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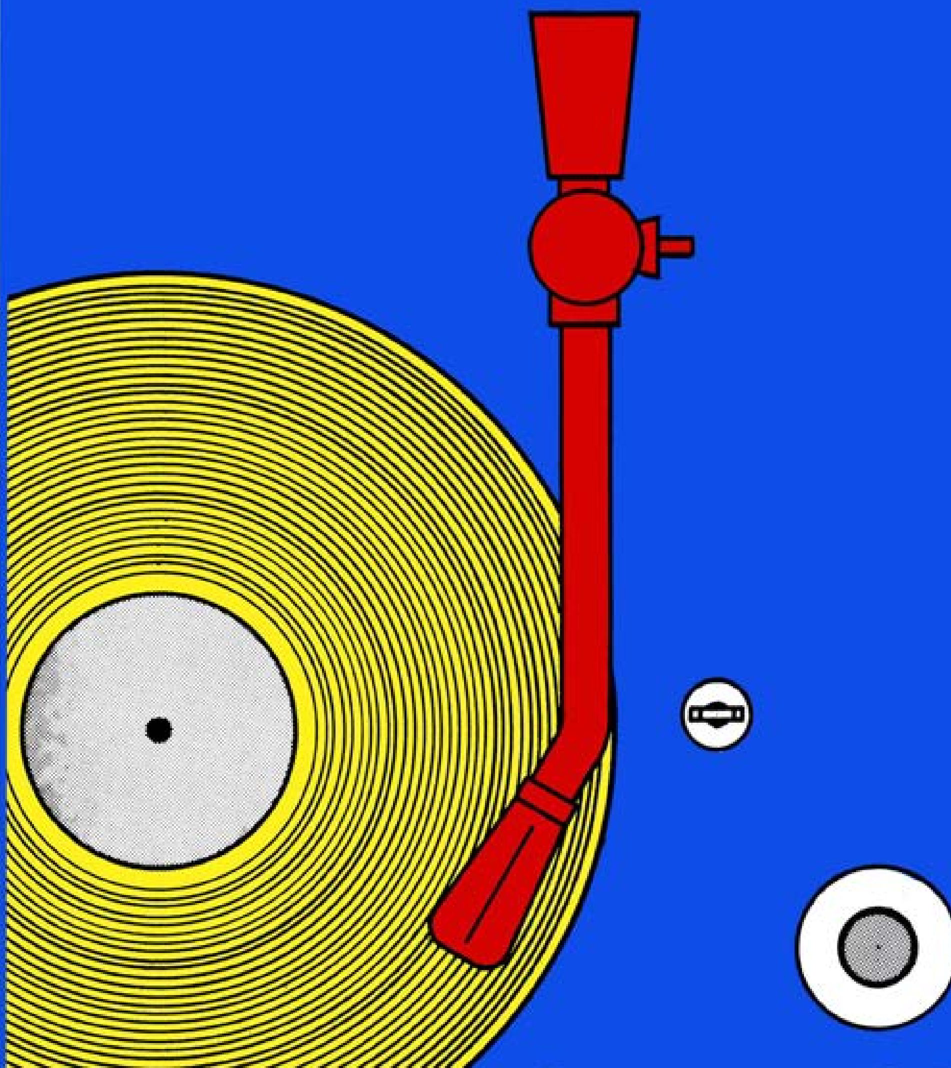
HI-FI FOR THE ENTHUSIAST



by M. L. GAYFORD

HI-FI FOR THE ENTHUSIAST-GAYFORD

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Hi-Fi
for the enthusiast

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Preface

The numerous people in many countries who are Hi-Fi enthusiasts aspire to enjoy the reproduction of music in their homes from equipment which not only offers better quality than that of normal commercial radio sets but also is suited to their individual needs as regards technical flexibility and the arrangement of the various elements in the home.

This book is intended for these enthusiasts and sets out to explain the most important technical factors in a straightforward though not over-simplified manner. The latest techniques used for pick-ups, turntables, amplifiers, tuners, tape-recorders and loud-speakers are highlighted and the important points to consider when buying or building units for home interconnection are stressed. Many pictures, diagrams and up-to-date references are included and some interesting and practical home-built systems of professional quality and appearance are illustrated.

I should point out, incidentally, that although this book has been written in good faith as a guide to the successful use of Hi-Fi equipment it is not implied that I can accept responsibility for unsatisfactory results! I should also emphasize that the fact that I have been given permission to reproduce certain designs and layouts does not mean that these are not protected by patents.

A wide variety of suitable units and components are now readily available and many home assemblers will find that the realization of a true Hi-fi system need not overtax their skill or their pocket. The lasting satisfaction obtained will be their reward.

M L Gayford

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Introduction

Music of all kinds has now become part of our lives. We have music while we work, music in the shops and music in the open air, and in the home we enjoy pop music, folk music, light music or serious music, according to our age and inclination. Never before has there been such a widespread desire for music and such a keen appreciation of it.

The production and reproduction of music now depends increasingly on electronic and electro-acoustical equipment and the technical standards of all types of system have risen enormously in the last few decades. Recording studios and broadcasters are equipped nowadays with expensive professional apparatus which is capable of very high standards of sound quality when handled with proper skill. Stereophony has added acoustic perspective to reproduced sound and, when skilfully used, gives a dramatic improvement in realism and emotional impact.

Contrary to opinions often voiced, jazz, modern pop and folk music benefit from high standards of sound reproduction as much as serious music. Complex rhythms, ingenious echo effects, vivid tone colours and vocalization can gain from high-quality reproduction as well as the more subtle but none-the-less compelling and dramatic performances of more serious or classical music.

Commercial radio sets and record-reproducers in the mass-produced cheaper field today can offer remarkable value for money in spite of enormously increased costs of production and distribution. Quality of sound reproduction and reliability are now both greatly improved as a result of many years of research and skilful engineering. Nevertheless, as is the case with most commodities, value is generally in proportion to the price and there are many people who are prepared to pay more to get increased enjoyment from a higher standard of sound reproduction.

An experienced Hi-Fi dealer is able to prescribe and install the best equipment for any individual need, but there is a considerable jump in price between mass-produced commercial sets and the best custom-built equipment. Many people, however, like to assemble "do-it-yourself" high-fidelity equipment from bought units, modules, kits or components, and it is here that great cost savings can be made without prejudicing satisfactory results, provided that the technical approach is along correct lines.

In this book it is proposed to suggest good outline designs and to stimulate ideas so as to enable you to avoid some of the pitfalls involved and to get good results from equipment which you can buy or make up. Although there are many excellent detailed designs published for amplifiers, tuners etc, both in the technical periodicals and by manufacturers of kits, there is a dearth of advice on the satisfactory integration of equipment units and in design of the cabinets and built-in enclosures which are so essential to the proper use and enjoyment of high-fidelity systems. It is here, particularly, that you can produce attractive individual arrangements to suit your own taste and domestic layout without being involved in exorbitant expense.

Television is also an established part of home entertainment and should be considered when a comprehensive entertainment assemblage is being planned. Television sound and quality is potentially of a very high standard and is worth feeding to a high-quality audio system.

The acoustic properties of listening rooms is of considerable importance, particularly for stereo reproduction, and a great deal can be done by comparatively simple acoustic treatment and rearrangement of existing rooms. If you are considering building a new house it is desirable, so far as possible, to design your listening room for the optimum properties.

In this book we will deal primarily with sound reproduction in the home or in small halls, together with aspects of sound recording which may also be of interest to those who wish to produce their own programmes. Recording should logically be treated first, but it is not of primary interest to many people who are building equipment, and so it is dealt with in a subsidiary manner.

We will consider first the sources of programme available, the equipment concerned being radio and TV tuners, record-players, tape-recorders and replay units, film sound-tracks etc. Amplifiers, control units, equalizers and filters of both valve and semi-conductor types are next dealt with, and then cabinets, housings and loudspeakers of various kinds, including multi-unit assemblies using electrical cross-over networks.

Before going into detailed consideration of the various parts of the system, however, we will deal briefly with the nature of speech and hearing, the acoustics of rooms and buildings, and the way in which loudspeakers may be used to produce the right sound.

1

Sound and hearing

The voice and other sound sources

When we inquire into the art of sound reproduction we must begin by studying the nature of human speech and hearing, as these are the basic elements involved. The making and appreciation of music came much later in human evolution but are now of equal if not greater importance as far as the high-quality reproduction of sound is concerned.

The first sound source with which we are concerned is therefore the human voice, as used in speaking or singing. It is an extremely complex mechanism, as regards both the rapidly varying waveforms produced and the complicated form in which the sound is radiated from the mouth, nostrils and, to a certain extent, from the walls of the throat and chest. The accurate re-creation of the speaking or singing voice requires, in the first place, an understanding of the mechanism in order that correct microphone and loudspeaker design and placement techniques may be employed.

Physically, the power of the human voice is produced by the air flow from the lungs, which is expelled by the action of the muscles of the diaphragm and rib cage. Fig 1 shows the parts involved in speech and clarifies the dominant modes of operation of the various parts of the throat, mouth etc. Vocal utterances are broadly recognized as "voiced" and "unvoiced" sounds. "Voiced" or semi-continuous waveforms are composed of discrete frequency components, used for vowels or a sung vocal output. These are generated by the air-flow modulated by vibrational opening and closing of the glottal vocal slot, producing the fundamental voice frequency, the harmonics or formants being augmented by the natural resonances of the other parts of the vocal tract, represented by the various cavities and constrictions. "Un-voiced" sounds such as "plosive" transients or hissing sibilants are produced by either closing off or releasing the air flow abruptly by the tongue, lips etc, or by causing turbulent "edge tones" etc by forcing the air-flow through narrow constrictions formed as required, for example, by the tip of the tongue against the teeth.

In fact, the character of a person's voice and much of the basic intelligence or substance of the vocal message are conveyed by

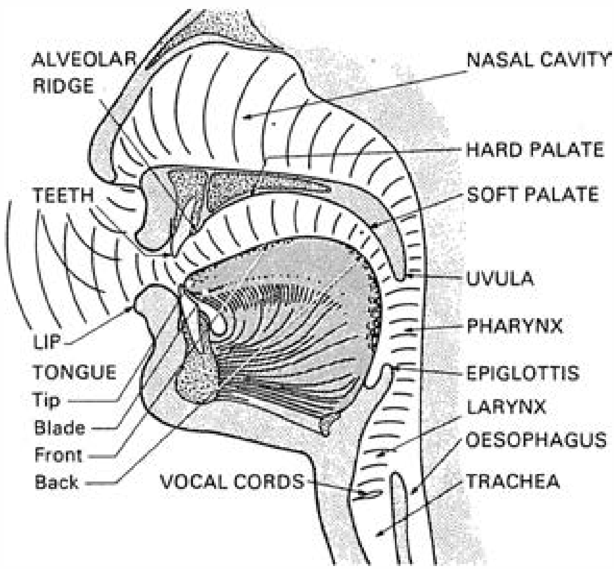


Fig 1. A representation of the speech organs showing the vocal-chord sound-source, the formant-producing vocal tracts and the radiating mouth and nasal outlets.

quasi-continuous utterances of near syllabic length (ie appreciable fractions of a second or a greater length of time). The distinctness or articulation is largely due to the unvoiced, transient, or sibilant sounds.

The range of variation of the human voice with regard to intensity, frequency range and emotional impact contributed by variations in stress, pace of utterance, pitch etc is enormous and it is not surprising that near-perfect reproduction of the human voice represents a searching test of a sound system. In fact, ideally a dynamic intensity range of 60 dB (1000 to 1 pressure ratio) and a frequency response from about 50 Hz to nearly

20 kHz are required, though it is true that it is possible to transmit intelligible, as opposed to completely natural, speech with a mere fraction of these values, as represented by the parameters of many systems designed to provide basic communication rather than the highest quality.

In order to pick up the various sounds produced by a human voice in the correct manner, a microphone must be designed to accept sounds from the mouth, nose etc accurately, without being influenced by the unidirectional blasts of breath which inevitably accompany speech, although these do not form part of direct sound when heard at the normal listening distance of a few feet. The microphone must also be placed so as to obtain the correct "acoustic perspective" for the voice in the particular room or surroundings where speaking is taking place. Broadly, this means that the microphone must not be so close as to react directly on the characteristics of the voice, by presenting an obstacle near to the mouth opening, nor must it be so far away as to pick up an excessive amount of reflected or reverberant room sound. In this connection, even a fairly directional microphone may not approach the remarkable discriminating power shown by the human auditory system when listening directly with two ears under comparable conditions.

Musical instruments of various kinds, on the other hand, differ considerably from the human voice and from each other, each type tending to have its own peculiarities. In general, they represent larger and more powerful sound sources than a voice, possessing horns, sounding boards, resonators and other sound volume-increasing devices. The waveforms of music are generally more constant and more continuous but there may still be important transients. Instruments often exhibit more markedly directional properties than a voice and this affects the required microphone placement techniques. Instruments may also produce subsidiary unidirectional blasts of air and other minor effects such as blowing, or mechanical operating sounds due to pedals, hammers, valves etc. These may also be picked up and exaggerated by an excessively close microphone.

Further, the whole question of stereophonic reproduction involves, in effect, another dimension in sound reproduction and may require complicated microphone techniques to produce a good subjective stereophonic effect when reproduced by two channels and finally listened to by two ears in the room where the sound is reproduced. It is important to distinguish between inter-channel differences existing in the necessarily artificial two-channel stereo system and the inter-aural effects existing around and within the head of the human listener. Two-channel electronic stereophonic reproduction is, in fact, an artifice which produces an illusion of sound source directionality and position by providing the human auditory system with enough of the right kind of clues.

The hearing process

The human auditory system is even more complex than the vocal system. The ear consists of an outer area, in which the head, the pinna and the ear canal are all important acoustically,

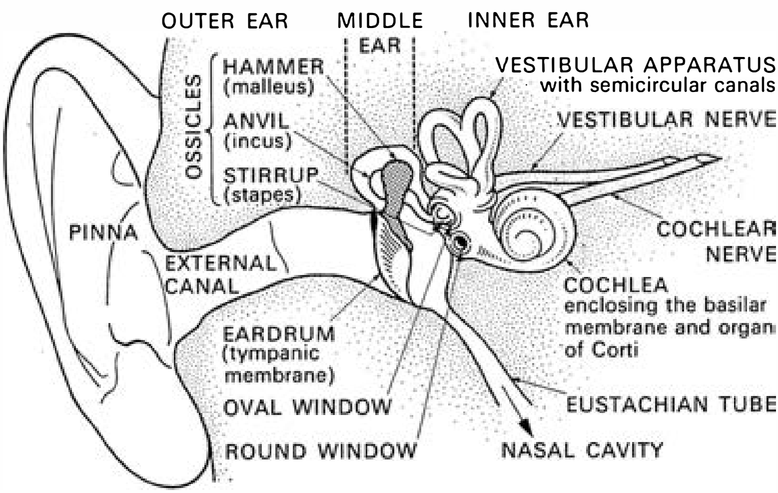


Fig 2. A diagrammatic view of the human ear which gives some idea of the complexity involved. The higher co-ordinating functions of the brain involved in the auditory functions are even more complicated and less well understood.

the middle ear system, the inner ear and neural system and, finally, the cortical and brain functions at various levels which are concerned with all aspects of the hearing process. The various parts of the ear are illustrated in Fig 2.

This shows a section through one complete ear and reveals the outer, middle and inner ear structure. The acoustic obstacle effect of the head modifies the response of each ear above about

1000 Hz, increasing the response of the ear to sound from the same side by up to 6 dB and reducing the response of that ear to sound from the opposite side. This difference in response with angle plays a part in directional hearing and also means that there is a difference between the pressure and the free-field response of the ear, as with a sound-pressure-operated microphone. Thus an earphone which is made to generate constant sound pressure in the ear canal will not produce the same sensation at all frequencies as will a freely incident plane sound wave which has access to the complete head, owing to the difference in response.

The pinna or convoluted outer ear structure protects the ear canal entry and supplies some directionality and a resonant cavity gain at middle-to-high frequencies for free-field listening.

The ear canal (meatus) is about 2·7 cm long and 0·7 cm diameter. It is terminated by the tympanic membrane or eardrum, which is an inwardly-directed cone of about 0·8 cm^2 in area. The first acoustic resonance of the outer ear and canal occurs at about 3000 Hz and provides a sound pressure gain at the eardrum. The adjacent middle ear cavity contains air (equalized to atmospheric pressure via the Eustachian tube) and the ossicular bone linkage system. In addition to providing mechanical coupling between the eardrum and the cochlea-coupled oval window, the ossicles provide an impedance transformation by lever action.

It may be noted that an earphone placed externally on the pinna is coupled to the ear canal in a complicated way; firstly because the individual convolutions of the pinna (which vary from person to person) represent a series of communicating passages and cavities feeding into the ear canal and, secondly, because a normal hard substantially flat ear-cap does not represent a constant sealed connection to the pinna, but one which usually involves a substantial leak to the outer air, the exact amount of leak depending on the way in which the earphone seats on the ear and the force which is applied to flatten the pinna. This leak can cause a serious loss of lower frequencies below 1000 Hz in the earphone response, whilst the acoustical passages can seriously modify the higher frequency response above 1000 Hz. These latter effects are even more severe in the case of large circum-aural ear-pad muffs, which are often provided in order to give better wearing comfort and improved noise exclusion.

The need for the maximum coupling efficiency between an earphone and the ear and the desire to eliminate external noise has led to the development of the insert-type earphones where a small tube or boss is fitted into the ear canal entry so as to make a seal. The small size of such transducers has tended to restrict the low-frequency power available and thus they have been largely used for the hearing aids and general communications rather than for high-quality reproduction.

If the ossicles become incapable of motion through disease, conduction deafness results, making air-path hearing aids useless. Bone-conduction receivers can be used to overcome this disability.

The inner ear includes the fluid-filled cochlea spiral, of length about 35 mm in all. The cochlea lengthwise partition consists of two discrete spaced membranes, one of which is the tapering basilar membrane with which the organ of Corti is in contact. The latter consists of many thousands of sensory cells terminating the auditory nerve. The inner and outer hair cells are among these and are in contact with the tectorial membrane. Deflection of the basilar membrane, due to a travelling wave excited by the stapes at the oval window, causes relative motion between itself and the tectorial membrane. This in turn deflects some of the hairs and produces electro-neural discharges in the associated nerves. The basilar membrane consists of tapered distributed elements and constitutes a non-reflecting dispersive medium. The amplitude and phase response of the various points along the basilar membrane simulate a succession of tuned filters of approximately constant sharpness or Q, linear increments of length along the membrane corresponding roughly to logarithmic frequency intervals, at least for middle and low frequencies. Thus a given frequency excites a maximum response at the appropriate distance along the membrane. The latter is therefore a frequency-analysing device with rather broad-band tuning. The reason for the extremely fine increments of pitch which can be discriminated must be sought in the higher levels of neurological and mental auditory processing.

Temporary deafness due to loud sounds is thought to be due to evanescent chemical changes in the hair cells. Prolonged exposures may cause permanent deafness due to damage and irreversible changes in the hair cells. Thus prolonged listening to music or other sounds of excessive loudness must be indulged in with caution in order to avoid auditory damage.

The cochlea fluid is subjected to compressional waves via the stapes and oval window, the fluid pressure waves ultimately being relieved by motion of the membrane covering the round window, which is in contact with the fluid on the remote side of the membrane complex which divides the cochlea spiral down its length. The vestibular apparatus and semicircular canals which act as a balance-sensing system communicate with the upper end of the cochlea above the oval window.

The physiological and mechanical functioning of the outer and middle ear and also the electrical neural potentials produced in the cochlea have been established and explained by G. Békésy in his prize-winning work,(1)* and by many other workers. The exact functions occurring in the brain in all the complex sensing, correlating, processing, association and memory activities which are concerned in the various aspects of our understanding, appreciation and use of the sounds which we hear are still largely a matter for speculation.

The basilar membrane thus performs only a rough kind of frequency analysis, whilst the middle ear muscles perform an automatic intensity-limiting function by reducing the mechanical coupling via the ossicular bones. The intensity of the perceived sound is transmitted to the brain in the form of electrical coded nerve pulses, the number of nerves "fired" in a given bundle and the pulse repetition rate conveying the information. Thus we see that auditory neural signals resemble a form of coded "digital" transmission rather than the "analogue" electrical waveforms with which we are familiar in sound-reproducing systems.

The brain is able to analyse the sounds incident on each ear, as regards frequency and intensity, with a remarkably fine resolving power. It is also able to conduct a running correlation on the signals from each ear to obtain directional sensations with regard to the incident sounds. The further running analysis of the complex and rapidly varying waveforms of speech alone and the comparison with previously stored memories of earlier sounds, together with all the other mental functions associated with the understanding and appreciation of sound is scarcely short of miraculous when compared to the common mechanisms with which we are more familiar.

* Superscript numbers in parentheses refer to the list of books and references on pages 165–6.

Psycho-acoustical aspects of hearing

We have considered broadly the way in which the human speech and hearing system functions, apart from the mental processes involved, the precise working of which can only be guessed at. However, a great deal of experimental work has been done to systematize the exact way in which we react to sounds of various types. It is desirable to have a knowledge of these characteristics in order to design and use high-quality sound-reproducing equipment.

The main aspects under which the psycho-acoustical hearing characteristics are studied are—

1 The perception of pitch (frequency).
2 The perception of intensity (sound pressure).
3 The appreciation of time duration, both in regard to transient sounds and in the recognition of small time intervals, particularly with regard to the arrival timing of wavefronts and the spatial localization of sound sources.
4 In the subjective impression we receive of the loudness of complex sounds, which depends on whether the sounds are impulsive or continuous in time and also on the frequency spectrum of the sound, as opposed to a simple objective measurement of the average or peak acoustic pressure.

Of these various psycho-acoustical factors, apart from those affecting the fixed performance parameters of the reproducing equipment, such as the basic frequency response and transient response, the main factor where adjustment may be required is the "scaling" of the shape of the frequency response when the reproduced level differs greatly from the original sound level at the microphone position. The response curve contours change with the incident sound pressure level, the most marked feature being the loss of sensitivity to low frequencies. This phenomenon is particularly noticeable when quiet male speech is reproduced at an unnaturally high level and gives rise to the so-called "loudness control" on some amplifiers, whereby the bass is automatically adjusted in level as the volume control is turned up. Unfortunately, the design of this type of control is complicated by the different low-frequency characteristics of speech and music.

Other types of "shaping" of the amplifier response curve are aimed mainly at minimizing various unwanted noises, such as low-frequency rumbles, hum etc, and high-frequency electronic noise, tape or disc noise etc, as well as higher-order distortion products. Some "equalization" or correction of known deficiencies in transducers etc is also used.

Stereophonic reproduction and binaural listening

It has been known for a long time that more than one reproducing channel is required if sound is to be reproduced in another place in a really satisfactory manner. Until recently economic considerations have limited nearly all communication systems to a single channel.

Thus the telephone is restricted to a single link, broadcasting is still largely performed through a single-channel transmitter, and a monophonic gramophone record is endowed with a groove executing lateral excursions only.

Nature has provided us with two ears and the nerve impulses from both ears are correlated by the brain so as to give definite subjective impressions as a result of almost any given acoustic stimulus, however slight. It is natural for us to listen with two ears to sounds approaching from various directions and we are able not only to form an accurate impression of the direction and character of sound sources but also to discriminate in a remarkable fashion in favour of wanted as against unwanted sounds.

Thus a monophonic reproduction of anything but the simplest sound source in idealized surroundings is a somewhat unnatural experience and we are unable to exercise our full psycho-acoustical powers. In many cases, the slight subconscious disturbance which is likely to accompany any unnatural experience would appear to be likely to prevent our full enjoyment or appreciation of monophonic reproduction of sounds, such as music, which have an aesthetic appeal. It is also common experience that we are to some extent unable to discriminate against unwanted background interfering noises or excessive reverberation when listening to normal monophonic sound reproduction, eg radio speech etc.

Present-day stereophonic sound reproduction might thus be partly defined as an attempt to provide extra directional information so as to make the result more natural and so to enable more normal psycho-acoustical processes to function for the listener. Economic factors limit us to two channels for domestic stereophony. More channels enable better stereophonic effects to be obtained but the gain in realism obtainable with two channels as compared to one can still be very marked.

To obtain the full benefit of binaural listening it is not sufficient to feed separate channels to each ear by earphones. The ears must be immersed in the "free" sound field of a room and the head must be free to turn and to make other subconscious movements which are vital to sound localization and correct natural auditory perception. Binaural listening on headphones can, however, give many of the benefits of stereophonic reproduction, although there will be certain shortcomings.

Early stereophonic experiments used a number of microphones spaced in a line across the front of a sound stage. A corresponding line of loudspeakers in another auditorium produced an approach to a "curtain of sound." It was found that three separate channels gave a fairly good stereophonic effect as regards naturalness of reproduction and spatial localization of sounds.

The three channels were carefully balanced to have identical frequency and phase delay characteristics. When a sound source is nearer to one microphone than to the others, two effects operate. Firstly, the direct sound intensity will be higher on the corresponding loudspeaker and, secondly, the sound will emanate from it sooner than from the other loudspeakers by a time dependent on the relative microphone distances. The velocity of sound-wave propagation is such that a path difference of one foot introduces a relative time delay of one millisecond. Thus we see that the stereophonic effect depends both on interchannel intensity differences and on interchannel time differences. It will also be noticed in a successful stereophonic system that a sense of spaciousness and direction is imparted to the reflected or reverberant sounds picked up by the microphones as well as to the direct sound.

In order to get good stereophonic reproduction over a large-audience seating area in a fair-sized hall or cinema it has been found necessary to use more elaborate systems utilizing a larger number of channels. Good results may be obtainable on or near the centre axis of the loudspeaker system when a limited number of channels is used, but a loss of stereophonic illusion and a defect described as "concaving" or "hole in the centre" are liable to be observed at some off-axis seats. The latter defect is manifested when a sound source moves across the stage. The source appears to recede or follow a concave path between the loudspeakers.

Five-channel stereophonic film systems using six magnetic sound tracks at the sides of 70-mm film have been used to give good stereophonic effects in cinemas. Five loudspeakers are spaced across the stage, the remaining channel being used for extra sound effects, produced in some cases by loudspeakers at the rear of the auditorium. This is done in order to get over some of the shortcomings of a stereophonic loudspeaker array which originates sound in a line in front of the listener. A few additional loudspeakers at suitable places can be erected as required and fed by an extra sound-track. Also, the full natural effect of reverberant sound cannot be reproduced unless some loudspeakers are distributed about the auditorium. Generally these must be fed with a signal having a deliberately introduced time delay as compared to the frontal loudspeakers, when used to enhance the reverberant quality. No artificial delay is added to the signal fed to the supplementary loudspeakers when it is desired to give the impression of a direct sound to the rear or above the audience. In this case, a common delay might be added to all the frontal stereophonic channels. The effect whereby the sound appears to originate from the source which originates the first sound to arrive at the listener was first investigated and analysed by de Haas.(2)

At about the same time as the early three-channel experiments were carried out, A. D. Blumlein of EMI proposed a different form of stereophonic microphone technique. In place of microphones spaced apart along a line or arc, coincident directional microphones were used with their major lobes of polar response oriented so as to point respectively to the left and right of the sound stage. The results are generally applicable only to a two-channel stereophonic system. No significant time differences occur between the microphones, as these are mounted as close together as possible. A sound source at an angle to the centre axis produces different amplitudes in the two channels as a result of the particular shape of the directional polar characteristics of the microphones used. It might be thought that the lack of inter-channel time difference information would seriously reduce the stereophonic effect obtainable. However, an effective amplitude-to-time conversion occurs at the listener's head due to the finite spacing between his ears.

The spaced microphone system is prone to some spurious effects, particularly when moving sources are involved. For example, the "Doppler effect" due to a rapidly moving source passing spaced microphones is reproduced at each microphone in turn, thus giving rise to several virtual sources when the sounds are reproduced. The crossed microphone system is free from this type of defect, though it has other limitations of its own. Thus it is sometimes difficult to obtain a proper stereophonic effect from sources spread across a wide sound stage unless the crossed microphones are moved well away from the stage, so giving an excess of reverberant sound. This follows from the fact that the entire stage width must be covered within the $\pm 45°$ angle between the main lobes of the crossed directional microphones. Any sources subtending angles greater than $\pm 45°$ may be reproduced at a reduced intensity on one channel and with a phase reversal on the other. In the case of crossed bidirectional (figure-of-eight) microphones, reverberant and other sounds emanating from behind the microphones will be reproduced with a 180° change of phase (an inherent property of a simple gradient-operated microphone). This may give the reproduction a disturbing or unnatural quality. Crossed unidirectional or cardioid microphones are somewhat better in these respects. It is not, of course, possible to have two distinct crossed microphones at exactly the same point in space. They are commonly mounted closely one above the other, often in the same casing. It has been found that the necessary finite vertical displacement between the microphones may introduce a small but sometimes noticeable inter-channel time difference for sounds emanating appreciably above or below the centre axis of the microphones, which may be interpreted by anyone listening to the final two-channel stereophonic reproduction as a horizontal displacement. Thus a person's voice and his footsteps might appear to come from different parts of the stage. A small horizontal displacement between the microphones is therefore considered preferable. In the case of very small capacitor microphone capsules, the vertical spacing can be made so small as to be negligible. Nevertheless, simple crossed ribbon or cardioid microphones will often give excellent amateur stereo recordings.

In practice, when an ambitious stereophonic recording is being made of, say, a large orchestral, choral or other performance involving a wide sound stage, it is possible to resort to a mixture of crossed and spaced microphone techniques.

Domestic stereophonic reproduction

Most producers of stereophonic programmes intended for domestic reproduction assume that the listener will sit near the centre axis of two loudspeakers spaced about 10 ft apart in a reasonably well-furnished living room of moderate size, the listening position being about in the centre of the room. It is advisable to have some acoustic absorption (eg curtains etc) in the side walls and on the wall behind the loudspeakers so as to minimize any loss of stereophonic positional accuracy due to first reflections of sound from these walls. Fig 3 (page 17) shows a plan view of such a room. It is possible that two loudspeakers with a fairly broad polar sound distribution are less critical as regards room conditions generally, but an improvement in definition and better stereophonic positional accuracy may be obtained if two loudspeakers with an optimum degree of directionality are used, eg two bidirectional electrostatic units or other loudspeakers with a carefully controlled higher frequency distribution.

The present means of general presentation and distribution of stereophonic programmes are by radio broadcasting (originally by two separate transmissions but now by multiplex modulation of a VHF transmitter), by dual-track tape-recorder or by stereophonic gramophone records.

Recently, attempts have been made to carry more directional information on the two tracks by making further use of the instantaneous relationship between the sum and difference signals in the left and right channels. Extra derived signals have then been fed to subsidiary front and rear loudspeakers in order to give a better acoustic ambience or sense of direction to the reverberant sound. The name "quadrophony" has been given to some of these developments.

Room acoustics: sound wave patterns

The sound pattern set up in a room is markedly different from the effects of sound propagation in the open air. Any source of sound causes alternate regions of increased and reduced pressure to travel outwards from its emitting surface. The source may be a pressure generator, ie a device which introduces alternating "puffs" of air under pressure into the surrounding air. An organ pipe, most "blown" musical instruments and certain specialized types of loudspeaker, eg the ionophone[16] or the pneumatic "loud-hailer," are examples of this type of sound generator. However, a more common type of source is a vibrating surface of a size sufficient to move enough air particles in contact with it to radiate an appreciable pressure wave. The normal domestic loudspeaker is of this type, in common with most stringed or percussion musical instruments. The human voice is, of course, a pressure type of source.

A sound wave started in the open air spreads out indefinitely, only suffering reflections from the ground, buildings, hills etc. Wind or thermal air currents will bend the wavefront, causing increased or decreased attenuation in the line of propagation, according to which way the wavefront is bent. For example, sound propagated with the wind is bent towards the earth and attenuation is reduced, ie the sound "carries better." The exact opposite occurs for sound propagated against the wind.

On the other hand, a sound wave propagated in a room is continuously modified in a very complex manner. It is reflected from all the boundary surfaces of the room and is selectively absorbed at each reflection, to an extent dependent on the porosity or other energy-absorbing properties of the surfaces.

The wavefronts are scattered or "diffused" by the presence of solid objects in the room or irregularities on the walls or ceiling. Some objects or surfaces in the room will be set into vibration by the sound waves and, if they are strongly resonant, their vibrations may persist long enough for them to be audible as subsidiary sound sources. Where walls, floors, ceilings etc, vibrate appreciably the effects may be described as structural vibrations of the room or building.

All these phenomena are studied in more detail in works dealing with room acoustics.[2,3] Note in particular, sound waves in a room sooner or later in their progress get into certain established "tracks" or room-resonant modes, which occur at various frequencies in one, two or three dimensions.

The overall effect on human hearing of these complex sound-wave phenomena is the crux of our problem. We have seen that the human hearing process may be thought of as having three main functional processes. The physical outer and inner ear mechanism may be regarded as an acoustic transducer of great efficiency and sensitivity, which responds to sound pressure variations and transmits equivalent nerve pulses to the brain. This in turn may be regarded as producing a running display of the instantaneous sound patterns at each ear. The human mind may for this purpose be regarded as a conscious scanning system associated with a memory store. The memory, built up by heredity and a lifetime of experience, enables the mind to carry out a continuous correlation, recognition and appreciation process on the sounds incident on the two ears. Thus, for example, the direction of a sound source may be almost instantaneously estimated, primarily as a result of the slight time difference taken by an oblique sound wave to travel around the head.

For present purposes, it is sufficient to note that we are able to assess the rapidly changing sound pattern is a room with extraordinary accuracy. For example, we are still able to locate a small sound source in a far part of the room, in spite of the fact that the direct sound from the source is to all intents and purposes swamped by a multitude of sound reflections from the walls. This is due to the "precedence effect." This means in general that if the direct sound arrives a few thousandths of a second (milliseconds) in advance of the wave reflections from the walls we are able to identify it and to locate the source, as long as we are listening with two ears. This power is lost when a normal monaural single channel is used, eg by putting a microphone in the room and listening at some other place via a loudspeaker. Stereophonic reproduction via two microphones and two loudspeakers can go a long way towards restoring the directional discrimination lost in monaural sound reproduction.

Loudspeakers in rooms

We see that the human speech and hearing processes are extremely complicated and that, strictly, it is necessary to study them in some depth in order to appreciate the factors involved in the use of loudspeakers in rooms. The complicated effects of room acoustics are also involved here. We wish not only to re-create speech and singing but also to reproduce realistically musical works by large numbers of instruments. When one considers the disparity between the many sound sources and the large and complicated waveforms set up by a symphony orchestra in a concert hall and the vibrations produced by a paper or plastic loudspeaker cone in a living room, some idea of the difficulties involved is brought home to one.

The heart of a loudspeaker is the transducer or motor unit. On its own it is unable to produce effective sound waves at low frequencies; it requires either a large radiating surface, as in the electrostatic loudspeaker, or some form of baffle or cabinet. Two distinct basic types of loudspeaker exist. In the first type, the rear of the unit is completely closed off by a sealed cabinet or box, so that only the front side of the cone is able to radiate sound into the room. In all other types of loudspeaker the rear side of the diaphragm, which is generating pressure waves in anti-phase to those at the front, is connected to the air in the room either directly, in the case of a flat baffle or a shallow open-backed cabinet, or via the port or other radiating openings in the cabinet walls.

The manner in which sound is radiated from a surface or an opening depends very much on the frequency. At the lowest audible frequencies the wavelength of sound is about 30 ft, which means that our sound sources are so small as to constitute omnidirectional radiators of spherical waves. An enclosed cabinet loudspeaker consists of one such source, whilst the open type represents two point sources, one behind the other working in opposition. These tend to neutralize each other and to produce less bass. They also have a bidirectional response, in that there is no resultant output in the plane of the real or imaginary baffle between them, producing a figure-of-eight polar curve similar in shape to that of a ribbon microphone. Thus we see that the purpose of a port or acoustic labyrinth must be to reverse the phase of the back radiation so as to aid that from the front, and to maintain the bass response.

At extreme high frequencies the wavelength of sound is less then one inch, which is considerably smaller than the average loudspeaker cone. The effect here is that, instead of being omnidirectional, sound is radiated in the form of directional beams. Where more than one sound source exists in near proximity, the effect is that interferences are produced at intermediate frequencies in particular, causing an irregular response.

We can summarize two of the main problems of loudspeaker reproduction as, firstly, maintaining a reasonably good efficiency at low frequencies and, secondly, obtaining a fairly consistent polar response, that is, a good omnidirectional or bidirectional distribution of sound at middle and low frequencies, together with a sufficiently well-spread and even polar sound distribution at higher frequencies.

There still remains the problem of how best to place the loudspeakers in your room in relation to the listening position.

No two rooms are ever exactly the same as regards their size, shape or acoustic properties, and the quality of the sound reproduction may depend to a large extent on the siting of the loudspeakers. A number of theories exist as to the ideal arrangements for various given cases. One is that most loudspeakers act as sound-pressure generators at middle and low frequencies and that if they are mounted in a corner, the converging walls act as an extension of the loudspeaker and increase the low-frequency response. Also, a room has a large number of natural resonant modes of the air and most of these have maximum pressure points near to the corners. Thus the maximum number of these

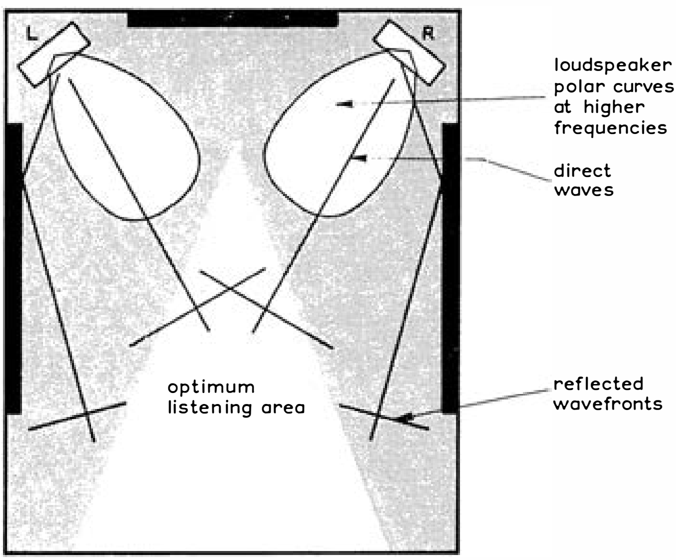


Fig 3. *An idealized room plan showing corner placement of two stereo loudspeakers in a symmetrical arrangement. Only two reflected waves are shown. In practice an almost infinite number of reflections blend to make up the reverberant sound. Excessive reverberation may "colour" the sound reproduction and can blur the stereo images.*

natural resonances should be excited by a pressure source in the corner, resulting in a smoother response of the reverberant sound in the room.

