining pedal can be used. The piano seems loud and reverberant enough by itself, and therefore needs less support from the surroundings. (When sound is reinforced with powerful amplifiers and loudspeakers on the stage or auditorium, the artists employ 'foldback' loudspeakers, which direct sounds back to them with a balance similar to that radiated into the main body of the audience. Hence they adjust the relative contribution of different electronic instruments according to how they wish the audience to hear them. This, of course, has less to do with the acoustics of the hall than with the sounds being reproduced over the audience sound system. However, the performers will no doubt take account of any quality of the hall acoustic which affects the overall balance.)

The whole orchestra will be affected by the conductor's awareness of the hall acoustic. Rehearsal in an empty hall does not give him the same impression as its sound with a full audience. This leads us to an important fact about the design of a hall. It is that, in a well-designed building, the audience will provide most of the absorption of sound. Obviously, a great many listeners have a clear view of, and are directly exposed to, the orchestra, in order to enjoy the music making, and a large part of the sound energy which flows into the hall is absorbed by them. A normally dressed person has an average absorption coefficient of about 0·47, and an effective surface area exposed to sound of about 1 square metre. Thus an audience of 2000 persons provides no less than 940 square metre absorption units, i.e. equates to 940 square metres of perfect absorber. If the hall itself were treated to provide several times this amount of absorption–to prevent the reverberation time from changing significantly with variations in audience size–then the acoustic would be far too dry for the music to benefit. The alternative is to arrange that the seats offer the same absorption as the audience, so that if a member of the audience leaves, he is replaced by the exposed area of the seat whose acoustic absorption is similar. This approach means that the total absorption is fixed and is in rough proportion to the size (i.e. seating capacity) of a hall.

There are many properties of a concert hall which relate to how the listener perceives the sounds it directs to him. These are judged in terms of musical qualities such as fullness of tone, definition or clarity, intimacy, tone colour and dynamic range. Thus a well-designed hall must preserve the balance, both in frequency and amplitude, of the many different sounds of the orchestra. It must give continuity and fullness to the musical line through having an appropriate degree of reverberation; it must project pianissimo passages and sound forth on the loudest dynamics so as to preserve both the delicacy and the impact of the music. The acoustic design must be such as to give the listener an impression of the size of the hall in relation to the music being played. This quality is very closely related to what is called the *initial time delay gap*, i.e. the time which elapses between the listener hearing the direct sound from the source and its first reflection. Very important in establishing this is the arrangement of the reflecting surfaces above and to the sides of the orchestra.

The proportions of a concert hall are dictated by some very basic considerations; for example the maximum distance from which people can see the stage directly in front of them and the width of the seats. Many of the older halls were designed for quite a small number of seats, and were made relatively long and narrow. Fortunately, this narrowness leads to reflected sound from the side walls reaching the audience very soon after the direct sound, which is an advantage. The first reflections arrive within a few tens of milliseconds of the direct sound, and the initial time delay gap is of the right order to give the listener some sense of space, size and closeness of the sound. More recently, it has been found that *lateral* reflections, i.e. those which are directed to the listener horizontally, from the sides of the hall, contribute greatly to his impression of space and size, even if their time of arrival varies from between 10 and 80 milliseconds after the direct sound. This important quantity has, of course, little or nothing to do with the reverberation time, but is more an indicator of the hall's transient behaviour.

The geometrical construction of the ceiling in a concert hall is another important consideration. By arranging the roof in sections, each acting as a reflector inclined differently according to the distance from the stage, it is possible to direct early reflected sound to all parts of the audience. Hence sounds of every frequency are properly projected, and this ensures that the full tonal quality and spectral balance of the music is maintained.

The aim in a large concert hall must always be to ensure that direct or early reflected sounds reach the audience with minimum obstruction and delay. Also that part of the sound energy which is reflected repeatedly from every boundary of the hall, i.e. the reverberant sound energy, must be random or diffuse, and must not form too large or too small a part of the total energy from the source. Problems can occur when these essential conditions are not maintained. For example, in very large halls reverberant sound can sometimes reach the listener so long after the direct sound that the two become disengaged. This creates an echo, which can upset the music badly. An echo is heard if the time difference between direct and reverberant waves exceeds about 50 milliseconds. There is a famous example of this in London's best known concert hall the Albert Hall, described below. Another kind of echo, called a flutter echo, can occur if multiple reflections take place between two, parallel walls in a hall. Many readers will have had the experience of hearing their footsteps while walking along a narrow, reflective corridor. The sound 'flutters' or rings, and can be heard as a series of pulses.

The problem can be serious in a concert hall, because the offending walls are much further away, and the multiple echo or flutter echo will confuse the music. Echo effects of all kinds are best attacked by trying to diffuse the incident sound, for example by projecting the boxes in which listeners are seated, outward from the walls. This is quite common practice in modern acoustic design.

The above points are best illustrated by reference to a few examples familiar to readers. There is an old myth in acoustics that says that the sound qualities of a hall improve with its age. This has no doubt been reinforced by the knowledge that many of the halls that are best loved were built in the last century. It must be said, however, that unless any of the constructural or sound absorbent fittings are changed, the acoustics of a hall remain the same throughout its life. There are good and bad acoustical designs in every age, and this is well illustrated by the case of The Royal Albert Hall in London. It was opened in 1871 by Queen Victoria, in one of her rare public appearances. The great organ was built by Henry Willis, and rebuilt by Harrison & Harrison in stages, the main part in 1924 and the remainder in 1933. It is one of the largest halls in the world, and certainly the largest in Europe. Its air volume is over 85 thousand cubic metres, with a seating capacity of 5 080 persons. After the destruction by bombing of the Queen's Hall in 1941, the Albert Hall became London's principal concert hall. Its uses are many; pageants, presentation ceremonies, organ recitals, symphony concerts, even athletic events. The Royal Albert Hall is almost elliptical in shape, quite different from the rectangular form of most other halls (Fig. 99). The initial time delay gap for listeners near the centre of the main floor is about 65 milliseconds.

