THE PRACTICAL HI-FI HANDBOOK

by

Gordon J. King

Assoc. Brit. I.R., M.I.P.R.E., M.T.S.

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Foreword

THE field of high-fidelity sound reproduction in all its aspects is a vast one, and one of the most difficult problems in planning a book of this kind is that of deciding just how far each branch of the subject can be explored within the space available. A whole volume could, in fact, be written on the subject of each chapter, but then a good deal of perspective would be lost. It has been my aim in the present work not only to give a kind of bird's-eye view of the subject as a whole, but also to provide the reader with a fair idea of the operation of the various items of equipment, and knowledge which will enable him to secure the best results from his own and—if he is a service technician—his customers' equipment.

The book is not concerned solely with servicing matters, though it contains much information of a servicing nature which, it is hoped, will assist the reader who is already a radio service technician, but has only just entered the high-fidelity field. Such information should prove useful to the many dealers and their engineers who have recently entered, or intend to enter, this branch of trading and servicing work. It is hoped that the book will clearly bring out the difference which exists between accepted standards in trading in radio and television receivers and the retailing and servicing of high-fidelity equipment. The enthusiast in this field invariably possesses a keen understanding of high-fidelity matters, which makes it essential that the knowledge of the dealer and technician should attain the same standard.

Owing to the very nature of "hi-fi", the enthusiast can only get the best from his equipment by understanding how the various sections work and how they are integrated into a perfectly matched system; it is also important to understand the adjustments which are required to maintain optimum performance. In this respect, the book is directed also to the enthusiastic amateur. It explores in a practical manner all the links in the chain, from the original sound at the microphone to the reproduced sound at the loudspeaker. It does not embrace frequency-modulation tuners, however, since these have been described in detail in my book "F.M. Radio Servicing Handbook", issued by the same publishers.

FOREWORD

I have included a full chapter on the subject of stereophony, which will undoubtedly become of major importance in the future. During the preparation of this book, stereophonic sound reproduction graduated from tape to disk, and I have thus been able to include the very latest information on this development.

Owing to its essentially practical presentation, one or two slight ambiguities may be noticed in the text; it is hoped that these will not be held against me by the purist, but ignored in the interests of simplicity of description.

My thanks are due to many manufacturers of high-fidelity equipment for their co-operation in placing so much information at my disposal and for supplying photographs of their equipment. I also wish to record my thanks to my wife Barbara for her tolerance and encouragement during many late hours spent in the writing of this book, to my colleague Peter Berry for his splendid work in preparing certain photographs, and to Mr. N. S. Hyslop, who has prepared excellent drawings from my very rough sketches.

Finally, I am indebted to Mr. L. C. Holmes who has been faced with the formidable task of editing not only the MS. of this work, but also my previous books, "F.M. Radio Servicing Handbook" and "Television Servicing Handbook".

Oxford,
1959.

G. J. K.

CHAPTER 1

Hi-Fi Fundamentals

THE servicing of high-fidelity ("hi-fi") audio equipment demands of the service technician and amateur enthusiast a more critical sense of appraisal of audio reproduction than that required in the case of ordinary sound equipment, where quantity rather than quality is the dominating factor. While the experimenter-enthusiast will be fully aware of this higher critical faculty, and will generally possess it, the man whose job it is to service domestic electronic equipment for a living, taking hi-fi equipment in his stride along with radio and television receivers, may not have such a sensitive ear. If he is not closely acquainted with the foibles of the modern hi-fi enthusiast, the service technician may well be excused his doubts and irritation on encountering the insistence of such an enthusiast that a high degree of distortion is being produced by his apparently excellent amplifier!

Accustomed to a standard of reproduction based on years of servicing radio sets of considerably limited audio fidelity, the technician may feel that the enthusiast's request for service of equipment which even in its alleged faulty condition is capable of reproduction of a high order, is far from warranted. This problem of differing standards can make life exceedingly difficult for the service technician when he starts to undertake the repair of hi-fi equipment. To be really successful at the job, the technician must himself develop "hi-fi" standards of judgment. This is usually automatic, as anyone dealing with the servicing of hi-fi equipment works in close liaison with the enthusiasts who operate it.

A technician new to the field quickly becomes initiated, and quickly realizes that (for instance) where a close-tolerance 50k resistor is stipulated by the maker, replacement cannot be made satisfactorily with a 47k resistor of mediocre tolerance, as can often be done in less exacting equipment with little adverse effect.

Hi-fi equipment does not just happen. It is created in the laboratory by a large number of small points being given a great deal of attention. The net result is hi-fi. Slight disturbance to one or more of these small points, either as the result of alteration in value of a component or unskilled service, will unbalance the design and possibly cause distortion. To the uninitiated service technician the distortion may hardly exist, but to the hi-fi perfectionist a world of difference will be discernible. The technician will have to use instruments on which to base his judgment of reproduction; listening tests waste time and lead to frustration.

Essentially, there are three types of hi-fi enthusiast. First, there is the music lover who wishes to play his favourite records with the minimum of distortion. This type is less technically exacting, since a reasonable quality of reproduction is sufficient to re-create in his mind the atmosphere of the concert hall—slight distortion thus goes unnoticed. Then there is the technical perfectionist whose observations are keenly focused on the various responses of the equipment. This type may not possess a highly developed aesthetic interest in music, but he is able to judge with curious accuracy just how much harmonic distortion there is, how the equipment is handling transients, whether or not additional damping would go to improve the overall results and similar technical matters. He obtains great satisfaction from listening to sounds of large magnitude with little distortion, and when he says that distortion is present it is most desirable for the technician to agree with him—until he can *prove* otherwise, of course!

Finally, there is the type who is a compromise between the other two—he represents the large majority of hi-fi enthusiasts, who are enthusiasts because they are not only technically interested in obtaining distortion-free reproduction, but are also interested in music in itself.

It is as well for the technician new to the hi-fi world to familiarize himself with these three types of enthusiast; this is equally as important as having the technical know-how, for anyone actively engaged in the servicing of hi-fi equipment will soon become aware that he has to be something of a psychologist as well as a technician—and it is most desirable to know one's subject. This is because sound is a *subjective* thing, and since it is this in which we are ultimately interested, it is most important to learn a little about it and its effects before moving on to more objective technical matters.

SENSATION OF HEARING

Sound is the stimulus which when applied to the ear gives rise to the sensation of hearing. It is not wholly true to consider sound as emanating from any particular source. Sound is essentially a function of the listener's ear, nervous system and brain. There would be no *sound* from an explosion, for example, occurring in a place without an ear, nervous system and brain to record it, though there would be considerable air disturbance, to say the least.

The source of any stimulus producing the sensation of hearing is always in some state of vibration. This can be demonstrated by the piano string, the tuning-fork or, to keep in line with our present theme, the cone or diaphragm of a loudspeaker. The vibration may be so slight and so rapid that it is not visible, or it may equally be so large and relatively slow as to be easily observed, as in the case of a loud mains hum affecting the cone of a loudspeaker. It is of little purpose in trying to alleviate the latter condition by securing the speech coil of the loudspeaker cone to the magnet pole piece with good-quality glue—a condition which was once observed by the author when investigating for lack of signals! (However, when questioned, the owner was true to principle in remarking, "but I got rid of the terrible hum which was caused by this cone thing vibrating." A true story!)

In the case of an organ pipe and other wind instruments, the source of the stimulus is a column of air. This can be realized from the considerable agitation of fine dry sand on a piece of paper when brought over the mouth of the pipe. The same effect can be observed by placing the sand-laden paper over the vent of a vented loudspeaker enclosure when the system is fed with low-frequency signals to which the vent is tuned, or resonated. In fact, it represents a good method of discovering the vent resonant frequency—assuming that an audio generator is at hand to feed a variable audio signal to the loudspeaker—and the free resonance of the loudspeaker cone. In the latter case, of course, the sand-covered paper is held over the loudspeaker cone. The reason for the agitation is that the air is moving in and out of the pipe or vent rapidly, and so sets the paper vibrating.

In many cases the vibration can be felt by placing a finger on the string or loudspeaker cone. It is surprising how sensitive the finger can be in this respect; some engineers check for mains hum by lightly placing the finger on the loudspeaker cone. Air vibration can also be felt. Standing in front of a large loudspeaker fully loaded to, say, 10 watts of low-frequency signal readily illustrates this fact.

Any stimulus of sound (in future we shall refer to it as sound in terms of both cause and effect) may vary in three ways, that is, in *frequency*, *loudness* and *quality* or *timbre*. The number of complete vibrations made by a sound-producing device in one second is called the frequency and determines the *pitch* of the resulting *note*. As an example, the string corresponding to bottom A in the piano vibrates at 27·5 c/s.

The loudness of a note or sound is governed by the amplitude of the vibration which, of course, determines the energy applied to the ear. The quality or timbre, which distinguishes between notes of the same pitch sounded by different instruments, results from the presence of harmonics in the make-up of the sound. For the present, these can be considered as subsidiary vibrations whose frequencies are exact multiples of the fundamental vibration.

AUDIBLE FREQUENCY RANGE

As the frequency is reduced, the resulting note eventually becomes resolved into the separate impulses of which it is composed. As the frequency is increased, however, the note becomes very shrill, and at about 15,000 c/s it is little more than a hiss. The high-frequency limit of audibility varies widely with different individuals. The limit is usually higher with young people, often extending to the region of 20,000 c/s, while with increase in age the limit may fall to some 9,000 to 10,000 c/s. Some people are highly conscious of the 10,000 c/s note produced by television receivers, while others, usually older people, are not at all disturbed. The high-pitched squeak of a mouse is often inaudible to people in their fifties, but often very disconcerting to young people.

At this point it should be made clear that a person who is virtually deaf above, say, 7,000 c/s is still able fully to appreciate music containing harmonic components extending well above this figure. It is still necessary for hi-fi equipment employed by such a person to be capable of reproducing all frequencies to the limit of the audio spectrum (the frequency range of good-quality equipment usually extends well beyond the accepted audio range, for technical reasons which will be explained later). Tests have revealed that distortion-free reproduction of music containing harmonic components up to some 18,000 c/s gives the sensation of considerable mutilation, when passed by way of a filter which chops off all frequencies above 7,000 c/s—not only to a person whose hearing is unimpaired up to 18,000 c/s, but also to one who is essentially deaf at 7,000 c/s.

The reason for this, as we shall appreciate better later on, is that a large part of music is composed of steep, rapidly occurring wavefronts (*transients*), produced by harmonic components of the fundamental frequencies of the various instruments. Cutting the higher frequency components has the effect of spoiling the desirable steepness of the wavefronts as well as reducing the overall amplitude of the sound. Since transients are responsible for the "attack" attributable to music, destroying these in a way that impairs the corresponding accelerations of the wavefronts is obvious equally to persons with and without extended frequency range.

There is another important characteristic of the human ear which gives the impression of dissimilar volume to sounds of different pitch. The sensitivity of the ear rises to a maximum in the region of 3,000 c/s, and falls off at frequencies above and below this range.

THE DECIBEL AND THE PHON

While the ear is considerably sensitive to small changes in pitch of a sound it is much less sensitive to changes in amplitude (volume). Instead of following a linear law, the sensitivity of the ear to changes in volume is logarithmic. This simply means that the impression a listener receives when a sound of certain volume is suddenly increased is proportional to the logarithm of the ratio of the energy or *power* of the two sound levels. The common logarithm of the ratio of two powers gives their relationship in *bels*. In mathematical form $Nb = \log_{10} (P2/P1)$. This holds for a decrease in power as well as for an increase in power, so that when $P2$ is less than $P1$ the value of Nb becomes negative.