In practice the natural resonances in a room are not always evenly spread and structural vibrations of plaster-boards, floors and other large areas may exist. Undesirable resonances may be excited and so in practice it may be necessary to experiment to find the best loudspeaker positions, having regard to the limitations imposed by the doors, windows, fireplace and furnishings. The height of the loudspeakers should be such that the listener's ears are not too far off the centre axis of the tweeters or the main cone in the case of single diaphragm loudspeakers. It is here that bookshelf loudspeakers rather than floor-mounted cabinets may have an advantage.

For good stereophonic reproduction, too, one must obviously attempt to set a symmetrical "sound stage" about a central listening area in the room. This means that the two loudspeakers must usually be about ten feet apart, equally spaced with respect to the side walls, which should preferably be symmetrical as regards their acoustic absorbing or reflecting properties. Most living rooms cannot be re-planned especially for "stereo listening" but fortunately, we can appreciate many of the benefits of stereo even if some of the exactitude of sound image positioning is lost.

2

Program input sources

In the previous chapter we have briefly surveyed some of the overall considerations which concern our hearing and the reproduction of music on a small scale. We will now proceed to more detailed consideration of the individual parts of the sound-reproduction chain, beginning with radio input units.

Radio tuners and sets

Sound broadcasting is carried out on the short, medium and long wavebands, in addition to VHF (very-high-frequency) bands. Television radiations are on the VHF and also the UHF (ultra-high-frequency) bands. A very large number of stations now operate in most countries on officially allocated wavelengths. In addition there are sometimes some unofficial transmitters.

Some important factors that militate against high-quality broadcast reception are—

1 Interference from other stations (programme break-through, "hash," carrier beat whistles etc).
2 Noise interference (crackles, man-made and natural static etc).
3 Distortion (modulation and demodulation distortion, sideband cutting, selective fading).
4 Tuning drift and signal variations (circuit tuning drift, multi-path propagation fading, aircraft reflection flutter).

Some of these troubles can be dealt with by sophisticated receiver design and by the use of elaborate aerial (antenna) systems, but it unfortunately remains true that for the best results a fairly high signal strength from the wanted station is usually necessary. This implies a powerful transmitter or fairly close proximity to the station, the preferred distance depending to a fair degree on the frequency.

The relatively close frequency spacing of stations in the long, medium and short wavebands and the long range of interfering transmissions makes it difficult to obtain true high-fidelity reproduction on these wavebands. The transmissions have perforce also to be amplitude-modulated (AM). This type of modulation makes it difficult for deeply modulated signals to be rectified without the introduction of some intermodulation distortion.

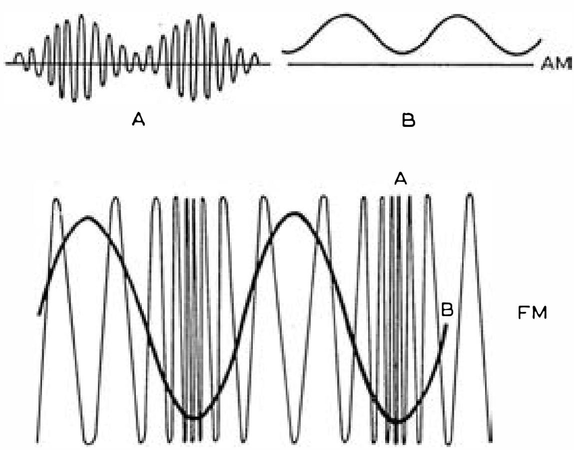


Fig 4. Amplitude modulation and frequency modulation of a radio carrier wave. A, *Modulated carrier.* B, *Audio-frequency signal received.*

(After Briggs and Roberts, *Aerial Handbook*)

The much greater "ether space" available in the VHF broadcasting bands and the limited range of transmissions at these frequencies enables frequency modulation (FM) to be used. This allows distortion-free demodulation of relatively loud audio signals to be carried out. As the programme matter is carried entirely by changes to the instantaneous frequency of the radio frequency carrier, interference can be greatly reduced. Most man-made and natural interfering radio signals are manifested as variations in the carrier amplitude; thus an efficient limiter in an FM receiver can cut out or greatly reduce most of these types of interference. Multi-path varying signals due to reflections of the radio wave from aircraft remain, however, a serious and indeed a growing trouble with FM.

The detailed design of good AM and FM tuners should incorporate various circuit refinements which help to obtain the best results. These will be considered later.

An audio-frequency feed can of course be taken from most radio and TV sets by using the loudspeaker outlet connections, but this is not always the best way of obtaining a high-quality sound output from a set. Sometimes better results may be obtained

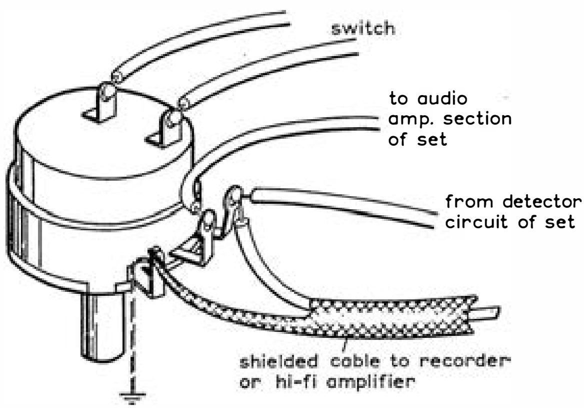


Fig 5. A connection taken from the audio-volume control of a commercial radio set in order to avoid the distortion which may be introduced in the output stages at higher signal levels. A suitable isolating transformer must be used for TV and radio sets used on mains without a mains transformer (ie normal "transformerless" sets) in which the chassis and shields may become "live" if the mains connection is made the wrong way round. It is safest to consult a Hi-Fi dealer if in doubt. (Many TV rental companies will supply isolating transformers by arrangement.)

by making special connections inside the set, so as to take an audio feed from a point ahead of the output stage. In some cases, the signal diode load resistance can be "tapped down," so as to reduce non-linear distortion by improving the diode operating conditions at deep AM modulation excursions. Note that safety precautions are necessary with mains-operated sets. Isolating transformers may be required.

We may summarize by saying that the sideband cutting on an AM set is due largely to the sharply tuned IF (intermediate frequency) filters which have to be introduced so as to make the set selective enough to separate adjacent station channels at

8 kHz intervals. In the service area of a powerful local station during daylight, when distant stations are usually not received owing to the absence of the upper-atmospheric reflecting layer which forms after dark, higher-quality AM sets can be used in which the selectivity is reduced. This enables the more extreme sidebands to be received, with consequent improvement in the high-frequency sound reproduction.

Television sound Bands I and II are AM in Great Britain, but there is no need for the extreme selectivity required on the normal AM medium-wave broadcasting bands, and the IFs can be tuned fairly broadly so as to give a reasonably good high-frequency response. It is necessary, however, to see that the "skirt" of the response on the side adjacent to the vision channel is not so wide as to allow picture-line and frame-synchronizing pulses to break through.

Not all TV sets may be suitable for modification to give a high-quality sound output in this way. A further difficulty may be that the amount of mains HT supply smoothing provided may be adequate for reproduction on the small loudspeaker in a table TV set cabinet, but the fuller bass response afforded by a subsequently added high-quality amplifier and loudspeaker may give intolerable hum, unless the TV set power supply smoothing is considerably improved. Many better-class TV sets can be re-arranged and/or modified to give a really good high-fidelity sound feed to a separate amplifier and loudspeaker system. As we have mentioned, the sound quality obtainable from modern TV studio sound equipment and transmitters is potentially very good and is worth higher-quality reproduction than is possible on the modest budget available for many TV set manufacturers to spend on the audio side of the average table set, having regard to the present highly competitive market conditions.

Stereophonic sound broadcasting is carried out economically on FM by using a sub-carrier on the normal single FM transmitter. AM stereo experiments have usually demanded two separate transmitters and are not regarded as an economic proposition. There have been other suggestions for AM stereo transmissions but these do not seem to have progressed beyond the experimental stage. Regular stereo broadcasts are now put out by a number of stations in Europe and the USA, using the FM sub-carrier technique on single transmitters. Comparatively simple additional decoder circuits can be added to the discriminator circuits of many FM receivers in order to produce two high-quality stereo channels, but the best decoders involve fairly elaborate circuits.

The main carrier system of an FM transmission will give a good single-channel (monophonic) programme on a normal set, where the stereo result is not required. The system is thus termed "compatible."

We have noted that it is unfortunately true that, except for listeners residing in interference-free areas close to transmitters, fairly advanced radio-receiver circuits and good outdoor aerials are required if high-quality reception is desired. Normal AM broadcast receivers have long incorporated features such as band-pass IF tuning filters, image-suppression circuits, automatic volume control (to combat fading), anti-drift oscillators etc. A good modern FM receiver should have many other features, such as precisely designed band-pass tuning filters of adequate bandwidth, an accurately stabilized oscillator and/or automatic frequency control, proper limiting action, adequate capture-ratio and a carefully designed discriminator; preferably also including inter-station muting or "squelch" reduction.

A good set, however, cannot give its proper designed stereo or other performance when fed from an inadequate aerial system, particularly in areas where the field strength is low owing to distance or screening effects etc.

It is important that the buyer of high-fidelity equipment should be able to assess the merits of the various types of commercial receiving equipment on the market. Examples of well-designed aerial systems and receivers are therefore described in the following pages.

Aerial systems

A good broadcast band AM aerial system basically consists of an adequate vertical length of wire extending outdoors to a reasonable height above ground. Any horizontal extension of the vertical wire from its top end adds capacity and thus helps to complete the circuit back to earth, but is not usually a critical part of the aerial design. Many good broadcast (long, medium and short waveband) aerials thus consist of a single 10–15 ft vertical rod or wand erected as high as possible on top of a building or on a mast. The down-lead through a building to the receiving set does not contribute much to signal pick-up and may, in fact, pick up locally generated electrical interference within the building. The

Belling-Lee "antiference" broadcast band aerial was designed as a vertical wand of the above type with a screened down-lead. Impedance matching transformers were used at top and bottom of the down-lead. These transform the relatively high aerial impedance to a lower value, so that the relatively high-capacity down-lead does not cause any appreciable shunt loss. The impedance is stepped up again at the set terminals to a value which suits the normal receiver's input circuits. A coil on a ferrite rod a few inches long can be used to replace a longer wire, particularly in areas of high field strength.

It must be noted that broadcast radio waves are transmitted by radiating wires or elements that produce electromagnetic radiations which are "polarized" either in the vertical or in the horizontal plane, according to the channel requirements allocated by the ruling authorities. This plane of polarization means that a vertically polarized wave will theoretically give the maximum induced signal in a vertically disposed aerial conductor and, similarly, a horizontally polarized wave will give the maximum signal in a horizontal conductor. Most broadcast band aerials are in the form of large vertical masts which radiate vertically polarized waves. There is some evidence that electrical interference often tends to have more vertically polarized power than horizontal and thus horizontally polarized transmitting arrays and receiving aerials give a slightly better signal-to-interference ratio than vertical aerial systems. The question of polarization and orientation of aerials becomes much more critical on TV and FM VHF and UHF transmissions. Mutual interference between adjacent TV stations can sometimes be minimized by adopting vertical polarization for one station and horizontal polarization for a neighbouring station. Owing to the "capture-effect" on FM, interstation interference between adjacent FM service areas is not usually a problem and horizontal polarization is almost universal for FM broadcasting. The VHF and UHF aerials differ from the long-, medium- and short-wave broadcasting single-wire aerials described, in that basically they consist of symmetrical "dipole" rods and precisely spaced adjacent reflector and/or director rods. The feed connection to the set is taken from the adjacent centre connections of the two dipole rods. The simple dipole is "bidirectional" in one plane, with a "figure-of-eight" polar pick-up pattern. The addition of front director and rear reflector rods spaced correctly from the signal dipole make the aerial more sharply unidirectional, according to the number and spacing of the subsidiary rods. Dipoles are fairly sharply tuned "wave-resonant" arrays, which give a fair measure of selectivity to the transmission for which they are designed. The length of the rods determines the resonant frequency and the diameter controls the bandwidth. A structurally strong aluminium alloy rod of about $\frac{3}{8}$ in. diameter usually gives an adequate TV bandwidth. The overall length of a dipole is given by the following formula—

$$L = \lambda \times 1{\cdot}57 = \frac{474}{f}\ \text{ft approx}$$

where λ = radio wavelength in metres
f = radio frequency in megahertz.

Figs 6*a* and 6*b* give dipole reflector and director details for British FM broadcast channels, and show one type of combined AM/FM array.

Thus if more than one channel is to be received by tuning the receiver, either different arrays or a composite multiple array must be used. The array must also point towards the desired station and, particularly in the USA, elaborate remotely controlled rotatable aerial systems are used. Mast-head transistorized RF amplifiers can also be used in order to boost the signal from the aerial before it traverses the down-lead, in which a certain loss of signal strength inevitably occurs and which may pick up some electrical interference. The down-lead cable from a dipole is of either the 75-ohm screened coaxial or the screened low-capacity twin-lead type. In either case the spacing and insulation of the conductor(s) and/or shield is critical and must be correctly designed. The input circuit of the receiving set must also be correctly designed to match the cable impedance, and the cable must not be "tapped" or looped on to provide another outlet without the use of special matching coupling units.

We see from the above considerations that, for most high-quality enthusiasts, there is no substitute for a well-designed and properly sited aerial system. The results which can be attained and the type of aerial required in any given district can vary considerably, particularly for VHF and UHF transmissions. Advice from local radio amateurs and Hi-Fi dealers may be very helpful in this respect.

In general it pays to buy a good-grade commercial aerial and

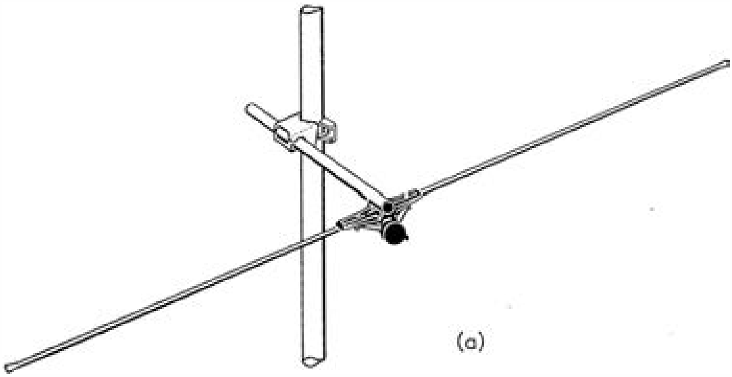


Fig 6(a). *A simple half-wave dipole VHF FM aerial for use in areas of moderate to high signal strength. The FM band II extends from* 87·5 *to* 100 *megahertz and the total width over the rods is about* 5 ft. *Standard low-loss* 75 *ohm coaxial cable is normally used but the equivalent shielded twin may also be used in some cases.*

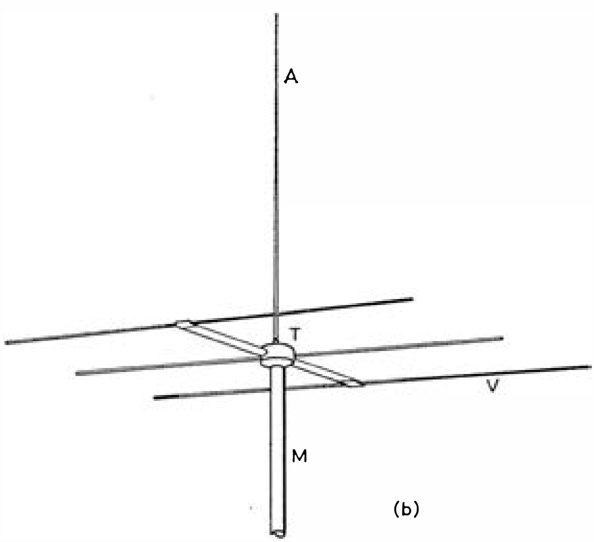


Fig 6(b). *A combined AM/FM aerial array. A is a* 10 *ft conducting fibreglass wand AM rod coupled to the* 75 *ohm coaxial cable by a transformer* T. V *is a three-element FM array also coupled to the cable.*

have it properly erected, using, needless to say, the correct specified down-lead and plug connector at the set input terminal. Aerial manufacturers will often provide instructions and technical advice in cases of difficulty.

The question of the risk of damage by lightning may be considered, particularly in very exposed sites. Modern aerials are not unduly susceptible to lightning strikes, but some small risk still

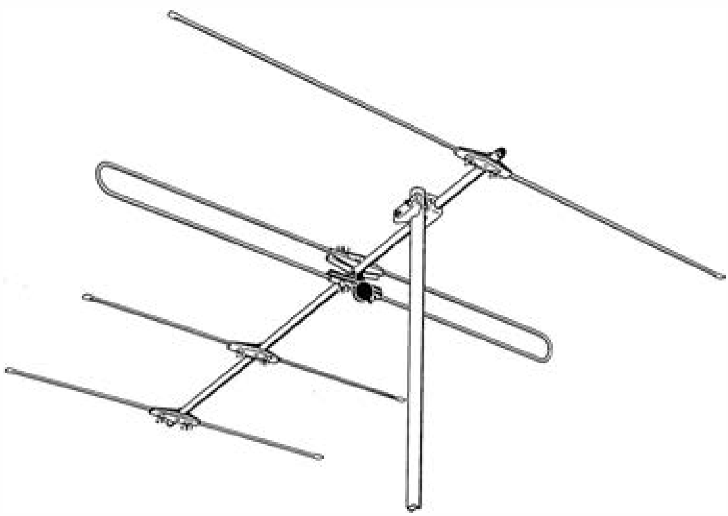


Fig 7. An efficient folded dipole with a reflector and directors, giving a suitably directional response for areas of lower signal strength or where excessive interference or multi-path propagation problems exist.

remains. It is thought that transistor mast amplifiers may be damaged by comparatively small lightning discharges which may not cause any other noticeable damage. It is often possible to insure aerial systems against lightning damage for a very small premium, or the risk may be covered in comprehensive domestic insurance policies issued in some cases. The possibility of damage by lightning discharges has caused anxiety in some districts as to the likelihood of damage to input transistors and masthead transistor booster amplifiers. In practice it seems that the isolating effect of input transformers and the series inductance of leads is sufficient to avoid such damage except in particularly unfortunate cases.

Tuner principles

The circuit line-up of a good FM tuner might be along the following lines—

1 The 75-ohm coaxial or balanced twin low-loss feeder from the aerial array is accurately matched to low-noise input stage, and further tuned band-pass stages provide adequate selectivity ahead of the mixer/oscillator stage, so as to provide adequate rejection of unwanted incoming signals, particularly those of the IF frequency (10·7 MHz) and the IF image frequencies corresponding to the signal being tuned in. The oscillator is normally tuned 10·7 MHz above the incoming signal. The image frequency will be 10·7 MHz above or below the oscillator and thus can pass into the IF amplifier unless attenuated.

2 The stability and accuracy of the oscillator are important, whether the circuit is of the fixed-tuned channel or the continuously variable tuning type. Oscillator temperature drift may be compensated by the adoption of negative temperature coefficient capacitors, by crystal control, by the use of a quarter-wavelength tuned line or "trough," or by the use of automatic frequency control (AFC) provided by a reactance tube or diode varactors whose characteristics are varied by the discriminator output in such a manner as to bring the oscillator accurately back on tune in the event of any oscillator drift occurring. Tuning of the RF and oscillator stages is accomplished by moving low-loss sintered ferrite cores axially inside the tuning coils, ie permeability tuning, by ganged tuning capacitors or by varying the bias on varactor tuning devices.

3 Two or three IF stages coupled by band-pass filters centred on 10·7 MHz follow the mixer stage. The first stage may have variable μ characteristics so as to enable a measure of automatic gain control to be applied. The last IF stage should be arranged to saturate sharply above a certain threshold signal so as to give an effective limiting action. A good limiter stage is a very important attribute of an efficient FM tuner, as all traces of unwanted AM signals, including most types of interference must be eliminated.

4 A carefully designed discriminator circuit of the Foster-Seeley phase-discrimination type or the ratio-detector type will produce an undistorted audio signal from the constant-amplitude frequency-modulated signal fed to it from the limiter.

The carrier frequency components are eliminated from the output, but a DC signal is produced which drops to zero when the receiver is exactly on tune. This DC signal has a magnitude proportional to the amount of mistune and the polarity depends on whether the receiver is tuned above or below the correct frequency. This DC signal from the discriminator can be extracted and used to operate a tuning indicator, as well as controlling an AFC reactance device. The polarity of the DC signal is used to operate some excellent forms of precise tuning indicator. Normal tuning indicators may only be a DC meter indicating a minimum "dip" for exact tuning. These do not indicate the direction in which the tuning control needs adjusting in order to restore or achieve correct tuning and it is necessary to de-tune in order to check that a minimum dip is indicated.

The AF response is given a "pre-emphasized" rising characteristic at an FM transmitter in order to improve further the signal-to-noise ratio, as the high noise frequencies can be attentuated by an equivalent amount after the receiver discriminator. These simple RC series equalizing combinations are specified most easily in terms of their time constant, ie the time in microseconds for a given charge on a capacitor to fall to *I*/*N* times the initial value (*N* is the natural logarithm base value 2·71 . . .). European and British FM transmissions use a 50-microsecond (μs) pre-emphasis. This fact should be remembered when considering imported sets.

Some designers used to prefer to use vacuum tubes (valves) throughout FM tuners, some used a mixture of tubes and semi-conductors, but the majority now use semi-conductor (solid-state) devices exclusively. Good designs may be achieved in each case and it is difficult to do other than to judge a product on its merits. New pulse-counting detectors are used.

5 Inter-signal quieting or squelch circuits are used as it is desirable to prevent or inhibit the action of AVC for inputs below the usable minimum, so as to avoid an up-rush of noise when tuning between signals. The "squelch" circuits may provide an amplitude threshold or "delay" (so called) in the AVC voltage generating stage which, in effect, provides an input "gating" effect which prevents the receiver coming into action unless the input signal exceeds a certain minimum value. This value ideally should represent the point at which limiting commences.

Typical high-quality FM tuner quantitative specification figures

Frequency range: 88 MHz to 108 MHz (Band II).
Sensitivity: 2 microvolts for 100 millivolts output and 20 dB quieting.

6 microvolts for full quieting (40 dB).

Audio response: Flat to ± 1 dB from 30 c/s to 20 kHz (allowing for the standard 50 microsecond de-emphasis characteristic).

Limiting characteristic: limiting commences at inputs above 2 microvolts.

Constant output level produced for inputs above 3 microvolts.

AFC characteristics: signal held constant to ± 1 dB for tuning centre frequency changes of ± 120 kHz approximately.

Image and IF rejection: 40 dB and 80 dB respectively.

Capture ratio: 3 dB.

AM suppression: 35 dB.

Stereo-channel separation: better than 35 dB at 1 kHz.

Fig 8 shows a high-quality FM tuner which should exceed this specification and will give the best results under adverse conditions. Local conditions and the number of FM signals receivable to some extent determine the elaboration of the tuner which you will wish to use. Again, your Hi-Fi dealer can often advise you on the most suitable tuner.

FM stereo multiplex receiver units

We have mentioned previously that the standard method of stereo broadcasting now used for radiating programmes in Britain, the USA and several European countries, notably France, Germany and Holland, is the FM "pilot tone" system in which the original two stereo sound channels (A and B) are subjected to a multiplexing or encoding process by which the instantaneous sum ($A + B$) is in effect transmitted by frequency modulation of the main carrier, whilst the instantaneous difference ($A - B$) is transmitted by a sub-carrier, in addition to a supersonic pilot tone of 19 kHz. Standard monophonic FM receivers produce the sum ($A + B$) of the left- and right-hand stereo

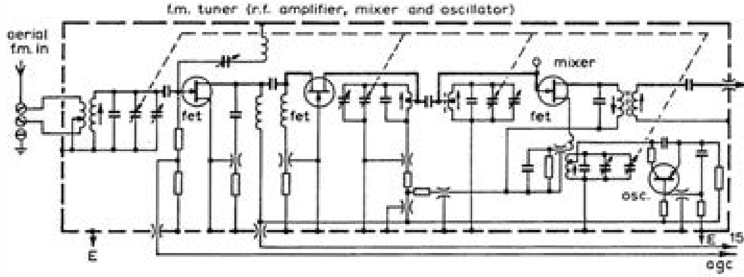


Fig 8. *The schematic of an advanced modern stereo FM tuner* (continued on the following pages). *This includes a number of advanced design features which may be essential to obtain the best results in areas of moderate or low field strength. Simpler tuners may be used where the field strength is high, but there is a trend towards the use of sophisticated tuners of this type when the best stereo results are desired. The RF and mixer stages use three low-noise field effect transistors. Tuning is by a ganged variable capacitor.*

channels at the discriminator output without any circuit modifications being needed. If the stereo microphone and mixing technique is skilfully arranged to this end, the sum signal can usually be made to give an acceptable high-quality monophonic sound signal. There may, however, be cases in which the need to produce the best possible stereo sound "picture" may mean some sacrifice of the monophonic sum signal acoustic "balance." This may possibly be justified on the grounds that listeners who want the best reproduction will be prepared to invest in stereo equipment and that the need to produce the best possible stereo result must be the paramount requirement. Generally, a good FM receiver will have sufficient band-width to accommodate the pilot tone and the difference channel in addition to the normal main carrier. An add-on decoder unit can usually be fitted on to a suitably designed normal discriminator circuit in order to extract the two original left- and right-hand stereo *A* or *B* signals. Most of the leading manufacturers of tuners allow chassis space or otherwise make provision for the easy addition of a stereo decoder unit which they have available. Decoder circuits have also been published by manufacturers and technical journals.

Fig 8 gives a tuner circuit using advanced modern design concepts. Many such designs include an electronically operated indicator which shows when a stereo programme is being received from any given transmitter. The non-linear distortion produced

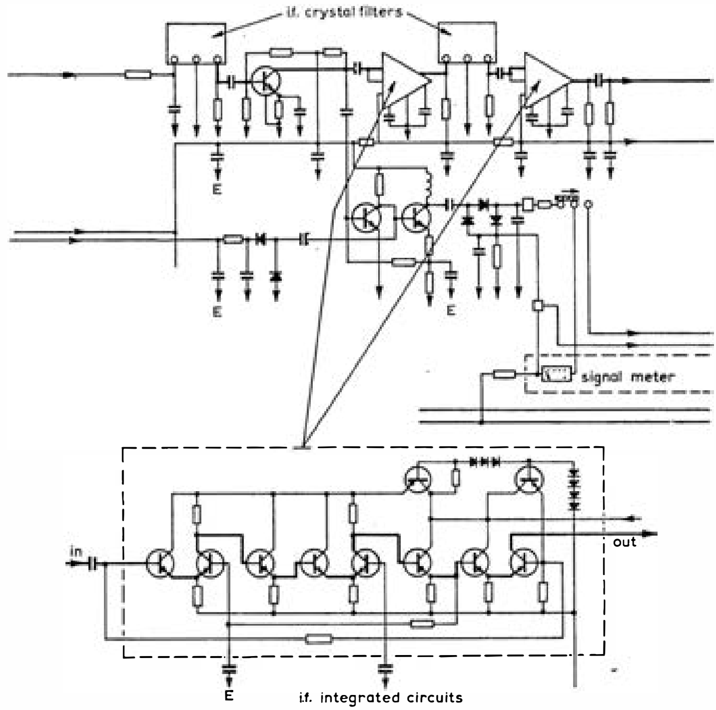
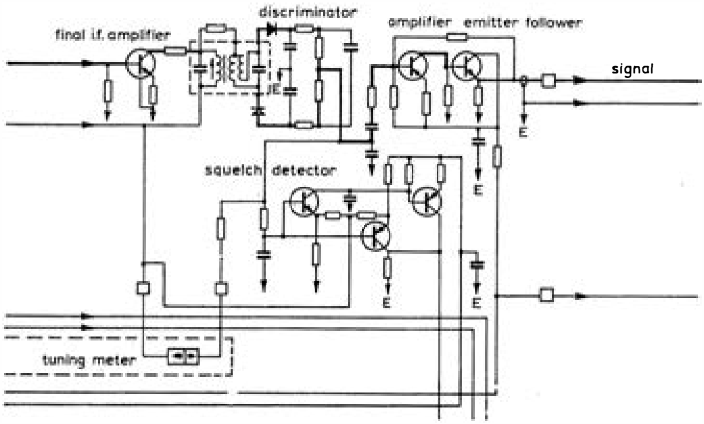


Fig 8 (contd.). *The IF stages use crystal band pass filters which give a stable wide-range response with high rejection of unwanted signals. Good limiting characteristics are obtained. An advanced discriminator and inter-station noise-reducing squelch circuits are used.*

by the encoding and decoding process is claimed to be substantially as good as that from the best monophonic broadcasts, and the stereo channel separation is usually between 35 and 45 dB. This is slightly worse than is achieved by the best stereo tape-recordings, but is substantially better than that of most stereo gramophone pick-ups.

From the broadcaster's viewpoint, the addition of the sub-carrier does not normally represent a very difficult modification of the transmitter, but some of the available power is diverted into the sub-carrier and thus the effective service area of the transmitter may be reduced. This is important if a national network is being planned, as extra transmitters may be needed to maintain the same coverage. The main complication is undoubtedly in the



radically different and much more demanding microphone studio and line transmission techniques which good stereo programmes demand. Stereo tapes and records do, however, provide a convenient source of programme if a sufficiently large library of recordings is available.

The two decoder outputs are, of course, fed to two separate stereo amplifiers and loudspeakers, which must be balanced in the usual way. The advantage of a panel indicator which shows when a stereo signal is tuned in is that the decoder and amplifier channels can then be switched to the stereo position. Otherwise, one may continue to listen to the $(A + B)$ monophonic version of the stereo broadcast. This may be desirable in marginal cases where noise is too severe for the full stereo result to be enjoyed.

Special characteristics of frequency modulation

We have seen that one of the great advantages of FM is that the effects of interference are greatly reduced. It is fairly easy to see that normal forms of AM electrical noise and static interference are greatly reduced by FM peak limiting. It is not so easy to see how the effects of adjacent channel station interference are also greatly reduced by FM, because of the so-called "capture effect," so that a slightly stronger signal can "capture" the receiver and reduce the weaker competing signal to complete inaudibility, though it may still cause some distortion of the "winning signal."

With AM broadcasting reception, an adjacent channel signal of 1/100 of the strength (−40 dB) of the desired signal can still

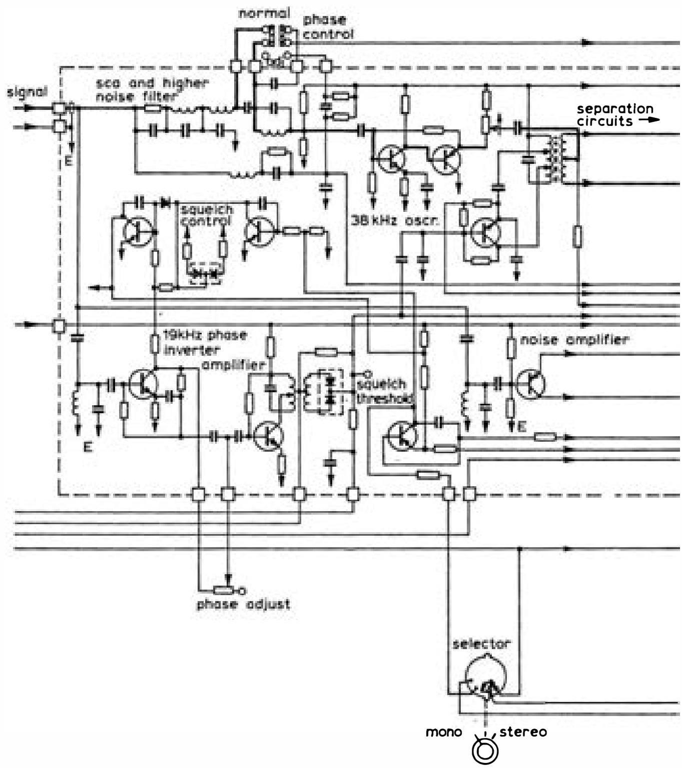
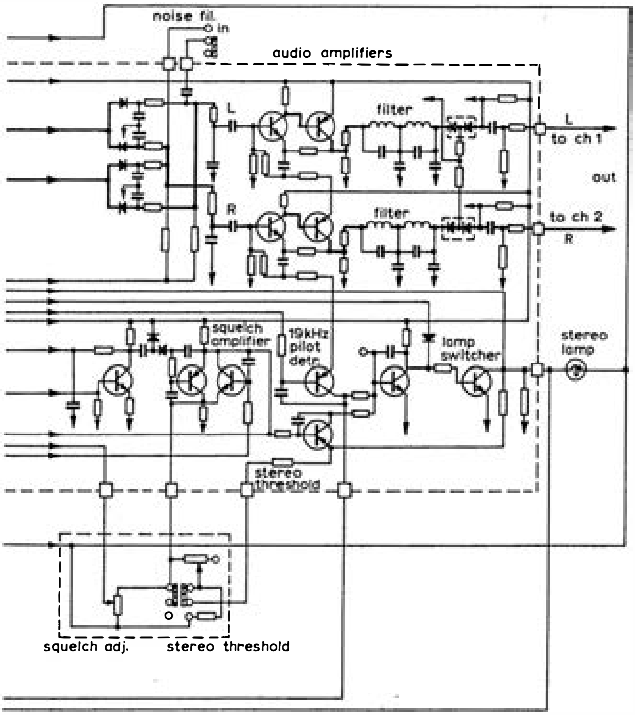


Fig 8 (contd.). *The decoder circuits produce L and R channel signals with low distortion and cross-talk levels. The sub-carrier frequency and harmonics are effectively suppressed from the output by suitable filter circuits.* (Contd. on facing page.)

give rise to an annoying inter-carrier beat frequency whistle, whereas with FM, a ratio of 2/1 or less in relative signal strength can be sufficient for full capture by the stronger signal. If two transmissions pass through the RF, mixer and IF stages of a FM set, as they will if they are of nearly the same frequency, the resultant RF wave will generally be both frequency- and amplitude-modulated. A good limiter will remove the amplitude modulation, but the phase modulation remains. However, it can be shown that the resultant vector becomes, in effect, frequency-modulated at the frequency of the stronger of the two incoming transmissions, even though the difference in amplitude is quite small. Thus the receiver locks on to the stronger signal, which is said to capture



the set. Owing to the selectivity of the RF and other stages and the relatively wide-frequency channel-spaced allocations normally given for stations within normal mutual range, it is likely that adjacent transmissions will produce much greater ratios of strength than 2/1 when a particular station is tuned in. However, the effectiveness of the capture action and the normally fairly short range of VHF transmissions mean that a number of stations can be operated at the same frequency provided that they are out of mutual range. Freak reflections from the ionosphere do, however, sometimes bring a distant transmission in at sufficient strength momentarily to capture a normally much stronger signal.

FM stereo decoder circuits and specifications

The unit may be electronically or mechanically switched on to the output of the discriminator without the de-emphasis circuits. A high-input impedance circuit is provided so as not to load the discriminator. In some cases the emitter-follower output from this stage drives a quadruple diode ring modulator via a low-pass filter designed to remove the 19 kHz sub-carrier frequency. The circuit is also designed to provide overall phase compensation of the signal. The diode ring modulator is "switched" by the output of a synchronized 38 kHz oscillator. A tuned circuit in the input stage feeds the 19 kHz received pilot tone via an amplifier and rectifier in order automatically to bias the diode signal switches to the "on" state for stereo. The 38 kHz pulses produced synchronize the oscillator switching signal correctly with the pilot tone. The left and right channels are recovered directly, the correct balance between sum and difference signals being automatically restored. Output filters prevent any sub-carrier leak to amplifiers. Other decoder circuits are also employed, using different circuitry.

Typical outline specification of decoder performance—

De-emphasis: 50 μs (European).
75 μs (N. America).

Output: 100 microvolts at 30 per cent maximum excursion.

Stereo cross-talk: better than 30 dB at 1 kHz between channels.

Pilot tone suppression: at least 30 to 40 dB suppression of 19 kHz pilot tone and 38 kHz switching tone in output. This is important as harmonics may cause whistles with tape-recorder bias frequencies.

High-quality AM tuner design

We have noted that, because of the close channel spacing, the general overcrowding and the large distances at which stations can be received, it is necessary for an AM radio receiver to be provided with a very high degree of selectivity when conditions make this necessary. This, unfortunately, leads to a severe loss of high frequencies because of the cutting of the outer side bands. When conditions are good and no severe inter-station interference is present, the band-width may be increased. Thus variable selectivity is highly desirable. This can usually be satisfactorily obtained by switching the IF band-pass filters, using tertiary or auxiliary windings designed to widen the pass-band.

AM reception is also much more open to noise interference and no effective form of noise-limiter can be adopted, unlike FM where very effective noise-limiting circuits can be used in the receiver. A switchable sharply tuned "dip" circuit tuned to around 8 kHz is also desirable to remove inter-channel "beat" whistle on AM.