A hall of tremendous character, and now more than a century in existence, it nonetheless has some quite significant acoustical

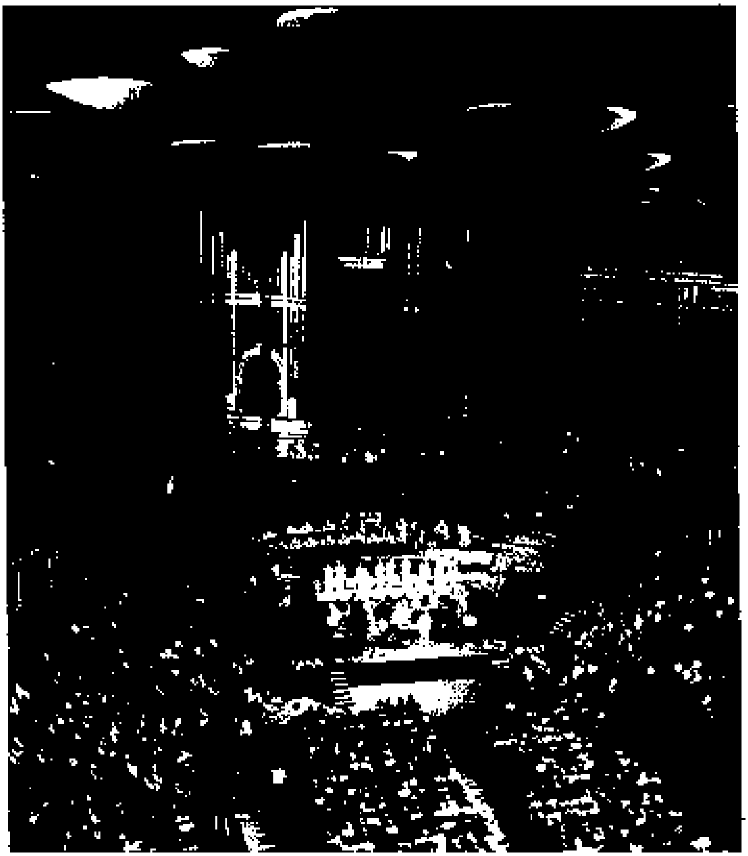
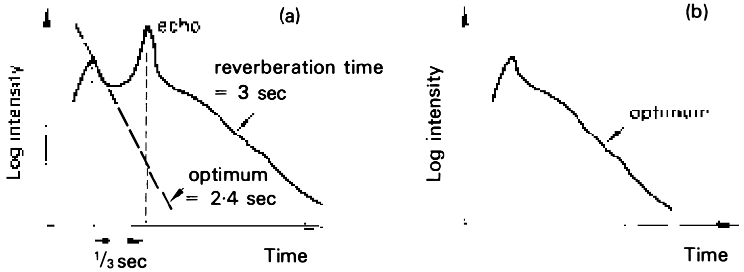


Fig. 99
The first night of the proms at the Royal Albert Hall. This picture shows the elliptical shape of the hall, the reflectors above the orchestra, and the saucers at the mouth of the dome.

problems. They arise, inevitably, from its great size, and over the years attempts have been made to improve things. The presence of an echo is almost unavoidable, due to the scale of the building and also to the dome, which represents about one-third of the total volume of the hall. Until about ten years ago, sounds which travelled up into the dome reached the listeners about 350 milliseconds after the direct sound, and produced a pronounced echo. The concave shape of the mouth of the dome caused focussing of these sounds, with the result that the echo had an even higher intensity than the direct sound (Fig. 100)! The problem was particularly troublesome to those seated near the back of the ground floor, to whom the focussed sounds were first directed. Absorbing all this energy would result in too low a reverberation time, so some means had to be found which would diffuse it, and reduce the path length of sounds which travelled the great distances to and from the dome.

An ingenious, economic and most effective solution to these problems was proposed by K. Shearer in 1969. He achieved it by suspending above the hall about 100 polyester/glass fibre diffusers, which he called flying saucers, whose diameters ranged from about 2 metres to 3·5 metres. They work effectively without

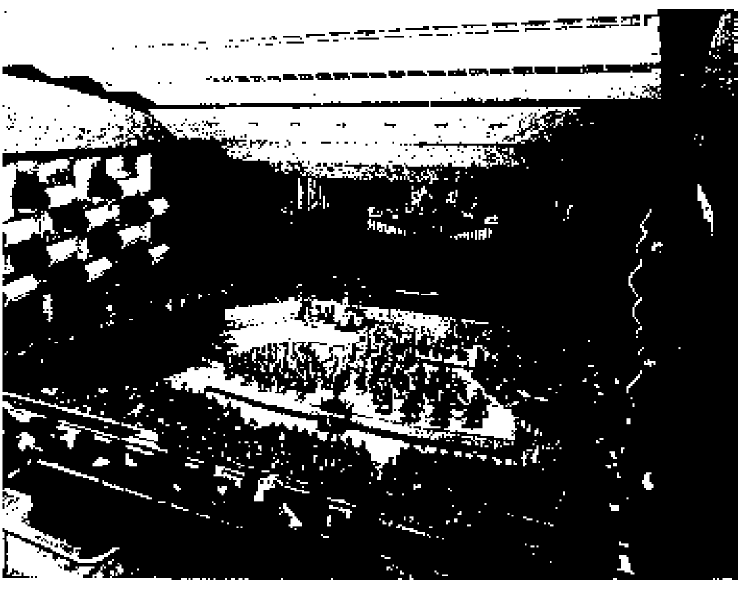
Fig. 100
The decay of sound near the back of the ground floor at the Royal Albert Hall, a) before the 'flying saucers' were fitted at the mouth of the dome, and b) after they were fitted. Notice how the echo is pushed into the general decay curve.



detracting from the appearance of the hall. Their lower convex surfaces disperse the sound that reaches them so that the intensity of the echo is reduced and a more diffuse sound field is returned to the audience. In addition, the diffusers reduce the distance of travel of these reflected waves, for they prevent many of the upward travelling waves from entering the dome volume. Hence the time interval between direct wave and echo is also reduced. Those waves which do travel up to the dome, impinge on the upper surfaces of the diffusers as they are reflected back, and are absorbed by the glass fibre material contained in them. The final result is that the important ratio of direct to reverberant sound energy at mid-frequencies is increased, the reverberant sound made more diffuse and the reverberation time optimised to about 2·5 seconds (Fig. 100). Recent experience confirms that the problem of the echo has been greatly reduced.

The experience of more recent generations is reflected in the design of The Royal Festival Hall in London. It was built as part of the development for The Festival of Britain in 1951. Its definition is extremely fine, indeed some have said that the acoustic is too dry, with a reverberation time of about 1·5

Fig. 101
The important ceiling design, irregular side walls and steeply raked seating are shown clearly in this picture of the Royal Festival Hall.



seconds at mid-frequencies. Its form is fairly long and narrow, with a steeply raked floor (Fig. 101). Note the important ceiling section, with its staggered reflecting surfaces and the boxes protruding from the walls, which help break up incident sounds and prevent flutter echo. The volume of the hall is 21 240 cubic metres, the seating capacity is 3000 persons, and the initial time delay gap in the centre of the stalls is about 35 milliseconds. Another feature of its modern design is the provision of an assisted resonance system in the ceiling. Briefly, it consists of a complete audio reproduction system, composed of a microphone, amplifier and loudspeaker. There are in fact about 170 such systems, each one covering a narrow frequency range of a few Hz, between about 60 Hz to 700 Hz. Each microphone is positioned by experiment at a point where a maximum in sound pressure at that modal frequency occurs. Some of the microphones are placed in Helmholtz resonators, tuned to the mode frequency. Hence when music is being played, sounds are picked up by the microphones, magnified and stored by the Helmholtz resonators, and reradiated by the corresponding loudspeakers to assist the hall acoustic (Fig. 102). The natural decay of sound