The whole range of hearing corresponds to a change comprising 13 bels, that is, starting at a power or intensity near the threshold of hearing to a point where the intensity begins to be painful. As 13 bels is too rough a scale for ordinary use, each bel is divided into 10 *decibels* (db). Thus, the difference in level between two powers ($P1$ and $P2$) in decibels is given by $N \text{ db} = 10 \log_{10} (P2/P1)$. This expression holds for any change of power, electrical as well as acoustical. Clearly, the ultimate effect of any change of electrical power in a hi-fi amplifier, for instance, is to produce a change of acoustical power from the loudspeaker. It is as well to become familiar with the decibel, as it crops up frequently in audio work.

As an example, suppose an amplifier delivering 1 watt into a loudspeaker is adjusted to promote an increase of 1 watt. The output is now 2 watts. Although the effect can be realized from the statement that "the power has doubled", there is little point in saying that "the power has increased by 1 watt" unless, of course, it is first clearly indicated that the original power was 1 watt. It is much better to say that "the power has increased by 3 db". Thus, doubling the power is equal to a 3 db increase (3·01 db, to be precise), and halving the power is equal to a 3 db decrease. In the latter case it is usually said that a -3 db power change has occurred.

A change of 2 db, equal to a power of 3 watts being increased to 4·75 watts or decreased to 1·9 watts, for example, is just about discernible by the average person, while a change in level of 1 db is hardly perceptible to the ear.

The decibel is also extensively adopted to compare two currents or voltages. When used in this way it must be ascertained that the resistances (R) in which the currents (I) and voltages (E) operate are the same. When this is the case:

$N \text{ db} = 10 \log_{10} (I2^2/I1^2)$ or $10 \log_{10} (E2^2/E1^2)$, these being equal to $20 \log_{10} (I2/I1)$ and $20 \log_{10} (E2/E1)$.

When the resistances are not equal due allowance has to be made:

$N \text{ db} = 20 \log_{10} (I2/I1) + 10 \log_{10} (R2/R1)$ and

$N \text{ db} = 20 \log_{10} (E2/E1) + 10 \log_{10} (R2/R1)$.

The decibel, as we have already seen, is essentially a unit for measuring relative powers, so when it is employed to express current and voltage gains and losses, allowance has to be made for the fact that power varies by

TABLE 1.1

CONVERSION OF DECIBELS TO POWER AND VOLTAGE/CURRENT RATIOS

db	*Power Ratio*	*Voltage Ratio*	*db*	*Power Ratio*	*Voltage Ratio*
1	1·26	1·12	15	31·6	5·62
2	1·58	1·26	20	100	10
3	2·0	1·41	30	1000	31·6
4	2·51	1·58	40	10^4	10^2
5	3·16	1·78	50	10^5	316
6	3·98	2·0	60	10^6	10^3
7	5·01	2·24	70	10^7	3160
8	6·31	2·51	80	10^8	10^4
9	7·94	2·82	90	10^9	31600
10	10	3·16	100	10^{10}	10^5

the square of the change of current or voltage. For example, an increase in current or voltage by a factor of two results in the power being increased by a factor of four.

When N db is known, the power, current and voltage ratio can be found as follows:

$P2/P1$ = antilog N db/10, $I2/I1$ = antilog N db/20 and
$E2/E1$ = antilog N db/20.

Decibel tables save the toil of making complex calculations, samples being given in Table 1.1 and Table 1.2. Table 1.1 gives conversion of decibels to power and voltage/current ratios. Figures not given in the table may easily be calculated. For example, if two db figures are added, their corresponding power or voltage/current ratios must be multiplied. Table 1.2 gives conversion of power ratios to decibels.

The apparent loudness of any tone is related to its pitch or frequency as well as to its amplitude or intensity. The *phon* is the unit of loudness level actually appreciated by the ear, and represents about the limit of difference in loudness of which the ear is sensible. At a frequency of 1,000 c/s, the loudness level of a pure tone in phons is equal to the number of decibels above the reference power, though this does not hold with any other frequency. It is this apparent non-linear loudness level over the audio spectrum which has recently encouraged the use of "loudness" controls on hi-fi amplifiers. As we shall see later, they function essentially to increase the bass response as the volume is reduced.

TABLE 1.2

CONVERSION OF POWER RATIOS TO DECIBELS

Power Ratio	*db*	*Power Ratio*	*db*	*Power Ratio*	*db*	*Power Ratio*	*db*
1·0	0·000	3·3	5·185	5·6	7·482	7·9	8·976
1·1	0·414	3·4	5·315	5·7	7·559	8·0	9·031
1·2	0·792	3·5	5·441	5·8	7·634	8·1	9·085
1·3	1·139	3·6	5·563	5·9	7·709	8·2	9·138
1·4	1·461	3·7	5·682	6·0	7·782	8·3	9·191
1·5	1·761	3·8	5·798	6·1	7·835	8·4	9·243
1·6	2·041	3·9	5·911	6·2	7·924	8·5	9·294
1·7	2·304	4·0	6·021	6·3	7·993	8·6	9·345
1·8	2·553	4·1	6·128	6·4	8·062	8·7	9·395
1·9	2·788	4·2	6·232	6·5	8·129	8·8	9·445
2·0	3·010	4·3	6·335	6·6	8·195	8·9	9·494
2·1	3·222	4·4	6·435	6·7	8·261	9·0	9·542
2·2	3·424	4·5	6·532	6·8	8·325	9·1	9·590
2·3	3·617	4·6	6·628	6·9	8·388	9·2	9·638
2·4	3·802	4·7	6·721	7·0	8·451	9·3	9·685
2·5	3·979	4·8	6·812	7·1	8·513	9·4	9·731
2·6	4·150	4·9	6·902	7·2	8·573	9·5	9·777
2·7	4·314	5·0	6·990	7·3	8·633	9·6	9·823
2·8	4·472	5·1	7·076	7·4	8·692	9·7	9·868
2·9	4·624	5·2	7·160	7·5	8·751	9·8	9·912
3·0	4·771	5·3	7·243	7·6	8·808	9·9	9·956
3·1	4·914	5·4	7·324	7·7	8·865	10·0	10·000
3·2	5·051	5·5	7·404	7·8	8·921		

HARMONICS

Most vibrating bodies execute a simple harmonic motion, giving a pure tone, or the vibration is composed of a combination of simple harmonic motions, giving rise to overtones, which are usually related in frequency to the fundamental. The sine wave (Fig. 1.1) is representative of simple harmonic motion, such as that produced by the vibration of a tuning-fork.

It is the presence of overtones or harmonics which is responsible for the

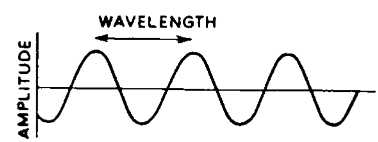
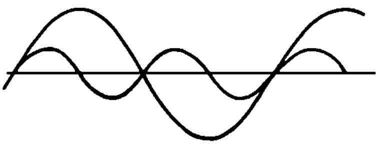


FIG. 1.1. *Simple harmonic motion, such as that produced by a tuning fork, can be represented by a sine wave.*



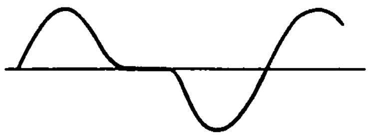


FIG. 1.2. *The two sine waves* (a) *are related in that one has twice the frequency of the other. In* (b) *the waves are compounded to give a composite wave. Here the sound is no longer pure, but has a high second-harmonic content.*

difference in the quality between the sounds produced by the various instruments of an orchestra. The human voice is also rich in harmonics, and since the harmonic content differs between individuals, it is often a simple matter to pick out a certain person by his voice. This is not always the case when contact is by way of the telephone, since this instrument is not wholly responsive to high-order harmonics, its high audio-frequency range being considerably limited, and causing a change in the quality of a voice. This effect is aggravated by speaking through a cloth held in front of the microphone mouthpiece.

Hi-fi amplifiers must be capable of responding fully to all high-order harmonics, and themselves must not be responsible for the introduction of harmonics which are not present in the original sound.

Harmonics consist of notes having 2, 3, 4, etc., times that of the fundamental. The violin, for example, is rich in harmonics at twice and five times the fundamental note to which the string is tuned. The amplitude of the harmonic is also important, and is relatively large in the case of a violin.

In Fig. 1.2 (a) two sine waves representative of simple harmonic motion, one of which has twice the frequency of the other, are given individually, the higher-frequency one being the second harmonic of the lower-frequency fundamental. In Fig. 1.2 (b) the sum of the two waveforms is given graphically, it being obtained by adding the ordinates of the fundamental and second-harmonic waves. A waveform such as shown in Fig. 1.2 (b), being obtained at the output of an amplifier as the result of a pure sine wave input (Fig. 1.1), would indicate most forcibly that the amplifier itself is producing a very large degree of second-harmonic distortion. Apart from being revealed on the screen of an oscilloscope, the distortion would be readily detected, since the ear is capable of recognizing the two sounds, even when they are compounded to form the wave of Fig. 1.2 (b).

TRANSMISSION OF SOUND

Any sounding body causes the surrounding air to be alternately compressed and rarefied in sympathy with the vibrations. As long as the vibrations occur, a wave of high pressure is followed by a wave of low pressure and again by a wave of high pressure, and so on. Compression and rarefaction waves are thus radiated in all directions from the sounding body at 1,088 feet per second, and an eardrum within range will be caused to vibrate in exact sympathy.

Air is the chief medium for the transmission of sound waves, as is clearly revealed by the classic experiment of extracting the air from a bell-jar in which is placed a sounding electric bell. As the amount of air in the jar becomes smaller, the sound of the bell gets weaker. To a lesser degree, all material substances can transmit sound waves. A wood rod, for example, is sometimes used to detect mechanical noises in a car engine. One end of the rod is held in contact with the ear while the other end is held in close contact with the region of the engine being checked for noise. The wood rod serves to transmit the sound waves in this case.

Sound waves in air are known as *longitudinal* waves. This term simply indicates that the particles of the wave-carrying medium travel backwards and forwards in a path whose direction is the same as that in which the wave is travelling. Electromagnetic waves, on the other hand, are known as *transverse* waves, indicating that the particles of the medium travel in paths at right-angles to the path of the wave as, for instance, the waves upon the surface of water.

Sound waves cannot directly be represented by a sine curve, since the particles of the wave-carrying medium remain in a straight line, being compressed and rarefied as we have seen. Nevertheless, it is possible to represent diagrammatically, to scale, longitudinal waves by means of a sine curve. The result is similar to the sine wave in Fig. 1.1. Such a wave possesses four distinct characteristics, which are (1) amplitude, (2) frequency, being the number of complete cycles per second emanating from the sounding body, (3) the velocity at which the wave travels from its source, and (4) wavelength, being the distance between each consecutive peak. The wave will also be endowed with the shape created by harmonics of the fundamental frequency (Fig. 1.2 b).

It is important to remember the relation between wavelength, frequency and velocity which, irrespective of the form of the wave, is expressed as the velocity being equal to the product of the frequency and wavelength, or velocity (V) = frequency (f) times the wavelength (λ). The wavelength can be found by dividing the velocity by the frequency, i.e.,

$$\lambda \text{ feet} = 1{,}088/f$$

This expression can be useful when investigating for standing waves in the listening room, as well as for other purposes.