For listeners residing at a distance from the desired stations and in areas where an appreciable amount of man-made electrical interference exists, a good high AM aerial rod is a valuable initial investment. This and other forms of aerial system have been dealt with in the section on aerials. The input circuits of an AM superhet receiver must have a good degree of selectivity, particularly at the IF frequency (470 kHz), as anything received at this frequency will otherwise pass straight into the IF amplifier. The "image" frequencies at ± 470 kHz from the oscillator frequency must also be attenuated.

The general operating principles of the superhet receiver and the detailed design of the various stages, filters etc is dealt with in comprehensive books and articles to which the reader may be referred.[4,6] The input-tuned circuits are usually combined with a low-noise RF input stage which is tuned by a ganged variable capacitor or a variable permeability system in synchronism with the oscillator, whose circuits have, of course, to be tuned exactly 470 kHz above or below the frequency of the incoming signal, in order to transfer the audio modulation to a carrier having the 470 kHz IF frequency. The production of this IF "beat" or difference frequency depends on the non-linear RF transfer characteristic of the mixer stage. This is a further reason for the use of a good RF stage with adequate selectivity, as this attenuates unwanted signals ahead of the mixer and reduces the chance of cross-modulation of wanted and unwanted signals in the mixer. In addition, the mixing process is more efficiently carried out and a better noise ratio is achieved if the wanted signal is raised to a suitable level by the RF amplifier.

The RF mixer and IF stages are used to give automatic gain control (AGC) by the use of variable-μ valves or alternative semi-conductor circuits. AGC is essential to prevent violent changes in the reproduced level due to fading and differing station strengths when tuning. AGC is generally derived at the detector stage by the use of a second diode with suitable RF decoupling stages. The impedance loading placed on the signal diode in an AM detector circuit is extremely critical if severe non-linear distortion of heavily modulated signals is to be avoided. The important criterion is the ratio of the AC and DC impedance

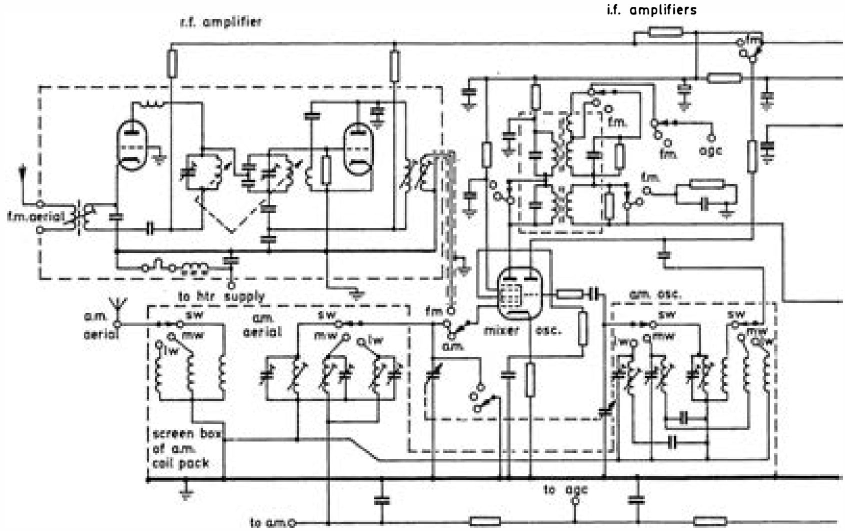
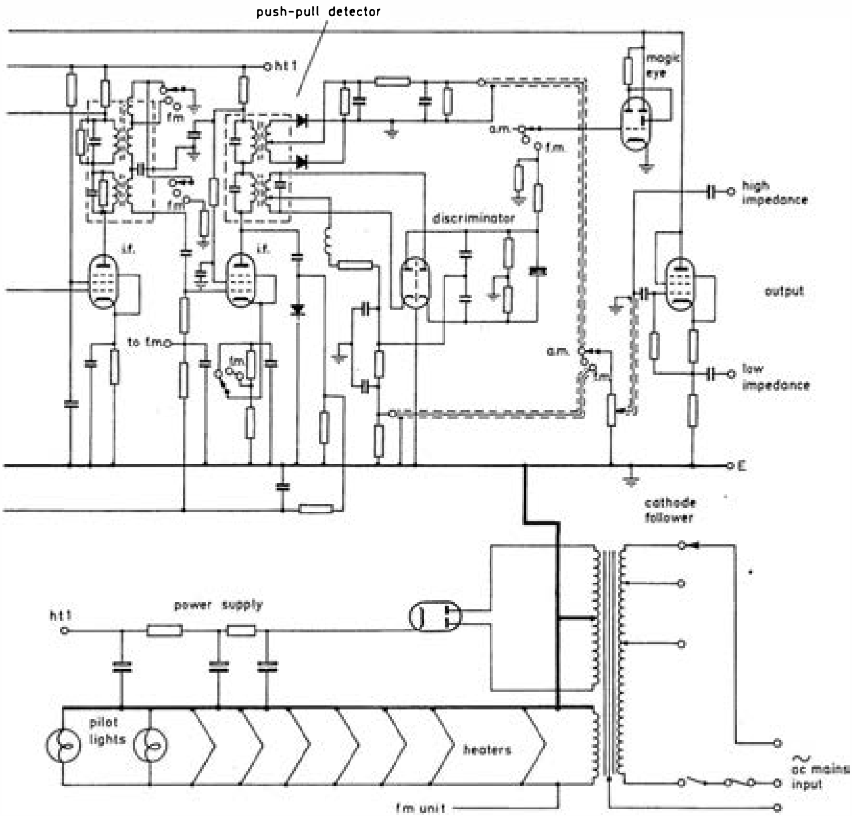


Fig 9. A modern high-quality valve AM and FM tuner functional schematic. Points to note are high selectivity in an RF stage ahead of the mixer, variable bandwidth IF transformers with tertiary windings and low-distortion diode detection system. These features, respectively, eliminate input image frequencies, whistles etc, reduce sideband cutting on strong local signals and enable heavily modulated loud signals to be demodulated without distortion. (Contd. on facing page.)

loading which follows the signal diode. Thus the value of the coupling components between the diode and the first AF stage and the input loading due to an AGC diode and its circuits may have to be carefully controlled. The effect of tuning indicator circuits may also have to be considered if these are effectively coupled at this point. A most effective way to achieve a high AC/DC ratio of diode load is to "tap down" the load by taking the AF signals output from a small fraction of the resistive load (25 per cent or less), thus "diluting" the effect on the diode of the AC load due to coupling capacitors etc. Push-pull full-wave diode detector circuits are also used.

AF noise suppressors

Various attempts have been made to eliminate or reduce impulsive noises which get through to the AF circuits. Generally these take the form of a preset peak-clipper designed to prevent the crests of large interference "spikes" being reproduced. Another system



is a "rate of rise" clipper designed to come into operation for a millisecond or so to cut out a "spike" with a very sharp wavefront. It is difficult to make either of these systems effective without affecting the quality of reproduction of music containing transients of the same nature as impulsive interference. The use of these circuits has thus largely been restricted to communications and short-wave receivers where high-quality reproduction is a secondary consideration.

Specification of a typical high-quality AM tuner

Tuning range: Long wave 2070–800 metres.
Medium waves 588–185 metres.
Short waves 1 2·2–6·6 MHz.
2 5·8–18·5 MHz.

AF response (*narrow band*): 5 kHz bridged T low-pass filter with 18 dB/octave minimum cut-off above 5 kHz. Wide band, minimum loss up to over 12 kHz (6 kHz and 12 kHz 3 dB points respectively).

Dip circuit tuned to eliminate 8 kHz with an attenuation of at least 20 dB. Loss at 7 kHz preferably not to be more than 6 dB. Output level 100 millivolts at 30 per cent modulation into 50 K ohms (approximately).

The tuning indicator may be automatically switched out for wide band response, as it will not normally indicate the centre point of a wide-band tuner and may thus be misleading. Delayed AGC is provided. The delay means that an initial offset or gating characteristic is provided so that noise and signals below a certain level do not operate the AGC.

Sensitivity: Better than 20 microvolts for 10 dB signal to noise ratio.
Better than 100 microvolts for 40 dB signal to noise ratio.

3

Records and equipment

The popularity of gramophone records as a source of home entertainment has always been considerable and the advent of long-playing microgroove records pressed from low-noise unbreakable vinyl plastic gave them a further sharp fillip. The gramophone record system has been the subject of very extensive development and at their best the results achieved are hard to rival by any other recording medium. Gramophone records are also very convenient to handle and store and they permit of readily locating any part of a recording by the use of a comparatively simple micrometer type of groove-locator operating on the pick-up arm.

Automatic record-players can now provide very high-quality reproduction and can be loaded up with long-playing records to give hours of entertainment of durations in excess of that obtainable from similar-quality tape-players. These, of course, may involve tedious rewinding, spooling and tape-threading procedures which have no parallel in gramophone disc equipment.

Gramophone records now, of course, provide two-channel stereo by modulating each groove wall independently and by employing a dual type of pick-up with mutually perpendicular sensing elements actuated by a single stylus. Such systems can now be made to work very well, but the magnetic tape system has an advantage over the gramophone record for stereo or multichannel recording as regards the number of channels and their mutual separation.

Recording and reproducing systems place very stringent requirements on the mechanical transport system which moves the recording medium past the transducers and a very high order of long- and short-term speed accuracy is essential. Long-term speed inaccuracy means that playback and recording speeds may differ. This can introduce errors of musical pitch and also alters the duration of a programme, which is sometimes important. Short-term variations appear as "wow" if they have a sub-audible cyclical repetition rate, or as "flutter" if the cyclical rate is in the audible band. As an indication of the stringency of the requirements, it may be noted that a frequency variation of 0·05 per cent wow may be noticeable at a critical part of the frequency band, such as 2 kHz.

To achieve this speed accuracy means not only that mechanical parts must be very accurately made to size but also that friction and vibration must be very low. This implies near-perfect shafts, bearings guides and pivots in all parts of the motors, drive and transport mechanisms generally. It is possible to achieve this sort of mechanical perfection by meticulous mechanical fitting and the use of very high-grade machine tools and gauging equipment. For quantity production of gramophone turntables and tape-decks, individual fitting of this type obviously is not possible, and carefully planned high-grade machining and testing equipment have to be installed.

From a user's point of view, it is obviously necessary to treat mechanisms with great care so as to avoid any possibility of destroying the accuracy by damaging surfaces, bending turntables or shafts etc. In particular, rubber or synthetic-rubber-covered drive wheels must not be left in engagement with shafts etc for long periods at rest in case they take a permanent "set," with disastrous results on the accuracy of their driving properties.

It is also very important to mount motor- or tape-decks in the manner recommended by the makers, using the exact fixings supplied, so as to avoid any risk of distorting frames or transmitting external shocks or vibrations to the turntable or deck. In some cases, the chassis or mounting plate is floated on very flexible springs for normal use. These are by-passed by rigid fixings or "shipping bolts" when the equipment is to be moved, so as to avoid damage to the springs or equipment when in transit.

Most modern units are designed to use permanently-oil-retaining bearings, PTFE low-friction bushes and other types of low-friction pivots, and thus may not need attention. If any lubrication is called for, it should be carried out strictly according to the makers' recommendations. The old adage of "just enough in the right place" should be observed. An excess of oil inevitably collects dust, thereby violating the other important precept, which is to keep all parts of the mechanism as clean as possible and free from all the dirt, dust and fluff etc which accumulate and may adhere to both fixed and moving parts.

We will now consider gramophone reproducing turntables and pick-ups in greater detail. Tape-decks will be dealt with later (page 77).

Design and use of turntable and pick-up units

MOUNTING OF TURNTABLES

The mounting and aesthetic integration of gramophone turntables will be dealt with in the section dealing with the arrangement of cabinets and complete systems. There are a number of general points to note. Firstly, the turntable should be accommodated so as to give easy and convenient access for changing records. Thus it should be at a convenient height and, if it is in a dark place, artificial light should be provided. Good frontal access is desirable via well-fitting doors without the need to slide or roll the turntable mounting board forward. In designing the cabinet a basic decision may have to be taken as to whether standing or seated operation is desired.

It is important to isolate the turntable from mechanical shocks due to foot-falls and the opening and closing of cabinet doors etc. Any appreciable transmitted shock may cause the pick-up to "jump" in the groove, with a risk of damage. If the loudspeaker is mounted in the same cabinet, feedback oscillations may occur at a low frequency through the pick-up functioning as a vibration-sensitive transducer. It used to be desirable to confine the pick-up in an acoustically damped enclosure with stout and leakproof doors when records are being played, but there is now virtually no high-frequency acoustic "needle-chatter" radiated from the record face on highly modulated grooves. A great deal of this was due to the vertical "pinch-effect" arising from the changing angle of a "Vee" groove section during the recording cycle, giving a double frequency (second harmonic) lift to the pick-up stylus. This represents distortion, and it is well established that small amounts of audible high-frequency distortion can give rise to annoyance far beyond that which might be expected from their acoustic sound level. It is still important to mount the pick-up as far as possible away from any mains transformers, smoothing chokes etc, or mains hum may be induced, particularly in the case of variable-reluctance or moving-coil pick-ups. The turntable base-plate should also be accurately level when in its final mounted operating position. Any error in level will cause side thrust in the pick-up and may induce serious groove-tracking errors and consequent distortion. Small circular spirit levels are available for motor plate mounting.

In the case of turntables used on ships, aircraft etc a carefully designed gyroscopic self-levelling base-plate may be used. If the pick-up arm is designed to use a pivot system balanced in both horizontal and vertical planes, the reproducer will be much less sensitive to levelling errors or variations.

TYPES OF TURNTABLE UNIT

These are basically divided into single-record-players and auto-record-changers, which will take up to ten or more long-playing records. Auto-changers are necessarily more complicated and costly than the equivalent single-record-players. Many LP records require changing only every fifteen minutes and, in the case of all but some very specialized products, auto-record-players only play one side of a pile of records, the pile having to be turned over to play the other sides. Thus, unless it is particularly desired to have fairly long periods of music from records without the trouble of record changing, a single-record-player may be the "best buy" for many amateurs today. The cheapest mass-produced turntables necessarily have limitations with regard to speed constancy, vibration and total running life before overhaul. The best professional "transcription" turntables may cost from £40 upwards but there is no doubt that a really good-quality unit is a worthwhile investment for the serious record quality-lover. However, there are now some good-quality single-record-playing turntables available at about half this price.

Good-grade auto-record-changer players are now produced at about the same price as the best single turntables. Generally speaking, for a given quality of turntable and drive, an auto-changer-player will cost one and a half to twice as much as the equivalent single-player.

Apart from the facilities offered such as the various playing speeds, fine adjustment of speed etc, and the performance specification (which can be authenticated by independent reviewers), one of the main attributes to look for in a turntable unit is solidity. A massive well-balanced turntable acts as a fly-wheel and helps to smooth out speed fluctuations, and a rigid base-plate will not be likely to have sharp mechanical resonances. These might be excited by vibration, shocks or even pick-up tracking reaction in the case of flimsy turntable base-plates. The result may be increased background noise coloured by the base-plate resonances.

Fig 10. *A modern high-quality turntable unit and plinth assembly. A very high-precision lightweight arm gives low pivot friction and provides accurate side-thrust compensation, enabling ultra-low-weight high-compliance cartridges to be used.*

Modern turntable and record-player units from established makers give very good results, approaching those obtained from the most expensive studio transcription turntables, and thus represent excellent value for money. High-quality pick-up arms capable of doing justice to modern low-tracking-weight cartridges are also provided in many cases.

The manufacture of automatic record-changers and -players*

The automatic record-changer dates from around 1930 and was invented to save the listener from having to leave his chair every two or three minutes to change a record. Records were then played

* Information for this section was kindly provided by Mr E. W. Mortimer and Messrs Garrard Ltd.

at only one speed (78 revs/min) and with the relatively coarse groove then used the longest uninterrupted playing time available was about five minutes. Nowadays, long-playing records have popularized the semi-automatic single-record-player, with mechanized pick-up lowering and auto-stop.

Since 1930 the art of recording and reproduction from gramophone records has changed so much that it has revolutionized the whole concept of record-changer design and production. The modern record-changer or -player is a precision instrument, and this section describes some of the intricate operations used in its manufacture to provide the means for achieving the high standard of sound reproduction which is expected today. It must be able to rotate records at any one of three speeds, $33\frac{1}{3}$, 45 or 78 revs/min, though the latter may become obsolete. It must handle three record sizes, 7-in., 10-in., or 12-in., and be able to sense which diameter record is to be played so that the pick-up can lower in the correct place. It must sense when the end of a record has been reached so that the pick-up can rise and move back to allow the next record to drop and be played. It must also be able to tell when the last record has been played so that it can switch itself off. In addition it should be able to play single records manually and have facilities for rejecting the record being played. A "pause" control is also useful for cueing.

The mechanical handling of gramophone records in order to play a number of them consecutively, although the basis of the automatic record-changer, is not, however, the principal criterion of its quality, which is the fidelity of the sound given by the system.

Among the many factors adversely affecting the quality of the reproduction obtained from gramophone records, three principal ones are wow, flutter and rumble, all of which are inherent to some degree in every rotating mechanical device used for sound recording and reproduction.

Wow is the term used to describe speed variations superimposed on the reproduction below about twenty Hertz and flutter occurs above twenty Hertz. Rumble is random unwanted noise below about 500 Hertz, often extending to sub-audible frequencies. These three factors originate in the mechanism used to rotate the record, and to reduce them to a practical minimum on a quantity-production basis calls for highly specialized engineering techniques, which are worthy of mention.

The high degree of amplification used between the pick-up and speaker will reproduce the slightest bearing noise as rumble and any imperfections in the driving system appear as wow and flutter. For example, a journal or thrust bearing normally considered good engineering practice would be too noisy in a mechanism used for sound-reproducing purposes, and electric motors used to drive the mechanism and turntables have to be made to much higher standards than would be required for purposes other than for the reproduction of sound.

The first essential is to make sure that all bearings are perfectly free-running, with the minimum clearance, and run with the minimum of noise or vibration. Whenever possible, oil-impregnated sintered bronze bearings are used, as these show little signs of wear over long periods of use and rarely need lubrication. The bearings need to be specially processed to meet the standard required. The bearing holes are accurately sized and finished by rotary burnishing and the self-aligning bearings as used in the electric motors are also spherically turned on the outside to make sure that the hole is true with the bearing and shaft location. Shafts which run in these bearings are specially finished, most of the turntable spindles and all rotor shafts being hardened, tempered, ground, lapped and superfinished to one micro-inch surface finish.

The clearance between the spindle and bearing bore must be very small to avoid noise which would show itself as rumble. The maximum clearance allowable in the bearings of the electric motor is $\pm 0{\cdot}0001$ in. and every motor may be run continuously for three hours on a running rack to bed-in the bearings to make sure that the shaft rotates freely, before being tested and assembled to a record-changer.

Special precautions are taken in the manufacture of the component parts of the turntable driving system to avoid rejection after assembly. The inside rim of the turntable is driven by a rubber intermediate wheel (see Fig 11) which in turn is driven by the motor pulley. The position of the rubber wheel and the angle of its tension spring must be carefully chosen so that the rubber wheel wedges just sufficiently between the motor pulley and turntable rim to drive without any trace of slip, but it must not wedge enough to cause undue loss of power and, incidentally, to increase rumble.

Should the motor pulley run more than 0·0002 in. out of true on any of its four steps it will cause flutter. This pulley must be interchangeable to accommodate differences in motor speed on various supply frequencies, and the required accuracy of the fit on the motor shaft is achieved by the use of special boring and turning machines developed for the purpose. The bore of the pulley may be "blind," owing to the fact that the step diameter for the speed of $16\frac{2}{3}$ is smaller than that of the motor shaft on which the pulley fits. The speed of $16\frac{2}{3}$ revs/min is used for certain special recordings such as talking books.

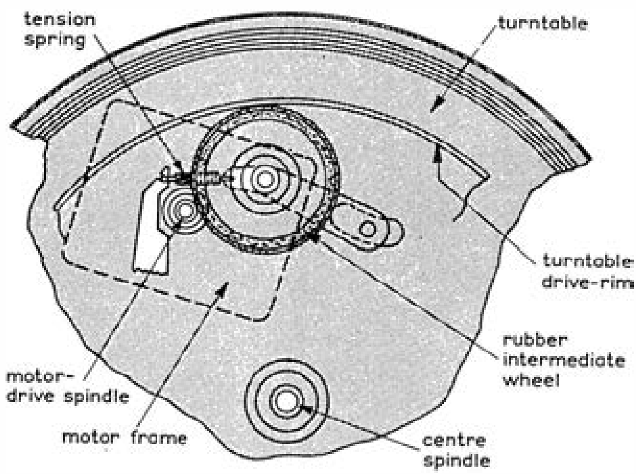


Fig 11. *An intermediate-wheel turntable rim-drive system. The angular relationship with the stepped motor pulley must be correct and the rubber idler must be clean, truly round and of the correct hardness. It is shifted to the correct motor pulley diameter for each speed.*

To give an accurately sized and parallel hole in such a pulley, it is bored to an accuracy of ±0·0001 in., but because of the very close tolerances, a small slot may be broached along the length of the bore to allow the air to escape so that the pulley can be assembled to the motor shaft. All four pulley steps are finish-turned by driving the pulley on a stationary arbor by means of a rubber wheel. When finished, the pulley is locked on the motor shaft by two diametrically opposite screws, which must be equally tightened to maintain the true running of the pulley.

The periphery of the rubber intermediate wheel is ground and then checked on a specially developed electronic instrument for truth and consistency of the resilience of the rubber. This check was found to be necessary in order to reject wheels having soft or hard sections in the rubber, which can cause wow and rumble.

The final link in the driving system is the inside of the turntable rim, which must be round and run true. Most record-changers have pressed steel turntables and, as it is not always practicable to machine the rim to provide a perfectly true driving surface, great care is taken in the forming of the rim to make sure it is round; the turntable is then held in an accurate centralizing chuck for piercing the centre hole.

On assembly, special attention is given to ensuring that all driving and driven surfaces are parallel to each other. For example, should the rubber intermediate wheel or motor pulley be slightly out of square it would cause what is known as "scrubbing," which would be reproduced as flutter.

The electric motor used to drive the record-changer can be a prolific source of trouble if certain precautions are not taken in its manufacture. It must run smoothly without hum, vibration or mechanical noise, and must have the minimum external magnetic field. It must also be robust and capable of running for long periods without the need for frequent lubrication, and even though it may be enclosed in an unventilated cabinet no over-heating should occur.

The smooth running of the motor is achieved by attention to the finish of the rotor shaft and bore of the bearings, as already described. Hum is avoided by making sure that the stator coils are, in the case of a four-pole motor, electrically in balance and that the rotor is exactly in the centre of the stator bore, this being one reason for spherical turning of the outside of the bearings, as previously mentioned. In addition all rivet- and location-holes in the motor laminations and bearing covers are held to very close tolerances.

Motor vibration, which can be a main cause of rumble, depends upon the true running of the rotor shaft and the balance of the complete rotor assembly. The rotors are assembled on the super-finished shafts, which are then stress-relieved. The outside diameter of the rotor is ground to size on an automatic machine and given a thin even coating of lacquer to prevent rusting. From experience, especially in the export markets, it has been found necessary to give special attention to what appears to be a simple operation, and the procedure adopted for lacquering rotors is worth describing. After grinding, the rotor assemblies are fed on to a ramp which conveys them through an electric oven having a controlled temperature. This drives off all traces of moisture or flux which may be trapped between the riveted laminations. The rotors, while still warm, are rolled over a ramp through a controlled depth of lacquer and passed through another drying oven to bake the lacquer.

The rotor assemblies are then checked for tightness of the shaft in the rotor by applying an end pressure of 250 lb in an air-operated jig, which also checks that the position of the rotor on the shaft is correct. The shaft is then checked for truth by rotating the assembly on its normal bearing locations and clocking the shaft at the point where the pulley will be fitted.

The rotor assembly is now ready for balancing, the accuracy of which is one of the main factors governing the final performance of the motor, especially in the reduction of flutter and rumble.

The rotors, which are dynamically balanced to an accuracy better than 0·031 cm-g on the latest type of automatic balancing machine, are fed into a magazine and are picked up one at a time by an arm which grips the rotor with the aid of a suction cup and places it on "V" bearings, where it is revolved by means of a rotating magnetic field. The amount and location of any out-of-balance is registered by the machine. The magnetic field is then switched off and the rotor is picked up by the arm, placed on a fixture and, if required, recesses are drilled in the copper end-plates at points previously registered by the machine to correct the balance of the rotor. A further position on the machine finally checks the rotor. If it is within the pre-set limits for balance it is passed; if outside the required limits it is rejected.

All the principal shafts on a record-changer run vertically and the thrust bearings on two of them, those for the turntable and rotor, can be a source of rumble. To avoid this the ends of the rotor shaft are lapped to a high finish so that when running on its ball thrust the minimum of friction and noise is generated. If a stationary record spindle passes through the centre of the turntable spindle a ball race must be used to carry the weight of the turntable. A normal commercial type of thrust ball race would be too noisy, consequently specially made races with five steel balls in a plastic cage between two hardened and lapped steel washers are used. The complete ball race assembly rests on a resilient plastic washer to cushion the weight of the rotating turntables from the body of the changer.

In addition to the various techniques developed and used in the production of components for the turntable driving system some of those used to produce the changer mechanism are also of interest.

The first requirement in the automatic handling of records is for the record-changer to drop them individually on to the turntable for playing. This is achieved by means of a stepped record spindle incorporating a lever operated by the record-changer mechanism to push the records off the step one at a time.

There are many types of such spindles, the example described being one of the most popular. The body of the spindle is stainless steel and four milling operations are required to produce the step and slots for the pushing lever and latch. The use of stainless steel was found necessary in order to withstand the abrasive action caused by records dropping and revolving around the spindle. Over a period of time this abrasive effect is sufficient to remove any plated finish.

The spindle assembly must be free from all traces of burr and these are removed from the spindle by brushing, and from the pushing lever and latches by roto-finishing. This latter process is used for removing burrs and sharp edges from practically all levers used on record-changers. The parts are placed in a drum with mineral chippings and water and are vibrated or tumbled for a given period. The method, size of chipping and time depends upon the shape of the levers and the material from which they are manufactured. The dimension limits for the record spindle have to be closely held so that a record on the top limit for thickness will not jam, and two will not drop together should they be on the bottom limit.

The most vital and sensitive mechanism on a record-changer is the automatic trip. This operates when the pick-up reaches the end of a record and engages the mechanism to complete one cycle. The mechanism used is known as the velocity type of automatic trip. As the pick-up approaches the end of playing a record a lever underneath is pushed inward by the pick-up arm. This lever, which has a friction pivot, intermittently engages with a projection on the revolving turntable hub. While the pick-up slowly moves inward, as it does while playing a record, the projection as it revolves pushes the lever away from it. However, when the pick-up reaches the run-out groove at the end of a record it moves the lever inward too far to be pushed away by the projection which then positively engages with it; this action causes the mechanism to engage and complete one changing cycle. The reliable operation of the trip is the basis of the whole mechanical performance of the changer and it must be sensitive enough to operate on a pick-up acceleration over $\frac{1}{16}$ in. within one revolution of the turntable. The action of pushing the lever away must be so light that it cannot be heard via the pick-up, and the

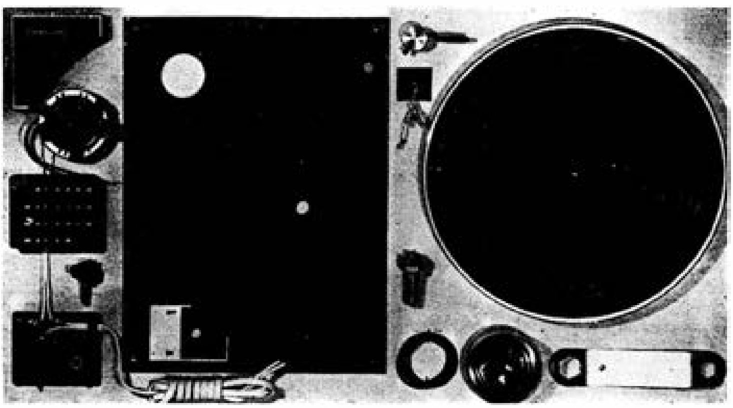


Fig 12. *A high-quality turntable kit for easy home assembly. A flexible precision-made rubber belt is moved by hand to the appropriate motor spindle step and drives the turntable rim.* $33\frac{1}{3}$ *and* 45 *rpm speeds are provided as being suitable for the vast majority of modern records. High-precision bearings and parts allow very low wow and rumble figures to be achieved.*

maximum side pressure which can be tolerated on the pick-up arm to operate the trip is about 0·25 g. It also has to cover a wide operating radius to accommodate all the types and sizes of records now available.

To achieve all this, the associated levers must be free from burr so that they move freely, and special lubricants are used, as ordinary oil would congeal and prevent operation. Clean and careful assembly is absolutely essential.

The final testing of record-changers calls for special instrumentation and most of the equipment used has been developed specially. Every changer is checked for correctness of all electrical and operational functions at inspection stations at the end of the assembly lines. Among the tests given are those for wow, flutter and rumble, all of which require the use of special test records and sensitive instruments. In addition there may be twenty-five operational functions to be checked on each record-changer and the failure of any one will cause it to be rejected.

Every changer should be checked for correctness of the following functions, but not necessarily in the order given—

1 Dropping one at a time, a stack of mixed 7 in., 10 in., and 12 in. records.
2 Operation of record-size selector mechanism.
3 Switching off after last record has played.
4 Height of pick-up lift, adjusted if necessary.
5 Lowering position of the pick-up using a special test record giving correct landing radii.
6 Operation of automatic trip when pick-up reaches end of record.
7 Stylus pressure; adjusted if necessary.
8 Operation of pick-up muting switch, if fitted.
9 Effectiveness of switch-click suppressor.
10 Lateral and vertical freedom of pick-up arm. Point friction must be of a very low order for modern low-weight cartridges.
11 Correct operation of speed-change control; check that the intermediate wheel is in the centre of the appropriate motor-pulley step.
12 Use special stroboscopic discs to check that, while the pick-up is playing a record, all speeds are within ±2 per cent of nominal.
13 Operation of "off," "manual" and "auto" control.
14 Operation of pick-up-arm bias compensator.
15 Angular location of record spindle.
16 Satisfactory operation when using large record spindle for dropping 7 in. records having 1·5 in. diameter centre hole.
17 Absence of overall mechanical noise.
18 Satisfactory operation on 10 per cent below minimum rated voltage.
19 Motor current consumption.
20 Phasing of stereo pick-up connections.
21 Angular position and freedom of record overarm.
22 Cartridge output and channel separation on stereo when a cartridge is supplied.

23 Insulation between motor and pick-up circuits and frame of changer.
24 Check setting of links in voltage-changeover terminal block and marking on motor (corresponds to its voltage and rating).
25 Finish and completeness.

Fig 39 shows the Garrard AP 75 semi-automatic high-quality single-record-player integrated with an amplifier system and plinth.

It should be noted that there are available basic transcription or other high-quality turntables without any refinements such as automatic lowering or stop facilities. This saves money and the value may be concentrated on the essential motor and turntable bearings and finish. The additional cost of adding a high-quality tone-arm must, however, be added on, and the need for careful manual lowering etc of the pick-up has to be considered.

The system as a whole

The gramophone disc-recording system is capable of giving extremely faithful reproduction provided that the system is designed and considered as a whole. It has been demonstrated that it is possible to record and reproduce television pictures on gramophone discs and to reproduce them with a pick-up basically similar to those in normal use at the present time. The pictures were necessarily not transmitted at the full rate possible with the normal electronic television systems or with rotary head cross-scanned television magnetic-tape-recorders, as these demand a frequency response extending to 3 Megahertz or more. Nevertheless, a rapid sequence of high-quality pictures was produced on gramophone records, which brings home the remarkable potentialities of the system. Improved disc TV recording systems are now being demonstrated.

As the gramophone disc and pick-up combination is predominantly mechanical in operation, it is likely to suffer a drastic loss in quality unless it is correctly understood and meticulously maintained by users, particularly as regards cleanliness. Top-class recording companies can be relied upon nowadays to cut their master records on very high-precision machines, and to comply with the international standards, with due regard to such factors as correct groove profile and surface finish, avoidance of excessive modulation, particularly avoiding excessive values of peak acceleration of the cutter tip.

It is obviously of paramount importance that the pick-up should at all times trace accurately the groove modulation. The greatest single source of deterioration of quality in gramophone record reproduction lies in the failure, for one reason or another, to achieve accurate tracking or tracing of the groove modulation by the pick-up.

These errors in following the groove modulation fall basically into two categories—

(*a*) Those errors due to the fundamental geometry of the groove, stylus and pick-up generally. The study of these limitations is usually classed under the heading of "tracing distortion." Various measures can be adopted to minimize this trouble, both in the recording and the reproducing processes.[7]

(*b*) Errors due to faulty record maintenance and inadequate cleaning; also to faults in pick-up design, maintenance and wear of stylus and record etc.

The whole subject is extremely involved and it is really necessary to have a firm grasp of the fundamentals in order to operate and maintain your gramophone equipment, records and pick-up properly, so as to realize the full capabilities of modern recordings.

The first essential is to appreciate the various properties of the groove and record materials. The recording process is an extremely involved branch of precision engineering which will merely be outlined here in so far as it has a direct bearing on the properties of the groove and thus affects reproduction.

Properties of monophonic and stereophonic record grooves

Record masters are cut by a chisel-shaped "Vee" profile sapphire cutter which is very carefully ground to the desired groove section and is highly polished. Modern cutter heads are multi-winding moving-coil instruments designed to drive the cutter over a wide frequency range, both vertically and laterally or in any desired combination of these motions, so as to produce stereo or mono recording. Feed-back windings provide inverse voltages which are injected into suitable points in the driving amplifiers, so as to flatten the frequency response and to reduce distortion in the amplifier-cutter head combination. Hard metal record-stampers are produced by successive plating operations, starting from the master. Many programmes are originally recorded on tape, and, in this case the maximum record-playing time can be achieved by driving the recording traverse lead screw from an electrical system fed from an advance tape replay-head, which enables the record groove spacing to be pre-set to suit the signal amplitude about to be recorded. Sophisticated electrical devices such as non-distorting volume limiters and devices designed to avoid unduly high recorded values of acceleration are now widely used in recording studios. In addition, it is possible, by using the "Dyna-groove" type of technique devised by RCA,

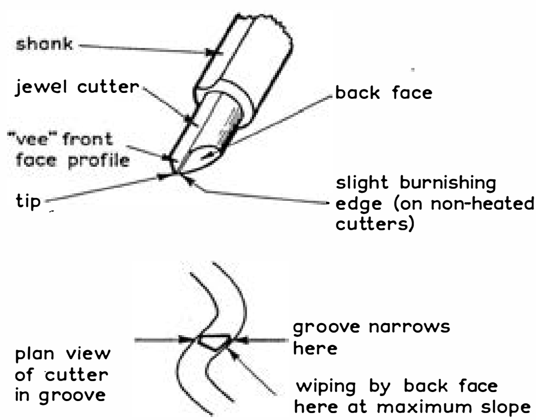


Fig 13. *The action of a gramophone recording cutter when producing the groove. The chisel-ended cutter moves both laterally and vertically to produce a stereo recording. The groove angle narrows at the centre point, where the velocity is a maximum and the absolute value of slope is limited by the shaving of the groove wall by the back angle of the cutter rear faces.*

to reduce the basic amount of tracing distortion, which is due to the finite groove and stylus sizes which have to be used. This technique is only fully effective if the results of record wall elasticity and the exact pick-up stylus size are accurately known.

It is necessary to cut a very highly polished groove, without any tendency to produce rough edges by "tearing" of the swarf (the removed ribbon of material) along the top edges of the groove. In order to engrave the higher audio-frequencies without loss, the cutter must be very sharp with a correct back angle behind the cutting surface. A dead-sharp cutter is liable to leave a rough groove, unless the stylus is heated to the correct temperature by a small DC heating coil wound on the top part of the cutting sapphire. In many cases the cut swarf is blown clear of the groove by an air jet and is then removed by suction.

The plastic material selected for record pressing must have a good thermoplastic flow, a high surface finish, low friction and good mechanical and thermal stability. Vinyl blended with certain additives meets the requirements fairly well, and is almost universally used today. Its main limitations are that it is somewhat elastic, it is subject to some variation of properties with temperature changes, it is electrostatic (thus picks up dust) and it may be subject to warping or "cold flow." Some of the additives used today are designed to minimize these shortcomings, but, for the most part, it is incumbent on the user and the pick-up designer to play and store records in such a way that the best results can be obtained from the present material.

New plastic materials are continually being investigated and when an improved record material becomes available at the right price it will doubtless replace the present materials used.

The record groove

Both mono and stereo recordings are made with the same standard microgroove whose section is typically defined—

Width of groove = 0·0025 in.
Width of inter-groove land = 0·0015 in.
Depth of groove = 0·0013 in.
Radius of groove bottom = 0·00015 in.
Angle between groove walls = 90°.
Groove width + land width = 0·004 in.
Number of grooves/inch = 250.
Maximum recorded amplitude = 0·00075 in. (up to 0·002 in. may be recorded when automatic groove widening is used during recording).
Recorded wavelength at 10,000 cycles/second—
(i) 0·003 in. at outside of disc.
(ii) 0·0015 in. at inside of disc.
Monophonic modulation is lateral.
Stereo modulation is +45° and −45° on each side of the vertical for the two channels.

The dimensional accuracy of the groove profile and the surface polish of the walls must be of a very high order indeed. This is brought home to us if we realize that the smaller audible amplitudes reproduced by the pick-up may be only a few millionths of an inch in magnitude (ie a few micro-inches). A surface with irregularities less than a micro-inch normally represents a very high polish.

The leading recording companies take steps to check the dimensions and surface finish of their recording cutters at frequent intervals with high magnification shadow-graphs and microscopes, so as to ensure that the grooves in their master records are smooth and accurate.

It was discovered very early in the history of recording that the groove had to be cut with an almost dead-sharp chisel-ended cutter, which removes the surface of the master record in a smooth continuous thread of swarf. Attempts to emboss a groove into the surface of the record by a spherical-ended stylus have not been successful for high-quality recording, partly because of the large amount of force required to be generated by the cutter head and also because the walls of material thrown up on each side of the groove have proved to be a source of trouble in the record-manufacturing processes.