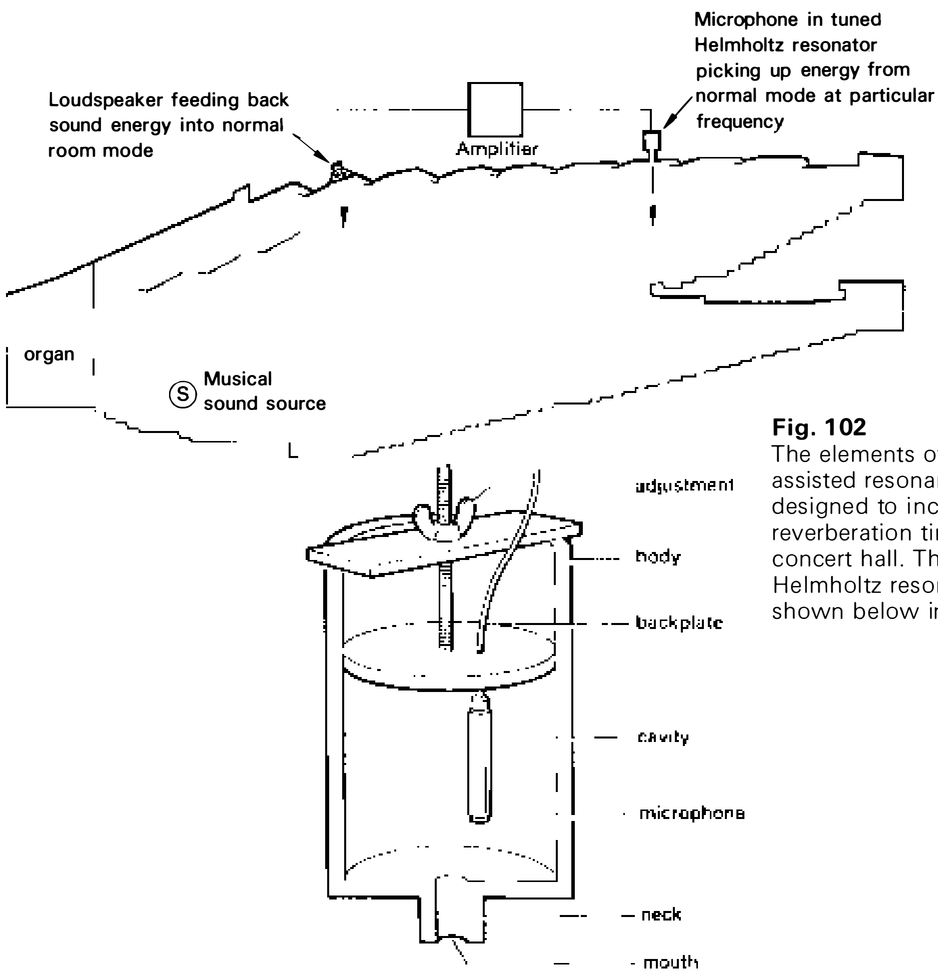


Fig. 102
The elements of an assisted resonance system designed to increase the reverberation time in a concert hall. The Helmholtz resonator is shown below in detail.

in the Helmholtz resonators means that the assisted radiation form the loudspeakers is further delayed, after direct sounds in the hall have subsided, so that the reverberation time is artificially prolonged. Experiments in the Royal Festival Hall began in the 1960s, and the assisted resonance system was found to be sufficiently helpful for it to be made permanent in 1969.

Another example of this system is the one installed commercially in the Central Hall of York University in 1973. Figure 103 shows clearly the resonators suspended at different heights, i.e. at points of pressure maxima at the different frequencies covered by the system. The volume of this hall is only a little over 7079 cubic metres, but its natural reverberation time of about 1 second made it a little on the dry side. The assisted resonance installation provided a useful extension of this in the lower frequency range, and was quite well received when first auditioned by experts in the Summer of 1973. The assisted resonance system allows changes to be made in the reverberation qualities of a hall, to suit, for example, music of different ages and style.

The town of Croydon, just outside London, has been famous for its fairs since the fourteenth century. This tradition led to the development of the Fairfield Halls in 1962. One of these is a concert hall whose acoustical qualities have been highly praised. Of somewhat similar general form to the Royal Festival Hall, the Fairfield hall (Fig. 104) is smaller, about half the volume. The initial time delay gap is about 25 milliseconds. The canopy above the auditorium and the ceiling construction are extremely successful in projecting sounds rearward. Notice, also, the alcoved panelling, the protruding boxes and columns at the sides, which aid diffusion. Although a fairly small enclosure, the reverberation time is still a little over 1·5 seconds in the middle frequency range, which gives adequate fullness of tone to music.

There must be many of us who have wished we could play

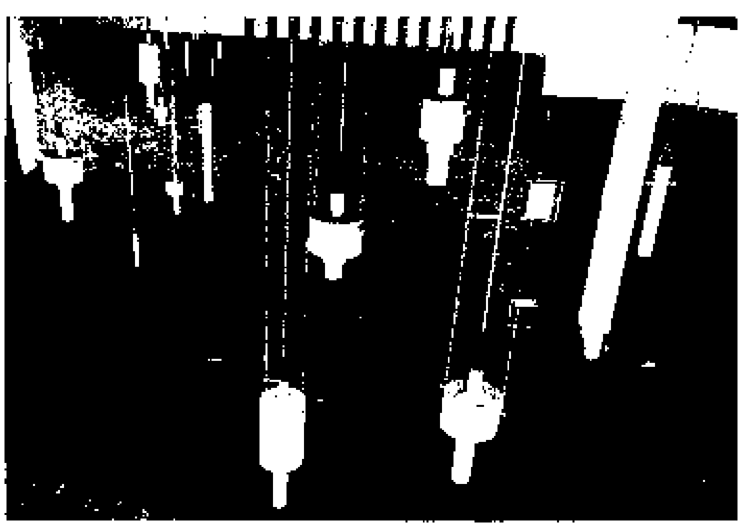
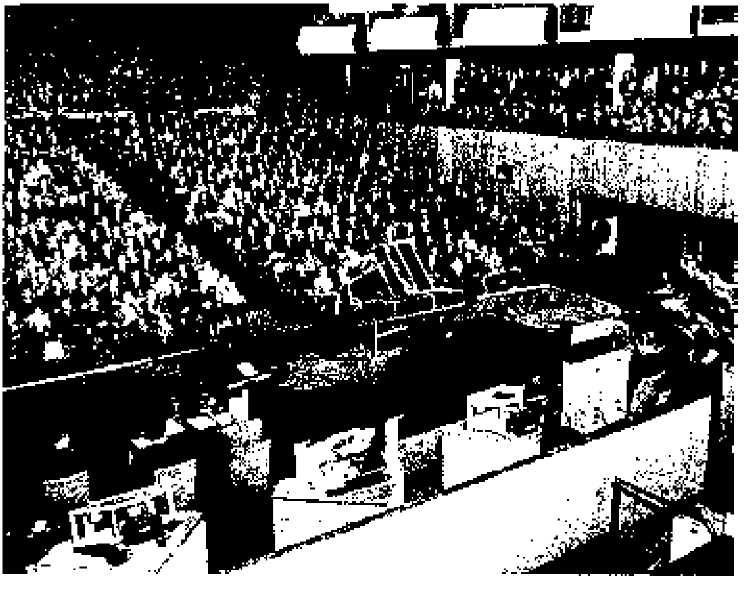