It sometimes happens that the service technician, hi-fi enthusiast or sound engineer is called upon to supply sound reinforcement in the open air

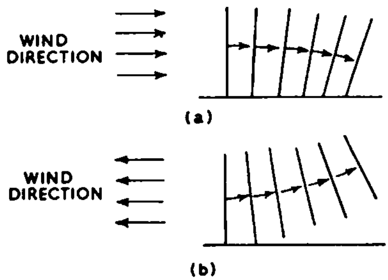


FIG. 1.3. *Showing at* (a) *how sound waves are inclined downwards when the wind is in the direction of the sound, and at* (b) *how the waves are given an opposite tilt when the wind is against the sound. Similar refraction occurs due to temperature variation of the atmosphere.*

—at a fête or garden party, for example—when the question may arise of the effect of wind upon sound waves from the loudspeakers. When the wind is fairly strong it is desirable to place the loudspeakers (with due consideration to the other factors involved) in relation to the listeners so that the sound is travelling with the wind. This is not because the wind affects the intensity of sound, though the velocity would be changed.

The reason that the sound is more clearly heard when it is travelling with the wind than if there were no wind, and vice versa, is that the sound waves are tilted as the result of increasing wind velocity with increasing altitude. This effect is illustrated in Fig. 1.3, where at (*a*) is shown how the waves are inclined downwards when the wind is in the direction of the sound and, at (*b*), the opposite tilt when the wind is against the sound. It must be remembered that the waves always travel at right-angles to their own planes and, under the influence of wind, their velocity is altered with increasing height.

The velocity of sound waves is also affected by temperature. A temperature rise promotes an increase in velocity, and the effects shown in Fig. 1.3 are often produced from this cause. During a hot summer's afternoon, for instance, sound waves may be tilted skywards as the result of the air temperature being greater at lower levels than at higher levels. The converse effect is often experienced when the lower air layers are at a lower temperature than the upper air layers. For this reason, distant sounds are often clearly heard on a cool, still evening, the effect being particularly noticeable over the surface of water.

BEATS

When one is slowly overtaking a noisy heavy goods vehicle in a car whose engine is not unduly quiet, a drumming or beating sound may develop and vary in frequency as the car engine is increased in speed in order to overtake the other vehicle as quickly as possible. When this effect is first experienced, one may incorrectly conclude that the back axle is due for renewal! The disturbance, however, is quite natural, being caused by a beat tone created as the result of sound waves from the two engines combining, the frequency of the beat being equal to the difference in frequency of the two sounds involved.

Such beats are sometimes produced in amplifiers, pick-ups and loudspeakers, and may give rise to spurious tones, referred to as intermodulation distortion, which may or may not be harmoniously related to the tones from which they arise. The distortion usually gives considerable harshness to audio reproduction, as well as to the sound of car engines!

The hi-fi technician will encounter many problems in which resonance plays a leading part. If an audio oscillator is connected across the terminals of a hi-fi loudspeaker system, and the oscillator is tuned fully over the audio spectrum from about 15 c/s to 15 kc/s (15,000 c/s), it will be found that at various frequencies different objects in the room will start vibrating vigorously in sympathy with the sound produced by the loudspeaker. (Let us hope that the loudspeaker enclosure is not subject to such disturbance.) When the sound has a frequency equal to the natural frequency of an object, then the object will vibrate in sympathy with this sound. This process is called resonance.

Heavy damping of the object, due to its design and firmness, will greatly reduce the intensity of the resonance. Loudspeaker enclosures are usually made so as to reduce their natural resonance to the minimum, though at the time of writing a speaker enclosure is undergoing development that is designed intentionally to resonate or flex at certain frequencies. The enclosure panels are designed to resonate at different frequencies as a means of damping the air column resonance within the enclosure, and so spread the effectiveness of the damping over a wider frequency range. It is reasoned that the more conventional method of acoustic damping wastefully converts sound energy at the resonant frequency into heat.

The usual arrangement, which is often adopted for hi-fi, is to use sand-filled panels, or panels of concrete, for speaker enclosures. In this way complete rigidity is secured, and there is little fear of the enclosure walls flexing, even when the alternating sound pressure within the enclosure is at a high level.

Resonance effects are at their height in the small, popularly-priced record-players, often colloquially referred to as "pop boxes"—not a hi-fi term! Here the loudspeakers (or loudspeaker) are contained within a portable housing along with the amplifier and record-player—often an auto-unit is employed. If an audio oscillator is connected across the speaker of one of these devices, things really start resonating within the box. After the case itself has ceased to resonate up to 200 c/s, the valves in the amplifier take over, then the various metal levers of the record unit at about 2,000 c/s, and so on.

When the instrument is used as intended, the box resonance enhances the bass response in a synthetic manner, and when "bop" records are played the other higher-frequency resonances undoubtedly merge with the general background effects. There are on the market, however, quite good portable record-players in which undesirable resonances have been damped as far as is possible. These instruments, of course, are more expensive than the single-valve outfits which are produced essentially for the reproduction of popular music in current demand. Nevertheless, true hi-fi equipment is demanded for true fidelity reproduction, and portable equipment is then completely out of the question. Separate units are essential, and pieces of equipment which are prone to resonance should as far as possible be removed from the loud-speaker system.

The power of resonance is illustrated by the traditional order "break step" which is given to a company of soldiers about to cross a bridge. If the troops' normal rhythmic step happened to coincide with the natural frequency of the bridge, vibrations of large magnitude would be promoted and there would exist a definite possibility of the bridge breaking up.

Apart from the resonance effects of objects, air itself can be caused to vibrate at certain frequencies under controlled conditions. As an example, tuning-forks are sometimes mounted upon hollow boxes so as to increase the volume of sound. The normally feeble sound from a tuning-fork is considerably amplified because the size and shape of the box is arranged so that the air inside possesses a natural vibration period equal to that of the fork. Thus, both the vibration of the fork and the vibration of the air, at the particular tuned frequency in both cases, contribute to the total energy of sound applied to the ear. Such a box is known as a resonator.

This particular effect must not be mistaken for the increase in volume which can be obtained by holding the stem of a tuning-fork in close contact with a table-top or board. In this case, the table-top of board simply serves as a sounding board; *forced vibration* is produced by the fork, and as a consequence the overall vibration is communicated to a much greater quantity of air than when the fork is vibrating unaided.

A well-known resonator is that due to Helmholtz. It was developed some hundred years ago for the purpose of harmonic analysis of a note, and it is still used for this purpose. Such resonators consist (in the original) of a brass spherical shell on which is formed a taper containing a small hole for the purpose of inserting into the ear. Diametrically opposite is a larger opening for presenting to the source of sound.

The air in the resonator resonates to one particular frequency—that to which the resonator is tuned—and when a sound is applied, the resonator picks out and amplifies only that component of the sound to which it is tuned. In this way components of a complex note too feeble to be detected by the ear alone become easily audible and can be checked for relative strength.

Resonators of this kind are made in sets, the note of each being set to the required standard. The resonant or resounding frequency is governed by the volume of air and the area of the pick-up aperture. The frequency is decreased by increasing the volume of air or by decreasing the area of the aperture.

The phase inverter or reflex loudspeaker enclosure adopts the principles of the Helmholtz resonator at the low-frequency end of the audio strectrum.

STANDING WAVES

Resonances also occur in the listening room, as the hi-fi service technician will undoubtedly discover for himself during the process of investigating for poor results in a customer's home on equipment which has previously worked with excellent results in the demonstration room! Such resonances, sometimes referred to as *eigentones*, are produced by multiple sound reflection between the opposite walls, and occur at the frequency at which the distance between the opposite walls is exactly one half-wavelength. This condition gives rise to standing waves at the critical frequency, whilst also considerably accentuating the response at the resonant frequency. In effect, the room serves as a resonator, and the air resounds at the frequency to which the room happens to be tuned.

Further resonances occur as the result of the other two parallel walls and the ceiling and floor, and others governed by the dimensions of the diagonals. The worst conditions occur when the room approximates a cube, with the speaker situated in the centre of a wall. Apart from the chief low-frequency resonance or eigentone at a half-wavelength, others, though possibly less disturbing, present themselves at all harmonics of the basic frequency. Thus, with the main resonance at, say, 40 c/s, created by a cube-shaped room with 14 ft. sides, additional resonances at 80, 160, 320 c/s and so on will also result. Reciprocally, it follows that the reproduction will be exaggerated at frequencies for which the walls are a *multiple* of half a wavelength apart.

ELECTRICAL REPRESENTATION OF SOUND

In all forms of sound broadcasting, recording and reproduction a means must always be provided to convert the sound energy into electricity. Such a conversion device, capable of receiving energy in one form and passing it on in another form, is known as a transducer. The microphone comes under this classification.

All microphones possess a thin diaphragm on which the sound pressure operates, and the resulting vibrations create currents of electricity which rise and fall in precise sympathy with the sound waves. For example, a sounding tuning-fork held in front of a microphone will give rise to a current waveform of frequency coinciding with that of the fork (Fig. 1·4). Similarly, a complex sound wave, composed of a number of tones and harmonic parts,

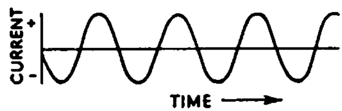


FIG. 1.4. *A sounding tuning fork held in front of a microphone will give rise to a current waveform, as illustrated, of frequency coinciding with that of the fork.*

will be electrically reproduced with equal accuracy. Within limits governed by the design and purpose of the microphone, the electrical output will depend upon the intensity of the sound applied to the instrument. Increasing sound intensity will result in increasing output, and vice versa. The electrical output will also vary with the frequency of the applied sound, though for high-quality work the microphone must respond evenly over the whole of the audio spectrum.

The output of a microphone is conveniently expressed in decibels relative to a fixed reference level. The reference level chosen is sometimes 0 db = 1 volt (open-circuit) with a sound pressure of 1 dyne per square centimetre. Thus, a microphone with an output expressed as −74 db below 1 volt/dyne/cm^2 would have an open-circuit voltage of approximately 0·0002 volts r.m.s. The output is sometimes expressed in terms of power for a stated sound pressure. The RMA rating is defined as the ratio in db relative to 0·001 watt dyne per square centimetre.

At this point it should be noted that a sound pressure of 0·0002 dyne per square centimetre corresponds to the limit of audibility of a 1,000 c/s note. This in turn corresponds to zero phon, and to give the reader some idea of the loudness scale, a quiet room is rated at 20–30 phons, average conversation 60 phons, interior of a tube train with the windows open 90 phons, proximity to an aeroplane engine 120 phons, while 130 phons is approaching the threshold of feeling or pain.

SOUND REPRODUCTION

To be of practical use, the very small power available at the output of the microphone must be considerably amplified, and this has to be performed without alteration of either the character of the electrical waveform, due to the sound waves, or of the response over the entire audio spectrum. With regard to the latter consideration, however, poor acoustics of the room in which the microphone is used (the studio) can sometimes be countered by the use of a frequency-selective network between the microphone and amplifier input. For example, the exaggerated response at low frequencies due to a room of small dimensions is sometimes mitigated by the introduction of a filter network which attenuates the bass frequencies at the microphone, in relation to the higher frequencies, before the signal is applied to the amplifier. This process is known as *equalizing* for room acoustics. Similarly, the equalizing function may take place somewhere in the amplifier chain.