The reproducing pick-up stylus must be a highly polished sphere *or* ovoid of exactly the correct radius, in order that it may fit accurately in the low part of the record groove without either actually touching the groove bottom or riding on either or both top edges. Styli of the normal outside permitted manufacturing tolerance limits will fit in an international standard groove. A careful study of the geometry of the motion of the V-shaped cutter, the moving groove and the spherical pick-up stylus shows some interesting and important factors—

1 The cutter crosses the centre axis of the groove at its maximum modulating velocity twice per cycle. The groove can be regarded as moving at an angle to the plane containing the cutting face and thus the angle of the groove is narrowed twice per cycle. This means that the reproducing stylus is squeezed upwards by a small amount. This affects the design of pick-ups in that the vertical mechanical impedance must be kept low so as to avoid excessive forces on the groove wall. Also, in a mono pick-up, if the transducer mechanism produces an output due to this vertical "pinch effect," spurious signals appearing as noise or distortion will be generated. In stereo recording the pinch effect is an inevitable slight complication which can cause

distortion and inter-channel cross-talk at high values of modulation velocity.

2 It is necessary for both the cutter and the pick-up stylus to have a minimum tip radius of a few tenths of a thousandth of an inch in order to ease manufacture and increase durability. However, if the groove has a sine-wave excursion, the centre of a

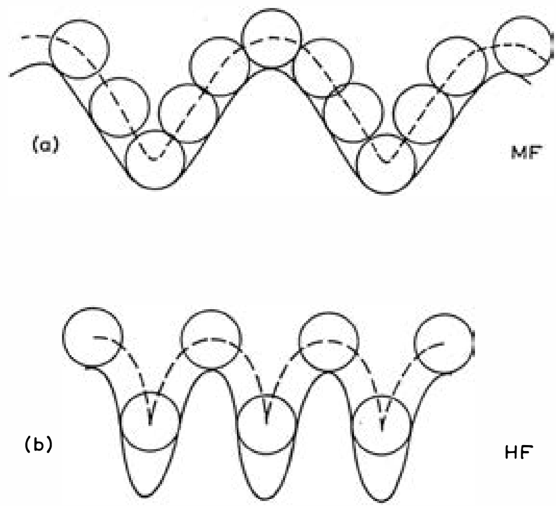


Fig 14. *The tracing of the groove by a hemispherical-ended stylus. A sinusoidal stylus excursion will not be obtained from a high-frequency groove produced by a sinusoidal cutter excursion, giving rise to "tracing" distortion. This is minimized by the use of an elliptical-ended pick-up stylus in which a very small side radius is used. An additional technique is to "pre-distort" the groove, as cut, in the opposite sense, so that a stylus with a given spherical radius will generate an undistorted wave. This assumes that the stylus reaction force produces no appreciable elastic deformation of the groove wall.*

spherical stylus tracking the groove will not trace out a sine wave; this is illustrated in a very simplified way in Fig 14. There is some cancellation of the tracing distortion for lateral mono reproduction owing to the fact that one wall of the groove to some extent compensates for the errors generated by the other wall. This is not so in stereo recording and the "reproduction-correlator" or "Dynagroove" recording systems have therefore been evolved. As previously mentioned, these consist in adding a carefully predetermined amount of inverse distortion to the recorded wave as a result of making certain assumptions as regards pick-up stylus tip radius, record elasticity and plasticity, pick-up mechanical impedance and stylus tip mass etc. This type of system is by no means easy to apply owing to the unknown pick-up factors involved in different reproducing systems. Record companies may rely on keeping tracing distortion within

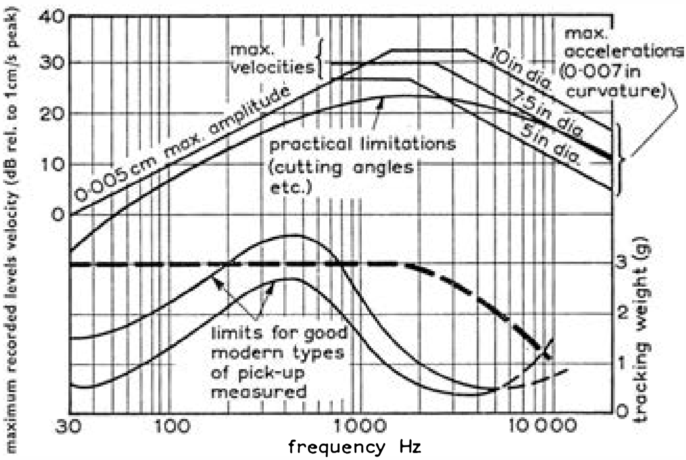


Fig 15. The maximum permitted levels recorded on gramophone records in terms of the "tracking ability" of practical high-quality pick-ups. (The dashed line shows the maximum permitted weight.) A lack of sufficient lateral compliance causes the stylus to climb the inner wall at peak excursions, whilst an excess of mass acting at the stylus tip may cause the stylus to climb the outer wall at peak excursions, where the acceleration is a maximum. The stylus may also fail to maintain contact after the peaks of vertical acceleration on stereo recordings. The broken line shows the maximum tracking weight ideally allowable if deformation of groove walls is to be avoided at the frequencies shown for the maximum recording levels encountered.

reasonable bounds by avoiding excessive modulation levels, particularly at high frequencies. This can be done automatically to a certain extent by using "limiter" amplifiers with appropriate characteristics. Fig 15 shows maximum recorded modulation levels for standard recordings and styli as well as some of the relevant pick-up properties.

Stylus and arm angular effects

Several of these geometrical effects are involved in recording and reproduction. One is the so-called "tracking angle," which is the angle between the vertical plane in which the stylus tip swings and the radius from the disc centre. Recording machines have an accurate lathe-type radial traverse slide, but most domestic pick-up arms swing about a vertical pivotal axis. This inevitably involves some tracking angular error, which causes a certain amount of distortion of the reproduced waveform. This can be minimized by mounting the pick-up cartridge at a suitable offset angle to the axis of the arm (usually between 15° and 30° according to arm length etc). The vertical operating angle of a stereo recording cutter is usually set at about 15° positive angle, as opposed to the negative angle of the old-style acoustic gramophones. If the pick-up stylus is not set at a similar angle, some distortion will occur. There is now a move to standardize the vertical recording angle. It is generally accepted that distortion is imperceptible if lateral tracking error is held to less than 5° and the vertical stylus angle is within $\pm 10°$ on $+15°$.

The above considerations illustrate the way in which the mechanical limitations and geometrical errors of our recording and reproducing devices introduce noise and distortion into gramophone record reproduction. It would obviously be better if infinitely sharp tips could be used on the recording cutters and pick-up styli. In practice the minimum realizable values are about 0·0001 in. tip radius for the cutter and about 0·0002 in. to 0·0003 in. for a spherical-ended or ellipsoidal pick-up stylus. These values, together with the high polish and good surface finish which can be imparted to record grooves on modern vinyl pressings are in fact capable of giving extremely good-quality sound reproduction. It is other factors such as the mechanical properties of the pick-up moving system and the record material, record warping, dirt etc that are likely to set the practical limits at present to gramophone reproduction.

The record material and the stylus

The plastic material used for record pressing is, of course, not infinitely rigid and it will be deformed slightly at the points of contact of the stylus tip. Any yielding of the groove walls means that the mean path traced by the stylus centre will be slightly distorted. The distortion depends on the force between the stylus and the groove. This is primarily dependent on the mechanical impedance of the pick-up system referred to the stylus tip and the downward weight which has to be applied to the pick-up in order to keep the stylus continuously in contact with both groove walls at all points of any modulation cycle which is recorded. The pinch effect and the vertical components of 45°/45° stereo recording mean that the vertical mechanical impedance characteristics of the pick-up system must be considered as well as the horizontal components. The effects of the indentation of a flat surface by a static load applied to a spherical indenter have been fairly extensively studied. The dynamic effects present when a pick-up stylus rides the two walls of a modulated groove with continually varying groove-wall curvature and instantaneous forces are very much more complicated and are very difficult to analyse. Up to a certain load, the record-groove wall surface will be elastically deformed, that is, it will spring back to its original form without any damage after the stylus has passed. Above a certain critical value, given by the yield point of the record material, plastic deformation represented by permanent surface or sub-surface damage and indentation will occur. In other words, the pick-up will leave marks on the groove walls and a greater or lesser degree of record wear will take place.

An enormous contribution to our knowledge of plastic and elastic record wall deformation is due to two British experts each working independently along his own particular line. The first was Cecil Watts, who first perfected the art of record groove photomicrography and analysed an enormous number of records of all kinds, both played and unplayed. The second was John Walton,[7] who, in addition to design work and theoretical studies of pick-ups, arms and grooves, developed a remarkable technique of taking electron-microscopic photographs of parts of record groove walls at magnifications of up to 20,000 times. These show up in a most revealing manner the effects of instantaneous plastic deformation of the groove walls at points in the modulation cycle and make it clear that the downward force and the mechanical impedance of the stylus load must be kept to a very low value if plastic deformation and wear of the record walls are to be avoided. It is, of course, desirable to avoid even any appreciable elastic deformation of the groove wall. The pick-up mechanical impedance is largely governed by the restoring stiffness together with the effects of pick-up arm resonances at low frequencies, by the system resistive damping at middle frequencies and by the effective mass at high frequencies, all values being referred to the stylus tip. It is particularly difficult to keep the moving mass to a low enough value, as the mass of any but a small stylus jewel rondel is liable to be excessive, without allowing for the other necessary moving parts of the transducer mechanism such as a moving magnet or armature, coil, crystal etc. The effect of any appreciably excessive mass may be to "wipe off" the high-modulation from the record and cause a loss of brilliance on subsequent reproduction. Reference[7] gives a wealth of detail and references on this subject, together with some most revealing electron-microscopic photographs, showing, for example, that in one case the effect of 300 playings with a 0·6 milligram stylus-tip mass was merely a polishing of the slight score lines and roughnesses left on the groove walls by even the highly polished cutter used for recording, whilst the effect of a 3 milligram tip mass was to obliterate the original high-frequency recording and to leave a distorted indentation of its own on the groove wall. The achievement of very low values of stylus-tip mass taxes the skill of the pick-up designer and requires the use of artifices such as the mechanical leverage action between the tip jewel on the stylus arm and the transducer element or armature.

Pick-up design and performance specifications

The mechanical impedance of any pick-up moving system depends on the size, weight, layout and mounting of the various parts concerned. The analysis and development of these parts are a complex and exacting task. This fact probably accounts for the fact that there are as yet relatively few really first-class designs of pick-up produced in the world.

Pick-ups are nowadays mainly electro-magnetic or piezo-electric (crystal) types. The first category usually depends on the induction of a voltage in a coil by either a moving armature or a moving magnet. Both types of element may be made very small, whereas a moving coil tends to be larger and more difficult to mount. A piezo-electric crystal or ceramic element tends to be relatively large but the high transduction efficiency allows the use of a flexible coupling between it and the stylus, thus affording mechanical decoupling of the transducer mass without encountering an excessively small output voltage level from the pick-up. The effects of temperature and time on the various plastic mountings, stylus arm and damping elements etc must be considered by the maker so as to ensure a consistent performance and a satisfactory life for the pick-up cartridge under all conditions likely to be encountered.

We have mentioned the three régimes on the frequency scale in which the stiffness, resistance and mass of the moving system predominate, at low, middle and high frequencies respectively. The effect of any mechanical reasonances may be to increase or reduce the impedance according to whether they are effectively "parallel" or "series" resonances when referred to the stylus tip, in an equivalent electrical circuit analogy. The placing and damping of these resonances is thus as important as the realization of suitably low values of stiffness and mass at the stylus tip. The applied downward weight offsets the reaction due to the mechanical impedance. The restoring stiffness reaction at low frequencies tends to cause the stylus to climb the near wall (ie that closest to the centre line) of the modulated groove at its peak excursion from the the mean position, whilst the mass reaction at high frequencies causes the stylus to attempt to climb the far wall at peaks, owing to the kinetic energy acquired at the maximum velocity (centre crossing) points. This causes a tendency for the outward motion to be continued, with the result that the outer wall may be mounted. In both cases, mis-tracking will occur.

The smallness of the downward weight required for the maintenance of correct two-point stylus-groove contact at *all* points of the cycle is a direct measure of the extent to which the mechanical impedance of the system has been kept down to a suitably low value at any given frequency. The "break-point" at which intimate contact is lost is manifested as a sudden increase in distortion. The procedure for testing for the break-point is described later.

Stereo recording and reproduction

We have seen that the standard form of stereo gramophone recording is a two-channel 45°/45° system in which each groove wall in effect moves independently at 45° to the vertical groove centre line to give the left- and right-hand channel signals. The recording is made with a cutter head with two windings which drive a normal cutter with a composite motion which corresponds to the instantaneous resultant of the two channels. When the signals from the two channels are "in-phase," this resultant motion is represented by a lateral motion of the groove. When the two signals are in "anti-phase," the groove undulates vertically.

In all normal recorded works the in-phase signals are stronger than the anti-phase signals and the lateral excursions predominate.

The stereo pick-up must thus have two independent sensing systems and a mechanical system capable of tracking the groove correctly in both horizontal and vertical directions. Thus it requires a lower vertical mechanical impedance than is the case for a mono pick-up. This poses difficult design problems on the

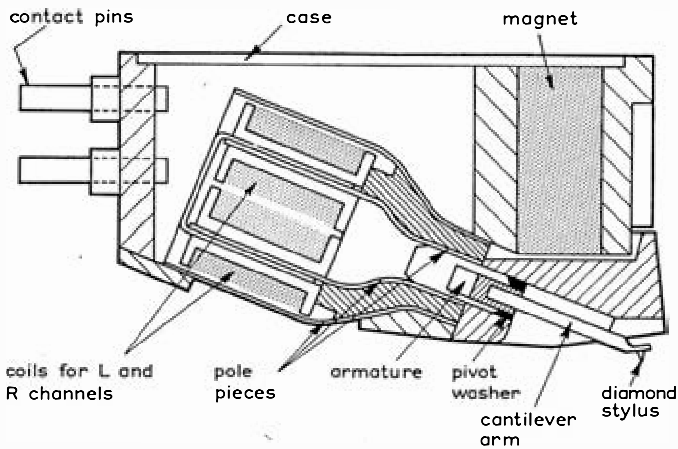


Fig 16. *A cross-section of a modern high-quality "induced flux" type of stereo magnetic pick-up. Flux from the fixed permanent magnet is fed to the two pairs of poles, which are mutually set accurately at right angles and carry the right- and left-hand stereo windings, by a miniature ferrite armature driven by the cantilever and stylus, the whole moving system being pivoted at its centre of gravity in a lightly damped elastic mounting ring.*

transducer unit mechanical layout and suspension, but fortunately, the low-frequency signals in particular tend to be in phase in the two channels and thus do not require such a high degree of vertical compliance as for lateral. Typical values might be—

Lateral pick-up compliance at stylus = 20×10^{-6} cm/dyne.
Tip mass less than 1 milligram.

Fig 16 shows the construction of a representative modern stereo pick-up of high quality in which both lateral and vertical mechanical impedance is kept at a low value referred to the stylus tip. Different models offer a range of cartridge constants above and below the values above. The correct unit should be selected for a given arm and turntable, bearing in mind that very high-compliance low-weight cartridges require near-perfect conditions in order to realize their full performance.

Checking procedure for correct pick-up tracking

Assuming that all avoidable side-thrust due to sources such as excessive arm pivot friction, inaccurate turntable levelling and record warp has been eliminated, and also that offset angle bias torque has been compensated in the pick-up arm design (page 69), then the mechanical impedance characteristics and groove tracking may be checked by the following method. The downward weight on the pick-up head is gradually reduced until distortion just becomes perceptible when heavily recorded passages are being played. The onset of severe distortion is quite sudden as the stylus tip loses the proper two-point contact with the groove and commences to climb the groove wall. Listening checks of this may be supplemented by examination of the reproduced waveform on an oscilloscope. The "break" represented by the onset of bad groove tracking should be clearly visible. Frequency records may also be used.

If there is excessive high-frequency distortion, excessive tip mass at the stylus point or a high-frequency resonance may be indicated.

Procedures such as the above can, with experience, give quite good assessments of the quality of a pick-up and whether it is likely to meet the criteria already outlined on groove deformation and wear.

The maximum recorded peak values of displacement amplitude (x), groove velocity ($\dot{x}$) and acceleration ($\ddot{x}$) encountered on standard lateral microgroove recordings are indicated in Fig 15 (page 60).

If the values of the pick-up lateral mechanical constants are referred to the stylus tip, the value of the stylus reaction force at the groove wall can be calculated to a first order of accuracy. It must be assumed that the stiffness of the stylus-restoring (-centring) system is the dominant factor at low frequencies, but that the stylus-tip mass is dominant at high frequencies. The stiffness-reaction force is given by the lateral stiffness constant times the peak amplitude. Thus, a compliance constant of 5×10^{-6} cm/dyne at a peak excursion of 0·005 cm (0·002 in.) produces a lateral reaction force of 1000 dynes or approximately 1 g. The vertical force required to maintain groove contact will be equal to the lateral force for groove walls inclined at 45° to the vertical, but extra weight is advisable as a factor of safety to allow for record warp etc.

At high frequencies, the peak groove acceleration may be as high as 1000 *g*. Thus, a stylus-tip mass of 2 milligrams will cause a reaction force of 2 g.* These simplified calculations may be in error owing to the effects of resonances, as well as the effects of damping resistance, which may increase the reaction forces considerably. The resonance between the pick-up arm mass and transducer restoring stiffness occurs at a very low frequency, and if it is not damped it may cause bad tracking and may exaggerate the effects of rumble. Warped records, poor levelling of the turntable, side thrust due to pick-up-head offset angle and other effects can increase stylus reaction forces and cause bad tracking.

Thus, in order to allow a reasonable factor of safety, a downward tracking weight of up to 3 g may be needed under adverse conditions, even when the pick-up has almost ideally low tip mass and high suspension compliance. It is also necessary for the pick-up movement to have a low vertical impedance so as to allow for the twice-per-cycle pinch effect lift due to groove narrowing. Unfortunately, there does not appear to be much latitude in regard to pick-up constants and downward weight. The tip radius should be as small as possible so as to minimize tracing distortion, but it would appear that we are then limited to something like the mechanical impedance and downward weight given above in order to avoid the possibility of plastic (permanent) damage to the groove wall at points of high acceleration. In practice, taking all factors into account, including the effects of tone-arms, record-changer mechanisms etc, the manufacturer has to arrive at some compromise between tip radius and the other factors involved. It is thus necessary to choose a cartridge with constants appropriate to the duty involved.

Checking for stylus wear and damage

We have stressed the need for a precise spherical or ellipsoidal highly polished sapphire, ruby or diamond tip for the pick-up stylus. With suitably low tracking weights, many thousands of record playings should be possible before any significant wear of

* See note on page 163.

a jewel tip occurs. A good diamond point should be regarded as substantially permanent for a low-weight pick-up. It is possible, however, for some stylus tips to fracture or to become badly worn in the course of time. Severe tip damage may be audible as noise or distortion and the groove may be scored and left dull in appearance.

Fully satisfactory visual checks unfortunately require a microscope or shadowgraph of higher magnification than is normally available in the home or in any but the larger shops (about 200 × magnification is required for 0·5-mil tip radius).

It is thus best either to use a diamond tip or to keep count of the number of playings by sapphire tips if these are used and to replace by a new tip after a great many records have been played. If the tip is appreciably worn, a change to a new tip will give an immediate improvement. Small microscopes of about 50 × magnification are used to give some guide to stylus condition in home use.

Needless to say, the correct replacement tip and attachment (usually a mounted cantilever spring) must be obtained for the model of pick-up and it must be carefully fitted after removing any dirt or fluff which may have accumulated inside the pick-up mechanism.

STYLUS DOWNWARD-WEIGHT CHECK

It is important to check the downward weight on the pick-up stylus point and also to check the vertical and lateral arm pivots for freedom of movement. A spring stylus pressure gauge reading from 0·1 to 5 g is suitable. Many modern arms have a calibrated micro-adjustment of the tracking weight and the bias adjustment.

Pick-up arm design

The arm is a very important element of the system and must fulfil several functions—

1 It must provide a light, rigid and resonance-free support and attachment for most standard types of cartridge.
2 It must be freely pivoted in both horizontal and vertical planes, without any shake or lost motion in the pivots. Frictional force at the stylus point due to the pivots should not exceed 0·5 g. The best modern arms give much lower figures than this.
3 The correct downward weight to give accurate tracking for

any particular cartridge must be accurately adjusted. An adjustable counterweight is preferable to a spring.

4 The arm must angle the cartridge so as to minimize the tracking angle error for the particular mounting distance specified between the record centre and the arm vertical pivot centre. A typical offset angle is 15°.

5 The use of an offset angle, whilst minimizing tracking angular error, inevitably introduces a torque tending to pull the pick-up towards the record centre. It is desirable to compensate for this, and the best arms incorporate a drawstring and small weight, a magnetic bias, specially angled tension springs or other devices which can be set exactly to balance the offset torque, ie to give side thrust or bias correction.

Certain points in the design of arms are of interest. The arm itself should be rigid and well-damped without being heavy. If it is flimsy or of too thin a section, torsional resonances may be present and, in extreme cases, these may become sufficiently strongly excited to upset the response of the pick-up or to add a ringing "coloration" to any background noise. Substantial aluminium rods, mouldings, die-castings or hardwood are successfully used for arms. Pivots are a particular problem. They should be fairly close to the record surface to avoid back-and-forth movement of the stylus which might cause wow on slightly warped discs. The vertical and horizontal pivots should preferably be as close together as possible or the effect of any slight errors in levelling of the motor board and turntable will throw the arm out of balance and exaggerate the side thrust produced. The ideal type of pivot is a polished hard single point or knife edge which will thrust any incidental dirt particles aside and whose effective frictional radius relative to its centre is very small, thus minimizing the effects of pivot friction.

We have mentioned that the effective mass of the arm resonates with the pick-up transducer suspension compliance at a low frequency, generally between 10 Hz and 40 Hz, in both the horizontal and vertical directions. This resonance determines the lowest frequency reproduced and, unless it is properly damped by some mechanical resistance around the transducer suspension system, rumble may be exaggerated and an excessive mechanical impedance presented at the stylus point, possibly causing bad tracking or shake. If the pick-up and arm system is correctly designed as a whole, the low-frequency response will be flat for both lateral mono and stereo recordings down to below 20 c/s (ie below audibility) with a sharp cut-off below this point. This is effectively a "built-in" rumble filter with good impedance and response characteristics.

Figs 10 (page 45), 38 (page 148) and 39 (page 149) show good modern pick-up arm designs mounted on motor boards.

Record care and cleaning

NEED FOR CLEANLINESS AND FLATNESS

We have seen from the previous sections on the record groove that quite small traces of dust and contaminant will be sizeable compared to the groove and stylus tip, and thus can cause loud "pops" in the case of discrete particles struck by the stylus and may cause distortion and bad tracking in the case of "sticky" surface contaminants. Even small accumulations of dirt will cause bad tracking and distortion in lightweight cartridges.

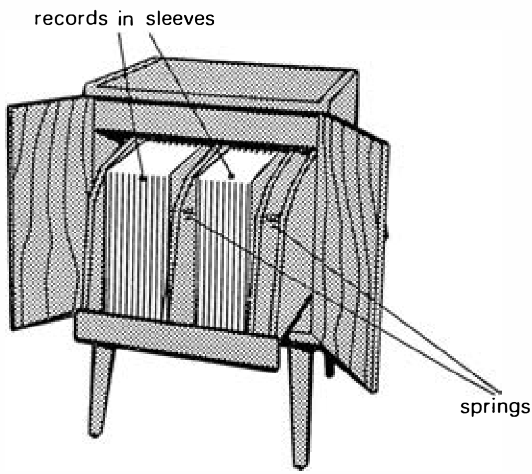
It is obviously essential to keep records as clean as possible and to replace them in their covers as soon as possible after playing.

Warping is another grave defect which can arise only too easily if records are not stored and used correctly. Anything more than a very small amount of warp will cause serious wow on reproduction because of the cyclical "stretching" of the track and the varying stylus-groove geometry introduced during rotation.

Records should not really be left in flat horizontal piles, as is often done, because any irregularity in the stack or on the surface below a record tends to cause the record to go out of flat because of plastic "cold flow." Leaving records in the sun or on top of a warm amplifier case is, of course, a certain way of producing bad warping. Once warped, it is very difficult to flatten records. It would strictly be necessary to place the record between two flat plattens or sheets of plate glass and to heat the assemblage up to just the correct temperature, cooling everything down again before removing the record from the plattens. However, storage on edge under light pressure helps correction.

RECORD STORAGE

It is best to store records in a cool clean cabinet, on edge, with a slight lateral clamping action to hold the stack of records in their cases gently but firmly erect without any tendency to lean. The "toast-rack" type of bent wire or slotted record rack should really only be used as a short-term stand while waiting to play records.



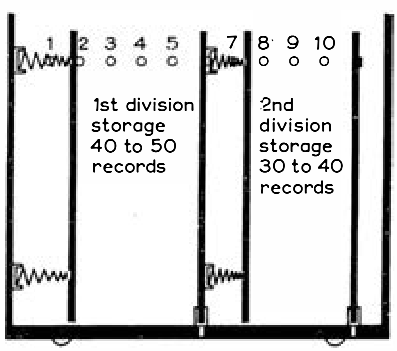


Fig 17. A new type of edge-storage system which houses the records vertically under a light spring pressure which prevents and helps to remove warp, one of the major hazards to which records are subject when not in use. Moving clips and partitions decrease or increase storage space by 1–10 records. Below, a plan view.

Figs 17 and 18 illustrate commercial systems of record storage on edge in a case or a cabinet where the records may be kept clean. The desired sideways pressure is maintained in the first system.

Record "grips" or handling pads are also sold to prevent finger-marking records (by Highgate Acoustics Ltd, London, W1).

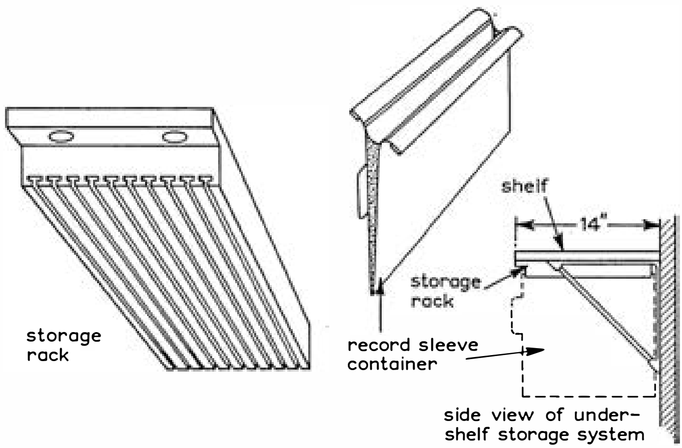


Fig 18. *A convenient and safe system of record storage in automatically sealable transparent plastic pockets which are suspended from a multi-track unit into which they slide.*

TYPES OF RECORD CONTAMINANT

We have noted the need for extreme cleanliness of record grooves, but it is as well to know as much as possible about the subject before attempting any groove-cleaning process, as the wrong treatment may make matters worse.

Records should not be fingered or blown on. Nor, in general, should they be polished, rubbed or washed except with the correct equipment and media.

Record contaminants have been found commonly to consist of (*a*) particles of dust, grit and fluff, (*b*) deposits left by soot from domestic fires, household sprays (when used), solid residues from impure antistatic and washing solutions where these have not been based on properly distilled and de-mineralized water, etc, (*c*) oily deposits from cooking vapours, tobacco smoke, diesel smog etc.

When one realizes the minute size of the irregularities which the stylus can sense on the groove wall (less than the wavelength of light) it is obvious that vestigial traces of oily deposits from the atmosphere may trap minute dirt particles firmly, so that these are forced into the groove wall by the stylus instead of being pushed aside. Even if the foreign matter is subsequently washed away, an audible click may be reproduced from the small crater left in the groove wall.

It is obvious that the less records are exposed to the atmosphere, whether being played or not, the less the risk of all types of contamination. Positive measures aimed at maintaining record cleanliness consist of "dust-bug" types of cleaner which track in the groove whilst it is being rotated for playing, and specialized record-cleaning machines or treatment kits which scientifically clean dirty or contaminated records. Velvet pads with the correct bristles are also used.

The extra pleasure and increased life obtained from really clean records emphasize the value of taking proper steps to achieve and maintain cleanliness.

THE DUST-BUG RECORD CLEANER

This device is produced by Cecil Watts Ltd, the original suppliers. Basically, it consists of a forwardly inclined brush of about two dozen nylon bristles with carefully-pointed ends of the correct tip radius to scour the bottom and sides of the grooves. They are mounted on a light freely pivoted tracking-arm ahead of the pick-up. A total tracking weight under 1 g is possible and a trace of antistatic fluid with a distilled de-mineralized water solvent may be applied via the bristles. It is really desirable that the particles of contaminant collected from the record groove should be removed at once, before they again lodge in the groove.

RECORD-CLEANING MACHINES AND MATERIALS

These range from more elaborate devices based on the "dust-bug" to ambitious routines which apply groove scrubbers, suction mops, spinning polishing brushes and controlled-heat driers. The record should be left in a completely clean condition with the merest trace of antistatic surface film to inhibit the acquisition of further electrostatic charges in the future.

Unless records are given a measure of anti-static treatment it is impossible to prevent the attraction of dust, and all users should apply a scientifically designed and formulated antistat and record-cleaning routine. Once treated and carefully stored, a light wipe with a special clean pad will usually suffice at each playing.

A stylus-cleaning brush should be used with care at each playing to clear the cartridge stylus tip of any dirt.

4

Magnetic tape recorders

The recording medium is a long strip or reel of plastic tape coated with a magnetic material. The tape is carried past the erase-, record- and replay-heads by the tape-transport equipment on the tape-deck or machine. It is taken from the supply reel and spooled on to the take-up reel during either the recording or the replay process, rewinding being necessary in each case before another recording or replay operation.

The heads consist of electrical windings on a ring core of Mu-metal or Permalloy with a small accurately formed gap across which the magnetic-coated side of the tape passes. The precise design of each of the three types of head has been established as a result of long experience and the manufacture and testing of the tape is a highly sophisticated process. An understanding of tape-recording involves a study of the magnetic properties of the tape coating and the interaction between the magnetic system of the heads and the permanent magnetism established on the tape.[8]

Magnetic recording tapes

Magnetic recording was originally carried out on metal magnetic strip or wire, but about 1940 Pfleumer in Germany made paper- and plastic-based tapes coated with a dispersion of particles of ferromagnetic material. Modern recording tapes have been directly developed from these experiments. Until about twenty years ago, the magnetic coating consisted of small spherical particles of ferric oxide (jewellers' rouge), Fe_2O_3. This material is normally non-magnetic, but was specially treated to produce magnetic properties. These earlier tapes had to be run at 30 in./sec in order to get a good frequency response. Ferric oxide particles were later produced in a long, needle-shaped form, rather than in the original spherical form. These produced tapes with much better magnetic properties (higher coercivity and higher remanence). These give good sensitivity and response at present-day tape speeds. Even higher-coercivity tapes are now produced using needle particles of ferrosoferric oxide, Fe_3O_4. These tapes are difficult to erase and are mainly used as master standards.

The plastic materials mainly used for the tape base are acetate, PVC and polyester. Acetate was the original material used as it is strong, does not stretch much, and the coating adheres well. It may, however, become brittle with age and it may wrinkle up if it absorbs too much moisture. PVC and polyester are less affected by ageing and moisture, but may stretch more readily, thus causing changes in pitch of a recording. The extra strength of a polyester base is useful for the thin extra-long-play tapes now available.

Standard tape widths are now slightly under $\frac{1}{4}$ in. (0·244–0·248 in.). The following table shows the tape classifications now produced commercially. Available playing time is from a few minutes to over 12 hours, depending on tape speed, spool size and the number of tracks successively traversed.

CATEGORY	THICKNESS (OVERALL)
Standard Play	0·002 to 0·0018 in.
Long Play	0·0016 to 0·0013 in.
Double Play	0·0011 in.
Triple Play	0·0007 in.

A number of other properties of magnetic tape are of great importance and need to be carefully controlled by tape manufacturers and users. Some of these are—

1 "Cupping," or the tendency for tape to develop a concave curvature across its width with the coating inside.
2 "Bias" or "Curl." These are waviness of the tape edges or deviation from straightness. Unless held within limits, these conditions may cause variations in output or speed "flutter."
3 Poor edges. A tape which has been badly slit to width during manufacture may have frayed edges which can shed pieces of coating or lead to tearing.
4 Coating quality and adhesion. The density of the magnetic particles should be as high as possible and as even as possible, the surface polish should be very high, so as to reduce head

wear and friction, and the coating thickness should be even and its adhesion to the base should be reliable. If these points are carefully watched by the manufacturer, the tape should have a good response and signal/noise ratio, free from "drop-outs" or momentary gaps in the signal output.

5 "Blocking" or "Sticking." Turns of tape tend to hold together on the spools because of electrostatic charges or gumminess of the surface, causing snatching and wow during playing. Friction or stickiness can also cause squealing or vibration of the tape at the guides. Care should be taken to use correct non-sticky splicing tape for jointing.

6 Tape stretching. All tapes should stand a steady pull of 4 lb without breaking, but the stretching yield may occur at half this value. The tension during normal recording or replay should not exceed about 6 oz. Greater tensions may occur during spooling or, in some cases, effects of moisture on tightly spooled tapes are said to cause stresses sufficient to stretch the tape. A reasonably good tape should not elongate more than about 0·15 per cent when a tension of $1\frac{1}{2}$ lb is applied for 1 minute.

7 "Print-through." When a recorded tape is stored on a spool undisturbed for a while, weak copies of the recording may be impressed magnetically on adjacent layers, causing pre- and post-echoes. The level of printing may be increased if any accidental stray AC magnetic fields happen to influence the tape (hence store away from mains units, TV scan systems etc). High temperatures, tight spooling and using thin (long-play) tapes are all factors tending to increase "print-through." Re-spooling tapes before playing will often reduce the magnitude of the echoes substantially, but a carefully applied partial-erase current applied during re-spooling has been found to remove the print-through without appreciably affecting the signal. This is said to be due to the fact that the printed echoes lie only on the extreme outer layers of the tape coating.

8 Noise. Ideally it is thought that the noise on a tape could be −70 dB on the signal, but in practice, owing to a variety of causes, it is unlikely to be better than −50 to −60 dB. Some of the principal causes are—

(*a*) Variation in particle size and distribution.
(*b*) Accidental DC magnetization.

(*c*) "Modulation noise," due to oscillator or other circuit noise which appears when a signal is recorded.

The best recording studios now use special dynamic noise-suppressing "compandor" techniques in order to reduce low-level tape noise to the vanishing point (the Dolby system).

Tape Table

Playing Time per Track

Type	*Diameter of reel (inches)*	*Tape length (feet)*	$1\frac{7}{8}$ *in./sec (h m)*	$3\frac{3}{4}$ *in./sec (h m)*	$7\frac{1}{2}$ *in./sec (h m)*
Standard Play	5	600	1 4	32	16
	$5\frac{3}{4}$	900	1 36	48	24
	7	1200	2 8	1 4	32
Long Play	$3\frac{1}{4}$	300	32	16	8
	5	900	1 36	48	24
	$5\frac{3}{4}$	1200	2 8	1 4	32
	7	1800	3 12	1 36	48
Double Play	$3\frac{1}{4}$	400	42	21	10
	4	600	1 4	32	16
	5	1200	2 8	1 4	32
	$5\frac{3}{4}$	1650	2 54	1 27	44
	7	2400	4 16	2 8	1 4
Triple Play	3	450	48	24	12
	$3\frac{1}{4}$	600	1 4	32	16
	4	900	1 36	48	24
	5	1800	3 12	1 36	48
	$5\frac{3}{4}$	2400	4 16	2 8	1 4
	7	3600	6 24	3 12	1 36
Quadruple Play	3	500	1 4	32	16
	$3\frac{1}{4}$	800	1 24	42	21
	4	1200	2 8	1 4	32

Tape-recorder decks and machines

There are quite a large variety of tape-recorders available, either in the form of decks for building-in or as complete portable units which may be "free-standing" in a suitable compartment of a Hi-Fi ensemble. The accompanying electronics may be fixed to the deck or may be arranged for adjacent mounting.

The cheapest commercial units available cannot reasonably be expected to give completely flutter-free music recording nor can they be expected to have an indefinite life if they are subjected to hard and fairly continuous use. At the other end of the price scale, the fully professional studio or instrumentation type of tape-recorder can be very costly and will normally be unnecessarily robust and elaborate for the average user. A few "replay-only" tape units have been introduced, but in view of the fact that pre-recorded programme tapes are both expensive and limited in availability they have not yet found widespread use. A number of semi-professional tape-decks and cassette machines are available which are capable of giving very good musical reproduction.

Although they are less sensitive than gramophone reproducers to external shocks and vibration when running and do not normally require levelling, so far as the need for general cleanliness and correct mechanical treatment are concerned tape machines call for the same order of care, if the best results are to be obtained. It might be thought that tape machines are more "fool-proof" than disc machines, but in fact many things can occur to prevent the tape from being transported properly or from making proper and intimate contact with the heads. Some tapes may shed dust, and accumulations of this and other foreign matter on the tape-drive, -heads or -guides may cause considerable trouble. Perfection of mechanical alignment of all the essential parts such as heads and guides, and the accuracy of the "running fit" of the drive spindle bearings are of paramount importance. Expensive hand fitting on a mass-produced machine is not to be expected and although mass-produced machines can be brought up to quite high standards of performance if an expert with proper equipment is able to spend some time perfecting alignments, trueing shafts, bearings etc, this is obviously not a job for the amateur.