Fig. 103
Helmholtz resonators suspended above the Central Hall of York University form part of the assisted resonance system there.

Fig. 104
Fairfield Hall. Showing the irregular alcove panelling on the side walls, the protruding columns and boxes for diffusion, and the canopy above the stage for early reflected sounds.

ourselves in one of these great halls, or, even better for the audio enthusiast, have our musical efforts recorded as we play and then make a 'live-recorded' comparison before a large audience. A



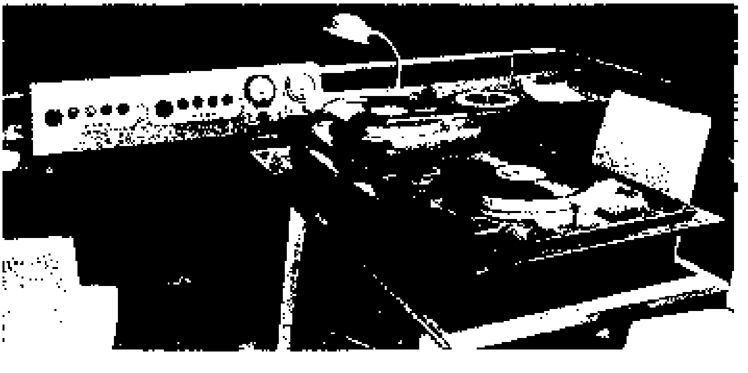


Fig. 105
A recording is made of a harpsichord during one of the Briggs concerts in the Royal Festival Hall. Later the music is reproduced over the audio system shown in the foreground, (above). The disc playback equipment and Quad amplifiers situated in Box 25 in the Hall, (below).

marriage, as it were, of the music of yesterday with the science of today. It was with thoughts such as these, and as a demonstration of how well audio equipment could serve the cause of music appreciation by reproducing it in its original surroundings – that Gilbert Briggs organised some special concerts in the 1950s. Gilbert Briggs founded the famous Company Wharfdale, and the success he enjoyed was entirely due to his personal enthusiasm and understanding of audio and music. The music concerts, as some readers who may have been there will know, were held at the Royal Festival Hall in London and at the Carnegie Hall in America. It was certainly a new idea, and a great success. Who had ever heard of 3000 people being invited to hear, as well as live music, the début of a hi-fi system? There were four concerts in all held at the Festival Hall, and in only one case, in 1956, did the demand for tickets not quite meet the number available. Some very distinguished musicians took part in the concerts, for example Leon Goossens the oboeist and Ralph Downes the organist; whole orchestras were booked for some of them. The

Fig. 106
Gilbert Briggs addressing an audience in the Carnegie Hall in America.

music making was recorded, often in stereo, by Peter Walker of Quad, and John Collinson who then worked with him. The recitals also included a variety of disc record material, with lectures and announcements by Gilbert Briggs. The controls and playback equipment were situated in Box no. 25 in the Festival Hall, while on stage with the presenter were the loudspeakers and power amplifiers. Among the models used were the Quad corner ribbon loudspeakers and Quad 15 watt valve amplifiers. As Gilbert Briggs said, 'it always surprised me that these were adequate for filling the large hall with reproduced sound, but the most outstanding feature of the RFH is that you can hear well no matter where you sit or stand'.

These evenings were fascinating and, thanks to their presenter, humourous as well as being unique musical experiences. Even so, they were costly. No profits were made and often there were net losses which had to be met by Gilbert Briggs himself. In New York, the cost of hiring the Carnegie Hall and the artists was even higher but this did not prevent the concerts from being a complete success. Some very fine recordings were made by Columbia and the Americans gave the team enormous technical help in ensuring a very close match between the playback installation and the live performance. Apart from hearing good music and what could be achieved from sound reproduction equipment of the time, the audience loved the spirited enthusiasm the team showed. The artists and musicians entered into the spirit of the experiments with similar keenness. They must have been very enjoyable occasions. One of the recordings they made in the Festival Hall, that of the Vaughan Williams Sea Symphony, was played to me only recently. It is a remarkable recording, and I think captures something of the wayward feeling of those early days of audio development. But I am glad to say that the verdict of every demonstration was, as it should be, that 'the live performance is best'!

THE DOMESTIC LISTENING ROOM

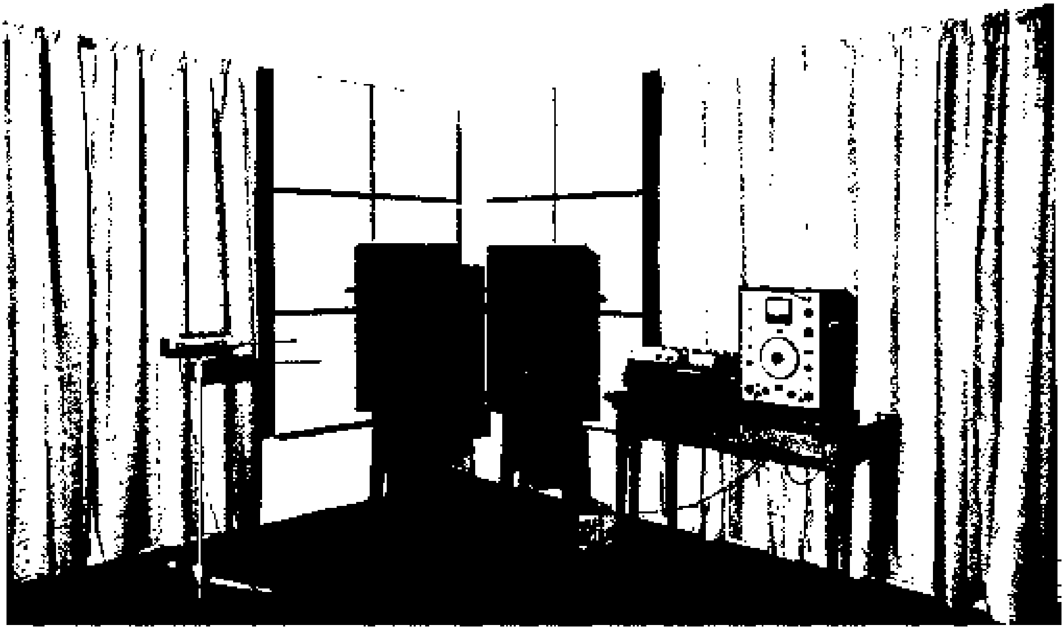
Those who travelled home from the Briggs concerts must have wondered how different it all sounded from their own domestic experience, and not just because of the replay equipment. There must have been many who were using the same set-up in their own listening rooms. The difference, of course, was in the listening environment, the size, shape and treatment of the room in which the music was being played.