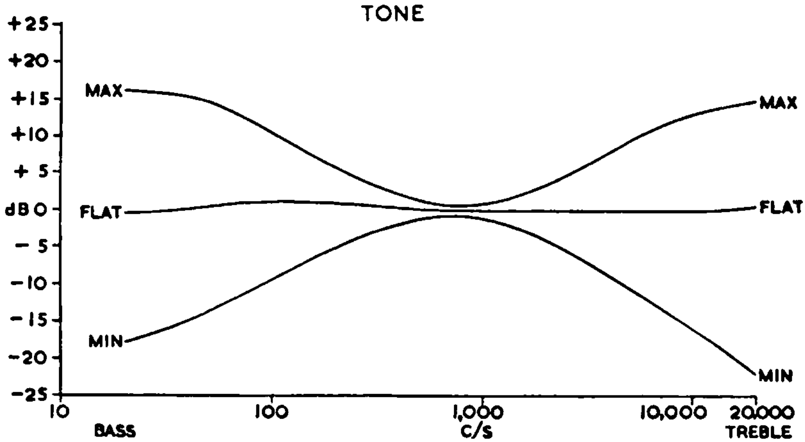


FIG. 1.5. *Curves showing how the bass and treble response of an amplifier can be altered to suit the acoustics of the listening room.*

Most amplifiers are composed of three distinct sections. First there is the voltage amplifier whose purpose is to step-up the small audio-frequency (a.f.) voltages occurring in the varying sound input to a workable level. This section may also contain equalizing networks of suitable form to cater for the various signals for which the voltage amplifier is going to serve. Next comes a tone-control section, in which controls are available for adjusting the degree of amplification of the treble and bass frequencies of the signal, usually relative to 1,000 c/s.

The idea is illustrated in Fig. 1.5. It will be seen that the bass is continuously variable from —12 db to +12 db at 40 c/s, and that the treble is continuously variable from —15 db to +12 db at 10 kc/s. Having such a control of the response of the amplifier aids considerably in the correction of impaired room acoustics from the reproducing point of view. The presence of low-frequency resonances, for instance, can be prevented from over-emphasizing the bass from the aspect of the listener by applying a suitable degree of bass cut and, possibly, treble lift. Conversely, some rooms may be acoustically "dead"; they have a tendency to absorb more of the lower and higher frequencies and thus seem to require more bass and treble than average rooms. Tone controls serve to correct such deficiencies of the listening room and maintain the faithful balance demanded by the hi-fi enthusiast.

Finally, the equalized, amplified and tone-controlled signal is passed on to the power amplifier, by way of a volume or loudness control, and is changed from voltage to power for operation of the loudspeaker. Some equipments have the power amplifier as a unit completely independent of the voltage amplifier and tone-control section, while other smaller amplifiers are complete in themselves. A block diagram of the three sections we have discussed is given in Fig. 1.6.

The loudspeaker is also a transducer, but it operates in the opposite way to that of the microphone; it receives an electrical representation of the sound which was applied originally at the microphone, and passes it on in the form of sound energy. We shall discuss both microphones and loudspeakers in some detail in later chapters.

We now have a complete picture of the whole chain of events, from the sound waves to the microphone, from the microphone through the amplifier to the loudspeaker, and from the loudspeaker to the ear. Let us always bear in mind that the results heard are a function of the mind of the individual, and that they are coloured not only by the equipment used for the reproduction of the sound, but also by the studio and listening-room acoustics. Although it is impossible to match the acoustics of the ordinary listening room with those of the concert hall, it is surprising what can be done synthetically by equalizers and tone controls, not to mention loudspeakers and enclosures!

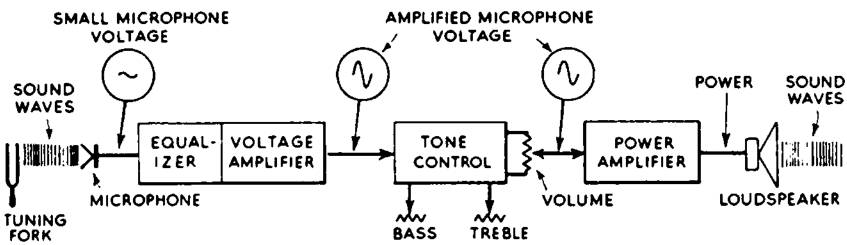


FIG. 1.6. *Block diagram of the three main sections of a hi-fi amplifier.*

We have so far considered "live" reproduction of sound, that is direct from the microphone to the loudspeaker. Of course, sound can be "stored" and used when required. The most popular medium for storing sound in this way is the gramophone record.

Instead of actuating the cone of a loudspeaker to produce sound waves coinciding with those at the microphone, the microphone-amplifier set-up powers a recording head whose purpose is to cause lateral vibration of a sapphire or diamond cutting tool in sympathy with the sound energy applied at the microphone The recording head is tracked radially over the recording blank, while the blank is carried on a heavy turntable which is arranged to rotate at a perfectly even speed. The recording head is also pivoted in such a way that the cutting tool is pressing on the surface of the disk. A spiral groove is thus cut upon the surface of the disk, running from the circumference to near the centre. The sound vibrations impart upon the groove a wavy lateral effect, which is clearly visible on any gramophone record. Sound is thus "stored".

Playback is accomplished by rotating the disk at the same speed at which it was recorded, and by the use of a pick-up carrying a stylus having a hemispherical tip, which rests in the V-section groove cut by the cutting tool. The pick-up is mounted on a tone arm which is free to rotate about a centre some distance from the centre of the record, and is free to move only in an arc which is approximately radial with respect to the record.

The pick-up is thus carried across the record under the control of the groove, while at the same time the stylus is caused to vibrate laterally in sympathy with the lateral waveform imparted during the recording process. The lateral vibration of the pick-up stylus gives rise in the pick-up to an e.m.f. (electromotive force) having the same pattern as the waveform on the record and, in some cases, in proportion to the velocity of the lateral movement of the stylus. This e.m.f. is applied to the input of an amplifier, is suitably equalized, and ends up as sound waves from the loudspeaker—ideally, as a replica of the sound waves applied to the microphone during the recording session.

STEREOPHONIC SOUND

A single-channel (often referred to as *monaural*) reproducer system, i.e., one microphone, one amplifier system and one loudspeaker system into which the single channel is working (more than one microphone and loudspeaker may well comprise a single sound source from the monaural aspect), can never give true fidelity of reproduction. Highly satisfactory reproduction of an orchestra cannot be secured if all the sound is radiated from a hole in the loudspeaker cabinet. The use of two or three speaker systems does not help much in this respect when they are all connected to the output of a common channel.

The "range" of the orchestra can only be realized by the use of two or more completely independent channels. With a two-channel system, which is highly suitable for domestic use, there are two loudspeakers each fed from a separate microphone (or from a separate signal source) through separate amplifiers. The basic idea is to place the loudspeakers relative to one another as the microphones are placed in front of the orchestra, or as they were placed during the recording of the programme.

In this way both ears of the listener are brought into operation in a selective sense. The orchestra appears to be spread in correct proportion across the room, between the loudspeakers, and the listener can pick out the individual instrumentalists as readily as if he were in the concert hall. The "muddiness" of the monaural system disappears completely, and a third-dimensional sense of presence is created.

At present the normal method of stereophonic recording on disks is what is known as the "45/45 system" where the two stereo channels are carried in one groove. This system (described in more detail in Chapter 10) has taken the place of the system where one channel is recorded by the "hill-and-dale" process, as adopted 80 years ago by Thomas Edison in connexion with the phonograph. With this, the cutting tool of the recording head was arranged to oscillate vertically in sympathy with the sound vibrations, so that the depth of the groove corresponded to the wave pattern of the sound; hence the term "hill-and-dale". The other channel was recorded in the same groove laterally, and a special recording head was used to modulate the groove both laterally and vertically in accordance with the two-channel signals applied. On playback, a pick-up functioning electrically opposite to that of the recording head gave two outputs corresponding to the two recorded channels. The signals were amplified independently, and were fed to the two loudspeakers to give the effect of stereophony. Hill-and-dale/lateral and 45/45 stereo recordings differ essentially only in the way in which the signals in the two channels are phased.

MAGNETIC RECORDING

Wire and tape coated with a magnetic material are also used for recording. The wire or tape (wire is now rarely used) is drawn steadily over the pole of an electromagnet, the current in which is caused to follow the wave pattern of the sound. The wire or tape thus becomes magnetized, as each section passes over the pole piece of the electromagnet, to a degree dependent upon the electrical representation of the sound applied at the microphone, and a magnetic wave-pattern is imparted upon the medium.

On playback, the medium is again drawn at the same speed across the pole piece of an electromagnet which this time is not energized, but which has induced in it small voltages corresponding to the varying flux in the core as the result of the magnetized medium. The voltages, representing the recorded sound signal, are applied to the input of an amplifier and end up as sound from the loudspeaker. When required, the recording can be easily erased by passing the medium over a permanent magnet or an electromagnet which is energized by a pure signal having a frequency above the audio spectrum (30–50 kc/s).

This system lends itself readily to two-channel operation, it being a simple matter to record one channel on one half of the tape and the other channel on the other half by the use of slightly displaced electromagnets for record and playback.

Sound can also be stored on film, on the principle adopted for the sound-track on ciné film, but a description of this method falls outside the scope of this book.

CHAPTER 2

Voltage Amplifiers, Feedback and Control Circuits

THE weak signal voltages at the output of the microphone, pick-up or tape recorder have to be increased in magnitude, altered in response to suit the acoustics of the listening room, and finally changed into power for adequate operation of the loudspeaker. The hi-fi amplifier system is required to perform these functions without changing the character of the original signal to any large degree, whilst also maintaining a level response over the whole of the audio spectrum and catering for a wide range of signal levels, from the smallest to the largest, without distortion. The dynamic range of a programme signal may well extend to the region of 60 db, which means that the signal range in terms of input voltage may extend from 2 microvolts to 2 millivolts, resulting in a few milliwatts of output from the loudspeaker on the quietest signal or several watts on the loudest signal.

The first duty of the amplifier is to step-up the signal voltage to a level suitable for operating the output stage, or power-amplifier section. Bearing in mind that the power amplifier may require a signal level approaching 1·5 volts to fall within its driving range and that the programme signal may only average 1 millivolt, one can clearly realize the necessity of voltage amplification of some 70 db.

At this juncture we should note that the term "voltage amplifier" indicates that the re-creation of the amplified input signal across a comparatively high impedance load is the essential function of this section. Obviously, since an infinite output impedance does not exist in practice, it must be concerned also with the amplification of power. However, the output impedance is usually of the order of 0·1 to 1 megohm, so *voltage* is the predominant factor. We need not consider here when an amplifier ceases to be concerned with voltage and changes over to power, but simply regard power amplifiers as those used to drive a loudspeaker.

A typical pentode voltage amplifier is shown in Fig. 2.1. There is nothing

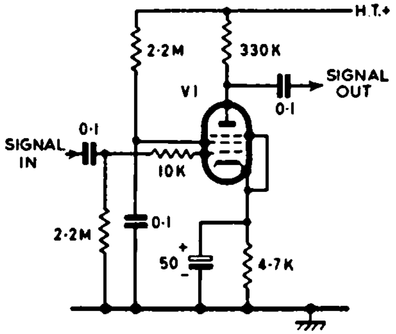


FIG. 2.1. *Pentode voltage-amplifier stage.*

about this circuit as it appears on paper to indicate that it is associated with hi-fi equipment. In practice, however, it will be found that possibly the anode and grid resistors are of the "low noise" variety, also the valve (i.e. Mullard EF 86 or GEC Z729—low-noise types). Careful choice of circuit parameters ensures that the stage is operating at the lowest distortion figure within its voltage range. Hum problems are also taken care of by the use of carefully placed wiring, and by having a common "earthing" point so as to avoid producing common impedances across which hum voltages may develop.

A triode voltage amplifier is shown in Fig. 2.2. In practice, V1 is usually one half of a double-triode valve, the other half being employed also in the voltage amplifier, with a frequency-selective feedback loop (usually switched) providing various degrees of equalization. This method is adopted in the Pye "Proctor" pre-amplifier. Generally speaking, there is little to choose between a modern pentode and triode. Greater gain is usually possible from the pentode, though such a stage does not give such freedom from intermodulation distortion at high output voltages as does the triode. Nevertheless, this is of little moment when the signal to be handled is of very low level.