Cassette-loading tape-decks have been produced, but, for the most part, domestic tape-decks use small cine-type spools. The larger professional machines use either the US NAB spools or the German "Magnetophon" original type of single-sided plate spool. All these types of spool are the subject of British and International standards so as to ensure interchangeability.

The best professional types of tape machine usually have a three-motor drive system, the tape-transport capstan being driven by a synchronous type of motor and each spool being driven by a hysteresis or induction type of motor. A light braking action on the feed-spool motor and the right amount of driving torque on the take-up-spool motor enables correct tape tension to be maintained during recording, reproduction, re-winding etc, the correct conditions on the motors being established by the controls.

There are, however, quite satisfactory machines in which a single synchronous motor drives both the capstan and the spools, a "slipping clutch" and an idler or belt drive being provided to give the correct driving torque to the spools. This idea of a slipping-clutch friction drive has been used for many years on professional cine-projector spool drives in order to maintain constant film-spooling tension.

The usefulness of a domestic tape-recorder can be further extended by synchronizing it to home cine equipment, thereby adding a sound track to otherwise silent homes movies. Various methods of doing this are available, eg the Bell and Howell Filmsound system.

Some aspects of the care of magnetic tape and the various types available have been covered in the preceding section on the tape medium. It may be reiterated here that tapes should be carefully stored on spools in boxes in a cool place away from any stray magnetic fields. This means that a suitable compartment away from mains equipment etc should be provided for keeping tapes in when an integrated Hi-Fi system and its cabinet work are being planned.

The way in which signals are recorded on magnetic tape has also been introduced in the preceding section. The need to apply the correct high-frequency bias is of paramount importance in order to obtain the maximum sensitivity and the lowest distortion from the tape. It is also necessary to ensure that any earlier recording or noise is erased from the tape prior to making a recording. This is done by applying the full bias-oscillator output to the erase-head, which precedes the recording-head. A much higher signal level is applied for erasure than for biasing. Theoretically, the bias level should be accurately set for each different type of tape used. The optimum bias level is generally taken to be that which gives the maximum output for the lowest value of distortion. This usually occurs at a bias slightly in excess of the value giving maximum tape sensitivity, ie the value which gives the maximum signal output for a given input. There is sometimes a pre-set bias level control provided at the back of tape-recorders. If an output level meter is available the user can set the bias control slightly beyond the maximum output point. If he has an oscilloscope or more sensitive distortion-measuring equipment, he can set the bias in a more scientific manner. Fortunately, the

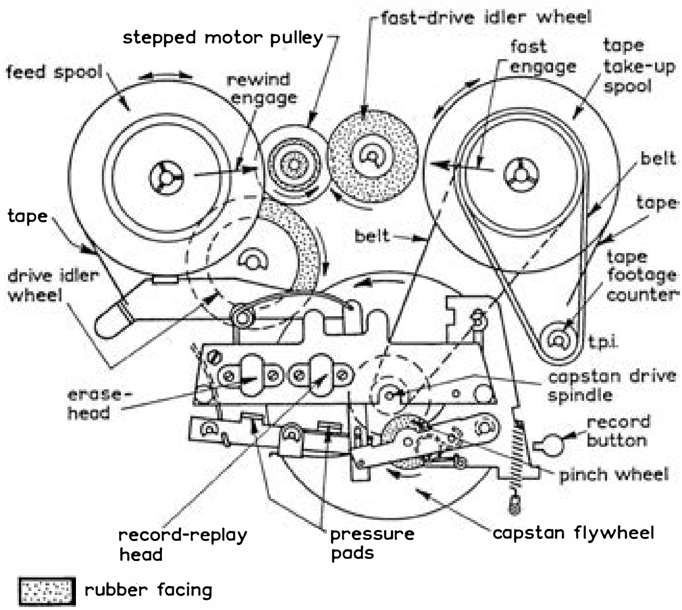


Fig 19. *A typical tape-drive traverse system used on a modern commercial tape-recorder. A synchronous motor drives the capstan flywheel via a rubber-tyred wheel in order to traverse the tape at a constant speed during recording or playback. The take-up spool and tape-footage counter are driven from the flywheel by flexible rubber belts. For fast re-spooling, a reverse idler is engaged, speed change being accomplished by engaging a larger-diameter step on the motor shaft. Slipping clutch arrangements are used to drive feed or take-up spools so as to maintain the correct tape tension.*

setting is not very critical in that the performance does not suffer very much if the control is fixed at one setting for different makes of tape, as is necessarily the case for most domestic tape-recorders.

Many professional tape-recorders have three separate heads. First, the erase-head, then the recording-head and finally the reproducing-head. The electrical and mechanical requirements for each type of head are different. The erase-head has to carry sufficient power from the bias oscillator to saturate the tape magnetically. It generally has a fairly wide gap (100 microns). The recording-head has to have a smaller but well-defined gap (10–20 microns) and need only carry the bias oscillator and recording-signal power. It must, however, be accurately made and aligned with its gap perpendicular to the tape length so as to give a good high-frequency response. The reproducing-head should have the finest possible gap (3–5 microns) and must also

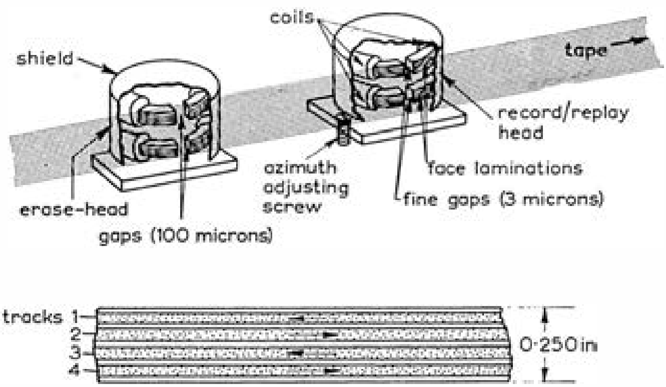


Fig 20. *A view of the head faces as seen by the tape in a modern four-track tape-recorder. The erase-head on the left has a wider gap and higher power windings. Recording and replay is accomplished by the other (dual-purpose) head, which has small very accurately defined gaps, sectionalized lamination and winding stacks for the separate tracks, and an overall magnetic shield which is cut away over the front face. Light pressure-pads are normally provided in order to keep the tape in close contact with the gap faces.*

be accurately aligned. It must generally be fitted with a magnetic shield so as to avoid hum pick-up. For multi-track recorders the heads must be suitably positioned and sub-divided so as to produce the correct tracks. The head faces which contact the tape must have a very high surface-finish, to ensure good tape-contact and low wear.

Most commercial recorders use an erase-head and a combined record/replay head with a fine gap of 3–5 microns.

We have noted that magnetic tape-recorders are arranged to run so as to traverse the tape past the heads at one or more of the various standard speeds: $1\frac{7}{8}$, $3\frac{3}{4}$, $7\frac{1}{2}$ or 15 inches/sec. An approximate guide to the frequency response available with good, well adjusted heads and good-quality tape is that the response in kiloHertz is approximately equal to twice the speed in ips. Thus $1\frac{7}{8}$ inches/sec gives a good intelligible speech recording with the maximum playing time, $7\frac{1}{2}$ inches/sec gives Hi-Fi recordings, whilst 15 inches/sec is seldom used except for some professional recordings. The majority of domestic tape recorders run at $3\frac{3}{4}$ and $7\frac{1}{2}$ inches/sec. They are normally set to record two tracks—the heads being offset with regard to the centre of the tape, so that a second pair of tracks are brought into play by turning the spools over and changing them left to right, giving four tracks in all. Where double vertically-in-line record- and replay-heads are provided, two stereo tracks can be provided. Double off-set heads will provide four-track recording. The use of only part of the $\frac{1}{4}$ in. wide tape is bound to have some effect on the output and hence reduces the signal/noise ratio, particularly for four-track recording. Four-track recording is, however, quite acceptable on a machine with good heads, using good tape, preferably a super-flexible long-playing polyester grade.

Volume-level indicators

The position of the volume control must be set for any recording so that the average modulation level on the tape is high enough to give a good signal/noise ratio, but not so high as to give rise to distortion through overloading. Volume indicators are usually of the electronic "magic eye" type or are fast-acting pointer indicator meters. "Riding" the volume control to maintain optimum recording conditions can be quite difficult, particularly if wide variations in sound level occur at the microphone. Electronic volume-limiter circuits are incorporated on some recorders, so as to enable a high average recording level to be maintained without fear of overloading. Circuits of this sort have long been used in disc recording, where it is essential to avoid the risk of running one groove into another. The audio signal is taken through a side circuit where the alternating voltage is rectified and used to bias a diode in such a way that its impedance is reduced when large signals occur. The diode may form the lower limb of a potentiometer circuit in the main signal path. A fairly fast-acting time constant is required (a few milliseconds) so as to avoid passing any appreciable overload peaks. The release time must not, however, be too short (preferably approaching ½ second), or audible "thumps" may be produced by excessively rapid changes of gain. Another type of limiter uses a low-current lamp of very low thermal inertia closely encapsulated with a photo-sensitive diode. The lamp is fed from the side-circuit voltage, as before, with the advantage that the slight thermal inertia of the lamp reduces the risk of "thumps." The attack time is reduced, however.

EQUALIZATION

The natural recording and playback processes on magnetic tape and various inevitable losses do not result in a flat frequency response. The playback equalization required differs for the various recording speeds and these curves have been standardized except for some variations to allow for differing losses in some designs (Fig 21).

TAPE-RECORDER CONTROLS

These have to perform both electrical and mechanical functions and various interlocks are essential. The bias-and-erase oscillator

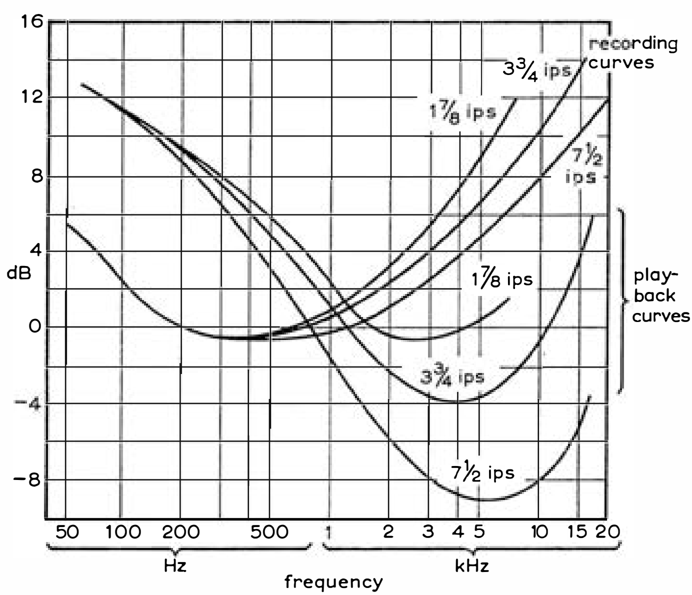


Fig 21. Tape-playback equalization curves which compensate for losses and for the natural rising response obtained when recording on magnetic tape.

must be cut off during rewind and playback, and the equalization must be changed to suit the speed engaged. The tape should preferably be withdrawn from the head faces, any pressure-pads being also withdrawn, during the rewinding operation.

Favourite methods of arranging controls are either by rotary switch, by a "joystick" type of control working in orthogonal slots or by "piano-key" types of push-button. The controls should be ergonomically sound in design, that is, they should come readily to hand and be easy to operate. They should also be clearly labelled and their various operations should be arranged in a logical sequence so that, for example, a recording is not likely to be accidentally erased by switching to the record position in mistake for the playback position.

ELECTRICAL CONNECTIONS TO TAPE-RECORDERS

Separate input terminals or sockets are usually provided for radio input, gramophone pick-up inputs and microphone inputs. Where stereo recording is catered for double input sockets have to be provided in each case. Correct polarity or "phasing" must be preserved throughout the system in the case of stereo. Overall acoustic/electronic professional phase-checking apparatus is used in large studios, but the amateur must usually rely on simple phasing checks such as trying a centrally disposed soloist or talker to check that a central stereo-sound image results on reproduction. Wrong phasing polarity results in a vague, blurred image and, in some cases, a lack of low-frequency response, ie audible as reduced bass or cancellation of any residual amplifier hum.

Pick-up and microphone input terminals were at one time often high-impedance, suitable for piezo-electric crystal pick-ups and microphones. Low-impedance pick-ups and microphones require extra amplification and must be fed via suitable input transformers or into low-impedance terminals. High-input impedance levels are now usually 50 K ohms to a few megohms and low impedances 30 ohms to about 200 ohms.

The advent of transistorized tape-recorder amplifiers operates against the use of high-input impedances on the score of noise. The optimum input impedance for many transistor circuits is 2 to 10 K ohms and some are now provided with integral input transformers of this order of impedance.

If external input transformers are used to match low-impedance pick-ups or microphones to tape-recorder inputs, the external metal cases or laminations of the transformers should be earthed to the input cable shields of the microphone etc and to the recorder earth terminal, if one is provided. Needless to say, no contact must be made with the chassis of any AC/DC or transformerless equipment on account of the risk of electric shock in the event of the mains plug being reversed, so as to make the chassis live to earth. A Hi-Fi dealers' advice should be sought in cases of doubt, as isolating transformers may be needed.

The output connections of tape-recorders are usually low-impedance terminals for external loudspeakers etc. These are normally quite suitable for connecting to the radio-input terminals of most amplifiers. Some tape-recorders incorporate links or special plug arrangements to mute the internal loudspeaker; professional machines have line output terminals suitable for feeding other amplifiers. Fig 22 gives a block schematic showing the functions of the various circuits incorporated in a portable professional recorder which gives full facilities.

Synchronization of tape-recorders and home cine equipment

The simplest type of synchronization consists in using stroboscopic bars either printed on the reverse side of the magnetic tape or on a pulley disc driven by the tape or capstan etc on the tape-recorder. The recording is made during the filming, using preferably a sound-and-vision start marker similar to the clapper board issued in film studios, thus enabling the starts of film and tape to be lined up for replay. The tape strobe bars must be illuminated by stray light from the cine-projector beam, diverted if necessary with a piece of glass or mirror in the beam. The usual cine shutter running at 16 frames/sec with a three-bladed shutter gives 48 light pulses/sec. Where the projector speed can be adjusted by a manual control it is fairly easy to bring it into synch with the tape strobes by making these appear stationary. Alternatively, the tape speed must be varied. This is not easy to do on most tape machines.

There are quite a number of commercial systems available for amateur and professional use, which offer automatic synchronization of magnetic sound and the film projector. One system is the "loop synchronizer" where a loop of tape from the tape-recorder is passed through a system of guides and rollers mounted on, or near to, the projector. A spring-mounted jockey roller is sensitive to the absolute speed of the tape and its deflection adjusts the

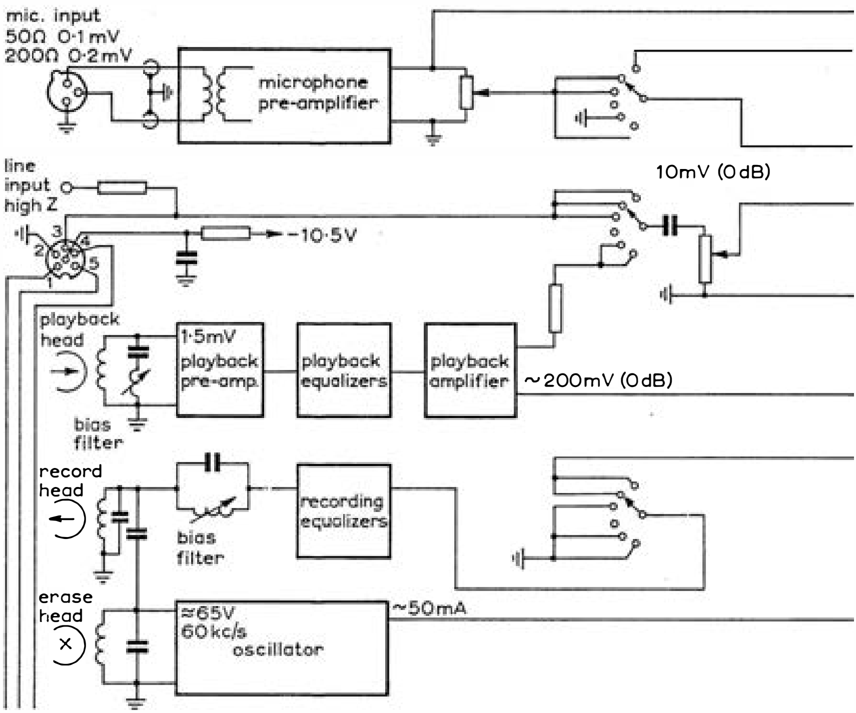
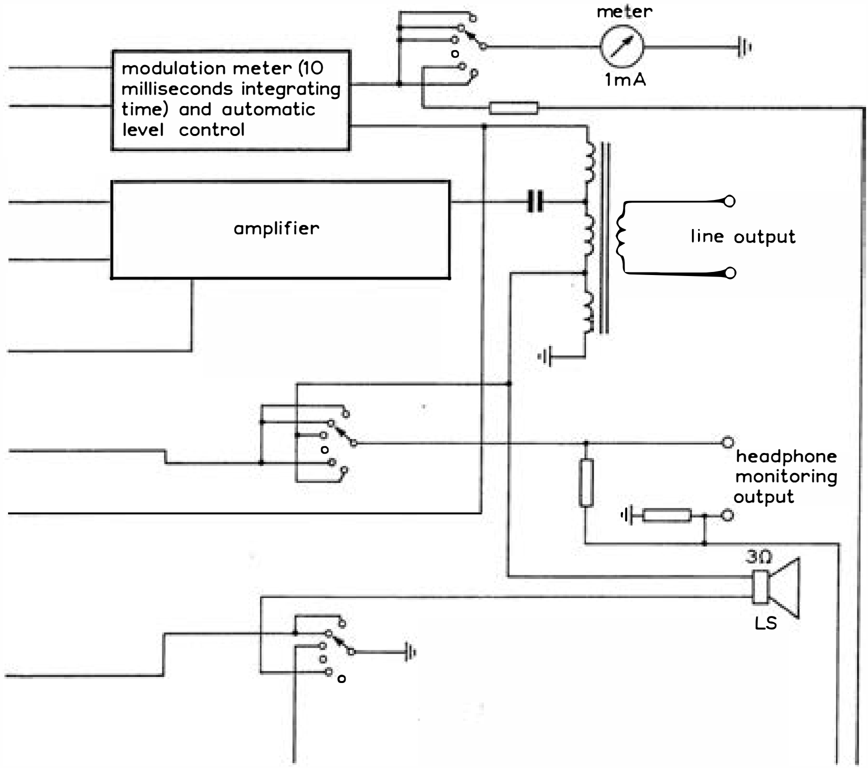


Fig 22. *A typical tape-recorder functional schematic showing*

value of a variable resistor in the projector-motor circuit, thus making the system speed-regulated and synchronous. Another system is the "Synchrodek." This is a differential gearbox which has a drive from the tape loop and another from the projector motor or sprocket shaft. The speed-difference differential gear output again is made to control the projector motor voltage so as to maintain synchronism. Full details of the various synchronizing systems and equipments can be obtained from books on cine equipment and recording.[8,9] A complete system is the Bell and Howell Filmosound.



microphone and other input circuits, extra outlets etc.

5

The use of microphones

The microphone function

Basically the function of a microphone is to pass on a correctly processed electrical copy of the sound field at a point of interest. This may be in the near sound field or in the distant sound field of a source. A microphone does not respond in the same way as the binaural human hearing system and thus its response to a given type of sound field depends on the microphone characteristics which, in turn, dictate its placement.

In general, microphones have to be placed fairly near to sound sources in order to obtain a signal having a correct "acoustic perspective" with regard to discrimination against unwanted reverberation, background sounds and other air disturbances having audible or surge-producing components. Relatively distant microphones may be used for speech and music in studios and other fairly large and quiet enclosures but a close microphone technique is necessary for speech pick-up in the majority of cases.

The performance of distant microphones may be optimized for plane wave incidence but close-talking microphones have to work under the complicated acoustic conditions prevailing around the human mouth and the secondary speech sources such as the nostrils, throat and chest surfaces etc. Human speech is normally heard at a distance of 1 metre or more where the wavefronts are more nearly plane. An ideal close-talking microphone should be designed to reproduce natural-sounding speech on either a loudspeaker or an earphone. In the latter case, the head-diffraction effect is absent and it is usually considered that a telephone microphone should have a rising response above 1 kHz to compensate for this.

Present two-channel stereophonic reproduction produces an acoustical illusion of source direction mainly by an amplitude-to-time difference conversion which occurs at the ears when two spaced loudspeakers are fed with different amplitudes obtained from crossed directional microphones. If spaced microphones are used, an inter-channel time difference element is introduced, but this is normally considered to be of secondary importance.

Hence we see that it is not easy to state the requirements of a microphone in simple terms and a considerable amount of subjective experiment has been involved in arriving at the specifications for given types of microphone. The extent to which ideal results are approached is limited by the overall state of the art and the potential performance obtainable from existing microphones.

Thus there is a fair amount of scope for experiment in microphone positioning and subsequent electrical equalization or tone control in any given set-up.

Types of microphone

The operating principles and the general forms of construction of most present types of microphone have been used for many decades and it is noteworthy that no widely applicable new principles have emerged. The tendency is to improve the performance and the reliability of known types by better design and the application of new materials.

Thus the carbon microphone is still almost universally used for telephones, as the large amount of research which has been applied over the years to the refinement of the carbon granules, the shape and material of the electrodes and other aspects of the design has improved performance and reliability to a very great extent. The carbon microphone also has useful subsidiary amplitude transfer characteristics such as lower-level noise suppression due to origin distortion, and high-amplitude clipping. If a linear type of microphone is substituted for a carbon microphone, amplification is needed and the lower-noise gating and upper-level clipping characteristics may have to be simulated in noisy surroundings.

Other forms of microphone transducer include silicon "strain-gauge" types and piezo-transistor types. Generally they have not approached the carbon microphone for efficiency. However, on account of its basic limitations as regards noise, linearity and inability to withstand certain types of vibration, the carbon microphone is not much used outside normal telephone applications.

Linear microphone transducers include moving iron, moving coil, piezo-electric and capacitor types. Efficient microphones may be made on all these principles but the moving iron is eliminated for use in high-quality wide-frequency range microphones for broadcasting, recording, acoustic measurements etc. All these types require amplifiers to raise their output level and in some cases to perform an impedance transformation.

In all cases either basically omnidirectional pressure types or pressure-gradient directional types may be produced. In the latter case the sound pressure has access to the front and to the rear side of the diaphragm, which is moved by the sound-pressure difference between the two sides at any given instant of time.

Present constructional trends and transistor microphone amplifiers

We see that microphone transducers may be basically divided into those in which the diaphragm drives the transducing element at a discrete point or area such as the apex and those in which the transduction mechanism is uniformly distributed over most of the diaphragm surface and is intimately associated with it. Such devices include ribbon microphones, capacitor microphones and piezo-electric plate microphones, in which a thin piezo-electric ceramic material is attached to the surface of a flat diaphragm. The potential gain in simplicity, reduction of mass, stability and robustness offered by distributed systems is apparent. They may also be more suited to the automated or integrated assembly methods which will be increasingly used in the future.

The availability of robust low-noise transistorized microphone amplifiers of very small size, which may in some cases be mounted in or near the microphone, has given more freedom to obtain an optimum compromise in the design of microphones. For instance, it may not be necessary to place so much emphasis on sensitivity, this being exchanged for robustness or a reduction in size. Head amplifiers may be powered over the microphone leads in various ways.

An important factor is always the signal-to-noise ratio obtainable at the input stage. When a microphone is associated with its own amplifier it is possible to obtain an optimum design compromise if a given minimum S/N (signal/noise) ratio is desired. The study of microphone and transistor noise performance is thus worth analysing in some detail. Special microphone amplifiers and mixer units are often needed if several microphones are used. These are expensive to buy but may be made up to published designs.

Noise considerations at transistor input stages

Transistors generate several types of electrical noise but have an advantage over valves in that they are substantially free from microphonicity and heater-induced hum. The resistive component of the base input impedance gives rise to thermal agitation ("white") noise and the carrier conduction or gate leakage effects give rise to shot noise, whilst other noise sources are due to recombination and surface effects, including leakage. The general effect is that one can consider the output noise as being due to several generators in series, one producing a "red" 1/f noise spectrum, and others a "white" noise spectrum. Any shunt capacity on the input circuit, such as that of a piezo-electric or capacitor microphone will again modify the input circuit noise to a "red" spectrum. The overall noise varies considerably with the type of transistor and also the transition point on the frequency scale where the higher low-frequency red noise falls into the constant white noise level.

The noise factor of a bipolar or a field effect transistor is the dB ratio of the actual noise from the device to the thermal noise produced in an equivalent resistor, referred to the input. The figure obtained depends on the centre frequency and band-width concerned. Audio devices are usually quoted for 1 kHz and a band of 10 kHz and may have noise figures between about 0·5 dB in the best cases to 12 dB or more. The S/N (signal/noise) ratio may be optimized for a given device in given circuit conditions by stepping up the microphone source impedance to a particular value, commonly between 1 K ohms and 10 K ohms. Operating the transistor at a fairly low power dissipation may help to minimize noise but there may be little improvement below a certain point and there may be a risk of overloading on large signal inputs.

It is seen that a microphone or other low-output transducer must be made up to the correct impedance so as to match the input transistor circuit to obtain the best signal-to-noise ratio. Ribbon and moving coil microphones are low-impedance devices owing to the sizes of wire or foil which must be used, whilst capacitor or crystal (piezo-electric) microphones represent high-impedance reactive sources. Input transformers are thus often used for dynamic microphones, whilst capacitor types require impedance-transforming head amplifiers, commonly using low-noise field effect transistors which can operate effectively at very high-input impedances.

Microphone properties and applications

Desirable qualities in a microphone are: small size, robustness, high sensitivity to desired signals with rejection of unwanted sounds (such as background noise, hum pick-up etc), uniform response to all frequencies, and suitable directional properties. No single microphone can meet all these requirements and the best results depend on the choice of microphone to suit the particular applications which the user has in view.

Certain classifications will occur to the user at once, for example, indoor and outdoor microphones, directional and omnidirectional microphones and microphones for close or distant talking. Detailed consideration of these categories will help to make clear the functional and constructional differences between the various types of microphones available. The distinction between high-quality and lower-quality microphones must be considered. Where cost must be taken into account, it should be remembered that a good microphone does not usually add much to the overall cost of an installation; a cheap one is thus a false economy. Good modern microphones are precision-made and utilize the best materials and manufacturing techniques. Each microphone is usually individually tested over its whole range of performance to a very exacting specification.

Directional properties of microphones

For many purposes a microphone needs to be sensitive to sound sources irrespective of the angle of sound incidence, as, for example, when the instrument is located centrally with respect to a group of performers. Such a microphone has been called non-directional since it has no favoured direction of acceptance, but "omnidirectional" is a better and more positive term to indicate that the microphone accepts sounds equally from all directions. It is well known that in air, as in water, the static pressure is the same in all directions; ie air pressure is naturally omnidirectional. In general, therefore, omnidirectional microphones are pressure-operated since they respond to changes in air pressure produced by sound waves.

Most early microphones were of the pressure-operated type, but later microphones were made which respond to changes in pressure gradient rather than to changes in pressure. In simple terms, if pressure is likened to the height of a hill, pressure gradient corresponds to the steepness of the sides of the hill and is thus a vector (ie directed) quantity. As air pressure varies with a signal, so does pressure gradient; thus an electrical device sensitive to pressure gradient changes can act as a microphone. Such a microphone is directional since, in its basic form, it is most sensitive to sound arriving from the front and back and least sensitive to sounds from the sides, top and bottom. The "figure-of-eight" directional pattern thus produced has certain advantages as it is sometimes possible to arrange for unwanted sound sources to be on the insensitive axis of the microphone. An even more useful sensitivity pattern is the cardioid (heart-shaped) polar curve. Cardioid microphones have the same discrimination between direct and random indirect sound as the bidirectional (figure-of-eight) types but they have the additional advantage of being insensitive to sound arriving from the back. This property is particularly advantageous for use on a stage, since the cardioid microphone helps to suppress unwanted noises coming from the orchestra pit or the audience and reduces echo effects from the back of an auditorium.

The growing popularity of stereo reproduction enhances the importance of directional qualities, since most stereo systems depend on the use of matched and accurately orientated directional microphones. Coincident "crossed axis" or spaced-apart microphones or a combination of the two types of mounting are used for stereo. It is worth experimenting with different angles and spacing so as to get the best stereo result for a given recording.

Close and distant talking

In a studio, particularly when music is being reproduced, the sound source is some distance from the microphone, whereas in public address systems, often used in a noisy environment, the talker is much nearer to the instrument. Some vocalists sing within a few inches of the microphone and obtain special effects by so doing. Finally, the commentator who has to provide a commentary during a live programme must put his lips close to the instrument.

These different requirements lead to the need for microphones with special characteristics. The single cardioid or bidirectional microphone gives a useful degree of exclusion of unwanted sound without the disadvantages of considerable size or complication. Microphones which operate on the pressure-gradient principle have another special property not found with pressure-operated types. With the former, the response to low-frequency sounds rises more rapidly as one approaches the instrument than do the middle- or high-frequency sounds. Ribbon microphones, which generally work on the gradient principle, are therefore well suited to studio use, but give low-frequency boost if used for close talking. This property can, however, be put to good use by introducing elements into the construction of the instrument which attenuate the lower frequencies so as to give an overall flat response when used for close talking at a prescribed distance.

Microphones with these characteristics can be designed so as to be eminently suitable for a commentator's use. Room or other noises, or the programme through which the commentator has to talk, besides being relatively distant, are robbed of most of their middle- and low-frequency content. The result is an effective suppression of everything but the commentator's voice. A microphone of this type is often fitted with a mouth-guard so that the distance from the speaker's mouth is accurately fixed. An alternative is the use of a microphone suspended on a neck halter. Small moving-coil types are best for this.

All microphones operate on the alternating components of air pressure of which the sound waves consist, so that puffs of air from mouth or nose and relatively large bursts of pressure which accompany certain consonants (such as "p" or "b"), can produce disastrous results when amplified from a microphone. One of the difficult problems of close-talking microphone design is to neutralize this "blasting" or "popping" without causing deterioration of the response. The commentator's microphone, as used by the broadcasting and television organizations, is so constructed that this unwanted characteristic is largely avoided and distortion-free speech of broadcasting quality is obtained. If this trouble is experienced when using a microphone close to the mouth, a simple windshield cut from porous foam plastic or a nylon stocking may be made, and this may also be used outdoors to protect against the noise of the wind.

Outdoor use of microphones

Some microphones, such as ribbon microphones, perform excellently in the calm air of a room or studio but are unsuitable for outdoor use. In general a microphone to be used outdoors should have smooth contours with no sharp edges to encourage air turbulence or edge tones, and a robust moving element. Pressure types are more suitable than gradient types, since any low-frequency turbulence produced by the action of wind on a gradient microphone is likely to be emphasized by the bass boost effect.

It is nearly always necessary to provide a microphone that is to be used out of doors with a windshield. At its simplest, this may take the form of a "glove" of foam material, sometimes silicone-treated to repel water. For use in high winds, however, the microphone must be enclosed in a cage of wire mesh closely covered with a suitable material, such as fine-woven wire mesh (about 120 meshes to the linear inch). Windshields act by attenuating the velocity of the wind air-stream, so as to prevent the setting up of serious turbulence round the body of the microphone, and, if carefully designed, shields give little attenuation of the much smaller amplitude waves of sound. Because the shield itself generates some turbulence, it should be 2 or 3 in. in diameter to be efficient in very windy conditions, so that this turbulence is kept as far away as possible from the operating elements. This is specially important for gradient microphones, which have an increased response for low-frequency turbulent sounds generated near the sensitive element. The simpler and smaller foam shields are still effective at low wind-speeds and are also very useful to suppress noises due to a speaker's breathing and explosive consonants or "pop" noises.

Fig 23 shows a number of representative modern microphones suitable for high-quality amateur use.

Impedance of microphones

Microphones are normally either of low impedance, as in the case of moving-coil or ribbon units, or of high impedance, as in the case of piezo-electric crystal or capacitor (condenser) microphones. Tape-recorders and pre-amplifiers may therefore have microphone input terminals or sockets for either high or low impedance. If it is desired to connect a low-impedance microphone to a high-impedance input socket, an external microphone transformer must usually be used in order to obtain an adequate

(a) *An omnidirectional moving coil microphone suitable for speech and general purpose recording.*

(b) *A compact unidirectional ribbon microphone with a polar pick-up characteristic which reduces unwanted background noise and reverberation.*

(c) *A high-quality ribbon microphone for indoor recording of high-quality speech and music. The accurate bi-directional polar curve helps to eliminate unwanted background sound. Two of these microphones may be mounted close together at right angles for "crossed microphone" stereo recording.*

Fig 23. A group of typical microphones.

signal from the microphone. Small shielded microphone transformers are available with various impedance ratios.

Low-impedance microphones are usually nominally 30 to 50 ohms, but a medium impedance of about 200 ohms is now becoming popular as well, in order to get a better match to transistor input stages. High-impedance microphones such as crystal units may in fact have to be used with a step-down transformer to match transistor inputs. Generally speaking, microphone impedance matching is not very critical, provided that sufficient signal can be obtained and that the frequency response is not affected. In cases of doubt the manufacturers concerned or the local dealers may be consulted.

Further references should be consulted for details of the design and construction of the many types of microphone available.[2,14]

Microphone placement

This is the art of placing the microphone relative to the performers so that a good "balance" is obtained between the different artistes, instruments etc, and the correct ratio between the direct and reflected (reverberant) sound as picked up by the microphone or microphones used. Generally, a close microphone position will give good "presence" and will eliminate excessive reverberation and the stray ambient background noises almost invariably present to some degree. The result may then be too "dry" or lacking in reverberant "colour," in which case use may be made of an artificial reverberation spring or plate equipment in order to add a controlled amount of reverberation to the final recorded result. Some of these devices are now within the range of the amateur's budget.

The reader is referred to books covering studio techniques and sound recording for more detailed treatment of microphone placement and studio recording techniques.[13,14]

6

Amplifiers and control systems

General properties of amplifiers

The old conception of an ideal amplifier as a device which merely increases the power in a circuit without modifying the signal in any other way has long since been superseded. Although it is still a prime aim to preserve the signal without distortion, modern amplifiers also perform quite a number of other functions. These can be broadly summarized as switching and control functions or filtering and equalization functions, together with a few useful subsidiary effects such as electrical damping of the loudspeaker. A good amplifier will also often provide smoothed and regulated power supply outlets for running radio tuners, pre-amplifiers etc.

Till fairly recently, the term amplifier mainly applied to a separate single-channel power-amplifier, fed from a radio tuner or a single-channel pre-amplifier. The advent of stereo led to the use of two electrically-separate amplifiers on one chassis sharing a common power-pack. A present trend is now towards single combined or "integrated" amplifier units which include a stereo tuner, control-amplifiers and power supplies. The enormous reduction in size, weight and heat dissipation resulting from transistorization and, in particular, the growing use of integrated circuits has made this type of combined equipment feasible and an economic proposition. Apart from a certain amount of metal required as a "heat sink" for output power transistors, most of the bulk of the new generation of amplifiers is in the power supply and in the control, equalization and switching functions. This overall concept greatly facilitates the design of compact, well styled and ergonomically satisfactory high-fidelity equipment which is easily placed in a living room and fits in well with modern furnishing arrangements. The elimination of the bulk, weight and large heat dissipation inevitably associated with valve-type power-amplifiers is particularly advantageous.

It was some time before all the difficulties associated with transistor circuits were eliminated and it could be said that the performance and reliability equalled that of the best valve amplifiers. The best modern transistor amplifiers embody a number of ingenious circuits and design features.

It is probably still true to say, however, that it is easier for the amateur to design and construct a good valve amplifier than an equivalent transistorized amplifier. Some of the specific difficulties which are encountered with transistors will be dealt with in the following pages. These include the inherently greater "origin-distortion" type of nonlinearity displayed by transistors as compared to valves and the means available to minimize the resultant "cross-cover" distortion in push-pull power output stages. Another phenomenon which has to be guarded against is the tendency in transistors towards "thermal-runaway" or overheating, as the result of which a transient overload or output short circuit may cause transistors to fail much more readily than valves. The advent of the more robust silicon types of transistors and the use of special circuits have greatly helped towards the elimination of these troubles.

Amplifier input and control functions

A well-designed amplifier must provide means for selecting, controlling and equalizing signals from any desired source. These normally include standard types of gramophone pick-ups, microphones, tuners or a tape-recorder replay outlets. Mono and stereo switching arrangements may also be needed.

Unlike most signals, which can be assumed to originate with a nominally flat frequency response, the majority of modern high-quality magnetic gramophone pick-ups generate an output with a response related to the "constant amplitude" disc recording characteristic, except in so far as this is modified by their own internal mechanism. Modern moving-magnet, moving-coil and variable reluctance pick-ups give an output substantially proportional to frequency for discs recorded strictly according to this characteristic, which is now a generally accepted international standard (BSI, RIAA etc). The input stages of amplifiers are thus generally designed to compensate for this curve when switched to normal gram inputs. However, certain pick-ups, particularly piezo-electric types are more conveniently designed so as to give a natural compensation for the standard recording characteristic and thus require a near-flat input amplifier position. Also a number of recordings have been made with different frequency responses, particularly those made before the introduction of the international standard. Again some records have unusual characteristics due to intentional special effects or unusual studio acoustics etc.