The reverberation time of the concert hall is several times greater than that of the domestic living room, and this means that far more sound energy is stored in the hall, to give more of a blending effect, even to reproduced music. Then, of course, the extreme bass performance of the concert hall is far smoother and more extended than it could possibly be in a small room, because of its greater number of modes. Even in the upper bass range the concert hall acoustic will be superior to the small room, because usually the acoustic design and treatment will have been optimised in this register, where most sound energy resides. Hence the principal differences between the two listening environments will lie in the low and middle frequency ranges, where the main body of the music is contained.

These major differences mean that the design of small rooms, such as talks studios and music monitoring rooms, present problems entirely different from those found in concert halls or large studios. In fact, the criteria to be satisfied in the successful design of small studios is no different from those which apply in domestic listening rooms. In a speech studio, for example, the announcer's voice is to be transmitted to the home listener, free from any intruding reverberance or colourations from the broadcast end, i.e. from the speech studio. In a small studio used for monitoring recorded music, the purpose is to hear the frequency balance and the recorded acoustic, *not* that of the monitoring studio. Exactly the same purpose should be served in the home listening room; the recorded acoustic should be heard without any acoustic problems of the listening room being evident. When music is played 'live' in the home listening room, the 'live' reverberance should be smooth and uncoloured, and this still requires that the room acoustic be a good one. Hence in the home listening room, good room design will ensure that the sound field received by the listener will be free from acoustic colouration, whilst good broadcast studio design will ensure that the sounds received by the microphone will be free from such colouration.

Ideally, the sound field in a small room should be as it is in a larger room, random and diffuse at all frequencies, for then changes in sound level will be smooth and unobtrusive. It implies perfect uniformity of absorption at each boundary, and in studios designed solely for music and speech transmission, every effort is made to ensure that this is the case. An example is shown in Fig. 107. Different materials are each distributed in patches over all wall surfaces, giving uniform absorption over most of the audible frequency range. The 2-foot by 2-foot modular absorbers include low frequency, middle frequency and high frequency types. Diffusion is obtained by interspersing different types, one amongst the others. The curtains are for decorative purposes, but behind them are further modular absorbers arranged similarly. The absorption provided by the walls, ceiling and carpeted floor ensure a reverberation time of about 0·4 seconds, about the same value as in a domestic room of similar size. It should be noted, however, that the studio is optimised for the sole purpose of sound reproduction whereas the domestic listening room must serve other purposes.

Just as in a concert hall, it is not only the reverberation time which is the significant factor. The reverberation time relates to

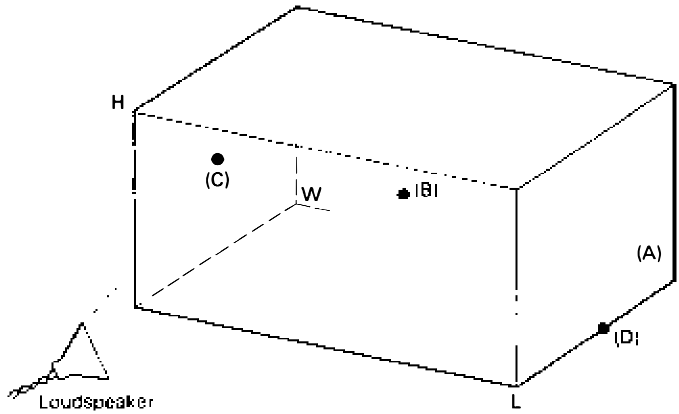


the build-up and decay of a diffuse sound field, from its steady-state value in a room. However, music is composed of many short (i.e. transient) sounds, and these are each very brief compared with the reverberation time, even of a small studio or home listening room. There are about twenty or so sounds made in every second, with durations varying from about one-tenth of a second to several seconds, depending on the music and the instrument being played. A similar situation exists in normal speech; although fewer sounds per second occur, their durations vary from about one-fiftieth to one-third of a second, and they are punctuated by longer time intervals between words and sentences. Hence steady-state conditions are not often reached; the durations of music and speech sounds are often shorter than the reverberation time, even of a small domestic listening room. Hence, the transient properties of the room, i.e. its behaviour during very short time intervals, is as important here as in the concert hall.

Fig. 107
Panel type acoustic absorbers distributed irregularly about the walls of the BBC's studio at Avenue House for good diffusion.

In a time equal to the reverberation time, 0·4 seconds, a sound wave can travel over 100 metres (about 400 feet). It could, for example, be reflected many times between two facing walls, and give rise to transient buzzes or rings, which will be clearly heard, especially if the walls are highly reflective. This is easily noticed when a short impulsive sound is made in the room, such as a hand clap. As sound is reflected to and fro, a rattle can be heard following the hand clap, or in smaller rooms with shorter distances between walls, a ring or buzz will be heard to follow the direct sound. If the walls, floor and ceiling are uniformly treated, such discrete reflections will not be formed, because

Fig. 108
Audibility of modes in a room. At A all modes have highest audibility, at B only one-eighth of all modes have high audibility, at C about one-quarter of all modes have high audibility, and at D about half of all modes are clearly audible.