NOISE AND HUM

Owing to the very low level of the signals to be amplified by the first stage, a problem peculiar to this section is that of "noise". Noise in this sense refers to all spurious disturbances which give rise to unwanted and undesirable signals across the output load along with the required signal. Since a quiet orchestral passage may give rise to only a few microvolts of signal across a pick-up or the playback head of a tape recorder, the stage will obviously

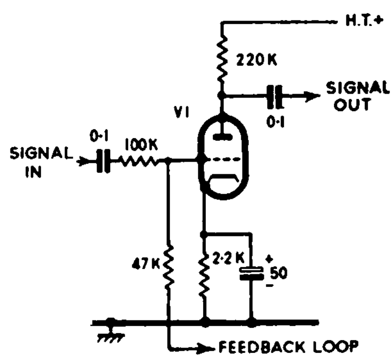


FIG. 2.2. *Typical triode voltage-amplifier stage.* V1 *usually comprises one half of a double-triode, the other half-section being used to complete the stage. A frequency-selective feedback loop may also be used for the purpose of equalization.*

be susceptible to the pick-up of stray hum signals and noise caused by incorrect or faulty components.

Ideally, the only noise voltage which should be present with the signal across the output load is that attributable to the random behaviour of electrons in the resistive components and in the valve. This is sometimes referred to as *white noise*, and is characterized by the hiss which emanates from the loudspeaker when the volume control of a very high-gain amplifier is turned full on. White noise is not confined to any particular frequency, but is distributed throughout the entire spectrum. If the equipment has peaks over its response, the effect of the noise will be considerably emphasized at the frequencies corresponding to the peaks. Because our ears tend to "peak" around 3 kc/s, white noise resolves as a hiss, focused on 3 kc/s. An interesting test is to apply white noise to the input of a hi-fi amplifier by way of switched filters serving to attenuate progressively the high- and low-frequency components of the noise. A filter tuned to around 600 c/s changes the hiss to a whistle, while a filter tuned to the low-frequency end of the spectrum gives rise to a roar.

With practice, white-noise tests of this nature permit rapid appraisal of the performance of hi-fi equipment, particularly if an oscilloscope can be used in the tests (remembering that the character of white noise is rather like that of transients, about which more will be said later).

Unfortunately, apart from white noise, there often exist other and more disturbing spurious signals across the output load of the voltage amplifier. Hum is a big bugbear in this connexion. Hum poses much more of a problem in hi-fi equipment than in ordinary domestic radios of limited low-frequency response. In the first stage, hum is invariably induced into the input circuit from either stray electromagnetic or electrostatic fields. In most hi-fi amplifiers, the control-grid circuit of the voltage amplifier is reached by way of the programme-selector switch, which gives the positions "gram", "tape" and "mic".

Hum caused by stray fields will diminish on backing-off the volume control, since the control is usually located in the circuit following the output of the voltage amplifier. Operating the selector switch also gives conclusive evidence as to whether the hum pick-up is common to all circuits or a shortcoming of one particular channel. If the amplifier is in good order, as can nearly always be proved by these simple tests, it is safe to assume that the hum signal is gaining admittance either by way of the leads connecting to the various sources of programme signal or by way of the external units themselves.

Electromagnetic induction occurs mainly in circuits of low impedance, and demands a complete loop into which induction can occur. For example, a low-impedance magnetic pick-up may enter an electromagnetic hum field as it traverses the turntable. The hum field may emanate from the gram motor or from a power transformer situated nearby. Whatever the cause, a small voltage (in terms of microvolts) will circulate the circuit comprising the pick-up coil and the primary of the matching transformer, but this voltage will appear at the grid of the valve stepped up in the same ratio as the matching transformer. Thus, an induced voltage as small as 2 microvolts will rise to 100 microvolts at the grid with a transformer having a turns ratio of 50:1, which is a reasonable value for an input transformer.

Electromagnetic induction of a similar nature may well occur in a low-impedance microphone circuit, in the circuit of the playback head of a tape recorder, or even at the coupling transformer. The overall loop effect can be obviated by employing either a tightly twisted pair of conductors or a coaxial line between the programme signal source and the low-impedance amplifier input. There is little purpose in using parallel conductors as these aggravate the loop effect, and a screening over such conductors offers little or no protection against electromagnetic fields.

The susceptibility of the inductor at the low-impedance signal source (such as the winding and core associated with a low-impedance magnetic pick-up) and the coupling transformer at the amplifier end in responding to electromagnetic fields, particularly those at mains frequency (50 c/s), can be reduced or almost eliminated by the use of high-permeability magnetic shields. The effect of such a shield at low frequency is shown in Fig. 2.3.

Where higher-frequency electromagnetic fields are present, more elaborate screening is usually called for. With a microphone transformer, for instance, it is often necessary to house it in a case formed of several shields, two of the type described and an intermediate one which operates by inducing into itself a field which opposes the offending field.

Electrostatic induction rarely affects low-impedance circuits, since the electrostatic charge is quickly dissipated around the low-impedance loop. However, in the case of a low-impedance circuit isolated from chassis, an electrostatic noise charge may appear at the grid as the result of the charge developing between the low-impedance circuit as an entirety and chassis. This can usually be cleared simply by earthing the low-impedance side of the circuit, preferably at a centre-tap on the primary of the transformer.

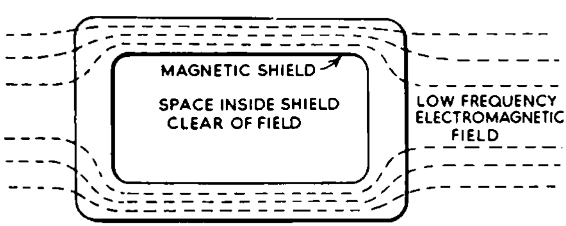


FIG. 2.3. *A magnetic shield serves to protect the transformer or inductor from stray electromagnetic fields.*

Long lines between the programme source and the input of the amplifier should always be avoided if hum is to be kept at the lowest level. If some reason or other nccessitates long leads, then 600-ohm transformers should be employed either side of the line, that is at the programme source and at the amplifier. In this way the danger of introducing hum due to different earth points can be avoided.

High-impedance connecting links should always comprise shielded cable having a low value of capacitance, bearing in mind that excessive capacitance at high impedance will impair considerably the high-frequency response of the programme signal. Television coaxial cable can be used successfully for this purpose, but if the lead is extra long and is liable to be moved about a lot—for example, if it is used as a trailing high-impedance microphone cable —microphony effects may become troublesome. These resolve as the result of small changes in resistance of the outer conductor as the cable is moved and give rise to a "brushing" noise from the loudspeaker. If the cable is dropped, the effect from the loudspeaker is similar to that obtained by tapping the microphone.

Enthusiasts living close to a powerful television transmitter often complain of hum caused by the pick-up of the vision signals. In some cases the sound transmission is also troublesome. This unwanted pick-up gains admittance to the equipment either at the input circuits of the voltage amplifier or on the negative-feedback loop, by way of the speaker leads. It can be cured without difficulty by the insertion of a television choke in the speaker leads, as close as possible to the amplifier, or in the control-grid circuit of the voltage-amplifier valve, as close as possible to the grid tag. A choke (or chokes) can easily be made up by forming a self-supporting coil from 18 s.w.g. tinned copper wire, with the wire equal in length to that of a quarter-wavelength of the frequency of the offending signal (67 inches for Channel 1, 59 inches for Channel 2, 53·5 inches for Channel 3, 49 inches for Channel 4, and 45·25 inches for Channel 5). ITA stations have not yet been reported as responsible, possibly because of the higher frequency used.

Trouble similar to this was reported on a hospital call system. In this case, though, the pick-up was from the v.h.f. transmitter employed in ambulances. The trouble was cured by the insertion of chokes and the making good of poor connexions on the call-system microphone. In the latter respect, the trouble was being aggravated by rectification (demodulation) of the signal caused by high-resistance joints.

VALVE MICROPHONY

Valve microphony is another factor which affects the voltage amplifier. The effects are ringing from the loudspeaker and a definite "ping" when the valve is tapped with a finger. Essentially, the trouble is caused by vibration of the electrode structure, promoting corresponding signal fluctuations across the anode-load resistor. High-slope triode valves are more susceptible to the effect than pentodes, particularly older-type triodes. Modern valves are less prone to the trouble, and circuit techniques help, as in these days it is not common practice to run the valve for maximum gain. Microphony is aggravated by vibrations from the loudspeaker, particularly when the speaker is situated in the same cabinet as the amplifier.

After the first amplifier stage, the signal is usually large enough not to be affected by problems of noise, since then the noise voltage is a very small ratio of the signal voltage. Apart from signal amplification, the duty of the first stage is that of securing the highest possible signal-to-noise ratio, and this is no mean task when it is considered that the applied signal voltage may well have a magnitude of only a few microvolts on soft passages of music.

As an aid in maintaining a good signal-to-noise ratio, the full signal voltage is invariably applied to the control grid of the first valve; the volume control being introduced after initial amplification when the signal is at much higher level. There are times, however, when the programme signal itself is at high level; for instance, when a high-output pick-up is used or when an amplifier is incorporated in a radio tuner or tape recorder. When this is the case, some form of attenuation is needed between the programme source and the amplifier input to avoid overloading the first valve.

FEEDBACK

One cannot progress far into hi-fi before coming up against feedback. There are two kinds of feedback, positive and negative. Positive feedback means that a portion of the output signal of an amplifier is fed back to the input in the same phase as the applied signal. The application of positive feedback results in an increase in gain of the amplifier, and instability and oscillation when the feedback exceeds a certain degree. Positive feedback is the *modus operandi* of oscillator circuits. Negative feedback, on the other hand, results in degeneration, and is arranged by feeding back a portion of the output signal in opposite phase to the applied signal.

There are also two modes of feedback, current and voltage. The former

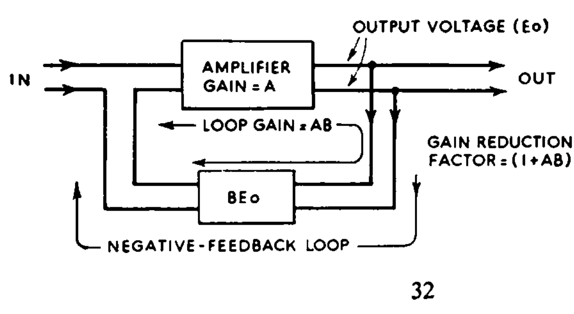


FIG. 2.4. *Block diagram showing the application of negative feedback to an amplifier.*

occurs when the feedback voltage is proportional to the output *current*, and the latter when the feedback voltage is proportional to the output *voltage*. The most common method of obtaining current feedback is by the use of an un-bypassed cathode resistor in a valve amplifier. Here, the signal voltage across the cathode resistor, being in proportion to the current, is reflected anti-phase into the control-grid circuit.

Now for a few simple expressions.

The gain of an amplifier stage is reduced by the omission of the cathode capacitor by the factor

$$\frac{1}{1 + Rk\dfrac{(1 + \mu)}{ra} + Z}$$

where Rk is the cathode resistor, μ and ra are the amplification factor and a.c. resistance of the valve, and Z is the anode coupling impedance. The expression can be reduced to

$$\frac{1}{1 + g\,Rk}$$

where g is the mutal conductance of the valve, when ra is large compared with Z, as is often the case with pentode valves.

The block diagram in Fig. 2.4 represents an amplifier with a negative-feedback loop. Writing A for the gain of the amplifier without feedback, and B for the fraction of the output voltage fed back, it can be stated that

$$\text{Gain with feedback} = \frac{A}{1 + AB}$$

The factor $(1 + AB)$ is known as the gain reduction factor, and may be expressed in decibels.