In other cases, some automatic limitation of the levels of low frequencies or high frequencies may have been imposed in order to avoid groove-wall breakdown or excessively deep or shallow cutting in stereo grooves, or, again, to avoid too high a value acceleration and consequent tracing difficulties with the pick-up stylus. Microphone and voice characteristics may also have to be compensated. Thus a fair degree of independent bass and top "curve-bending" type of tone control is desirable. Circuits which do this are described later, including the well-known Baxandall tone controls. It is also desirable to eliminate any extreme low-frequency rumbles and hum and to reduce any appreciable high-frequency electronic or recording noise; suitable stepped filter characteristics with varying degrees of cut-off slope are often provided for both low-frequency and high-frequency cut-off. These filters have usually been designed as passive (non-amplifying) four-terminal networks in the past, based on conventional inductance and capacity filter network design theory. These components tend to be rather bulky and expensive by modern "integrated" standards, particularly in regard to the case of sharp cut-off low-frequency high-pass filters designed to remove rumble. Active filters designed to form part of an amplifier circuit are now gaining favour. These may often be designed as resistance-capacity networks which may be placed in the feedback paths of integrated or other transistor amplifier circuits (Figs 28 and 29, pages 115, 116).

As well as allowing the connection of various inputs at the correct impedance, the amplifier input stages have to meet extremely critical requirements as regards the range of voltage accepted. The two bounding conditions are the electronic noise-level limitation on the lowest signal levels that can be accepted and the overload capability limitation on the largest input allowed. These two requirements are not always compatible, in that a minimum noise figure is normally obtained when a transistor or valve is run at a low current, which may involve too low an overload ceiling.

The possible dynamic range on modern recordings is now considerably greater than that of a few years ago, owing to the fact that more sophisticated types of control, such as the "Dynagroove" system, help to offset the groove-tracing distortion limitations and also avoid the risk of excessive groove amplitudes and acceleration values by means of advanced peak limiting devices. Dynamic-noise suppressors, such as the Dolby noise suppressor, mean that lower minimum signal levels can be accepted. The better modern gramophone pick-ups are also able to track higher recorded levels. It used to be assumed that the input stages of an amplifier could accept +20 dB above the nominal level at any input terminal. This figure now needs to be regarded as an absolute minimum and preferably should be considerably increased. If 30 dB is taken as a maximum requirement, then a high-output crystal pick-up with a nominal average output of 0·5 volt could give no less than 15 volts at peak levels. This is obviously in excess of the possible handling capacity of a transistor stage with a 12-volt supply and thus the input would have to be attenuated ahead of the first transistor. In other cases, the first transistor stage may have a considerable gain, possibly over 100 times as a voltage ratio. Thus an input level attenuated to 0·15-volt peak, which might be satisfactory for the input transistor, could overload the next stage with the output of 15 volts. This trouble might be reduced to negligible proportions by the use of suitable negative feedback over the two stages, and also by the use of 40-volts silicon transistors in the early stages. Thus the design of the input stages prior to the volume control point must be considered as a whole, and in fact this is now normally taken care of by the use of various forms of negative feedback circuit in modern designs. There is obviously considerable design flexibility as long as separate (discrete) components are used. However, some of the advantages of integrated circuits may be obtained by using commercial linear integrated transistor amplifier units and incorporating a volume-control in the external feedback circuit loop allowed for by the manufacturer. A volume control right at the input is not desirable on low-level circuits because, even when set to maximum, it may introduce enough loss to worsen the signal-to-noise ratio on a weak signal.

A number of special complete audio-amplifier integrated circuits are now made by leading manufacturers, who also supply details of the external components which have to be added in order to make pre-amplifier and output stages with tone control and volume control facilities etc.

Power-amplifiers

Many of the important properties and requirements of power-amplifiers have been well established over the years for valve amplifiers. Thus we are familiar with concepts such as matching to loudspeakers for optimum power transfer and proper damping of the loudspeaker bass resonance. Other factors are the power-handling capacity over the audio-frequency range, the amount and type of non-linear distortion produced, the response to normal and abnormal transients and surges. Other properties of push-pull valve output stages, in either class A or class A-B, concern normal centre-tapped output transformers or transformers with "ultra-linear" primary circuit taps for tetrode screen connections, tertiary feedback windings, tapped or series/parallel secondary connections etc.

Transistor power amplifiers offer a number of potential advantages over their valve counterparts, particularly as regards size, weight and the amount of heat dissipated. It seems natural that the transistor should form a good power device for driving a loudspeaker because a transistor is basically a relatively low-impedance current-operated semi-conducting device whereas a valve is fundamentally a high-impedance voltage-operated vacuum-tube amplifier. Transistor circuits lend themselves much more readily to the elimination of the loudspeaker-matching output transformer, which is one of the more bulky, expensive and critical items in valve amplifiers.

On the other hand, power transistors introduce a number of new problems of their own. They are inherently more nonlinear than valves and are more likely to be damaged by temporary current or voltage overloads. As the active parts of transistors are very small in area, it is difficult to dissipate any internally generated heat rapidly enough to avoid overheating and damage. They have to be intimately connected to metalwork, preferably with good heat dissipation in the form of fins etc, ie heat sinks. The heat dissipation problems caused by the relatively high standing current required by class A circuits and their comparative inefficiency has caused push-pull class B output circuits to be favoured for transistor amplifiers. Two particular problems arise here. Firstly, the amplifying transfer characteristic of transistors tends to follow a more nearly exponential curve than the equivalent valves. This means that the initial part of the characteristic near the zero point shows more origin curvature or "bottom bend." This tends to cause much greater "cross-over" distortion in the centre area of the combined transfer characteristic which is obtained in class B transistor push-pull output stages. Also, as transistors require an appreciable input-drive current, the position is again worse than for valves, which require virtually no input-drive current. The bottom bend curvature of transistors causes the effective input resistance to vary during the drive cycle. This means that the previous transistor drive stage must represent a low-source impedance in order to avoid distortion due to the drive current varying during the cycle in a nonlinear fashion. One solution is to use a transformer-coupled drive stage, but this is contrary to the general desire to eliminate transformers from transistor circuits as far as possible, on the grounds of cost, bulk and the small amount of non-linear distortion which is inherent in all iron magnetic circuits. It is also difficult to avoid some lack of balance between the two halves of the push-pull transformer windings, unless bifilar or multi-wire windings are adopted. The application of a large amount of negative feedback over the output and driver stages will reduce the distortion to a low figure, but there is still liable to be some detectable audible distortion of low-level signals which are in the high distortion region around the cross-over point between the two output transistors, unless special precautions are taken.

Unfortunately, it appears that this type of distortion on low-level signals, such as quiet voices or quiet unaccompanied string passages, represents a relatively large amount of the higher order harmonics, typically third harmonics and higher. These are objectionable to a much greater degree than their measured amounts might suggest. Factors of this sort have tended to cause transistor amplifiers to sound inferior to equivalent valve amplifiers, even though measurements of total harmonics might appear to be similar. Also distortion measurements are often quoted at full output, not for low levels, though some manufacturers do quote total harmonics at 10 milliwatts as well as at full rated output. Genuine comparisons should ideally quote these as well as intermediate levels.

It is possible to reduce all distortion to a suitably low level and to give results equal to the best valve amplifiers if careful circuit design and component matching is applied to transistor drive and output stages. Full complementary working of matched silicon output transistors and the achievement of constant-current drive and correct balance over the whole frequency range are among the methods adopted by the leading high-quality manufacturers. Good regulation of the power supply is also necessary to avoid voltage drop on sustained loud signals.

Precautions against output transistor breakdown and damage

Valve amplifiers are largely immune to accidental electrical damage if they are correctly designed and the components are properly rated. Thus overload by high inputs, surges, short-circuits or accidental disconnection of the loudspeaker under load etc cause no damage.

Transistors are much more vulnerable to permanent damage due to current or voltage overloads, and built-in precautions should be incorporated, either in the form of overload trips, fuses or automatic limiting circuitry. The possible combinations of overload conditions are not always easy to predict and considerable development has been carried out by the leading manufacturers. Their amplifiers are probably proof against most forms of accidental overload, due to the use of such devices as sensing transistors to monitor output transistor emitter currents or output voltage levels, the sensing circuits being made to operate quick-acting diode switches. This is a more sophisticated approach than the earlier latching overload trip relays. These generally protect against current overload only and may have to be specially designed in order to ensure fast enough operation. Unlike advanced electronic protection circuits, they usually put the amplifier out of action until they are re-set manually, the power supply being cut off.

If simple unprotected power transistor circuits are used, it is desirable to observe certain precautions. Before use, input and output terminals, plugs, wiring etc should be checked to ensure that no intermittent or short-circuited connections are likely. It is not always advisable to connect unusual output circuits involving excessive reactance, such as long extension leads, equalizers, piezo-electric or electrostatic units in place of the normal moving-coil loudspeaker speech-coil that the circuit has probably been

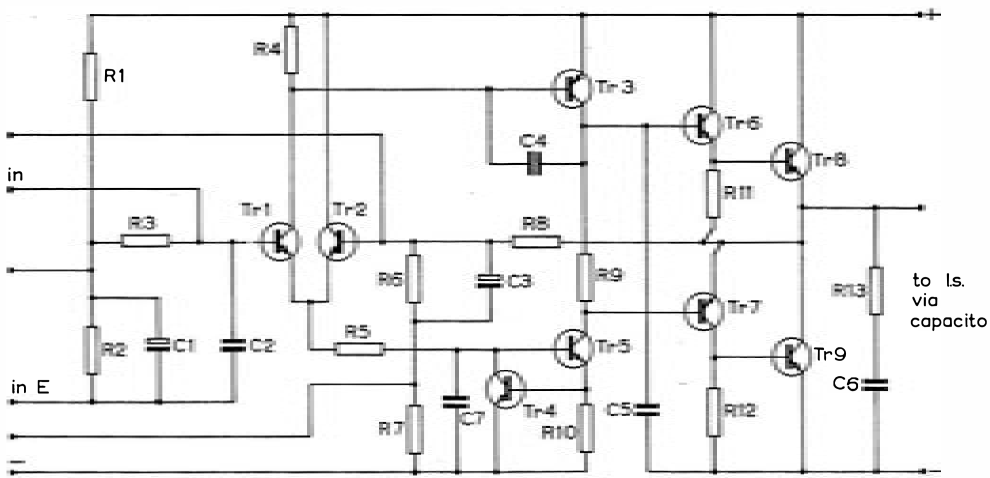


Fig 24. The schematic of a modern high-quality transistor power-amplifier module for Hi-Fi sound reproduction. Two complementary silicon driver transistors are used in a highly developed circuit using origin offset and constant-current drive with a large amount of negative feedback, to ensure that the nonlinear distortion at both high- and low-output levels is kept at very low figure; there are no transformers and the unit is accommodated on a compact printed circuit board.

designed to operate. Other reactances may upset feedback circuits operating over the output stage and cause ringing or even inaudible high-frequency oscillation. As many transistor circuits will operate efficiently up to radio frequencies, there is a risk of unsuspected thermal damage from this cause. If a separate external output transformer is used for any purpose, such as extension loudspeakers, earphones etc, and the circuit should happen to be broken whilst carrying a high signal level, the back-emf generated in the high inductance of the transformer winding may be large enough to cause voltage breakdown of the output transistors.

It is, of course, inadvisable to cause excessive surges by changing input plugs etc without first turning the volume control to minimum, as these may also damage loudspeaker diaphragms and suspension if the low-frequency response of the amplifier is unusually extended, as well as possibly initiating thermal overload conditions in the output transistors.

Practical amplifier considerations

We see that the big advantages of transistors lie in the enormous reduction in size and heat dissipation and in potentially increased life and reliability.

The transistor is basically a low-impedance bipolar current-amplifying device as opposed to the valve, which is basically a high-impedance unipolar voltage-amplifying electron tube. The term "bipolar" means that the current is conducted in a normal junction type transistor both by means of electrons and by "holes" or electron deficiencies within the crystal. A "unipolar" device conducts by means of electron movement only. All electronic valves and certain specialized semi-conductors (eg field-effect transistors) are of the latter type. We have noted that the inherently low impedance of semi-conductors as compared to the high impedance of vacuum tubes means that both input and output transformers can be largely dispensed with. This is a great advantage as regards size and cost and the small inherent non-linear distortion produced by any magnetic "iron" cored component is eliminated. This particularly applies to output transformers, where the distortion due to magnetic core saturation may be by no means negligible, unless it is offset by the application of a fairly large amount of negative feedback over a path including the output transformer and also over some or all of the power-amplifier stages. The distortion produced by input transformers is very much less, due to the minute current swing produced by a microphone or low-impedance pick-up.

Transistors have two major drawbacks from the point of view of the circuit designer: their operating characteristics are more nonlinear than those of valves, and the "production spread" or variation in characteristics from one transistor to another is also often much greater. This means that for very low noise input stage applications or where the maximum efficiency with low distortion is required in output stages, transistors may have to be specially manufactured or specially selected, with a consequent increase in cost.

Thus, in spite of the possible simplifications and the reduction in size, the cost of really high-performance transistor amplifiers may not be much less than that of valve amplifiers.

Performance specifications and functions of amplifiers

The basic function of an amplifier is, of course, to increase the minute amount of electrical power available from microphones, pick-ups etc to a value which will give satisfactory sound reproduction from a loudspeaker system. Subsidiary functions include switching, response equalization (tone control), volume control, impedance transformation (eg matching of a high-impedance device such as a crystal 'pick-up) provision for the elimination of unwanted non-essential signals from the reproduction (eg turntable rumble, surface hiss on old records, radio interference etc).

The specification and design of an individual amplifier to perform one specific job can be simplified and restricted to the bareest essentials required for efficient operation in the required manner. This might result in the cheapest possible design, but it might be found inadequate to perform other tasks for which the need might arise at some future date.

Thus it is usually found that, except in the very cheapest amateur-built designs and the cheaper commercial products for home sound reproduction, provision is made for some flexibility in the design with regard to such important parameters as the number and type of inputs accepted, possible variations of gain and response etc.

The power output of an amplifier must be decided, having regard to the efficiency of the loudspeakers, the size of the room and the type of programme to be reproduced.[3] To take two extreme examples, a bass-guitar amplifier would require a large power reserve in order to accept the enormous low-frequency transients fed into it and is normally required to be reproduced in a moderate-sized hall, whilst the reproduction of eighteenth-century chamber music in a small living room with an efficient high-flux loudspeaker would need only a comparatively small power output.

It is obviously sensible to allow an adequate margin in the power output of a given amplifier and, in fact, the reproduction of most types of speech and music demands a peak-power handling capacity very greatly in excess of that needed to handle all but the peak programme levels. For most domestic purposes a very general rule of thumb might be that 3 watts available power is about the minimum which should be considered for high-quality reproduction, whilst between 10 watts and 20 watts should be sufficient for all but the largest rooms.

The common reference to "available power" brings us to one of the first difficulties which the amateur encounters when trying to assess the relative merits of the various types of amplifiers which he can choose from. It is, in fact, difficult to specify the actual useful power which an amplifier will feed to a given loudspeaker under practical conditions. This power figure depends on a number of variable factors. One of the most important is the actual motional impedance value of the loudspeaker and its relative resistive and reactive values at any given frequency. This may be very different from the nominal impedance value quoted for the unit concerned. Difficulties of this sort have led to amplifiers often being rated in terms of the "steady-state" power dissipated in a resistive load equal in value to the nominal loudspeaker impedance which the amplifier is designed to feed. This is what is usually implied when an amplifier power is stated. The usual rating calls for amplifier power to be specified in this way. Other amplifier ratings are given which specify a peak programme music power rating which gives somewhat higher figures for most amplifier powers.

The above observations should be borne in mind when comparing amplifier specifications, together with the fact that a good power rating at middle frequencies does not necessarily mean that the performance is satisfactory at extreme low and high frequencies. The situation is further complicated by the fact that it is not possible to run some transistor amplifiers continuously at their full programme rated output, on account of overheating of the output transistors. Normally, there is no real risk of burning out the power transistors on programme because of the relatively enormous peak-to-mean-power ratio encountered on all usual programmes.

In fact, the design and full checking of modern efficient high-quality amplifiers demands a very considerable amount of expertise. Complete checking involves measurements of noise, pulse response and non-linear (intermodulation) distortion. The amateur obviously cannot carry out tests of this kind without having a lot of expensive professional test equipment at his disposal. He can, however, buy a good reputable amplifier or kit or build to a good published design with considerable confidence in the available power provided.

Amplifier controls are very important and their operation must be properly planned so as to be logical and convenient. This is an aspect where ideas vary and there is considerable latitude for experiment.

Typical high-quality amplifier specifications

An indication of good practice can be obtained by examining the design of representative power amplifiers, and preamplifiers. A monophonic or single channel chain is usually first considered. Stereo would demand duplication of the channels with probably a larger common power supply unit, though the power output requirements per channel could normally be approximately halved for domestic use.

A figure of about 12 watts for a monophonic power amplifier is generally found to be adequate for a good domestic installation. This power is considered to be the resistive load value, as measured.

The detailed specification of a preamplifier/control unit or a power amplifier in terms of performance figures gives a fair guide to the merit and capabilities of the equipment, but certain reservations should be borne in mind. Firstly, the exact meaning and validity of the figures depends on the method of measurement and the instruments used. The example of power-output measurements mentioned illustrates the point. The dilemma applies to other pieces of audio equipment and stems from the fact that speech and music represent extremely complex signals and thus instrumental (electrical) measurements are necessarily simplified and to some extent artificial. They have been framed to give as

Fig 25. A picture of the power-amplifier module of Fig 24. The printed circuit board is $2\frac{3}{16}$ by $2\frac{1}{2}$ in. The power output transistors at the top of the picture are mounted on a matt black heat sink large enough to dissipate the heat generated by moderate music powers. Larger powers may be dissipated if the heat sink is mounted intimately in contact with a larger metal chassis etc.

much information as possible, but cannot always supply a complete comprehensive picture of all aspects of amplifier performance. Any amplifier tests should therefore be backed up by special transient or square-wave tests and by listening tests on specially chosen programme material.

With these reservations in mind we might specify a typical high-quality amplifier channel as follows—

FREQUENCY RESPONSE AND STABILITY CRITERIA

Radio, tape and microphone input response should be ± 1 dB from 50 c/s to 15 Kc/s, ± 2 dB from 20 c/s to 50 c/s and from 15 Kc/s to 20 Kc/s, for flat response position of controls. Any response outside the extremes of 20 c/s and 20 Kc/s is not normally an advantage and may actually cause trouble by reproducing sub-audible turntable or microphone rumble etc below 20 c/s and radio carrier or bias oscillator leaks etc above 20 Kc/s. Although not directly audible, interfering signals of this sort can cause considerable trouble by overloading various parts of the system and possibly causing severe whistles or intermodulation distortion, in addition to introducing a risk of overheating transistor output stages etc. However, the desirable frequency response of an amplifier in the top cut-off region is intimately associated with its phase response, that is the extent to which the angle of the signal vector lags behind or leads the signal at lower frequencies. If an attempt is made to give an amplifier a very sharp response cut-off immediately outside the audible range this is bound to cause a violent change in the phase response in the cut-off region. The effect is to cause a damped oscillation to persist at the frequency in question. This will be excited by any electrical transient and may lead to "ringing" or a semi-continuous oscillation. A slightly more severe phase shift may in some cases actually produce a positive feedback path around some path of the amplifier. If some gain due to a tube or transistor etc occurs in this path, continuous oscillations will build up and the amplifier is then "unstable." A very complete circuit theory has been worked out for amplifiers involving negative feedback. This theory has a fundamental bearing on high-quality amplifier design, in that the application of a fair amount of negative feedback is essential in order to reduce nonlinear distortion to a low level as well as to control the frequency response. The theory shows that if a polar plot of the gain and phase characteristic around a circuit loop is carried out, the system will be stable unless the polar curve encloses the point (+1, 0). This is the well-known "Nyquist criterion," which determines the stability of all feedback systems, including all types of electronic and other amplifiers, and also electro-mechanical servo-systems such as high-speed voltage recording test instruments (eg Brüel and Kjaer level recorder) or large controlled dynamic structural vibration-exciting equipment. If the transmission properties of the path vary with signal level so as to cause the Nyquist plot to enclose the critical point on overload etc but not for normal signal levels, the system is only "conditionally stable." A good amplifier is stable under all conditions of load or temporary overload and is said to be "unconditionally stable." The rate of cut-off of the response outside the pass-band is bound up with the phase response and, in general, should not have too rapid a fall. If steep-cut filters, cross-over networks etc are desired, these generally should be introduced between amplifier or buffer stages so that they are not embodied in any part of a feedback loop. The whole system must still be carefully investigated with the aid of square waves or other transients in order to see whether any appreciable amount of "ringing" has been introduced. The application of large square-wave pulses at various repetition rates or frequencies and examination of the output waveform will show if the amplifier is running into oscillation on overloading peaks, ie is only "conditionally stable." A good check of stability for an amplifier with an output transformer within the feedback path (ie negative feedback taken from the loudspeaker or output terminals) is to apply a 30 Hz square-wave input signal large enough to produce a degree of overloading. The output waveform should not exhibit bursts of higher frequency oscillation at any part of the low-frequency cycle.

Enough has been said in the preceding paragraphs to show that the design of a high-quality amplifier to have a particular frequency response is not a simple matter and that a number of quite sophisticated tests should really be applied to see that the amplifier is stable under all conditions, in addition to having the desired frequency response.

When the negative-feedback loop of an amplifier is being designed or adjusted, various "tricks of the trade" can be used in order to avoid the necessity of analysing the circuit in detail on paper with full knowledge of all relevant stray or leakage circuit elements, or of using expensive professional equipment such as a Nyquist diagram plotter. One artifice is to open the feedback loop at a convenient point and measure the overall frequency response around the loop. The rate of fall of the response above the cut-off point should not usually greatly exceed 12 dB/octave for a good stability margin to be achieved.

GRAMOPHONE INPUTS ETC

These should be equalized to within $\pm 1{\cdot}5$ dB of the BS or RIAA standard record equalization curves for $33\frac{1}{3}$, 45 and 78 rpm on the assumption that pick-ups with ideal frequency response are used. The appropriate input selector switch positions should be handy and clearly visible.

BASS AND TREBLE CONTROLS

These controls should be arranged to give about ± 15 dB maximum lift or loss at both extreme bass and top frequencies. The response between the "starting points" of the boost or loss areas on the response curve should be flat and ideally the starting points should progress along the flat part of the curve according to the severity of the boost or cut provided. The well-known circuits due

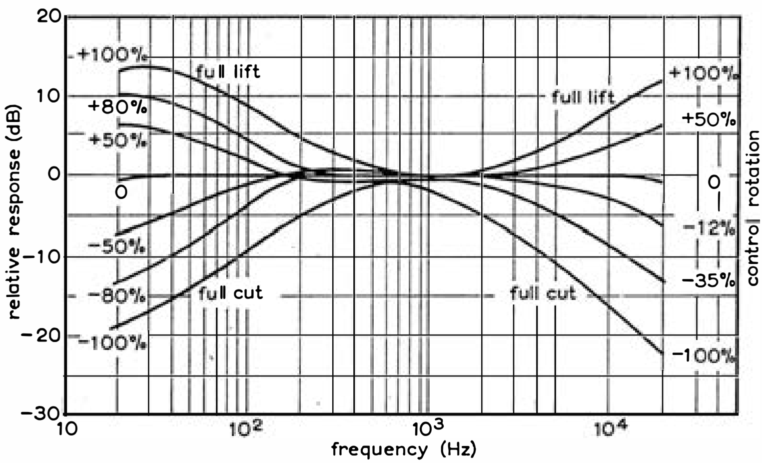


Fig 26. The range of independent continuously adjustable bass and top lift and cut tone control available with the well-known Baxandall passive tone control circuit which may be used between two stages of a pre-amplifier.

to Baxandall have this desirable characteristic, which avoids undesirable exaggeration or deficiency of middle, low or high frequencies when control of extreme bass or top is sought. The bass and top positions should either be switched in an adequate number of discrete steps or be controlled by accurate and easily set continuously variable controls. The central flat position should be well defined and easy to find. Fig 26 shows bass and treble control responses of this kind.

RUMBLE FILTERS

It is not desirable to reproduce sub-audible frequencies due to the presence of turntable rumble, wind or breath surges on microphones etc. In order to get a steep and effective low-frequency

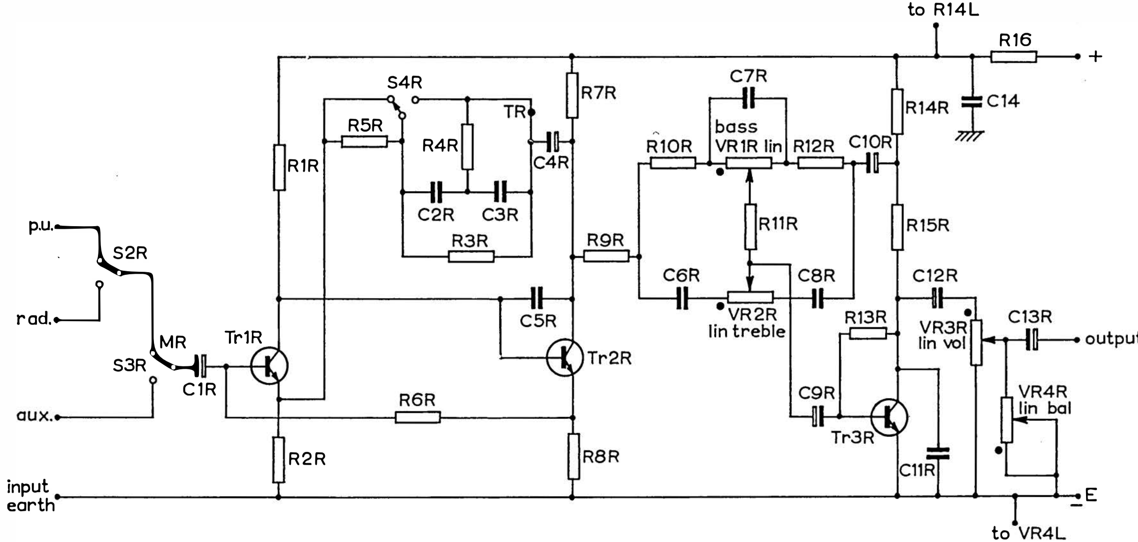


Fig 27. The schematic of a modern stereo pre-amplifier (one channel shown). The RC network between the first two transistors corrects the output of magnetic velocity-reponsive pick-ups for the standard recording characteristics of gramophone records. Continuously adjustable bass and top-tone control is provided by the potentiometers in the other RC network, which is sited between the second and third transistors. A stereo balance control is provided after the output volume control. All variable controls in stereo pre-amplifiers have to be ganged and matched to track accurately together.

cut-off at sub-audible frequencies without affecting audible low frequencies, it is necessary to use a multi-element scientifically designed and correctly terminated filter. This is liable to use relatively large and expensive inductances and capacitors. The former may also have to be carefully shielded in high-permeability screens of "Mu-metal" or Permalloy in order to avoid hum pick-up from the electro-magnetic stray fields radiated by mains transformers etc. As it does not incorporate amplifying elements

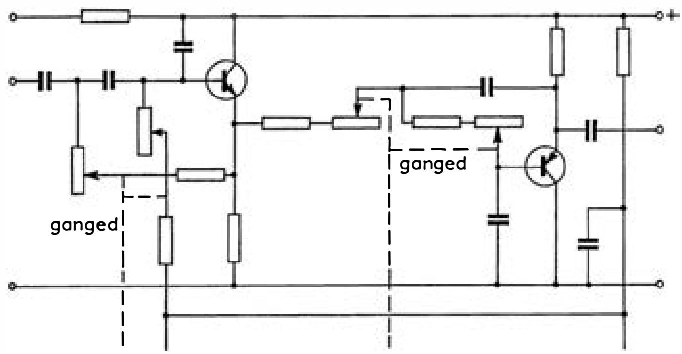


Fig 28. The schematic of a stereo two-stage active filter unit (one channel only is shown). This may be connected between the stereo pre-amplifier and the power-amplifiers, so as to give independent bass and top steep-cut filters to enable low frequency rumble and surges to be eliminated. A top steep-cut filter will reduce residual scratch, hiss and higher-order distortion products. The two sets of controls for the channels are accurately ganged.

such as valves or transistors, this is a "passive" type of filter. A high-class filter of this kind can be permanently connected in circuit as the full audible bass response is retained. For more compact and less expensive amplifiers an altogether simpler form of rumble filter is preferable. An RC network incorporated in a negative-feedback loop around one or more amplifying stages can be made to give attenuation of 12 dB per octave below about 70 Hz, without much "roll-off" at higher bass frequencies. So-called "active" filters of this kind are usually arranged to be switchable "in" or "out" and are often referred to as high-pass filters. Fig 28 gives typical circuits for the two types of filter used, ie rumble and top-cut.

TAPE PLAYBACK EQUALIZERS

Standard tape-recording characteristics have been laid down by the CCIR (the European radio standardizing body). These characteristics differ for the various speeds 15, $7\frac{1}{2}$, $3\frac{3}{4}$, $1\frac{7}{8}$ inches/sec. If suitable equalizers are incorporated in an amplifier in addition to gramophone equalizers, the amplifier can be used directly with a tape reproducing-head for tape playback or for tape-recording monitoring from a separate head as a check whilst tape-recording is actually in progress (Fig 21). The $7\frac{1}{2}$ and $3\frac{3}{4}$ inches/sec tape-head equalization characteristics do not differ enormously from gramophone equalization characteristics and in emergency these may be used following a tape replay-head if some sacrifice in response is accepted. The proper equalization is normally an integral part of tape-recorder playback circuits, which produce a flat output.

TOP-CUT FILTERS

These are designed to give fairly steep high-frequency cut-off characteristics in order to deal with programmes containing an intolerable amount of high-frequency noise, hiss or distortion. They are also referred to as low-pass filters or steep-cut filters. As in the case of rumble filters, conventional multi-element passive filters may be introduced between amplifier stages to give high rates of cut-off. As we have seen, if the rate of cut-off is too

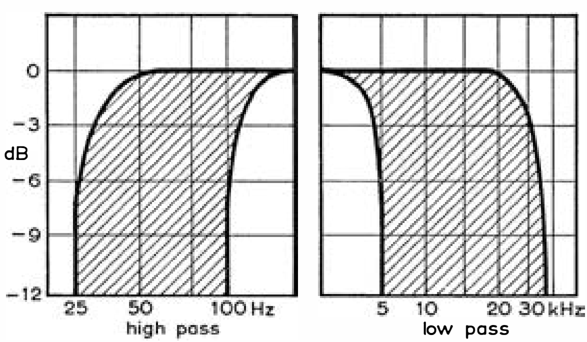


Fig 29. The range of responses available from the active filter unit shown in Fig 28. The shaded areas show the range in which the steep-cut filter characteristics are independently adjustable. In the widest response position of the controls, the system is flat between 40 Hz and 20 kHz, the narrowest response being between approximately 120 Hz and 4·5 kHz.

high, "ringing" or severe transient distortion near the cut-off frequency may be introduced and so in practice the rate of cut-off is likely to be restricted to about 18 dB/octave maximum. RC networks used as feedback elements in an active filter circuit are now used to provide top cut filters and will readily produce cut-off rates of 12 dB/octave or more, which is often sufficient for any but the most severe hiss, even from elderly shellac records etc. Popular values of nominal cut-off frequencies for switched filters are 4 kHz, 7 kHz, 10 kHz and 12 kHz. In some cases, the cut-off frequency or turnover is infinitely variable over a similar range by using a potentiometer as one of the resistance elements in an active filter circuit.

Equalization characteristics for older records

The majority of gramophone records made since 1954 are recorded to the standard pre-emphasis curve described by the three time-constants 3180, 318 and 75 microseconds for $33\frac{1}{3}$ and 45 rpm records. 3180, 450 and 50 microseconds were used for some 78 rpm records. For records issued before 1954, various recording characteristics were used and some trial and error may have to be used for best reproduction. A number of earlier records are recorded to various specified curves and some makers have published correct equalization characteristics for makes and labels issued prior to 1954, but they warn that shared or otherwise unspecified masters may have been used in some cases.

Parallel operation and unconditional stability of amplifiers

We have noted some of the factors affecting the stability and frequency response of amplifiers and the manner in which negative feedback has a bearing on the amplifier characteristics. Virtually all valve power-amplifiers, as distinct from transistor power-amplifiers, have to use an iron-cored output transformer in order to match the relatively high-impedance valves to low-impedance speech-coils. If amplifiers are to have closely controlled response, gain and distortion characteristics, a fairly large amount of negative feedback has to be applied over several amplifier stages, including the output transformer. This places very stringent requirements on the design and construction of the output transformer with regard to the phase shift introduced by circuit elements such as leakage inductance and winding stray capacity. It is possible to minimize these factors by subdivision of the various windings and by ensuring exact symmetry in winding and connecting the various coils. A good power-amplifier is stable with all types of load and a number of amplifiers may sometimes be operated in parallel without risk of instability if a larger power output is required. Generally, however, it is not advisable to attempt to run amplifiers in parallel in order to get more power output unless they have been designed and tested with this in mind.

Stereo amplifiers system matching

Ideally the two stereo chains should be accurately matched as regards gain, phase, frequency characteristics etc. Overall gain should generally be matched to better than 2 dB. Commercial stereo amplifiers are provided with accurately matched and ganged controls for switching, bass, treble, volume etc. In addition a balance control is provided giving a fine control of the relative gain between the two channels. Some amplifiers have been produced with dual concentric knob controls, so that the characteristics of the two channels can be individually varied at will. A phase-reversing switch to give reversal of polarity on one channel used to be a necessary feature in the early days of stereo records when record channel phasing was not standardized. This facility is still sometimes provided on amplifiers, although once correct phasing has been established it should not nowadays be necessary to use a reversing switch.

Stereo listening on earphones

The use of matched high-quality earphones for binaural listening to stereo programmes can be very well worth while, and earphone outlets in the form of special jack sockets or DIN outlets etc are often provided on amplifiers. Moving-coil earphones are often wound to about 300 ohms and may be shunted across loudspeaker outlets of 3 to 15 ohms, dummy-load terminating resistances preferably being provided if the loudspeaker is disconnected. Ganged earphone volume controls and "cross-mixing" facilities are additional refinements which may be added for earphone listening. Cross-mixing circuits are sometimes provided to dilute the left-right stereo separation, which some people may find to be excessive on stereo earphone listening.

Note that safety precautions may be necessary when connecting earphones to equipment. Again, a dealer should be consulted.

7

Loudspeakers and enclosures

The loudspeaker is one of the most important items in the reproducing chain. In addition to its electro acoustic role, it must be considered aesthetically as something which is necessarily an important item to be fitted in your furnishing scheme. Where stereo is concerned, the problem may be even more difficult to resolve. Thus the loudspeaker is one of the most difficult units to choose when deciding on a reproducing system.

It is, of course, possible to go for a monolithic system in the form of a single cabinet which houses the loudspeaker as well as the amplifier, turntable, tuner etc. Most enthusiasts, however, will prefer to have their loudspeakers housed in individual cabinets, as this allows the maximum flexibility in both performance and placement of the loudspeakers.

Very few will wish to attempt to construct their own loudspeaker motor units, though this is not impossible. Many will certainly wish to make their own enclosures or cabinets, as this can be both satisfactory and economical to undertake. It is then first necessary to decide on the type of loudspeaker units to use. These range in price literally from a few shillings to scores of pounds, and an optimum choice for any given projected system can prove very difficult to make.

Not only are loudspeakers difficult to apply correctly, but they are probably more prone than any other part of the system to introduce subtle forms of distortion and coloration of the sound. These are not only generally undesirable in themselves but may be very difficult to predict from response curves and casual listening tests. The art and practice of loudspeaker measurement is not generally as advanced as that of amplifiers and other parts of the system, and it is extremely difficult to make reliable comparisons between loudspeaker units, particularly at different times and different places.

It is pertinent, therefore, to list a few guiding principles when first considering the choice of a loudspeaker unit.

Choosing a loudspeaker unit

In the first chapter we discussed the general way in which loudspeakers perform in rooms and obtained some idea of what to expect from the various types. Later in the present chapter we will go more deeply into the design and construction of various types of loudspeaker and cabinet. It is, however, possible to assess and to select the type of loudspeaker best suited to one's own particular needs by considering a number of general rules. Once the field has been narrowed, it then becomes easier to decide between the remaining alternatives. Even when the more technical details of performance are available in the form of response curves, distortion characteristics, transient response criteria etc it is still extremely difficult to arrive at a comprehensive "figure of merit" for a loudspeaker. The final choice for most people is thus to some extent a personal decision based possibly in part on cost, on such listening tests and assessments as one is able to make and on the recommendations of reviewers, experts and friends.

Firstly, one should consider the exact purpose for which the loudspeaker is required, for example—

1 Low, medium or large power output. This is only partly dependent on the size of room. The dominant type of programme and use should be considered. For example, electrical instruments such as electric guitars, electronic organs etc are liable to give rise to sustained low-frequency notes of such power that the loudspeaker units must withstand larger cone excursions and speech-coil heat dissipation than normal.
2 Whether the unit is to cover the whole frequency range or only a part of it. The latter category will include bass woofers, mid-range units, high-frequency or ultra-high-frequency tweeters.
3 The type of cabinet or enclosure in which the unit is to be housed.
4 The impedance of the unit and the way in which it is to be mounted and electrically coupled to any other loudspeaker units in the same array.

Additional decisions involve the operating principle and whether the unit is a direct radiating system or horn-coupled. The great majority of loudspeakers today are direct radiating moving-coil types, with the exception of wide-range electrostatic units such as the Quad. Ribbon, electrostatic and ionic tweeters are also available. Electrostatic and most moving-coil tweeters are direct radiating types, but the other two types mentioned above are horn-coupled.

A final decision is the basic efficiency desired. In some units this is fixed by the designer but many of the more widely used direct radiating moving-coil units are available in low, medium and high magnetic flux densities. In these days of highly efficient transistor amplifiers, it is not so important to strive for the utmost in efficiency as it used to be, and medium-flux units are often quite satisfactory (about 9,000–11,000 gauss). Units with the maximum flux density will generally have slightly better damping and a better transient response, with the result that they are often specified for the highest quality.