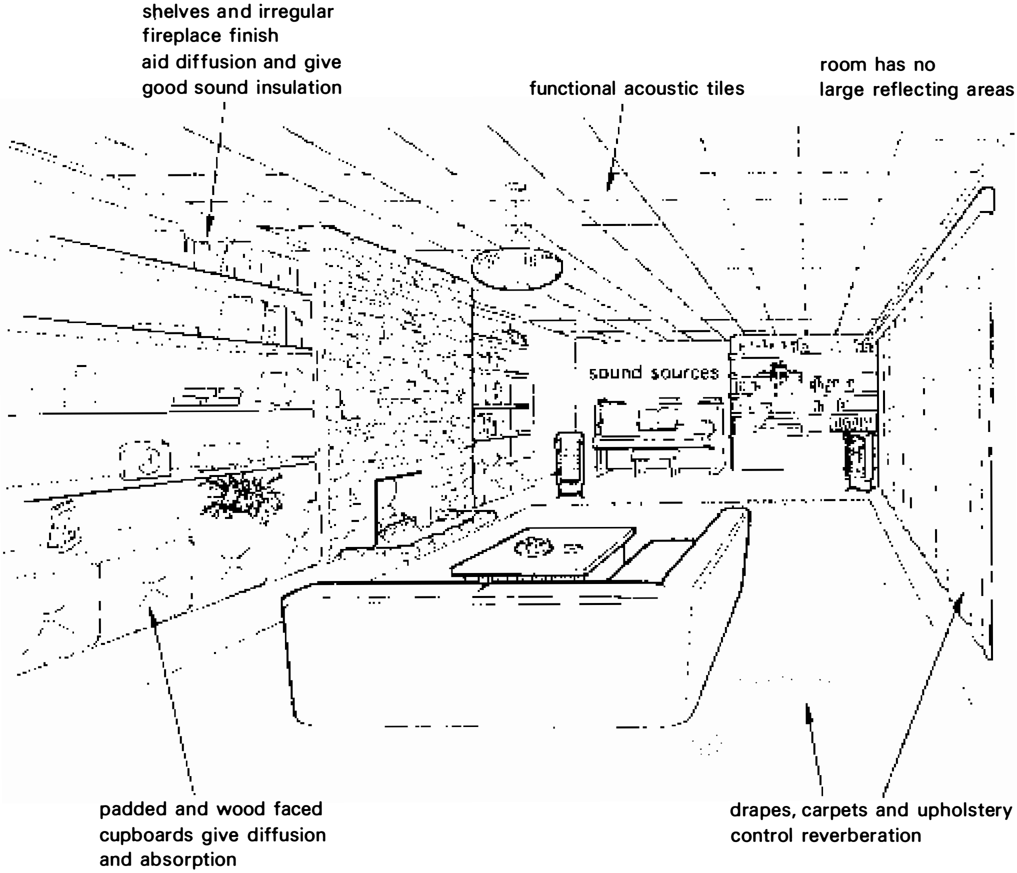
sound waves will be directed randomly in every direction. Hence uniform acoustic treatment also ensures that the room will transmit transient sounds faithfully.

A well-planned listening room must be one which produces an unobtrusive sound field, both under steady-state and transient conditions. If the room is of rectangular shape, it is desirable that at least one of its dimensions is in irregular ratio to the other two, even if they are simply related. If this requirement cannot be met by the basic structural dimensions of the room, then consideration might be given to fitting a suspended ceiling, carrying acoustic absorption material. The floor-ceiling dimension will be effectively that between the floor and the surface of the suspended ceiling.

An alternative to structural change is to provide sufficient deadening absorption for the room modes to be less prevalent. A point to remember, also, is that in the centre of the room, which is where listeners will normally be seated, the audibility of room modes is lowest. In fact only one-eighth of all modes have their maximum pressure at the room centre. In the room corners, on the other hand, every single mode frequency has a pressure maximum (Fig. 108). This is easily verified by the listener moving near to any corner of the room while listening to some music. He will notice that the music acquires a boomy, plummy quality to its lower-frequency register. It is sometimes said that the loudspeakers should therefore be positioned tight into the room corners, for the room will then augment their bass performance. Bearing in mind that the principal colourations of the room occur in the clearly audible *upper* bass range, between about 60 Hz and 200 Hz, I would say that this is an unwise practice. In fact there is a disadvantage in placing speakers too close to the room corners, especially if they are high-quality systems with well-extended bass performance. The room colourations will in fact *mask* the very deep bass notes reproduced by the speakers, because the room modes are of rather higher frequency. The result will be that natural, deep bass quality is spoiled, and replaced by irregular, plummy upper bass notes.

Figure 109 shows the layout of a domestic listening room of rectangular shape and fairly typical size that has some features which are important to good listening conditions. In this room transient and steady-state sound problems are minimised, and the reverberation is properly controlled. Firstly, the room is of approximately rectangular shape, and the sound sources are placed along one of its shorter sides. This means that early reflected sounds from the narrowly spaced walls will arrive soon after the direct sound, and so augment it. When the music is reproduced over the loudspeakers, this arrangement will lead to good stereo image stability and sensible image size. There will be the same advantage when music is produced 'live' in the room; the 'live' reverberance or reflected energy will be quickly directed to the listeners. Except at the far end of the room where sounds emanate, there are no large reflecting wall areas to strengthen room modes or support flutter echoes or buzzes. In fact the acoustic treatment in the room must never be unevenly distributed on the walls, floor or ceiling; this would mean a net flow of energy away from the reflecting surfaces and towards the absorbing ones, which must detract from good diffusion. If the acoustic treatment is

Fig. 109
A well-planned listening room showing features which are important to good listening conditions.



uneven, it could make transient phenomena or mode frequency problems associated with the untreated surfaces, even more evident than if the room had not been treated at all. This would happen because the uneven treatment would suppress some of the modes, but leave the few remaining even more conspicuous. It is the axial modes, i.e. those occurring between any one pair of opposite surfaces in the room, which contribute most to colouration. The room is not cluttered with complicated ornaments, such as trays, trolleys or delicate metal and glass objects, for these can rattle and resonate in all manner of ways. It is surprising how often these things become excited into distracting rattles and rings by the dynamics of the music.

The room features not only acoustic absorption materials, but irregular reflecting surfaces provided by the wall furniture. The bookcases, shelves (which should not be too tightly packed with books or papers) and reflective lower cupboards aid diffusion by scattering the incident sound and maintaining the right degree of reverberation. Another advantage of wall furniture is that it helps to isolate sounds made in the listening room from entering neighbouring rooms through wall vibration. Windows require a heavy drape to conceal the large reflecting area while listening, and the ceiling may carry functional acoustic tiles which contribute a little useful absorption in the middle frequency range. It is helpful if the floor is thickly carpeted, for this will take care of any mode problems between floor and ceiling. The absorption coefficients of almost every material used in furnishing today are available from the raw material manufacturers, (and see Table 14 p. 149). The acoustic absorption which the drapes, carpet and upholstered furniture provide, will control the reverberation time to about the 0·4 seconds required to give enough liveness for recorded acoustics to benefit.