As an example, in an audio amplifier we may have $A = 180$ and $B = 1/20$, giving $AB = 9$ and the feedback factor $(1 + AB) = 10$. The gain with feedback is then $180/10 = 18$. In this case, the feedback has reduced the gain of the amplifier 10 times, which is the same as saying that the amplifier has 20 db feedback. Clearly, the input signal required with feedback to secure the original output must be $(1 + AB)$ times the input signal without feedback. In other words, in the foregoing example, the application of negative feedback has made it necessary to increase the input signal 10 times (20 db) in order to obtain the original output signal.

The term "loop gain" often appears in relation to feedback. It refers to the factor AB, which is the gain that would be indicated by applying the signal at the grid of the first valve and measuring the signal at the output of the feedback loop.

Within limits, the application of negative feedback reduces distortion in the same ratio as it reduces the gain. For example, a small amplifier without feedback may well produce something like 10 or 11 per cent distortion (possibly made up of 10 per cent in the output stage and 1 per cent in the preceding stages). The application of 20 db feedback will not only reduce the gain of the amplifier by a factor of 10, but also the distortion by the same factor—in this case, bringing it down to something like 1 per cent. This is, indeed, a useful application and one which is practised extensively in all hi-fi equipment. We shall see later that modern equipment includes a number of feedback loops each serving a specific purpose.

Positive feedback can be added to improve even further on the distortion figure. If we again consider the small amplifier mentioned above, and apply, say, 6 db *positive* feedback over the stages preceding the output stage, both the distortion and gain of these stages will be increased by a factor of 2 (positive feedback increases both the gain and distortion, as would be expected). The distortion in the first two stages will thus rise from 1 per cent to 2 per cent.

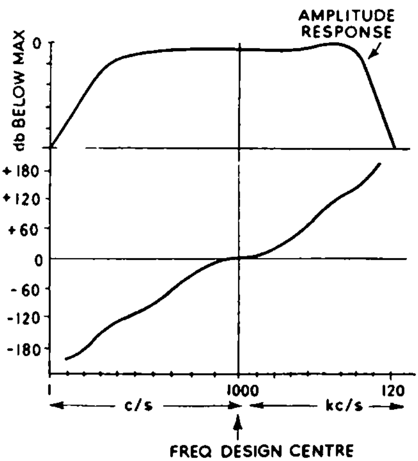
Without *negative* feedback, therefore, the overall distortion will now be in the region of 12 per cent. Now if the fraction of output voltage B fed back is maintained as in the original example, the negative feedback will be 26 db instead of the original figure of 20 db. This, of course, is because A has been doubled, as also has the loop gain (AB). This means that the distortion will be reduced by a factor of 20 (26 db), which brings it down to something like 0·6 per cent.

If the positive feedback is further increased the negative-feedback loop gain will rise in proportion and the distortion will reduce accordingly. As with all things, there is a limit to which positive feedback and negative feedback can be increased. However, this device lends itself admirably to the use of triode voltage amplifiers, as distinct from pentodes. With a double-triode valve, for example, positive feedback may well permit a stage gain of some 2,000 times, thus allowing the use of some 20 db more negative feedback as compared with a pentode.

FEEDBACK STABILITY

The capacitive and inductive circuits employed in amplifiers can really disturb the application of feedback. So far we have assumed that the voltage fed back as negative feedback is exactly 180 deg. out of phase with the input voltage, and that the voltage fed back as positive feedback is exactly in phase with the input voltage. Unfortunately, these ideal conditions hold for only a comparatively small range of frequencies. Series capacitive reactances, such as coupling capacitors, cause a progressive phase advance, while leakage inductance in transformers and odd stray capacitances in the circuit promote a progressive phase delay. The phase change between the input and output signals occurs progressively, positively and negatively, with frequency either

FIG. 2.5. *Typical amplitude and phase characteristics of an audio amplifier.*



side of the frequency design centre of the amplifier. The effect is illustrated graphically in Fig. 2.5.

Such phase change with frequency causes the feedback voltage to vary in phase in relation to the input voltage either side of the optimum for the feedback employed. However, for negative feedback to change to positive feedback its phase must change through 180 deg. Whilst this is unlikely to happen within the range of the audio spectrum, it may take place at a frequency well above audio (supersonic frequency) or at a very low frequency. At these frequencies, the response of the amplifier is likely to have fallen to a comparatively low level, and this, of course, will result in the loop gain (AB) reducing in a similar ratio. The point is that for oscillation to commence, due to the change from negative to positive feedback, the factor AB must be unity or greater. In other words, to meet the conditions of stability, the product AB cannot be greater than 1.

Feedback design margins are employed to ensure this condition, the "phase margin" being defined as the angle by which the phase differs from 180 deg. at the frequency where the loop gain (AB) falls to unity. A margin of 30 deg. is usually aimed at, giving a maximum phase shift of 150 deg. The "gain margin", on the other hand, is defined as the amount by which the loop gain (AB) is below unity at the frequency where the phase shift is 180 deg. A typical value for the gain margin is 10 db.

The application of positive feedback within a negative feedback loop, as already discussed, can play havoc with the stability margins, and extreme caution is needed when this form of feedback is applied. It sometimes pays to make the positive feedback somewhat frequency-selective so that it is applied in full force over one side of the spectrum only. Special attention can then be given in terms of stability to the circuit parameters concerned with phase shift at the frequency over which the feedback is applied.

When negative feedback is applied over more than two stages the possibility of instability is considerably increased. Very low-frequency oscillation may commence, causing the speaker cone to pump to and fro at one or two cycles per second. This effect may not be heard as such, though it can seriously degrade the reproduction. Similarly, very high-frequency oscillation is inaudible, but it can upset the operating condition of the output stage, causing the grid circuits to pass excessive current.

ADVANTAGES OF NEGATIVE FEEDBACK

Apart from reducing harmonic and intermodulation distortion to a very large degree, negative feedback also reduces spurious noises and signals, such as hum and microphony, in the same ratio as it reduces the gain. It also has a marked effect on the stability of the parameters of the amplifier due to random changes within the feedback loop. An example in this latter respect is an amplifier which without feedback has its overall gain reduced by some 10 per cent as a result of a drop in mains voltage; the same amplifier with 20 db negative feedback has its gain reduced by only 1 per cent, with the same drop in mains voltage. The reason is that the effect of gain changes is reduced by the feedback factor, which in the case cited is 10:1 (20 db).

Negative feedback also considerably improves the overall frequency response of an amplifier. In this case, however, it is not true to say that the feedback improves the frequency response in the same ratio as it reduces the gain, as in the other cases mentioned. The reason is that when the response of the amplifier falls the frequency is well removed from the frequency design centre of the amplifier, and at these frequencies the phase of the signal fed back differs somewhat from the ideal of 180 deg. The feedback is thus not 100 per cent negative, though its phase may be well within the range required for stability.

An important function of feedback so far as hi-fi amplifiers are concerned is the effect it has on the anode impedance of the output valve. The application of feedback does not alter the optimum load of an output valve, as is sometimes thought, though it does alter the *source* impedance or the impedance as "seen" by the loudspeaker. The source impedance is *decreased* by the application of negative voltage feedback or positive current feedback, and *increased* by negative current feedback or positive voltage feedback.

We shall see later that a low source impedance is desirable from the point of damping the loudspeaker so as to avoid the cone oscillating to and fro after the application of a steep transient signal. For negative voltage feedback the source impedance (sometimes known as effective output impedance or resistance) is equal to $\frac{ra}{1 + AB}$. For current feedback ra is multiplied by the feedback factor.

DAMPING FACTOR

The ratio of the nominal output impedance of an amplifier to the source impedance is known as the damping factor. A damping factor of 30 is typical for modern equipment. This would be due to the nominal 15 ohms output impedance divided by 0·5 ohm source impedance. Clearly, a low source impedance results in a high damping factor. A number of commercial amplifiers utilize a variable damping control so that the damping factor can be "optimized" to suit the speaker used. However, in practice it is found that little is to be gained by increasing the damping factor above about 20, particularly if a good-quality speaker is correctly loaded in an enclosure. Nevertheless, some claim that all types of loudspeaker call for different damping factors, and that a variable means of achieving the critical damping is essential.

One should take into consideration the resistance of the speech coil, which is effectively in series with the damping circuit and represents the dominant impedance at high damping factors. For example, an amplifier having a damping factor in the region of 10 will have a net damping circuit impedance of some 13 ohms with a speaker whose speech coil measures 12 ohms. An increase in damping factor to 40 will reduce the damping circuit impedance by less than 1 ohm, bringing it down a little above 12 ohms.

The use of positive and negative feedback loops in the output stage, and a means of varying the positive feedback, allows the variation of the damping factor from about 30 to infinity. An infinite damping factor means that the source impedance falls to zero ohms; beyond this point the source impedance becomes negative, and if the negative output impedance is greater than the load impedance the amplifier commences to oscillate.

It must be stressed that negative feedback is not a cure for all amplifier ills; a poorly designed amplifier cannot be made into a high-quality one simply by applying or increasing the feedback. Indeed, as has already been intimated, increasing the feedback on an amplifier of dubious characteristics may well turn it into an oscillator, even if the feedback appears to be of negative mode. The phase shift in the output transformer and coupling impedances, particularly in a not-too-costly output transformer, extends to high degrees at the limits of the audio band. This factor severely limits the amount of feedback which can be applied. There are one or two methods of reducing this phase shift which will be considered later.

Apart from instability, an amplifier of inherently high distortion will benefit but little by negative feedback. It may well happen that the *order* of the harmonic distortion may change by the application of feedback in a case such as this; for example, second and third harmonic distortion may be changed to fourth and ninth.

PROGRAMME SELECTION AND EQUALIZING

The section of the hi-fi amplifier which deals with voltage amplification, programme selection, the control of volume, loudness and tone, signal filtering and slope of filtering is known as the *pre-amplifier*, or control unit. This section may be independent of the power amplifier and connected to it by means of a cable, as with the Pamphonic Type 2,001 amplifier, the Pye HF25/A and many others, or it may be an integral part of the amplifier, as with the Pamphonic Model 1,004 and many smaller amplifiers of 10-watt rating.

Whether independent of or integral with the power amplifier, the function and general characteristics of the pre-amplifier are essentially unchanged. With independent units, features in addition to the basic requirements are sometimes embodied, additional filtering, a slope control and extra equalizing positions being typical in this respect. Hi-fi outfits comprising separate units are invariably more expensive than their composite counterparts. More scope can thus be given to the designer to facilitate the development of his pet feature; more money is available for the extra components needed and there is more room available on the chassis since size is not restricted as is the case with some composite units.

Most pre-amplifiers are designed to cater for four programme sources, namely, pick-up (gram), radio, tape, and microphone. The programme required is selected by a rotary switch (selector switch) and the signal eventually finds its way to the control grid of the voltage-amplifier valve. There are four input sockets, of course, corresponding to the channels available, and the signals can be present on each of the four sockets ready for immediate selection when required. To avoid a strong signal on a channel which is not selected from breaking through along with the signal on the selected channel, the sockets corresponding to the channels not in use are sometimes short-circuited by means of an additional wafer switch ganged to the selector switch. The Pye HF25 has such a feature.

In addition to the four programme positions, the selector switch may also have three or four positions relating to the pick-up channel, giving six or seven positions in all. The extra positions on the pick-up channel permit the selection of the most suitable equalization characteristic for the record being played.

Unfortunately, over the years, records have been cut with a diversity of recording characteristics, each demanding a slightly or greatly modified equalization characteristic. However, since the universal acceptance of the R.I.A.A. (Radio Industries Association of America) recording characteristic —known in Britain as British Standard No. 1928—there will soon be little need for complex switched equalizing circuits. The bulk of the long-playing repertoire in future years will have been recorded to this characteristic. For the present, three or four degrees of equalization are desirable to cater for disks which have already been cut to suit one or other of the re-play curves given in Fig. 2.6.