The cost of moving-coil units depends on the size, the complexity or any special treatment of the diaphragm system, the flux density and, a most important factor, the quantity in production at any given time. Hence lower-flux units for the mass-produced radio and TV market may be obtained quite cheaply. These may sometimes be used successfully in high-quality arrays, particularly when they are only required to cover part of the frequency range. However, it must be borne in mind that the response characteristics of these units are often tailored to suit a fairly restricted frequency range. A fairly high bass resonance is usually provided, partly as matter of economics, in that it is easier to provide robust cone surrounds on a mass-production basis than to make the necessarily more critical flexible edge rolls or other forms of termination required for the highly compliant cone systems which must be used in efficient "long-throw" loudspeaker units combining the properties of reasonable size and a low bass resonance. It is also undesirable to reproduce extreme low frequencies in the cheaper radio and TV sets in order to avoid placing too stringent a requirement on the level of mains hum, induced frame pulses, acoustic feedback to turntables and pick-up cartridges etc. Sideband-cutting may also sometimes be compensated by a broad resonance between 3 and 6 kHz in AM receivers, whilst inter-station carrier whistles may be reduced by means of a dip in the loudspeaker response at around 9 kHz.

In high-quality systems, broadcasts are preferably received on

VHF FM, and optimum electrical circuitry and loudspeaker cabinet arrangements are used; it is obviously desirable to use specialized loudspeaker units designed to produce a flat response over the widest possible band. The cheaper types of mass-produced units obviously have limitations in these respects, unless they are very carefully applied so as to utilize the flatter parts of their frequency range.

Generally speaking, the same axiom applies to loudspeakers as to other transducing elements, such as microphones or gramophone cartridges. This is that one should, in general, spend the maximum one can afford on these items because the range in performance between the cheapest and the most expensive is almost certainly much greater than that applying nowadays to other items such as amplifiers, cabinets etc. Furthermore, it is possible to save much money by constructing these latter items oneself, which is emphatically not the case with transducers unless you are exceptionally well qualified and equipped.

When deciding on the purchase of loudspeaker units the problem that is most difficult to resolve, apart from power handling, is probably the question of exactly how far one wishes to go in the reproduction of extreme low or extreme high frequencies. Assuming that you have considered the power-handling capacity of your amplifier at the frequency extremes, it is then necessary to examine the implications both in size and cost of providing a given extension of frequency range at either extremity.

It may, in fact, be debated how far it is necessary or desirable to reproduce the lowest and the highest octaves at the limits of audibility. For present purposes we may define the lowest octave as 30–60 Hz and the highest as 10–20 kHz. By reproduction, we imply reproduction at full strength. It must be borne in mind that really full low-frequency reproduction of this kind can add considerably to the size and cost of the loudspeaker units and the cabinets, to say nothing of the more stringent power output requirements of the amplifier. Full high-frequency rendering of this kind probably means not only using special tweeters but may also involve special measures to ensure adequate sound distribution or polar response.

There is no doubt a case for providing such a response if perfect reproduction of sound is to be attempted, but in practice, on a great many programmes and systems, the amount of sound energy present at these extremes may be small and, moreover, there may be undesirable extraneous or intruding noises which are so disturbing as to make it necessary to restrict the frequency range of the final reproduction.

Limiting factors in wide-range loudspeaker reproduction

If the bass response is extended to be really flat to the lowest audible frequencies, it is true that there may be enhanced enjoyment of certain musical sounds if, in fact, these are present in the incoming electrical signal. Actually, sounds of extremely low pitch are liable to be obscured by unwanted acoustic surges, due to the effects of draughts and air currents on microphones, residual traffic rumbles etc, and hence this part of the spectrum is liable to be removed prior to transmission or recording, so as to avoid over-modulation and distortion. In our own equipment, when reproducing records, turntable rumble and vibrational excitation of the very low frequency resonance between the tone-arm mass and the small restoring stiffness of a high-compliance cartridge may also generate an output of extreme low frequency. If this is reproduced at an audible level, we would normally resort to a bass-cut rumble filter, typically giving an attenuation of 12 dB per octave below various nominal cut-off frequencies switched to occur at 70 Hz and below.

Thus, for various reasons, most people do not in fact listen over long periods to extended low frequencies. The question one has to consider is whether it is worth the considerable cost and elaboration of having such a loudspeaker system. The answer is generally some form of compromise, with the smaller types of bookshelf loudspeaker at one end and the necessarily larger and more expensive loudspeakers at the other end of the scale. The efficient reproduction of extreme bass is costly in money and in space. The user has to decide whether it is worth it, and, if not, just how much bass he does wish to have available from his loudspeaker system.

At the high-frequency end of the spectrum similar arguments may apply, in that it is often necessary and desirable to cut part or all of the upper two octaves, so as to minimize the often very considerable annoyance caused by residual higher order distortion products and high-frequency noise and hiss from a wide variety of causes, ranging from radio static to thermal and other noise in electrical circuits, to say nothing of magnetic-tape noise and record hiss.

However, given a really clean programme signal, there is no doubt that an extended high-frequency response to a point well above the normal pure-tone audible limit is essential to give completely realistic reproduction to transients and other sounds rich in high-frequency components.

Really good programme conditions of this sort are increasingly available today in FM reproduction of live broadcasts in areas of good signal strength, and in both tape and disc recordings using modern methods of noise reduction and means of controlling the recorded level.

The provision of ultra-high-frequency tweeters is not particularly expensive in terms of cash or space, and so it seems reasonable to consider the provision of a flat and extended high-frequency response, as long as this is balanced by a reasonable low-frequency response.

Polar response considerations

One further parameter to be considered when choosing loudspeaker systems is the type of directional response which can be obtained from different designs. Enclosed types of cabinet loudspeakers are basically omnidirectional at low frequencies but the units become directional at high frequencies where the wavelength of sound is small compared to their size. Tweeters thus require to be small in order to give a well-spread beam of high-frequency sound.

Open-backed loudspeakers, such as wide-range electrostatic units, are fundamentally bidirectional with zero output in the plane of the diaphragm. In fact, it has been claimed that a good stereo effect is obtained if the figure-of-eight polar response is maintained up to high frequencies, together with a less critical listening position.

On the other hand, it may be found that the positioning of bidirectional loudspeakers is more critical in some rooms, and this should be borne in mind as a possibility when considering their choice. If in doubt, it may be possible to arrange a demonstration in your own listening room. This should not be hurried, as the excellence of the results which may be obtained justifies taking trouble to find the best siting and also listening to a wide range of programmes over a period of time.

In fact, this applies to listening tests on any new loudspeaker system. Established listening habits die hard and one may have to live with a new system for a while before appreciating the results completely.

The moving-coil loudspeaker unit

We have considered some of the main factors involved in the choice and use of the various types of loudspeaker likely to be used in home sound reproduction. We will now consider the design and the basic mechanism of loudspeaker units in more detail. The moving-coil loudspeaker driving unit is by far the most popular type in use today, but we have noted that other principles are used. The push-pull double-sided electrostatic system is used successfully for high-quality wide-range loudspeakers, and to some extent as tweeters. Ribbon loudspeakers are also used as tweeters, as also are ionic loudspeakers. It may be noted that the driving force in the case of moving-coil units is on the speech coil only and thus is at the apex of the cone. In the other types mentioned the driving force operates over the whole active radiating area, in other words, over the whole of the current-carrying ribbon or over the whole of the charged diaphragm surface of the electrostatic unit. In the ionic unit the force is generated due to thermal expansion of the electrically charged air particles in the sound-originating cell.

The general principles and constructional layout of the almost universal moving-coil unit have not altered very much since it was first introduced some forty-five years ago. The driving force is due to the direct interaction between the signal current and a constant radial magnetic flux in the gap and it is basically free from non-linear distortion. A conical or flared shape is given to the diaphragm in order to make it as light and as rigid as possible, so that the efficiency of sound radiation may be kept as high as possible. A flexible edge surround as well as a radial centring "spider" or a bellows member are usually required to keep the coil centred in the gap.

The exact construction of the various types of moving-coil loudspeaker unit naturally depends to some extent on the size, the frequency range covered and the type of magnet used.

Several typical units are illustrated in this section of the book. In the case of the electrostatic loudspeaker, the diaphragm does not have to be rigid and thus can be both light in weight and of a large area. In contrast, the cone of a moving-coil loudspeaker has to be relatively small in order to meet the need for rigidity, about 18 in. being the maximum size for bass units; wide range units seldom being more than about 12 in. effective diameter.

Modern units use cast or sintered permanent magnets producing a flux density of between 7000 and 17000 gauss in the gap. A die-cast or pressed-metal chassis is attached to the magnet and is extended forward to support the diaphragm surround and to form the mounting ring for attaching the loudspeaker unit to the cabinet front. The diaphragm is usually made from a special felted paper composition or, in some cases, of a light metal, a reinforced rigid foamed plastic or a solid plastic material. Speech coils are wound from copper or aluminium wire on varnished paper or plastic formers.

The moving system at its simplest may be considered as a rigid mass suspended on a restoring spring. An alternating driving current will give substantially the same driving force at all frequencies, but resonance will occur between the mass and the spring at a frequency normally arranged to be at the bottom end of the frequency range covered by the loudspeaker. Above this point the mass effect increasingly predominates over the spring as the frequency rises. The sound radiated from a rigid cone can be calculated by taking it to be equivalent to a flat circular piston mounted in a large flat baffle. The result shows that the acoustic radiation load offered to such a piston increases with frequency up to a distinct cut-off point. The air moved by the surface can be considered as having both mass and resistance components. The mass component adds slightly to the diaphragm mass and represents kinetic energy stored in the air at points in the cycle where the velocity is at a maximum. Some of this kinetic energy is returned and stored as potential energy in the spring at points representing spring deflection. The resistive component represents the sound energy radiated and is the useful component producing the sound output. The sound energy radiated by such a mass-controlled piston should be substantially constant between the bass resonant frequency and the top cut-off frequency. The response on the axis is slightly improved at higher frequencies because the sound energy becomes concentrated along the axis at the expense of off-axis sound. It is evident, however, that a strictly rigid diaphragm would not be able to cover the full frequency range, but in practice very few cones are completely rigid and in fact most are given a degree of controlled flexibility which allows a number of subsidiary break-up modes of vibration to occur in addition to the main bass resonance. Ideally it is possible to extend the response well beyond the theoretical top cut-off point, but obviously at some frequencies parts of the diaphragm will move mutually in-phase whilst other parts may move in anti-phase. The result is likely to be a response showing a number of peaks and dips. The whole effect is very complex, but with good design a satisfactory wide-range response may still be obtained.

When deciding the size of cone to be used, we note that a large cone can move a given volume of air with a smaller excursion and, with less nonlinearity, but it may rely to a greater extent on higher modes in order to cover the frequency range, and it may become excessively directional at high frequencies. Ideally, the size should be tailored to the frequency range the unit is required to cover. This is, in fact, what we usually find in the case of high-quality multi-unit loudspeakers, where we have woofers, mid-range units and high-frequency tweeters in a descending order of size. One additional point, however, is that for a given output a larger cone is liable to require a larger size of cabinet with a consequent increase in cost. This has lead to an improvement in the performance of smaller mid-range and wide-range units, particularly as regards the permitted excursion of the cone. Edge surrounds and centring bellows behave as nonlinear restoring springs at any excursions exceeding a small fraction of an inch, unless very special precautions are taken in their design. Also turns of wire on the speech coil which move beyond the limits of the pole pieces encounter the reduced fringing magnetic flux and again tend to introduce nonlinearity into the force on the coil. This can be minimized by careful proportioning of the pole pieces and by such devices as making the coil shorter than the gap depth. An idea of the magnitude of the problem is obtained when it is realized that a small cone may have to move through a total distance of ½ in. to reproduce a bass note at full concert hall strength in a large living room. The effect of nonlinearity is, of course, not only to produce harmonics of single tones but also to produce much more objectionable combination tones when many frequencies are being reproduced simultaneously, as in normal programmes. Additionally, sub-harmonics as well as harmonics and combination tones may also be produced by flexing of the cone material. The overall result is that the actual value of nonlinear distortion varies widely at different frequencies and can generally only be given conveniently in the form of a continuous curve. Another effect, known as Doppler distortion, occurs when a loudspeaker is simultaneously reproducing strong low- and higher-frequency tones. The high frequency will be increased or decreased in pitch according to whether the cone is moving back or forwards at the low frequency. This form of distortion is inevitable in wide-range loudspeakers but is fortunately not usually very severe. It is greatly reduced if the range is divided between a number of units.

The construction of moving-coil units

We have seen that in detail this is determined by the cone size and shape, by the magnet and, to a large extent, by the permitted cost of the unit. Cones may be circular or elliptical in plan with a "curvilinear" or a straight-sided section. Large or small integral-edge corrugations may be used or large single-edge rolls may be cemented on to the cone edge. Alternatively, foam or fabric edge materials may be used. Integral-edge corrugations may be treated with viscous compounds in order to add damping and to absorb the mechanical waves traversing the cone, and thus to prevent "standing waves" being set up along the cone section. The centring bellows are usually made as air-tight as possible so as to exclude dust from the outer coil gap. A fine-mesh dome made from stiff gauze is generally fitted over the coil end at the apex of the cone so as to seal the inner gap. This gauze allows some air passage so that the air inside the gap is not completely sealed off, as this might introduce an unwanted stiffness or might cause trouble due to atmospheric changes.

Magnets may be of the fully-columnar high-energy-content cast type, or may be of the large, flat ceramic ring type. The latter are cheaper but are larger in volume for a given gap flux density. Speech-coils are wound to the various standard low-impedance values (3, 8 or 15 ohms) and are connected via the rather critical flexible connecting leads. These must not fatigue or give rise to buzzes, and used at one time to be a considerable source of trouble. Some high-quality units make connection to the coil by means of a long flexible metal spring centring spider (Fig 30).

It is, of course, essential to ensure that the magnetic gap is permanently and accurately centralized. The older pole piece assemblies relied on screws, staking or welding, a special centralizing brass cup often being provided to ensure that the gap does not become eccentric if the unit is dropped or distorted. Such a

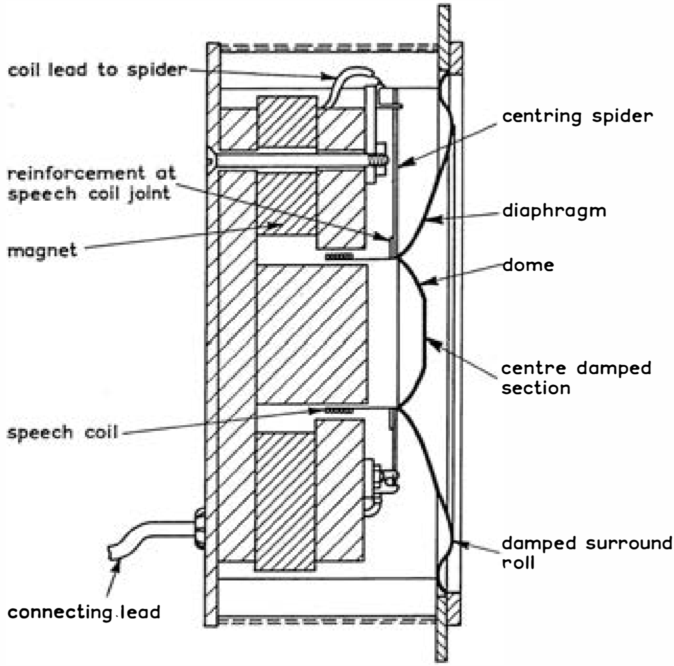


Fig 30. *A highly developed modern small wide-range loudspeaker unit. A high-flux ceramic magnet is used and considerable precautions are taken to avoid mechanical or magnetic nonlinearity. A curved titanium cone is used with a highly compliant surround which permits large excursions in relatively small enclosures.*

magnetic system is in a state of unstable equilibrium, the tendency being for the force to pull the centre pole over in the direction of any small initial eccentricity. Epoxy resins are now largely used to secure poles and magnets together with a strong and permanent bond, ceramic magnets being particularly suitable for securing in this way because of their large surface area.

The electrostatic loudspeaker

It is known that the modern wide-range electrostatic loudspeaker may achieve a substantially linear driving force over a large flexible diaphragm mounted between rigid perforated electrodes closely spaced on each side of the diaphragm. The theory shows that the diaphragm must not only be polarized via a high resistance from a high-tension supply of up to several kilovolts, but it should also have an optimum value of surface conductivity. The ideal is to maintain a constant value of charge on the diaphragm during operation, without any tendency for the charge to migrate or move about on the surface during the AC cycle of operation. The precise manner in which the various requirements have been met in practice has been arrived at by many years of development by the manufacturers and undoubtedly represents a triumph of both research and the scientific application of materials. The achievement of mechanically non-resonant driving electrodes with correct acoustic transmission properties, as well as the permanent sealing of the whole system against the entry of dirt or moisture, is a considerable advance.

The loudspeaker is almost a pure capacity as far as the amplifier is concerned and it is necessary to ensure that the power amplifier is designed to work into such a load without instability or ringing on transients.

Fig 31 shows the very successful high-quality wide-range loudspeaker made by Quad Ltd. This is a fairly large bidirectional

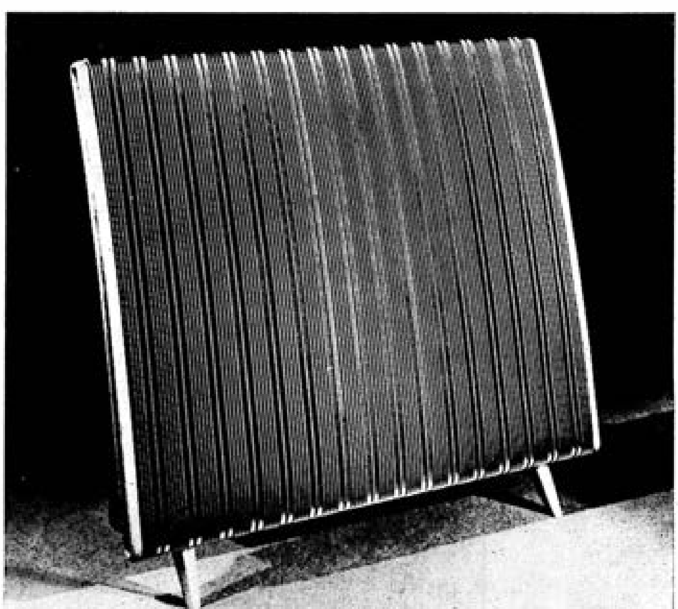


Fig 31. *The Quad electrostatic loudspeaker: a well-known wide-range unit of very high quality.*

unit which radiates a full bass response without the use of any subsidiary baffles or enclosure. Normally the higher frequency radiation of electrostatic units must be restricted by cross-over networks to a smaller narrow vertical strip so as to avoid undue focusing of high frequencies.

The extra complication, including the use of a relatively high-voltage polarizing supply, has so far restricted the use of electrostatic loudspeakers to the higher-quality systems, though the possibility of permanently polarized "electret" materials becoming a practical proposition may remove some of the expense and complication.

The Ionophone

This is a very interesting invention as it represents a unique form of loudspeaker without any solid moving parts, as movement is directly induced into the air itself by means of a quartz cell at the throat of the horn, which is an essential part of this form of loudspeaker. The air in the cell is ionized or electrically charged by the application of a modulated RF high-voltage signal at a carrier frequency of 27 MHz. The effect is to cause variations in the heat of the air at the rate of the audio-frequency signal which modulates the carrier. The result is to send sound-pressure wavefronts into the horn, which is matched to the outside air, so as to increase the sound volume in the usual way.

This is almost the only audio transducer in use which possesses no mechanical moving parts, with the result that the initial response is almost perfect, the properties of the horn representing the only limiting factor. The principle operates over the whole audio range and into the supersonic frequency range, the practical limitation at low frequencies being the size of the horn required subject to the limit of thermal linearity of the air in the cell. Messrs. Fane produce a very high-quality tweeter which operates from 3500 Hz to well over 20,000 Hz. It has an output which will accept the music power of a 20 watt amplifier in that range.

It must be noted that the electrical system of this type of loudspeaker is similar to the output stage of an AM radio transmitter, and thus it has to be carefully shielded in order to prevent unwanted external radio transmission. The Fane unit is claimed to be well shielded and, in addition, the 27 MHz band is chosen as representing an acceptable part of the radio spectrum for such purposes.

Moving-coil tweeters

We note that these may be horn-coupled or direct-radiating types. For domestic purposes where large amounts of power are not usually involved, the direct-radiator moving-coil tweeter has a number of advantages. The construction can be simpler and cheaper than that of a horn unit and the mounting in the main cabinet used for the other loudspeakers can be both simple and effective, provided that certain precautions are taken. The cabinet front is normally almost 1 in. thick. If a hole is bored in this and the tweeter mounted behind it in the same manner as for other loudspeaker units, a cavity resonance may occur in the high-frequency range and may produce audible irregularities, in the form of a peak followed by a dip at a higher frequency. Thus it is convenient to fill up the cavity if the tweeters are small enough to be mounted partly within the thickness of the front panel. It is also, of course, desirable to mount the tweeters as close as possible to the main loudspeakers so as to avoid any unnatural effects due to noticeable separation of the high- and the lower-frequency components of a sound source. Small tweeter units have sometimes been mounted in front of and within the cone cavity of the main moving-coil loudspeakers. The rear side of the tweeters has to be closed off from the sound radiation of the main cone, so as to make the tweeter a stiff sound source and to prevent it being driven acoustically at lower frequencies, with the result that it is difficult to make the unit small enough to avoid an appreciable obstacle in front of the main cone, which may cause irregularities in the frequency response. Co-axial mounting of a tweeter in this way can be successful if the unit is small enough to become effectively a part, or an extension of the main unit centre pole. Generally, however, it is easier to mount a tweeter separately within about 4 in. of the edge of the next lowest unit covering the frequency scale.

Fig 32 shows a modern ultra-high-frequency moving-coil tweeter which is designed to cover the frequency range from about 6 kHz to above the range of audibility. It exploits the use of a flat ceramic magnet which not only gives a high magnetic flux density but also achieves a very flat type of loudspeaker unit, which is thin enough from front to back to be accommodated partly within the cabinet front and thus to avoid any harmful cavity-resonant effects. In a typical design, the front may be between $\frac{3}{4}$ and 1 in. thick and the tweeter may be inset and mounted

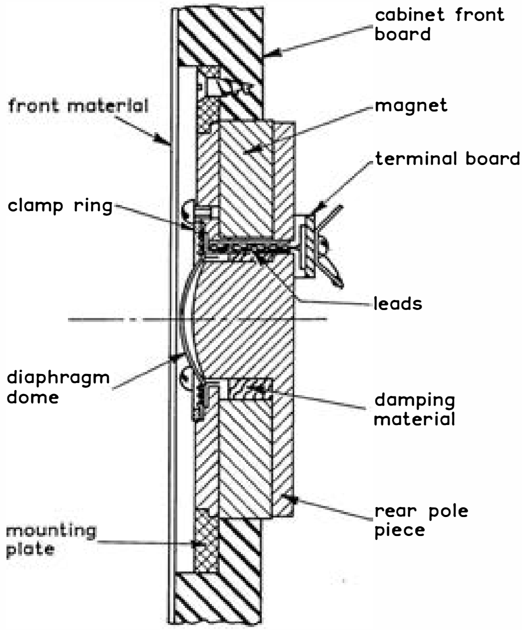


Fig 32. *A sectional view showing the construction of an ultra-high-frequency tweeter loudspeaker of the direct radiator moving-coil type. The response is maintained to over* 20 *kHz. This type of unit is used in the highest-quality loudspeaker systems to extend the range above that of normal tweeters and to ensure the best possible response to transient sounds.*

by means of the front flange so that the back is sealed and the diaphragm is clear of the cabinet front covering material, as shown in Fig 32. The front covering material may be a semi-rigid material such as expanded metal or may be a stretched open-weave material of the Tygan type. Either of these materials may be supported on the cabinet front by a plastic foam flexible backing material cut away round the loudspeaker openings. Owing to its absorbent nature, this will not result in any increased cavity-resonant effects, and it will give an increased clearance between the dome of the tweeter diaphragm and the front covering material.

The diaphragms of these tweeters and those of somewhat larger units (which have a response which can be used lower in the frequency scale, eg down to 3500 Hz) are more akin to those of

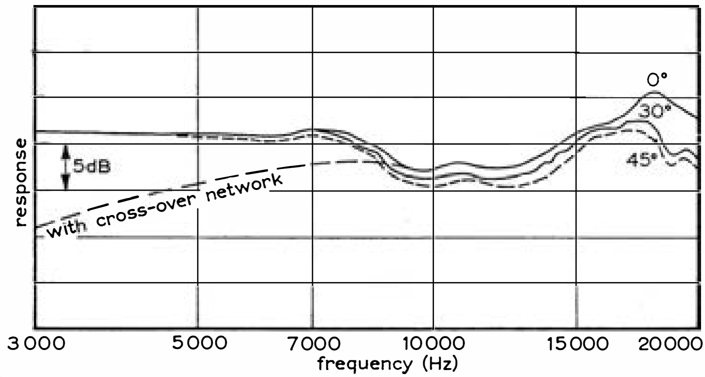


Fig 33. The response of an ultra-high-frequency tweeter similar to that shown in Fig 32.

moving-coil microphones than those of cone loudspeakers, in that a small diaphragm moulded from a thin rigid plastic material is used. The centre part of the diaphragm is domed outwards for rigidity, rather than made in the form of a concave cone, as in larger loudspeakers. The surround is relatively large and is given a controlled restoring compliance by means of tangential flutes or a flat radial roll section. The outside diameter of the complete

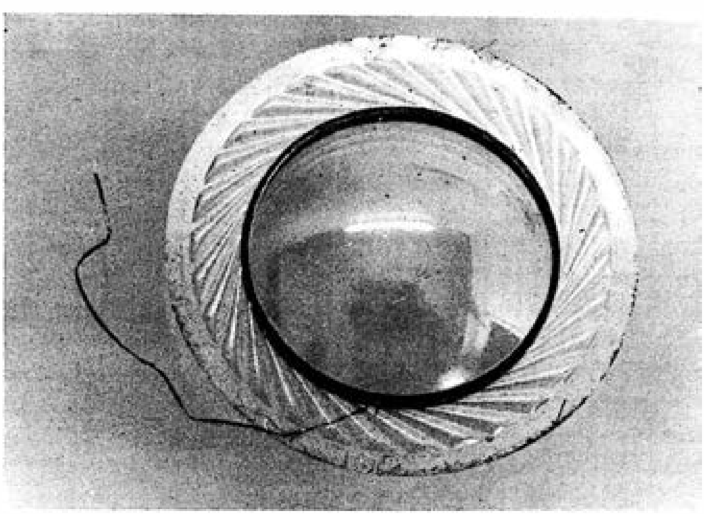


Fig 34. The lightweight 1-in. diameter plastic diaphragm and moving-coil system of the ultra-high-frequency tweeter of Fig 32.

diaphragm is usually not more than $1\frac{1}{2}$ in. so as to ensure freedom from break-up modes and a wide polar spread of high-frequency sound radiation (ie an approach to omnidirectional polar response). The speech coil is wound from copper or aluminium wire and is "formerless"; that is, it is wound as a self-supporting two- or four-layer coil cemented to the diaphragm. In the unit of Fig 32, some acoustic resistance damping below the coil slot is added in the form of a ring of porous plastic foam. The tweeter and its flange are preferably made a sufficiently good fit in the cabinet front so as to seal the enclosed loudspeaker cabinet and to avoid any air-leaks. A little putty or sealing compound may be added if necessary to ensure that the joint is sealed.

Loudspeaker cabinets or enclosures

The cabinet not only serves as an essential mounting and protecting frame for the loudspeaker units but also fulfils an important acoustical function. This is to control or to eliminate altogether the out-of-phase radiation from the rear of the loudspeaker diaphragm.

Many different types of cabinet, baffle or enclosure have been evolved in order to give the desired results, but certain problems are common to all. Apart from the difficulty of obtaining a close approach to the ideal acoustic function at low frequencies, difficulties arise owing to mechanical resonances of the panels forming the walls and back, and also to acoustic resonances of the air enclosed within the cabinet. Both these types of resonance will be excited by the loudspeaker units, the mechanism being that energy is absorbed when the loudspeaker sounds the appropriate note; after the note has ceased, the energy stored in the mechanical and acoustical resonances of the cabinet are given back and radiated as sound, either by driving the loudspeaker cone from behind or by the continued vibration of the cabinet walls. The effects occur at low and middle frequencies and tend to cause a "coloration" of the sound reproduction, which is often described as "booming" or "honking," when it is particularly obtrusive. Obviously, it is very difficult to eliminate these effects completely and they are undoubtedly a factor which influences the character of the sound produced by particular loudspeakers and gives them their individuality. Generally speaking, middle frequency resonances are the most objectionable but are more readily cured by using sufficiently thick and well-braced cabinet walls and by including a proper amount of cabinet internal acoustic damping in the form of sound-absorbing material.

Low-frequency mechanical vibrations of the cabinet walls are very difficult to eliminate completely, as it is not easy to make them thick enough to be completely rigid or so heavily damped as to be completely dead or inelastic. In fact, a certain amount of controlled cabinet wall vibration may be accepted as having a sounding-board effect, as with many musical instruments, adding some fullness to the low-frequency reproduction. Nevertheless, the effect must be under control and the modern tendency is to make the walls as rigid and as dead as possible, by using suitable thicknesses of material, braced if necessary.

To a fair extent, the final performance demanded of a loudspeaker is in the realms of art rather than science, the choice often depending on individual taste, and often to a large extent independent of such factors as the cost and complexity to which one is prepared to go. One must, however, beware of certain traps when making choices based on limited initial listening tests. Some loudspeakers may sound impressive on certain programmes only or may have defects which become increasingly obtrusive in time and become annoying or even fatiguing to listen to. Excessive or boomy bass or hard peaky top may well come in these categories, as certain programmes may sound effective whilst many others may be unsatisfactory. However, it is noteworthy that nowadays the loudspeakers produced by well-established and experienced manufacturers are generally of good quality and have obviously proved acceptable to a large number of listeners with many different set-ups. Different programmes may still benefit from an intelligent use of the bass and top tone-controls provided on good amplifiers. Some suggested forms of listening test are given in the later section on sound-quality judgments.

We will now examine various types of cabinet and consider some examples of good modern design.

It is a fundamental tenet that loudspeaker enclosures have to be designed to suit the type of moving-coil unit to be used, particularly as regards the effective area and mass of the diaphragm and the centring stiffness. To some extent, the opposite applies, in that loudspeaker units may have to be chosen to suit the cabinet.

Tuned enclosures are mainly of the Helmholtz resonator or reflex-ported type. Quarter-wave tuned pipe or labyrinth types of cabinet have been used but tend to be more expensive to construct.

Ported reflex enclosures

In the reflex system the loudspeaker cone and suspension forms a series mechanical resonant circuit and the enclosured air volume forms a parallel resonance with the mass of the air associated with the port (actually the mass of the air in the port pipe plus the radiation mass of the air added to the orifice of the port). If the two resonances are tuned to the same frequency (ie if the unit bass resonance coincides with the frequency to which the cabinet resonator is tuned) the result is a coupled double resonant "circuit," exhibiting two resonant peaks. One is below and one above the bass-resonant frequency of the unit. If these two peaks can be damped to the correct degree, the bass response of the loudspeaker is extended considerably below the bass-resonant frequency of the unit. In practice it is quite difficult to obtain the correct damping of both the resonant peaks in the overall bass response unless a fairly large cabinet is used, with correctly positioned acoustic-resistance material mounted in exactly the right places. A common defect with reflex cabinets of compact size is that the upper bass-response peak is not well damped, producing an unfortunate bass "coloration" at a relatively high bass frequency (often 120–150 Hz).

Nevertheless, it is possible to make bass-reflex cabinet designs which work correctly, always provided that the proportions of the port, the resonator tuning and the damping are precisely right. Generally, expensive laboratory equipment is needed to achieve this result, though simple impedance measurements at the speech-coil terminals can be a good guide. The enclosure walls must be thick enough to be substantially non-resonant and internal damping material has to be correctly disposed inside the cabinet, firstly to damp the resonant upper peak of the two which occur and, secondly, to damp air-column mid-frequency resonances inside the box which may otherwise cause coloration. These internal resonances may also be radiated out through the port opening as well as driving the loudspeaker unit from behind. It is necessary for all joints in the cabinet and such places as loudspeaker unit mounting flanges and terminal plates to be fairly air-tight, otherwise the port resonator tuning may be upset and sound leaks may occur.

A number of designs of ported cabinets are produced by the various loudspeaker-unit manufacturers, and the home constructor may obtain detailed plans from them. It is usually necessary, however, to follow the constructional details, dimensions and recommended materials fairly closely. In some cases, the proper damping materials and even complete properly damped port units are available from the manufacturers concerned.

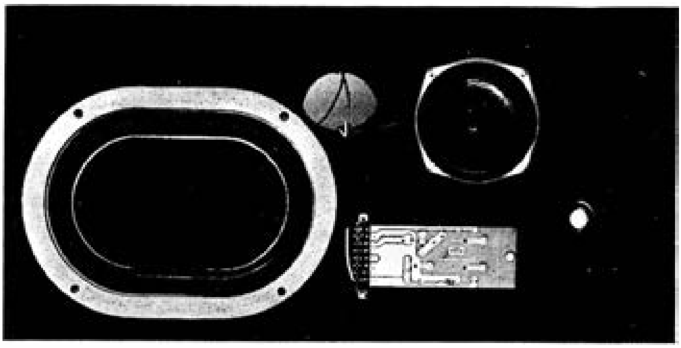


Fig 35. *A high-quality three-way loudspeaker module kit for assembly into home-constructed or ready-made cabinets to give a frequency range from* 30 *Hz to well over* 20 *kHz at a power rating of* 30 *watts. The large elliptical bass unit has an aluminium-reinforced expanded-polystyrene diaphragm and covers from* 30 *to* 300 *Hz. The* 5 *in. mid-range unit has a specially designed and treated diaphragm with a highly compliant linear-roll suspension. It covers the range from* 300 *Hz to* 3 *kHz here, though it may be used as a bass and mid-range unit in smaller enclosures. The ultra-high-frequency tweeter gives a response from* 3 *kHz to* 30 *kHz. The printed circuit three-way cross-over network is shown. The high and mid-range units are isolated at the rear in their own enclosing cases and the low-frequency bass port opening is seen by the bass unit. Cabinet details are provided.*

An interesting variant to the port or open passage is to provide a subsidiary "slave" driven diaphragm which is given optimum properties and may enable more compact cabinet proportions to be used.

Fig 35 shows a loudspeaker kit of high quality which is supplied with damping material and cabinet constructional details.

THE INFINITE-BAFFLE OR ENCLOSED CABINET

This is probably the most popular type of enclosure in use today. The enclosed volume of air acts as an "air spring" and adds directly to the total restoring stiffness of the cone, so specially designed units with low-stiffness surrounds and centring bellows have to be used with these cabinets if they are to be of reasonably small size.

The completely enclosed infinite-baffle box-type of enclosure offers some considerable advantages over most other types of cabinet as regards appearance, cheapness, compactness and performance, and a number of special loudspeaker units are produced for use in enclosed boxes. These designs now represent an advanced form of technique and have a wide frequency response and reduced nonlinear distortion. As we have noted, the stiffness of the air in the enclosure constitutes an extra spring acting on the diaphragm. It adds directly to the stiffness of the surround and centring bellows. The low-frequency resonance between the total stiffness and the diaphragm mass represents the lowest bass frequency which the loudspeaker can radiate efficiently. This resonance without any added box stiffness usually occurred at about 40–50 Hz on the older types of high-quality loudspeaker unit. Adding a box enclosure of normal volume could raise the resonance and hence the bass cut-off frequency to about 100 Hz. In order to radiate bass efficiently down to 40 Hz, the loudspeaker resonance in free air has to be reduced to about 15 Hz. This means that very free surrounds and centring bellows are needed. The centring bellows must still ensure that the speech coil remains centralized in the radial magnetic gap under all conditions. It must also give a linear restoring force and must resist the onset of mechanical fatigue. Similar considerations apply to the surround. This must also form an air-tight seal and be properly damped so as to act as a correct terminating mechanical impedance to wave motion traversing the diaphragm outwards from the speech-coil. If this termination is not correct, a reflected wave will travel back along the diaphragm towards the coil and wave interference will occur at certain frequencies, resulting in an irregular response.

Generally a modern "infinite-baffle" loudspeaker unit has a good all-round performance and is characterized by a stiffer, more rigid and heavier diaphragm than before. The extra weight helps to keep the bass resonance low without making the centring bellows too weak. A relatively large and soft roll surround is often used, which is arranged to have a resistive nature so as to provide the right mechanical impedance and damping properties. The radial magnetic gap may be increased so as to allow more speech-coil clearance to compensate for the weaker centring action of the bellows. The heavier diaphragm behaves more nearly as a piston and has fewer severe break-up resonances than before, but the increased mass and the wider gap tend to reduce the efficiency. However, modern magnet materials such as ceramics or columnar cast magnets are capable of providing a high magnetic flux without any increase in weight, so that the efficiency can be maintained at a good level.

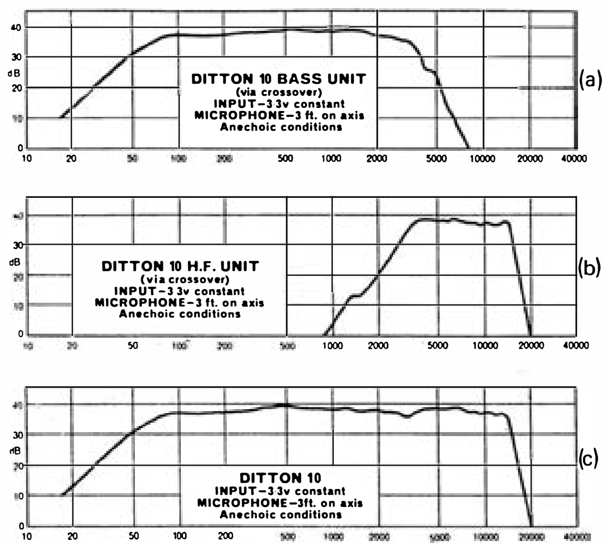


Fig 36. *The response curves of a modern compact bookshelf type high-quality loudspeaker.* (a) *The response of the bass and mid-range unit, which is a special* 5 *in. loudspeaker with a high-compliance cone suspension with a basic resonance of* 30 *Hz and capable of an excursion of* $\frac{1}{2}$ *in.* (b) *The response of the high-frequency moving-coil pressure direct radiating unit, which is* $1\frac{1}{2}$ *in. in diameter and is mounted close alongside the main unit in a compact enclosed cabinet.* (c) *The combined response of the two units when combined via a three-element cross-over network* (3·5 *kHz cross-over point*).