The listening area is near the room centre, but if the room is fairly long, say over 6 metres, it may be necessary to move this area nearer to the sound sources. The room modes will still have low audibility, but the ratio of direct to reverberant sound will be slightly increased. I do not recommend that the loudspeakers be placed tight into the room corners; place them one metre or so from the walls, and maybe inwards slightly. They should be so orientated that their treble axes cross at the room centre. Then, listeners will be within 30 degrees of both axes, at almost every listening position. This is illustrated in Fig. 110. The size of the room is fairly typical, but it can be seen that the listener can move a metre or so in almost any direction from the centre, and still be within 30 degrees of both treble axes. Any point within the solid lines is within 30 degrees of at least one of the speaker axes, whilst any point within the much larger area to the right of the broken lines and still within the solid lines is within 30 degrees of both axes. In practice, therefore, almost every listener in his home listening room is close to the axes of his speakers,

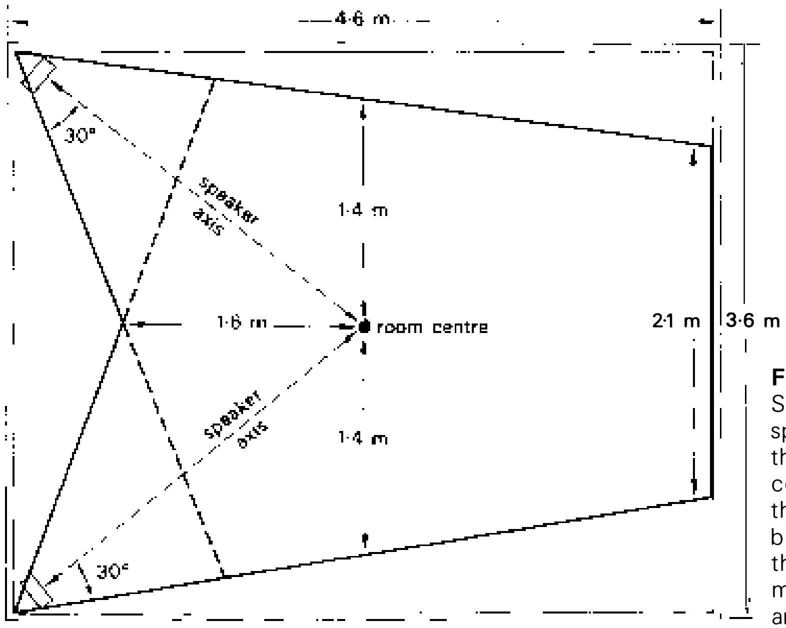


Fig. 110
Showing that when the speakers are angled so that their treble axes coincide at the centre of the room, the listener will be within 30 degrees of the axes when seated at most points of the floor area.

which is why the axial response curve of the speaker is a most important one.

If you have a listening room of something like this shape, remember some of these pointers when you are going to refurnish it. If you have a small room of regular shape, the best solution is to treat the room heavily, so removing much of its rather coloured reverberance and hardness. This might be done with curtains or wall tapestries, thick carpets and upholstered furniture. A neutral or even 'dead' listening environment will always be preferable to the rings and boom that are otherwise heard in small untreated rooms. See if I'm not right.

MUSIC LIST

We have come to the end of the book, and no doubt readers will be looking for a little light relief, for some music to enjoy. You may, remembering some of the points of the last few chapters, wish to lend an objective ear to your equipment, too. To meet both needs, I have compiled a record list, a discography of quite varied examples. All are of high technical quality but some I have chosen also on musical grounds.

Where there are specially good examples for listeners to make objective judgments of their hi-fi system, I have mentioned the special points to listen for in the comments. Many of the recordings are also available on cassette but the technical quality may be more variable and so they have not been included. The cassette can, of course, be easily traced from the disc number.

Whilst I have separated classical from other musical categories, this is purely for quick reference and I hope readers will take their choice from both sections. Great music is written in every age and style. A choice from this list would make an absorbing collection for the newcomer to recorded music. I hope that at least a few moments of special pleasure will be found from the records that you choose.

Note: An LP record is in preparation which is designed to complement this book and includes examples of special technical quality and musical interest.

Fig. 111
A selection of the records chosen for the music list.

Jazz, Middle of the Road & 'Pop'

Joan Armatrading *(A & M AMLH 64588)* Open, intimate voice sound, well balanced with group. Superb stereo imaging and extended bass.

Fleetwood Mac. 'Rumours' *(Warner Brothers KS 6344)* Open with good stereo and depth. Delicate voice and acoustic guitar sequences. Complex percussive tracks show up loudspeaker balance and amplifier smoothness.

Dire Straits Communique *(Vertigo 9102 012)* Excellent recording, fine treble presence and voice balance, slightly warm in the lower register.

10 cc. 'Bloody Tourists' *(Strawberry 9102 503)* Well-recorded, excellent music making from this famous group. Well-balanced vocal, with small intimate acoustic.

10 cc. 'Deceptive Bends'. *(Mercury 9102 502 BE)* A fine album. Record surface deceptively quiet. Bass firm but not hard, very 'clean' recording.

Dave Grusin. 'Discovered Again'. *(Sheffield Lab 5. Direct Cut Disc)* Remarkable drum sounds and tight transient control. Superb example of quality obtainable from an LP. Double bass number on side 2 is gruelling bass test for loudspeakers.

Bill Berry. 'For Duke' *M+K 'Realtime' RT 101 (Direct Cut Disc)* Remarkable recording of acoustic bass and saxophone; will show loudspeaker transient precision. Classic Duke Ellington music.

Rob McConnell & Boss Brass. 'Big Band Jazz'. *Umbrella UMB DD4 (Direct Cut Disc) (2 record set)* Effortless music making; exceptional recording technique takes the listener closest to the musicians.

Jay Berliner, Toots Theilman & other guitarists. 'The Guitar Session'. *Philips FDX 333 (Direct Cut Disc)* Unequalled recordings of the guitar; several artists and instruments. Listen for transient attack on the notes.

Andrew Lloyd Webber. Pagannini Variations for 'Cello & Rock Band. *EMI MCA MCF 2824* Tight sound, with close recording of solo 'cello. High Dynamics–good for testing dynamic capability of audio equipment.

Lincoln Mayorga and other musicians. *Sheffield Lab 1. (Direct Cut Disc)* Tremendous instrumental variety and spectrum; hear the musicians 'stand by' at the start of side 2.

English Sinfonia with Neville Dilkes. 'Grainger on the Shore'. *EMI ASD 3651* Pleasant music and rich tapestry of instruments which sound best on loudspeakers with careful middle and treble balance and smoothness.

Classical

Sir Adrian Boult, London Philharmonic Orch. Holst, The Planets. *EMI ASD 3649* Recording specially sponsored by KEF Electronics. Atmospheric, airy, superb low register.

Sir Adrian Boult, London Philharmonic Orch. Elgar, Symphony

No. 1. *EMI ASD 3330* The last and finest recording of this moving work by one of our greatest interpreters. Superb balance and tonal accuracy of the string section of the orchestra.