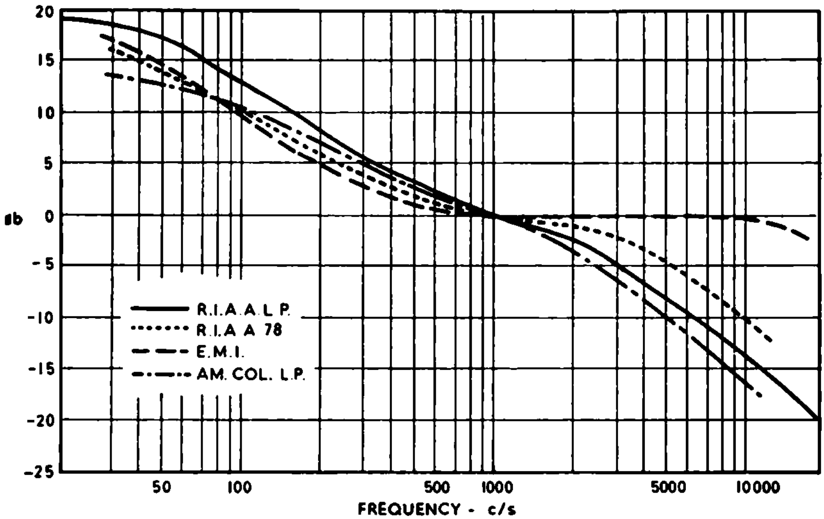


FIG. 2.6. *Equalization curves in current use.*

For reasons closely related to the design of pick-ups and signal-to-noise ratio on re-play, disks are cut with a rising high-frequency response and a falling bass, the opposite to the curves in Fig. 2.6. The reasons for this are given in a later chapter. Clearly, in order to secure a straight-line frequency response, equalization curves must be the opposite of the record curves.

PRE-AMPLIFIER FIRST STAGE

Let us now investigate a circuit providing the functions so far outlined. In Fig. 2.7 is given the first-stage circuit of the Pamphonic Model 2,001. In this, the circuit arrangement associated with the selector switch clearly follows the lines laid down in this chapter. The signal, after selection by SW1A, is applied to the grid of V1A by way of R2 (valve V1 being of the double-triode variety, Mullard ECC83). Pre-set controls are available on the pick-up, radio and tape channels so that the signal level can be pre-adjusted to avoid overloading the first stage where the programme-source signal is of high level.

The amplified signal appearing across R4 is coupled capacitively to the grid V1B through C2 and C10. A portion of the signal appearing across R4 is also fed back to the grid of V1A by way of the switched resistor and capacitor networks. This results in selective attenuation due to negative feedback and thus provides various degrees of equalization as selected by switch SW1B, which is ganged to the selector switch SW1A.

Bass correction is controlled by capacitors C6 and C8 (bass boost) with

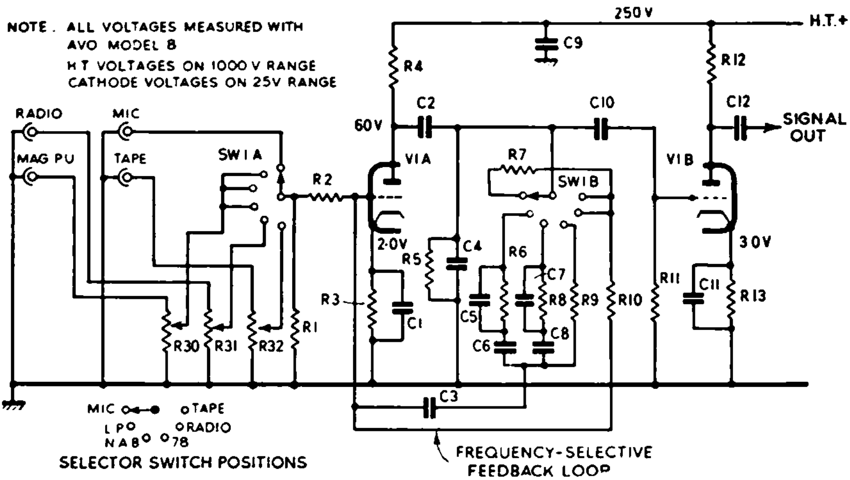


FIG. 2.7. *The first-stage circuit of the Pamphonic Model* 2,001 *amplifier.*

a very low-frequency roll-off, due to the progressive reduction in feedback at the lower frequencies. Capacitors C2 and C3 are concerned essentially with the coupling of the loop and have little material effect on the response, being of high value—32 and 50 mF respectively. Capacitors C5 and C7, along with their associated resistors, control the top-cut. A certain degree of fixed correction is also given by R5 and C4.

The arrangement does not apply any appreciable amount of negative feedback to the programme circuits, owing to the isolating resistor R2, and it represents one of the most popular equalizing circuits in hi-fi use today. It is extremely quiet, stable and efficient in that the unwanted gain is employed as a distortion-reducing agent.

On the "mic", "tape" and "radio" positions, the feedback is not frequency-selective, being controlled by resistors R7 and R10 respectively. Since high gain is required on the microphone channel relative to the other channels, the feedback is considerably reduced by R7 being of much greater value than R10, and by R7 and R10 being in series when the selector switch is set to "mic". Thus, apart from providing equalization, the negative feedback can be switched to facilitate a balanced output in spite of the various voltages on the four channels.

The equalized signal is further amplified by V1B, and appears in this form across the load resistor R12. From here it is conveyed through C12 to the following section of the pre-amplifier circuit, which is usually the tone-control section.

The circuit in Fig. 2.7 is designed to match or load most magnetic pick-ups of the hi-fi type. It must be stressed that every pick-up, irrespective of type, has an optimum load which provides the correct curve to work into the equalizing network as stipulated by the makers. With magnetic pick-ups, a load larger than the optimum results in an increase in high-frequency output, while an increase in low-frequency output occurs with crystal pick-ups.

The practice of incorporating facilities for correct impedance-matching of various pick-ups is increasing in popularity among amplifier manufacturers. The Armstrong and R.C.A. units adopt two input sockets, one for magnetic pick-ups and the other for crystal types, which can be switched independently. The Pye Mozart, on the other hand, features two controls (pre-set), one for providing attenuation and the other for matching. This is known as "Dialomatic Pick-up Compensation", and permits immediate matching for any pick-up. A list of settings for a very wide range of pick-ups is given in the instruction manual. A similar idea is used on the Decca FFR25. Other manufacturers use plug-in matching units.

Some simple equalizing networks rely on the fact that the response of the pick-up is altered with alteration in load resistor.

TONE CONTROL

The most popular tone-control system, giving independent control of both bass and treble, is that due to P. J. Baxandall (*Wireless World*, October, 1952). The circuit of the network is given in Fig. 2.8, from which will be seen that it is focused around the triode valve V1, this usually being one half of a double-triode. Basically, the operation of the circuit relies upon frequency-selective negative feedback, the feedback loop being by way of the anode of

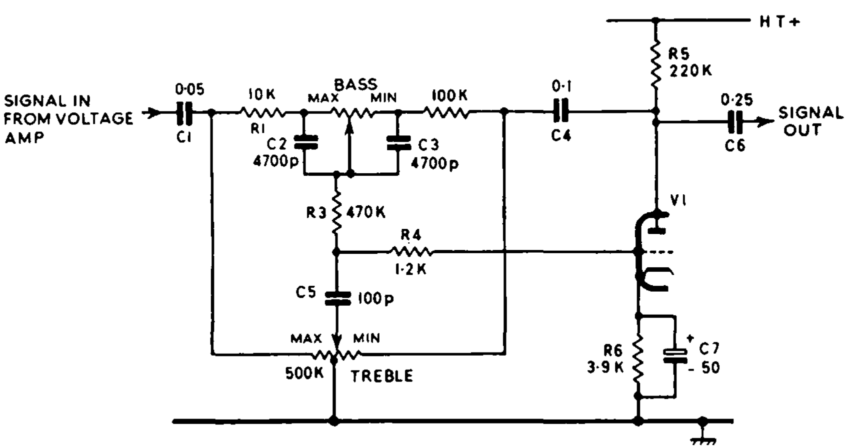


FIG. 2.8. *Baxandall tone-control circuit, with typical component values.*

the valve, through C4 and the resistor/capacitor network, and back to the grid circuit through R4.

The overall control is formed by the amalgamation of two somewhat complex independent treble and bass control circuits. In order to secure both treble lift and treble cut, the treble control has a tapped resistive element connected to chassis. The treble-lift elements are C5 and the section of the treble control connected across chassis and the junction of C1, R1, while the treble-cut elements are C5 and the section of the treble control connected across chassis and the junction of R2, C4. Bass lift and cut is given by the bass control in association with C2, C3, R1, R2, R3 and back to the grid by way of R4.

The lift and cut is confined to each end of the audio spectrum, thus permitting extreme frequencies to be lifted and cut to a large degree without disturbing the response at the centre of the spectrum.

There are a diversity of tone-control circuits which rely simply on filtering the signal to varying degrees to provide the necessary boost or cut either side of the spectrum. Such an arrangement is sometimes called a "passive tone control". Two simple resistor-capacitor networks of this kind are shown in Fig. 2.9; circuit (*a*) providing treble control and circuit (*b*) bass control. A combined treble- and bass-control circuit is also a fairly common set-up, and one which in some quarters is held in favour over the feedback arrangement, possibly owing to the greater flexibility of response over a wider frequency range.

Passive networks can be inserted between two voltage amplifiers since, in common with all tone controls and equalizers, inevitable attenuation results from the circuit, and this must be made good by additional amplification. Generally speaking, it is not a good idea to include the network in a low-level stage, such as in the input circuit of the first voltage amplifier, for the reason given above. This may not apply, however, to pick-up equalizers if the pick-up used has a large output voltage.

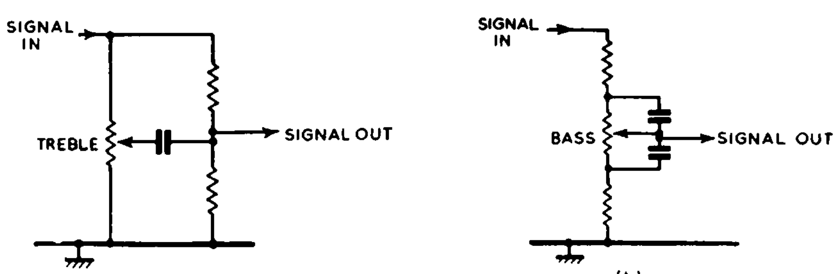


FIG. 2.9. *Two simple resistor-capacitor passive tone-control circuits:* (a) *providing control of treble and* (b) *control of bass.*

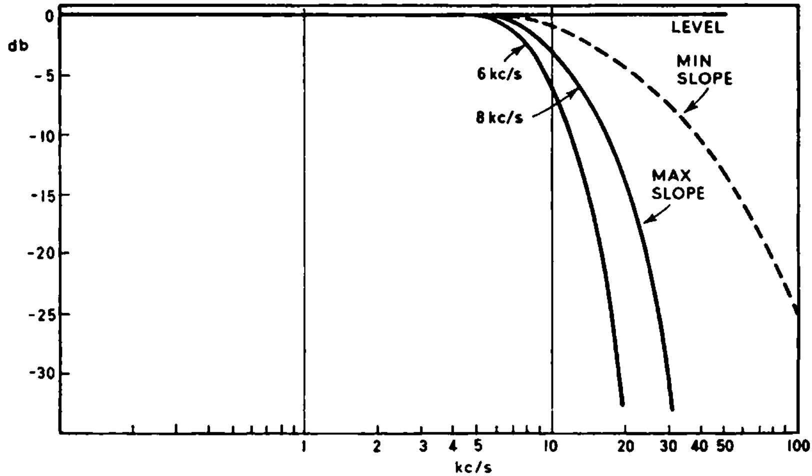


FIG. 2.10. *Two-position filter characteristic with a slope control.*

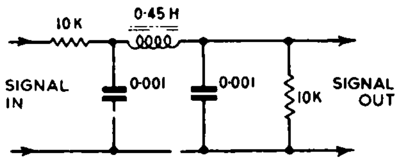
After suitable amplification by the valve following the tone-control network, the signal is usually fed into a filter system.