The result is a reduction in the size of the cabinet required to take even three-way loudspeaker systems using 12 in. woofer or bass units. A cost reduction, ease of housing, and a reduction in size of the cabinet bring other subsidiary benefits. Smaller cabinet wall panels are more rigid for a given thickness and hence do not require much damping or bracing; also acoustic edge diffraction effects are reduced in severity. Here the wave radiated from the diaphragm expands smoothly until the edges of the hemispherical wavefront meet the front edges of the cabinet; the pressure wavefront then suddenly has to diverge to propagate itself around the cabinet. The result is a discontinuity which, as always in wave propagation, causes diffraction irregularities to occur. The smaller the cabinet, the further up the frequency scale before the onset of severe diffraction effects is met. These effects cause measurable irregularities in the frequency response, but at higher frequencies the radiation from a diaphragm becomes concentrated in a narrower solid angle in the forward direction and cabinet-edge diffraction is then of small importance. Other precautions have to be taken in the design. The most prominent internal air-column resonances are the axial resonances between the three pairs of parallel walls. These resonances will occur at middle audio-frequencies in smaller enclosed cabinets and must be critically damped by draping or fixing suitable acoustic-absorbing material inside the cabinet. If undamped, these resonances will be excited by any corresponding frequency applied to the diaphragm and will affect the acoustic impedance loading on the rear of the diaphragm, causing an irregular frequency response. They may also persist after the exciting signal has died away, in which case, as we have seen, the diaphragm may be driven from the rear by the box-resonance to cause "honking" or resonant hangover. If the interior of this box is excessively damped by overfilling with absorbing material, then a large resistive load is acoustically coupled to the rear of the diaphragm which may then be over-damped, with a resultant loss of efficiency and a slowing-up of the rise time of transients. It is also desirable to maintain a nearly air-tight seal at all joints, screws etc. Effective sound radiation demands high power and the generation of very high sound-pressure levels in the enclosed box and air may escape at high velocity from even small leaks. Tweeter units mounted in the main cabinet must have their backs effectively closed off and must be well sealed into the cabinet round their periphery. It is desirable that the internal acoustic damping material used inside the cabinet should have the ability rapidly to take up the small variations in heat of the air which occur during the acoustic cycle. If there is no heat loss to the surroundings during an acoustic compression cycle occurring at low frequencies, then the process is classed as "adiabatic" and a maximum value of air stiffness will be provided by the air volume in question. It is well known that the work done in compressing air is stored as heat and thus the air temperature is raised. If this heat is transferred to a distributed material in contact with the air then on decompression less work will be returned from the air and the net effect will be that of a reduced

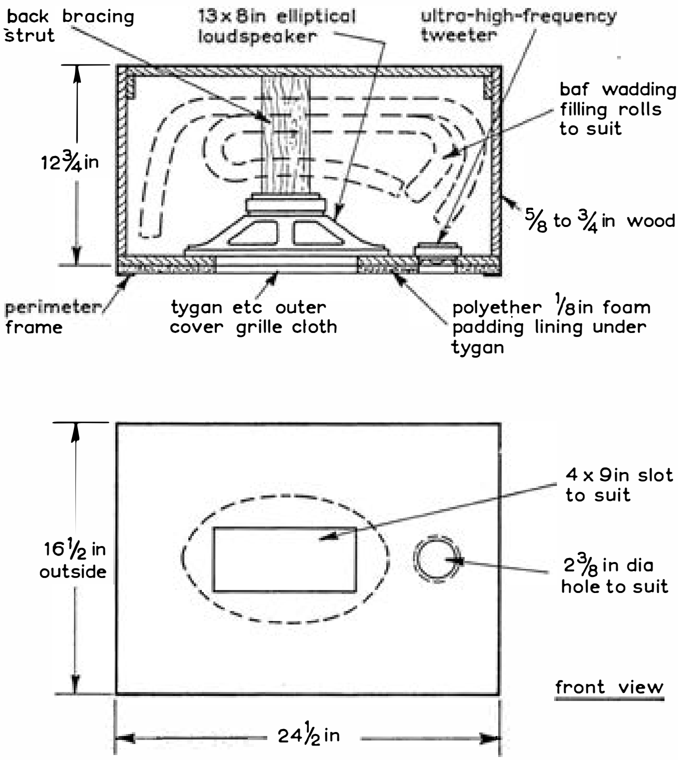


Fig 37. *An economical design by the author for a home-constructed two-way loudspeaker system. A wide-range elliptical paper-cone unit with a compliant surround is used in an enclosed cabinet with an ultra-high frequency tweeter fed by a simple blocking capacitor. Good-quality* 13 × 8 *in. commercial elliptical wide-range units are available with low bass resonance suspensions free of purchase tax* (*at the time of writing*). *Suitable tweeters are now available from several leading makers.* (Units by Fane, ITT etc.)

stiffness. Materials such as BAF (bonded acetate fibre) wadding have been found to be good cabinet-damping materials for this reason, as well as being light and clean.

Infinite-baffle loudspeaker design thus requires that the correct bass units are used and also that the construction and the electrical (amplifier) damping as well as the cabinet internal damping are correct. If all these factors are not properly balanced, the bass response may be peaked or even largely deficient.

Good examples of enclosed cabinet design are seen in the excellent performance obtained from "bookshelf" and "slim" loudspeakers now available using highly developed bass units capable of large linear diaphragm excursions, often combined with good tweeter units.

Unpacking and assembling loudspeaker units

Although modern loudspeakers are strongly made, it is important to take certain precautions when handling units and assembling them into cabinets—

(i) Never drop loudspeaker units even when packed, or subject them to any heavy air blast or suction, such as that which may result from slamming tightly fitting cupboard doors. A unit should not be abruptly placed or removed from a table with the cone face down.

(ii) Do not allow any ferrous metal objects such as tools, screws, scissors, needles etc to come near to a loudspeaker. Above all never place a loudspeaker on a work-bench where there is a chance of steel filings and swarf being attracted into the gap. Very fine swarf can penetrate dust-excluding gauze caps over loudspeaker gaps and it is impossible to remove unless the unit is demagnetized and taken apart—a job for the manufacturer.

(iii) It must be ensured that the cabinet front mounting surface is sufficiently smooth and flat to enable a good seal to be established without distorting the loudspeaker chassis or mounting ring by undue tightening of the securing screws. It is advisable to use brass or bronze screws in order to avoid the possibility of magnetic slivers being produced. The screws must be strong enough and long enough to support the unit. Any shocks during transit can generate large acceleration forces on heavy loudspeaker units.

Testing and quality judgments on loudspeakers

A number of important tests may be made on loudspeakers in order to eliminate defects and also to make satisfactory comparisons and judgments of quality. The tests proposed here may readily be carried out without the use of special test equipment.

1 Continuous tone testing. A large low-frequency tone may be obtained by such devices as placing a loop of insulated wire from the input of the amplifier near to a mains transformer in the power pack. It will give an idea of the bass capabilities and will tend to show up defects such as buzzes and rattles in the unit or cabinet, as well as checking on correct phasing of dual bass units. A phase reversal shows up as a drop in bass level. However, it is necessary to avoid applying constant tones to loudspeakers at a high level as there is a risk that the speech-coil may be over-heated and burnt. Normal programme peaks are of such small duration that speech-coil heating does not occur, except possibly from sources such as electric guitars or electronic organs, which demand heavy-duty loudspeakers.

Buzzes in loudspeakers may be due to loose washers or terminals etc, or may be due to loose front fabrics or expanded metal. Mounting these tightly over a foam underlay (with cut-outs for the loudspeaker openings in the foam) will often prevent these troubles.

2 A useful continuous random tone for testing is often readily obtained by tuning an FM set away from any station, if necessary removing the "squelch" noise-suppressing circuits. The resulting audio signal is close to "white noise," so called by analogy with white light, which contains all frequencies in the visual spectrum at equal intensities. If the lower frequencies are relatively increased in level by means of the tone-control bass boost, the result is "pink noise," which is sometimes claimed to be closer to the frequency distribution in normal programmes.

Listening to random noise on loudspeakers can give quite a lot of useful information. The higher frequencies should be well audible as a smooth rushing sound. Acoustic colorations or transient hangover tend to show up in the form of alterations in the evenness of the random noise. Listening off the axis of the loudspeaker gives a good idea of the distribution of high frequencies in the room.

3 Speech can be a good test of accurate reproduction, especially if the original recorded talker is able to speak in the same position as the loudspeaker in the room. However, male speech

in particular must be reproduced at the same level as the original or the low frequencies will be exaggerated according to the low-frequency auditory contours of the ear if the level of reproduction is too high. Sibilants may also be exaggerated in some cases.

4 When music is used as a test, apart from certain specially recorded passages on test records which are designed to show up certain effects, it is advisable to use well-balanced musical selections free from over-modulation which might give rise to tracing distortion on gramophone records, where these are used as a programme source. It is best to avoid any recordings in which special effects may have been used as these may prejudice the judgment of quality.

A tendency for all bass notes to sound muddled and indistinguishable can indicate resonant bass coloration and intermodulation. Piano music is a good test of quality. Poor transient response, any appreciable intermodulation and wow or flutter will all be shown up. If string tone is correct and if castanets, cymbals and the triangle stand out sharply and with good "presence," an extended top response is indicated.

In general, all reproduction should be improved by stereo. A sense of space and another dimension are added to the reproduction and the so-called monophonic "hole in the wall" effect is largely removed. The overall result should be more natural and presence should be improved. The effects of intermodulation and other imperfections are usually less noticeable and listening generally is less fatiguing and more satisfying. The spatial stability and apparent direction of individual instruments and soloists should be reasonably good and a sense of spaciousness should be associated with the reverberant sound as well as with the direct sound. On stereo, one is checking the audible effects which depend on channel matching, "cross-talk," loudspeaker characteristics and positioning, in addition to the general parameters of reproduction which also apply to monophonic reproduction. Dramatic pieces involving rapid movement and action can have an increased emotional impact when reproduced on a good stereo system. Any appreciable interchannel level mis-match or cross-talk may considerably reduce the dramatic appeal and sense of rapid transition involved in a programme.

8

Planning a complete high-quality system

If a complete installation is being planned from scratch it is useful to have a guide as to the best way to apportion one's funds so as to obtain a system of balanced quality.

As a very rough indication, it might be advisable to budget so as to spend approximately equal sums on—

1 The loudspeaker(s) and enclosure(s).
2 The amplifier system, including pre-amplifiers.
3 Gramophone turntable and pick-up.
4 Tape-recorder (if required).
5 Radio-tuner unit and aerial system.

It is, of course, possible to spend more or less money, according to the quality, power and facilities desired.

If the enthusiast is prepared to construct his own loudspeaker enclosures and other cabinet work a considerable saving will be made. If he is able to construct his own amplifiers, tuners and other circuitry, a proportionately larger saving in cost may be achieved. Other compromises between quality and cost can be worked out according to the power output required from the amplifiers and loudspeakers and whether, for example, extra features such as a high-grade automatic record-changer could be dispensed with in favour of a single-record player or whether a tape-recorder can be dispensed with.

In the following pages we illustrate some suggestions for accommodating possible systems designed to fit small and large rooms with various degrees of complexity and variety in the quality and number of services offered.

The housing of Hi-Fi equipment and units involves problems concerned with acoustics, power supplies, ventilation, ergonomics (the science of human control of machinery), aesthetics (appearance and form), space, portability (weight) and safety. This is a formidable list and in modern housing and apartmental living conditions it is obviously extremely difficult to find space for a technically well-planned system. Given enough space, one can obviously design large pieces of furniture or built-in cabinets which can house any desired amplifiers, tuners, turntables, tape-recorders, loudspeakers etc. Many examples of luxurious installations have been described in the Hi-Fi journals. They are often the work of dedicated enthusiasts who have been able to devote a fair amount of money and a large room to their hobby.

The average Hi-Fi enthusiast is not in this happy position and almost invariably has to make a number of compromises.

The possible approaches vary considerably, but they may generally be classed either as designs in which most or all of the equipment, usually with the exception of the loudspeakers, is housed in one enclosed cabinet, or, alternatively, the various units may be housed separately. It is difficult to integrate such an assemblage in a completely satisfactory manner owing to the differing sizes and shapes of the various units and the need for access to the controls and mechanisms. It is natural that the "bookshelf" type of approach should be gaining favour, as it offers a number of advantages to the amateur.

Equipment housed in cabinets

Fundamental decisions on the designs relate to the question of how the user intends to operate the equipment, and whether his approach is that of the armchair listener who wishes for the minimum disturbance of his comfort, or, alternatively, if he regards himself more as a technical operator who is prepared to cross the room and operate controls etc from a standing position. A number of constructors' designs and fully assembled cabinets have been produced which give a choice of arrangements of both kinds. In considering the merits of any particular design, such factors as the position, angle and lighting of controls and mechanisms for ease of operation from a sitting or standing position must be considered, together with considerations of bulk, appearance and the enclosing of equipment with doors and covers in order to keep it free from dust etc when not in use. It is generally well worth using two or more separate loudspeaker cabinets. The loudspeaker enclosure can then be correctly designed in its own right, the risk of microphonic feedback to pick-ups is largely eliminated and correct loudspeaker placement in the room is greatly facilitated, particularly for stereo. Figs 38–42 show some interesting commercially produced arrangements offering space for tuners, amplifiers, turntable, and in some cases tape-recorder and record-storage etc. Efficient dust protection is generally afforded when the units are not in use.

Fig 38. *A recent compact high-quality stereo tuner and amplifier plinth assembly which includes the BSR high-fidelity automatic record-changing turntable normally fitted with a smoked Perspex (tinted) lid. Two special omnidirectional loudspeakers shown here are wide-range column types which handle the output of* 15 *watts per channel.*

Fig 39. *An amplifier and turntable plinth assembly using the high-quality Garrard AP* 75 *automatic single-record-playing turntable and arm assembly on a plinth into which Sinclair stereo pre-amplifier and power-amplifier modules have been assembled. The modules of Figs* 24–9 *may be used or alternative power units may be fitted to give power output ratings of over* 40 *watts per channel. The AP* 75 *arm gives a micro-adjustment of tracking weight down to below* 2 *g with side thrust compensation. Automatic record-starting, groove-positioning and automatic lowering, raising and stop are provided as well as a pause-control.*

Wall and room-divider types of housing

These have considerable appeal today, as many tuners, amplifiers, tape machines and loudspeakers are available as free-standing units in their own cabinets. The size of all these units is now such that they can usually be accommodated in the "bookshelf" or "divider" type of structure. A number of commercial kit systems are available for making up those units to individual requirements. The vertical open-work structure is economical of room

Fig 40. *A convenient trolley type of cabinet* (*Imflex Mini-trolley*) *for tuner, record-player and amplifiers, which includes some storage space. A hinged Perspex lid covers the units when not in use. The height is convenient for armchair or normal operation.*

space as it keeps the floor clear. The systems are versatile, in that alterations and extensions to the system can generally be accommodated and the general effort can be made to harmonize with modern room designs and decorative schemes. It is desirable, however, to run the various interconnecting wires and cables neatly and preferably as far as possible out of sight behind structural members etc. Fig 42 shows wall and divider units of the kind described above.

Fig 41. *A compact and reasonably priced cabinet (the "Trio") for housing tape-deck, turntable, tuner and stereo amplifiers as well as providing alternative storage space.*

Detailed system planning

Having decided on the general scope and the approximate sum of money available for the system, one must consider precisely how it is to be realized. For example, a tuner and amplifier stereo system and gramophone turntable system with two good loudspeakers all in the form of separate but technically compatible units may be decided on, with the possibility of operating later with a tape-recorder and microphone. Use with slide or cine-film sound accompaniment may also be considered as a later project.

The obviously safe ways of buying recommended equipment from reliable dealers have been mentioned. It is of great importance to ensure that a major investment of this kind is not only going to be technically compatible as an assemblage of units but will

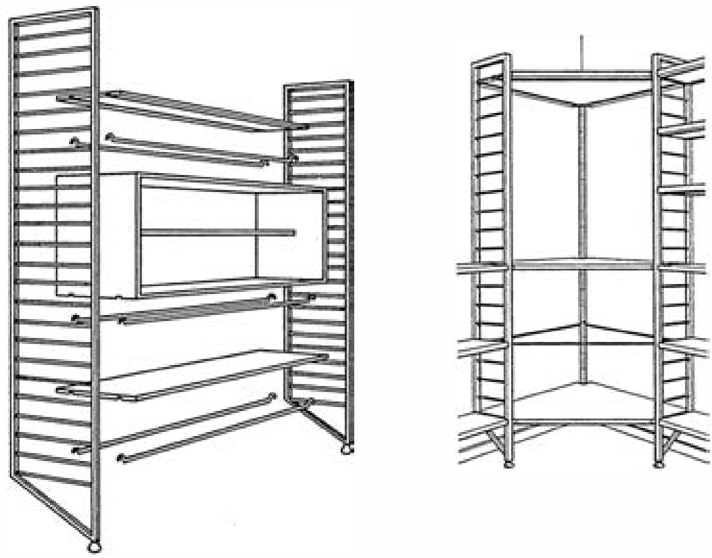


Fig 42 A comprehensive wall-fixed or room-divider unit (the Staples "Ladderax" unit) giving a flexible arrangement of modular shelves, panels and cabinets to accommodate a wide range of Hi-Fi and other home entertainment equipment, together with books, ornaments, tape storage cases, records etc.

also really suit your needs and tastes, to say nothing of the space available for housing and laying out the system.

Obtaining the very best equipment throughout is necessarily expensive and the cost of a complete outfit as above could now be £800 or more. On the other hand, obviously the cheapest commercial Hi-Fi equipment must have certain limitations.

Many enthusiasts will wish to save money and also to have good equipment by doing as much as possible of the construction themselves. The obvious possibilities of loudspeaker and other cabinet-making have been mentioned. The use of cabinet kits and ready-to-assemble units is an alternative to the rigours of the carpentry and cabinet-making from scratch which are otherwise involved. A number of such kits are now advertised by reliable makers.

It is worth while to consider building your tuners and amplifiers from established designs, kits or modules, a few of which we have illustrated in previous pages. These are now available from a number of reputable suppliers and we will now consider some of the factors involved in linking up and using units of this kind.

The integration of kits and modules

Apart from the housing of complete and finished amplifiers and other units in bought or home-made cabinets, the use of finished modules or sub-assemblies, the building of kits supplied and packaged to make up approved designs and, finally, the building of amplifiers or tuners to designs published in the technical journals all represent possible steps in the home construction of Hi-Fi equipment.

It is necessary to remember, however, that it is required to assemble, wire and interconnect finished commercial amplifiers and tuners or modules in accordance with the makers' recommendations, so as to ensure that the correct results are obtained. This particularly refers to freedom from hum, interference, unwanted break-through and cross-talk. The correct types of screened connecting leads must be used for aerials and audio-input leads, cables etc in order to avoid excessive capacity losses and so on. Professional advice from the supplier or dealer will enable the correct procedure to be followed. General rules are always to use cables of adequate size and diameter; proper shielded co-ax or twin low-loss RF cable of the correct impedance being used for VHF aerial leads, without any joints or dead-end extensions. These may cause disastrous internal reflections or standing waves on the cable, unless the proper fittings, plugs and terminating devices or baluns are used.

Audio-input connections must be as short as possible and must be kept away from output leads or mains cables. Low-impedance input connections may be run in shielded-twin-microphone cable or thinner equivalents, but high-impedance inputs such as those for the standard 47 k ohms pick-up inputs should be run in low-capacity shielded co-ax or other cables, unless the runs are so short as to avoid any shunt capacity losses of the higher frequencies. Earthing of cable shields, plug shrouds and the chassis via the mains third pin is of paramount importance. The correct choice of earthing points and the avoidance of hum-causing loops are among the most difficult decisions one has to make in the wiring of audio systems. To some extent, every system is different and it is difficult to give any firm rules. Generally speaking, good low-resistance earth connections must be made everywhere, the main earthing point being applied at one place only; each shield end or other earth connection is run back to this point, to avoid common loops of earth-lead in which small mains leakage capacity currents etc can couple to high-gain input points and cause hum. It is also important to ensure that the metalwork of cases, panels and turntables is earthed properly, not only to prevent hum, but as a safety measure.

Suppliers of amplifiers and complete kits usually include the necessary plugs and connectors with the exception of mains plugs, together with some wiring instructions, but the makers of subsidiary units such as turntables and tape-recorders often do not supply connector plugs, except possibly with a microphone etc. Coaxial "gram plugs" are standard and may be obtained colour-coded, but a number of DIN plugs and connectors are also used and it is important to obtain exactly the correct pin arrangement. One final point is that loudspeakers must be wired with suitably low-resistance leads so as to avoid a loss of power.

Any difficulties one may have in laying out and connecting up finished commercial amplifiers are naturally likely to be less than those which may be encountered when assembling modules or kits of one's own choosing. Here again, reputable suppliers are at pains to give very detailed hints and instructions for various types of set-up, but, of course, the more one follows different ideas the greater the risk of meeting difficulties. However, the makers of good kits and modules are usually very helpful in cases of difficulty, and are generally prepared to service or to correct faults and defects encountered by genuine home constructors, at a nominal charge.

The choice must be yours. Established commercial units will give you a known and usually high standard of performance. Modules and kits will normally be cheaper, and in many cases can claim a very good performance. The precise extent to which this is achieved in practice may depend largely on individual skill. There is also the often very great satisfaction of making and assembling as much as possible of one's own Hi-Fi system, which can then reflect personal tastes and ambitions to a considerable extent.

Names of suppliers of established kits and modules appear in the advertisement and other pages of the Hi-Fi journals and their products are often reviewed, in addition to the normal fully assembled systems.

Glossary of Hi-Fi and radio terms

Acoustic Feedback. An undesirable oscillation or howl caused by sound energy from a loudspeaker either being picked up by a microphone or causing a turntable and pick-up to vibrate.

AF (Audio-frequency). Frequencies within the range of human hearing. Generally defined as 20 Hz to 20,000 Hz. (See "Hertz.")

AFC (Automatic Frequency Control). A circuit designed as a refinement to a radio receiver to keep it accurately tuned to the desired signal.

AGC (Automatic Gain Control) or AVC (Automatic Volume Control). A circuit which automatically keeps the audio output of a radio tuner at the same level whatever the strength of the radio signal received.

Ambience. The persistence of sound in an auditorium which gives it a particular acoustical character. (See "Reverberant Sound.")

AM (Amplitude Modulation). The usual method of carrying an audio signal on a radio-frequency wave for transmission. It is the method used in most long-, medium- and short-wave broadcasts. The RF carrier amplitude is varied.

Ambient Noise. Surrounding noises, other than the sounds one is trying to reproduce, ie traffic noise, factory machinery, rain, wind etc, which may affect the microphone or may be present in the listening room.

Attenuation. The reduction in the level, loudness or strength of a radio or audio signal due to deliberate or accidental losses.

Attenuator. A device for providing attenuation (see above) of a signal. The simple attenuator is formed by two resistors. A volume-control is a simple form of variable attenuator. (See "Potentiometer.")

Audio. Relating to sound reproduction and recording.

Baffle. A large flat board on which a loudspeaker is mounted. It has to be large to allow good reproduction of low frequencies. Enclosed cabinets generally give satisfactory performance at a much smaller size. (See "Infinite Baffle.")

Balun. This is a device for offsetting the effects of unbalance when interconnecting UHF or VHF aerial cables. A half or quarter wavelength of line may be formed into a spiral and housed in a box for this purpose.

Bass. The lower notes or frequencies in the audio range.

Bias. In tape-recording, bias is the high-frequency signal (100 kHz approx) applied to the magnetic tape together with the signal to be recorded. This bias ensures that a strong, low-distortion audio signal is recorded on the tape. In valve or transistor circuits it is the small DC voltage applied to the grid, cathode, emitter or base etc to ensure that low-distortion, efficient amplification is achieved.

Bias Compensation of pick-ups is the provision of a small torque to offset the tendency for the stylus to experience an inward force due to the record drag and an offset angle in the pick-up arm.

Binaural listening involves the use of both ears, either directly in a free sound field or by the use of earphones.

Bulk Erasure. A method of erasing or removing the magnetic signal from a complete reel of tape in one quick operation. The reel of tape is passed through the AC magnetic field from a mains-energized electromagnet.

Capacitor (Condenser). A component made up of conducting plates separated by insulation. It presents an increasing restriction to the flow of alternating current as the frequency of the current gets lower. It will not pass direct current. The names given to the various types (paper, mica, ceramic, electrolytic etc) generally refer to the type of insulation material used in the construction.

Capstan. The motor-driven spindle or pulley in a tape-recorder which drives the tape at the required speed. It has to be very accurately machined to ensure the correct tape speed with the minimum of wow or flutter. (See "Pinch Wheel.")

Carrier. A radio-frequency wave which is modulated so as to carry audio-frequency or other signals. (See "AM" and "FM.")

Cartridge. The part of a pick-up which translates the mechanical vibrations of the stylus in a record groove into an electrical signal to be fed to an amplifier. It is fitted in the head on the pick-up arm.

Coaxial cable. A screened sheathed cable enclosing a single insulated conductor. A characteristic impedance of 75 ohms is used for VHF aerial and input cable.

Compliance. The reciprocal of stiffness. It is a measure of the flexibility of movement of a pick-up stylus or loudspeaker cone etc. The higher the compliance the more free or flexible is the movement. High compliance is desirable in a lightweight Hi-Fi pick-up or a low-frequency loudspeaker cone suspension. (See "Stiffness.")

Condenser. See "Capacitor."

Cross-over Network. A circuit made up of capacitors and inductors which divides the audio output of an amplifier into two or more bands of frequency, so that all lower frequencies can be fed to a loudspeaker specially suitable for reproducing low frequencies and the higher frequencies to loudspeakers specially suitable for reproducing higher frequencies.

Cross-over Frequency. The frequency at which a particular cross-over network separates the lower-frequency sounds from the higher ones.

Cross-talk. A break-through by part of an unwanted sound signal on to the wanted sound. For example, some left-hand stereo sound is heard on the right-hand loudspeaker together with the proper right-hand sound.

Cycles per Second (c/s or cps). The unit of measurement of frequency rate of vibration. The rate of vibration of a sound wave determines the pitch. Now generally replaced by the hertz, kilohertz, megahertz etc (abbreviated Hz, kHz, MHz etc).

Decibel (dB). A unit of measurement using the logarithmic ratio of two signals. Most commonly applied to electrical signals, sound levels, gains of amplifiers, loss in attenuators etc. dB are added or subtracted where gains are multiplied.

Decoder. In stereo, the circuits which separate the FM multiplexed signal into the required left and right channels for output to stereo amplifiers.

Detector. The circuit in a radio receiver which converts the modulated radio-frequency signal into an audio-frequency signal for application to an audio-amplifier.

Diaphragm. The sound radiating or collecting surface connected with the transducer or energy-converting element of a loudspeaker or microphone.

Dipole. A symmetrical dual-rod aerial for VHF etc reception.

Discriminator. The RF demodulator or detector stage in an FM receiver.

Distortion. If the signal out of an amplifier differs in any way (other than in level) from that being put into it, distortion has occurred. The commonest form is harmonic distortion or intermodulation distortion.

Dolby System. An automatic method of reducing recording noise and hiss by controlled volume compression and expansion in appropriate frequency bands during the recording and playback process, particularly of master tapes used to produce gramophone records.

Dynagroove. A method evolved by RCA for reducing tracing distortion on gramophone records.

Dynamic Range. The difference in volume between the quietest and the loudest passages of the music or other sound being played.

Equalization. The deliberate introduction of a particular frequency response into an amplifier to compensate for a non-flat response in the source of sound. For example, for technical reasons the frequency response produced during recording on disc or tape is not flat, so the amplifier which replays these recordings has to have a particular frequency response to compensate for this. Tone-control is a form of equalization.

Ergonomics. The science of the relationship between man and the machine. In this context it particularly applies to the use of controls and their functions.

Feedback. Taking part of the output of an amplifier and feeding it into the input. Negative feedback does this in a way that minimizes distortion, hum etc in the amplifier. (See "Acoustic Feedback.")

Filter. A circuit designed to remove or attenuate certain frequencies. In an amplifier filter circuits are often used to remove record hiss or scratch (low-pass filter) or turntable rumble (high-pass filter).

Flutter. A fluctuation in frequency of reproduced sound generally due to speed fluctuations in turntables or tape decks. Flutter refers generally to the faster fluctuations (say over 10 Hz). Lower-speed fluctuations are usually called "wow."

FM (Frequency Modulation). A method of applying an audio signal to a radio-frequency wave for transmission, so as to vary the carrier frequency. It is the method used by the BBC and others on their VHF sound transmissions. It allows good Hi-Fi quality reception with very little interference.

Frequency Response. The relative amplification of an amplifier or a complete audio system at various frequencies. If the amplification is the same at all stated frequencies then the frequency response is said to be flat.

Gain. The amount of amplification which occurs in an amplifier or system. If, when one volt of signal is put into an amplifier the output is 10 volts, then the gain is said to be 10 times. However, gain is often measured in decibels.

Harmonic Distortion. A form of distortion of a signal passed through an amplifier, which adds to the original sound harmonics or overtones of the original frequencies, ie if a 1000 Hz tone is put into the amplifier, the output would also be mainly a 1000 c/s tone but would have some amount of 2000 Hz, 3000 Hz etc added to it.

Heat Sink. A mass of metal, suitably shaped, on to which a transistor is bolted. Its purpose is to carry away the heat generated in the transistor, thus preventing failure of the transistor through overheating.

Hertz. The number of complete cycles per second executed by an alternating quantity. Named after Hertz, the discoverer of radio waves. Multiples are kilohertz, megahertz etc. Abbreviations are Hz, kHz, MHz.

High-pass Filter. A filter which will allow tones or musical notes above a certain frequency to pass but will block or attenuate lower tones. Such a filter can be used to minimize turntable rumble if it is designed to cut off frequencies below about 30 Hz.

Hum. An unwanted low-pitched sound produced in an amplifier from the AC mains or its harmonics; usually 50 Hz or its second harmonic 100 Hz. Can be caused by mains wiring or the mains transformer being too close to signal wiring, inadequate screening, inadequate smoothing in a power supply unit etc.

Infinite Baffle. (1) Ideally a flat board of infinitely large area with a loudspeaker mounted at the centre. (2) Generally used to mean a completely enclosed and sealed box, the only hole being that in which the loudspeaker is mounted. (See "Baffle.")

Integrated Amplifier. An amplifier in which the pre-amplifier, main amplifier and power supply are all contained in the one unit.

Integrated Circuit. An electronic package which includes a number of transistors, diodes, resistors and capacitors formed and connected on a common substrate by the use of modern microelectronic techniques.

Intermediate Frequency (IF). In a superhet radio receiver this is the frequency to which the radio frequency is converted for amplification before detection.

Intermodulation. A form of distortion produced by imperfect amplifiers, loudspeakers, pick-ups etc, whereby the mixing of two different tones produces unwanted tones which are not harmonically related to either tone and are therefore very unpleasant.

Kilocycle (kilohertz). A kilohertz is equal to one thousand cycles per second. Hertz is now the preferred term to describe cycles per second, when referring to electrical or wave-propagated quantities which are met in audio and radio engineering.

Level. The loudness of reproduced sound or strength of signal.

Low-pass Filter. A filter which will allow tones or musical notes below a certain frequency to pass but will block or attenuate higher notes. Such a filter is often used to reduce unwanted noise from worn records (scratch filter).

Mass. The amount of matter involved in the part or body under consideration.

Megacycle (Mc/s). A megacycle per second is equal to a million cycles per second. Megahertz now replaces this term.

Micron. A millionth part of a metre, ie 10^{-6} metre or approximately $\frac{1}{25}$-mil.

Microsecond. A millionth part of a second.

Mil. 1 thousandth of an inch, ie 0·001 in.

Millisecond. A thousandth part of a second.

Mixer. An electronic circuit designed to combine and control several different signals so that the "mixed" or combined signal can be applied to a single amplifier.

Modulation. A method of superimposing a sound signal on to a radio frequency for transmission. (See "AM" and "FM.")

Mono (monophonic). Describes any system of sound reproduction which uses only one channel, ie either one loudspeaker only or, if more than one, then the same sound produced from each.

Motor-boating. A low-speed popping sound sometimes produced from a faulty or badly designed amplifier, even when no signal is being put into it.

Multiplex. A method of radio transmission of stereo signals, ie two separate signals, left and right sound, can be carried on one radio-frequency FM main channel by the use of an additional sub-carrier frequency. (See "Sub-carrier.")

Negative Feedback (NFB). (See "Feedback.")

Objective. A measurement conducted by instrumental means which do not involve human decisions or judgments.

Oscillator. An electronic circuit which generates an alternating current. Used to produce tone signals in signal generators, electronic organs etc, and to produce radio frequencies where required in superhet receivers, radio transmitters etc.

Parameters. The fixed properties of a device which govern its performance.

Piezoelectric. The property of certain natural or synthetic crystals and ceramics whereby a voltage is generated when they are subjected to a mechanical force or strain.

Pinch Wheel. A rubber-tyred wheel holding the tape in contact with the capstan in a tape-recorder drive system.

Polar Response. The output of a microphone or loudspeaker plotted as a polar curve showing the variation at different angles relative to the main axis.

Potentiometer. (1) A simple attenuator using fixed resistors. (2) Often refers to the resistor with a continuously variable tapping point which is used in electronic circuits as volume, tone or other control. (See "Attenuator.")

Power Amplifier (or main amplifier) amplifies the required sound to a level suitable for connecting to the loudspeaker. It may be a separate unit for use with a pre-amplifier or it may be incorporated in a radio set, radiogram or integrated amplifier.

Pre-Amplifier. In audio it has the necessary volume and tone controls and amplifies the small signals from pick-ups, tape-heads etc to a level suitable for feeding into the main amplifier. It is often convenient to keep the size of the pre-amp (control unit) small.

Pressure. Acoustic pressure refers to the instantaneous variations of the atmospheric pressure above and below its static or undisturbed value caused by the passage of a sound wave.

Pressure Unit. A special type of loudspeaker designed to be mounted on a horn for production of sound. Often used in public-address work.

Quadrophony. This name has been given to experimental forms of stereophony in which front and rear loudspeakers are fed with appropriate signals derived from further multiplexing arrangements to give a better sense of depth and ambience than is provided by left- and right-hand stereo speakers only.

RF (Radio Frequency). Higher frequencies than audio. Used to carry sound signals in radio transmission.

Resonance. In electronic or mechanical systems it is a tendency for excessive response to a particular frequency. Bells or tuning forks are mechanical resonators. The tuned circuits in a radio receiver which give accurate selection of the wanted station display electrical resonance.

Reverberant Sound. The multiple reflected sound waves which persist in a room after a sound is originated.

Rumble. The low-frequency sound resulting from vibrations of the turntable being amplified.

Selectivity. The degree to which a radio tuner or receiver can reject stations or signals on frequencies close to the wanted one.

Semiconductor. An electronic device (such as a transistor or diode) which uses a solid material such as germanium or silicon as the medium instead of the vacuum used in valves. The properties of the devices depend largely on internal junctions which are formed between *p*- and *n*- type semi-conductor material.

Sensitivity. Describes the ability of a tuner or receiver to give reasonable reception of weak or distant stations. Generally given as the lowest number of microvolts of signal received which will give acceptable sound quality.

Signal-to-noise Ratio. The ratio of the wanted signal voltage to the voltage of unwanted signals (noise), generally expressed in dB. The higher the signal-to-noise ratio the lower will be the background noise on the programme.

Solid State. Describes circuits which use semi-conductors (transistors etc) rather than valves.

Squelch. Circuits on FM sets which automatically silence the inter-station noise received while tuning the receiver.

Stereophony. Dual or multi-channel techniques for giving sound-directional information with a potentially considerable increase in realism.

Stiffness. The reciprocal of compliance. Is normally defined as a mechanical spring constant.

Sub-carrier. An additional carrier wave transmitted in the FM stereo multiplex system which conveys the stereo channel difference information, the sum being conveyed by the main carrier wave.

Subjective. A personal judgment or impression which is used to give a quality or other rating of sound reproduction etc.

Tape-recorder. A magnetic tape-recorder used for sound or vision recording; single or multi-channel recording is possible.

Transducer. An energy-converting device such as a microphone, pick-up or loudspeaker which interchanges acoustical, mechanical or electrical signals.

Transient Response. The ability of a loudspeaker or amplifier to handle sudden bursts of sound such as when a cymbal or drum is struck, or when a string is plucked.

Treble. The high notes or frequencies in the audio range.

Tweeter. A loudspeaker designed specifically for reproducing high notes or high frequencies.

UHF (Ultra High Frequency). Radio frequency greater than 300 megacycles (higher than VHF). BBC 2 transmits on UHF radio waves.

Varactors. Special semi-conductor diodes which are designed to provide given changes in capacity when the DC bias is changed, thus enabling them to be used as radio tuning devices.

VHF (Very High Frequency). Radio frequency between 30 megacycles and 300 megacycles. BBC 1, ITV and BBC FM sound use VHF radio waves.

Wave. A propagated travelling disturbance of sound or any other disturbance such as radio waves.

Wavefront. The salient face of an advancing wave.

Woofer. A loudspeaker designed specifically for reproducing the low notes or low frequencies.

Wow. Slow fluctuations in the speed of a turntable or tape-deck. (See also "Flutter.")

Note. The abbreivation for grammes weight is g. The gravitational acceleration on a falling body is represented by italic *g*. In most cases the distinction may be inferred by the context.

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