Colin Davis, Concertgebouw Orch., Amsterdam. Stravinsky, Petrouchka. *Phillips 9500 447* Arresting, lively, fine attack and balance.

Robert Shaw, The Atlanta Symphony Orch. & Chorus, Stravinsky, The Firebird; Borodin, Polovetsian Dances. *Telarc DG 10039 (Digitally processed)* Wonderful album–fine example of quality available from an LP.

Zubin Mehta, Israel Philharmonic Orch. Mahler, Symphony No. 4. *Decca SXDL 7501 (Digitally processed)* Great attention has certainly gone into this production. Superb string and brass sounds; high dynamics. Fine interpretation.

Frederick Fennell, The Cleveland Symphonic Winds. Works by Holst, Handel & Bach. *Telarc 5038 (Digitally processed)* A recording revealing the varied tone colour of all brass and woodwind instruments. Tests so many aspects of pick-up and loudspeaker balance and definition.

Bernard Haitink, Concergebouw Orch., Amsterdam. Debussy, 3 images for orch; Danse Sacree & Danse Profane. *Philips 9500 509* Wonderful, reflective music, perfectly recorded. Also intricate textural writing. Any depressions in the frequency range of the system and some instruments will be lost. The two dances for harp show an airy deep bass which is a particular test for bass character in the loudspeaker.

Anne-Sophie Mutter, Berlin Philharmonic, Herbert von Karajan. Mozart Violin concertos no. 3 and no. 5. *Deutsche Grammophon 2531 049* Fine musicianship from this young artist. Listen carefully to violin tone and how it is warm and resiny in its lower register and strident in its upper range.

Neville Marriner, Academy of St. Martin in the Fields. Haydn, Symphonies nos. 45 and 101. *Philips 9500 520* Extremely quiet surface, and a delicate balance make this a most attractive performance of this early music.

Janet Baker, London Symphony Orch., Andre Previn. Chausson, Duparc songs. *EMI ASD 3455* It seems almost irrelevant to comment technically, but notice how the mezzo-soprano voice extends far down into the lower middle register where only the very best loudspeakers will render its true fullness and range.

Vladimir Ashkenazy, Chopin, Etudes, op. 10 and 25. *Decca SXL 6710* The piano is well recorded, having a fairly resonant ring in its upper range, and good dynamics. Fairly open and no harshness. See how a poor performing pick-up will 'tear' on recorded peaks.

Harold Farberman, Royal Philharmonic Orch. Gliere, Symphony No. 3. *UNICORN PCM 500/1. (2 record set). (Digitally processed).* An epic symphony from a 20th-century Russian composer. Most attractive music of great scale. Very interesting and accomplished recording, containing instrumental delicacy and big orchestral dynamics. Superb stereo image precision.

Hi-Fi Terms and Specifications

These terms, found in the book and in manufacturers' specifications, are listed here to clarify for immediate reference the basic factors and qualities which any hi-fi listener or purchaser should be aware of.

Arm friction Vertical or horizontal friction due to a gramophone arm bearing.
Bandwidth Frequency response.
Bi-radial Elliptical stylus.
Bookshelf Common name for small loudspeaker enclosures, usually of the 'pressurised' type.
Channel separation Cross talk.
Compatible cartridge Pick-up cartridge which may be used on mono or stereo records.
Conical Standard shape of gramophone pick-up stylus.
Crossover The frequency at which one loudspeaker begins to take over from another, e.g. crossover point between low-frequency woofer and high-frequency tweeter.
Cross talk The degree of isolation between two audio channels. Specified in dBs at a given frequency.
Decoder Necessary for an FM tuner to reproduce stereo programmes.
De-emphasis Network to attenuate high-frequency audio in an FM transmission.
Dispersion Vertical/horizontal radiation pattern of a loudspeaker at a given frequency.
Drive unit Loudspeaker chassis without enclosure.
Elliptical Gramophone pick-up stylus with elliptical cross-section. Lower tracking and distortion compared with conical stylus.
Equalisation Frequency-response correction necessary to produce desired response. Specified as a time constant.
Loudness control Frequency correction applied to a volume control to partially compensate for hearing characteristics at low levels.
Power bandwidth Frequency response of a power amplifier at full output, usually between half power points.
rms Root Mean Square. True indication of the useful power content of a signal. Equals 0·707 of peak to peak value of a sine wave).
Rumble Very low frequency noise produced by mechanical drive systems, usually specified as dBs below a given signal level.
Rumble filter Electrical filter to remove rumble, usually cut-off below 50Hz.
Scratch filter Electrical filter to remove high frequency noise, such as surface noise on a record. Often switchable to different frequencies, with specified roll-off slope. i.e. 7kHz 18 db/octave.
Sensitivity The signal required to produce a given output level.
Side-thrust compensation System of weights on a gramophone arm to compensate the tendency for the arm to pull towards one side of the record.
Stylus force Tracking force.
Tracking The ability of pick-up cartridge to reproduce certain signals for a given tracking weight.
Tracking error Distortion of a reproduced signal due to gramophone pick-up arm moving in an arc rather than a straight line.
Tracking force The optimum playing force required for a pick-up cartridge to correctly reproduce all material.
Wow and flutter Cyclic variations resulting in frequency modulation of a programme due to mechanical imperfections in a drive system. Specified as percentage but must state measuring techniques.

Acknowledgements

The author is indebted to the following for their kind help and encouragement in the preparation of this book. Mr R. A. Walters (Consultant) for his research and contributions to Chapter 3, Lever Audio Ltd and Rola Celestion Ltd for their co-operation on the loudspeaker design in Chapter 4, Mr Rex Baldock (Consultant and Technical Advisor, *Hi-Fi News & Record Review*), Mr Stanley Kelly, Mr Peter Walker, Miss Joan Coulson (EMI Records), Mr Robert Cribb (Polygram Record Services), Mr W. Bayliff (Decca Record Research Laboratory), Mr Carl Anthony (IPC Press), Mr Tony Briggs (Philips Press Department, Eindhoven). Finally, my thanks to my editors Miss Camilla Simmons and Mr Peter Furtado for their patience and understanding, and to many others who provided advice and illustrative material for the book.

Photographs

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Roger Driscoll MSc., PhD. is widely known in the audio world as a consultant and lecturer. He has contributed to all the major audio journals and during the 1970s was a leading reviewer for the *Gramophone.* He has acted as consultant to many companies in the audio industry, including Philips, and he has also undertaken other projects such as the sound system installation at the Fairfield Hall. As well as considerable scholarship in the fields of audio and acoustics he has a special enthusiasm for music reproduction in the home.

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How do we hear sound?

What are the features
of different musical sounds?

How does your hi-fi
reproduce them?

Which system
should you choose?

How can you make the best
of your listening room?

How much money
should you spend?