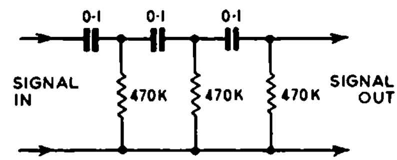
FILTERS

A filter is included to avoid the amplification of noise due to worn records and whistles caused by inter-channel interference when the hi-fi equipment is used with an A.M. radio tuner. Since the noise and interference frequencies are focused towards the high-frequency end of the audio spectrum, the filter is arranged to have a steep treble cut.

Most pre-amplifiers have a four-position filter switch, giving three positions of treble cut at 4 kc/s, 7 kc/s and 12 kc/s, and a "filter out" position. In addition, a control designated "slope" is often incorporated whose purpose is to vary the rate of treble attenuation from a minimum of some 8 db per octave to a maximum approaching 35 db per octave. The idea is shown graphically in Fig. 2.10. Here two filter positions are available, one at 6 kc/s and the other at 8 kc/s. The broken-line curve shows how the slope control serves to affect the rate of attenuation. Maximum slope indicates maximum attenuation rate.

When using new disks of recent pressing, it is desirable to commence operation with the filter switched out of circuit and with the bass and treble controls at "level", thus giving an extended flat response. Then, as governed by the acoustical environment, the bass and treble controls should be adjusted, bearing in mind that the ear is the final arbiter, as distinct from numbers on a dial!





(Left) FIG. 2.11. *Capacitive-inductive low-pass filter.* (Right) FIG. 2.12. *Each coupling section provides a low-frequency roll-off at the rate of* 6 *db per octave;* 18 *db are given by the three cascaded couplings shown.*

With worn records, the overall performance can be enhanced with the filter adjusted to give a cut-off at 4 kc/s or 7 kc/s coupled with the application of a little top-boost by the treble control. The same reasoning usually applies to noisy radio programmes. With music of a high transient content, it often pays, if a filter position is called for, to reduce the rate of attenuation by the slope control. This avoids "overhang" and "ringing" at high frequencies.

Filters come in two types. First, there is the tuned inductor arrangement in which an inductor connected in series with the signal source is resonated by capacitors. This is illustrated in Fig. 2.11. The circuit is tuned so that a sharp dip occurs at the high-frequency end of the response, and the falling side of the curve represents the treble-cut effect. At resonance, the circuit offers a very high impedance to signals at that frequency. The capacitors are usually switched, thus providing various filter frequencies, while the control of slope is by damping the circuit with the resistor. A variable resistive element permits a variable control of slope, as already explained.

Secondly, there is what is known as the "parallel T" circuit. This requires a large number of low-tolerance resistors and capacitors in order to give the desired high rate of attenuation at the various filter frequencies. Such a network is used in the R.C.A. pre-amplifier.

Combination circuits are also used, as also are less elaborate resistance/ capacitance networks in *pi* and M-derived configurations. It should be noted that resistance/capacitance inter-stage couplings affect the frequency response, but at the low-frequency end. A sharp cut at a low frequency is often desirable for eliminating gram motor rumble, and for avoiding unnecessary low-frequency distortion, particularly where the programme material possesses excessive low-frequency signal and the amplifier is not too good at the low-frequency end of the spectrum. It is far better to cut off sharply at about 40 c/s and achieve a "clean" bass than endeavouring to extend the response down to about 20 c/s and create unnecessary distortion.

Rumble filters, as these devices are often called, are switched on some units, giving only the positions "filter on" and "filter off". They are usually simple in design, often being built into an inter-stage coupling. A single resistor/capacitor coupling gives a roll-off of 6 db per octave. Since this is hardly steep enough for the purpose in hand, however, cascaded couplings are favoured. As each coupling gives the standard 6 db per octave roll-off, a slope of 18 db per octave is achieved by the use of three networks in cascade (Fig. 2.12).

The signal leaving the filter may be taken direct to the output socket of the pre-amplifier unit, or to another valve for further amplification and to facilitate matching to the power amplifier.

PRE-AMPLIFIER FINAL STAGE

A triode valve connected as a cathode follower is favoured as the final stage in the pre-amplifier unit. The advantage of the cathode-follower in this application is the low impedance across which the output signal is developed. This allows relatively long pre-amplifier/power amplifier connecting cables without causing undue attenuation of the higher audio frequencies, whilst also minimizing the pick-up of spurious signals, such as mains hum.

A cathode-follower stage is shown in Fig. 2.13. It will be seen that the load impedance is connected between cathode and chassis instead of between anode and h.t. positive, as in the more conventional arrangement. The cathode loading feature results in 100 per cent negative feedback, and as a consequence the distortion developed by the stage is at extremely low level, as also is the gain, being less than unity. This is of little moment, however, since adequate signal is usually available at the output of the filter, and the stage serves admirably as a matching device, for apart from its low output impedance, it has a very high input impedance and thus has little shunting effect on the circuit to which it is connected.

In the circuit in Fig. 2.13, Rg is the normal grid resistor, C the coupling capacitor, R1 the load and Rk the ordinary cathode-bias resistor.

The pre-amplifier volume control is invariably connected between the output signal at the filter and the cathode-follower valve grid.

There is still considerable controversy regarding the merits and demerits of the loudness control. It seems to be an accepted feature in the United States, though in Britain its use is by no means universal. It is not a new device, having been used many years ago in the form of a tone-compensated volume control in broadcast receivers.

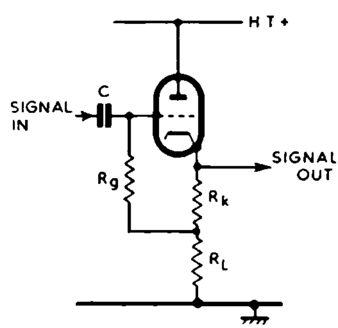


FIG. 2.13. *The cathode-follower. The stage gain is negative, being less than unity.*

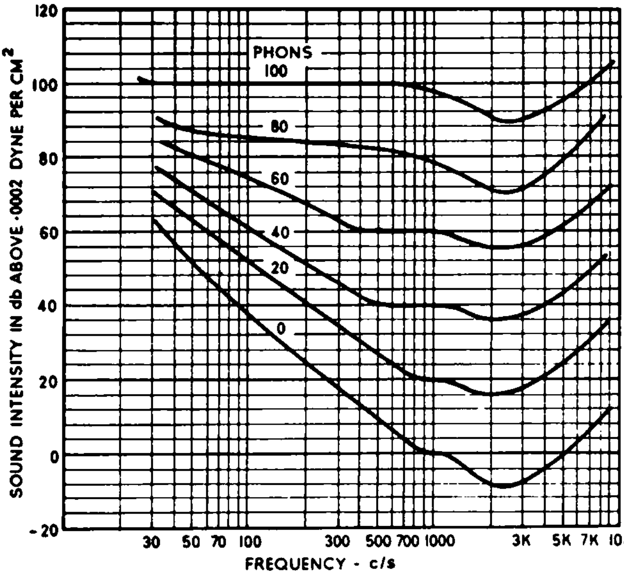


FIG. 2.14. *Fletcher-Munson equal-loudness curves show how the ear's response varies with intensity.*

It will be recalled from Chapter 1 that the ear does not respond in the same way to all frequencies. How the ear responds over the audio spectrum is revealed by the Fletcher-Munson equal loudness curves (Fig. 2.14). These curves show the required relative levels of sound at various frequencies for a *sensation* of equal loudness, being based on the reference frequency of 1,000 c/s. It will be recalled that the *loudness* of sound in phons is numerically equal to the sound *intensity* in decibels of an equally loud 1,000 c/s note, and that zero phon (corresponding to zero db at 1,000 c/s) is equal to a sound pressure of 0·0002 dyne per square centimetre (this, incidentally, is equal to 10^{-16} watts per square centimetre).

The curves demonstrate clearly that at low-level listening a considerable bass lift, and to a lesser degree top lift, is demanded in order to secure the sensation of the same apparent loudness over the full frequency range. There is little doubt that reproduction at a level of some 5 watts, with the treble and bass control adjusted to suit the room conditions, is of a far superior quality to that at 500 milliwatts with the tone controls left at their original settings —this condition can be created simply by backing-off the ordinary volume control. To secure anything like the original balance, considerable bass lift and a small amount of top lift is essential.

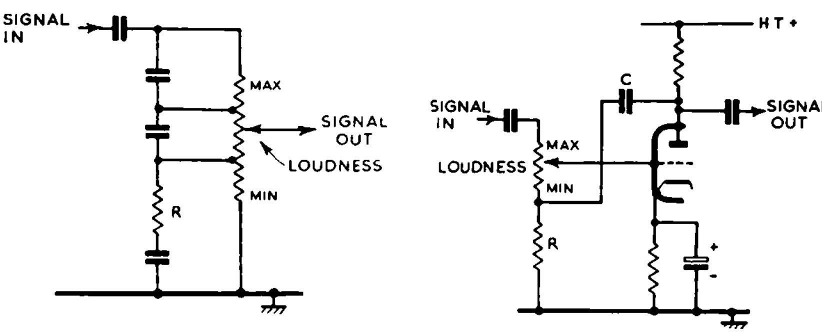
To avoid having to make these tone-compensating adjustments every time the volume control is adjusted, the loudness control has been evolved, and is designed around the Fletcher-Munson curves to quite a high degree of accuracy, at least from the low-frequency point of view. Retarding the control provides automatically the required degree of bass lift.

Two loudness-control circuits are given in Figs. 2.15 and 2.16. The first is arranged around a tapped volume control to which are connected capacitive elements. With the control in the maximum position the capacitive reactance shunting is at minimum, but as the control is rotated towards the minimum position the shunting increases progressively and the higher frequency components of the signal are attenuated with respect to the low frequencies, which effectively provides a bass lift. The cross-over point is somewhat governed by the resistance R.

The negative-feedback scheme in Fig. 2.16 makes use of an ordinary volume control and a frequency-selective negative-feedback loop by way of capacitor C. The loudness control and resistor R form a potential-divider in the feedback circuit, feedback being at maximum at the minimum position on the loudness control. Thus, as before, when the loudness control is retarded towards minimum the feedback of the higher audio frequencies increases, and a progressive boost of bass results.

It is usual to employ an ordinary volume control as well as a loudness control, the two controls often being connected in cascade, though in some cases they may be independently positioned in different stage couplings. Their actions are somewhat related, and for this reason their settings should be established with some care so as to avoid over-emphasis of the bass, possibly falling outside the range of the bass control proper.

The following procedure should be adopted where possible: turn the loudness control to maximum, or switch it out of circuit completely if a switch is provided for this purpose; set the volume control to a fairly high level; balance the sound to suit the room acoustics by means of the bass and treble controls; reduce volume to normal room level by backing-off the loudness control. A reasonable balance should be maintained throughout



(Left) FIG. 2.15. *Loudness control formed by tapped frequency-selective volume control.* (Right) FIG. 2.16. *Negative-feedback loudness control.*

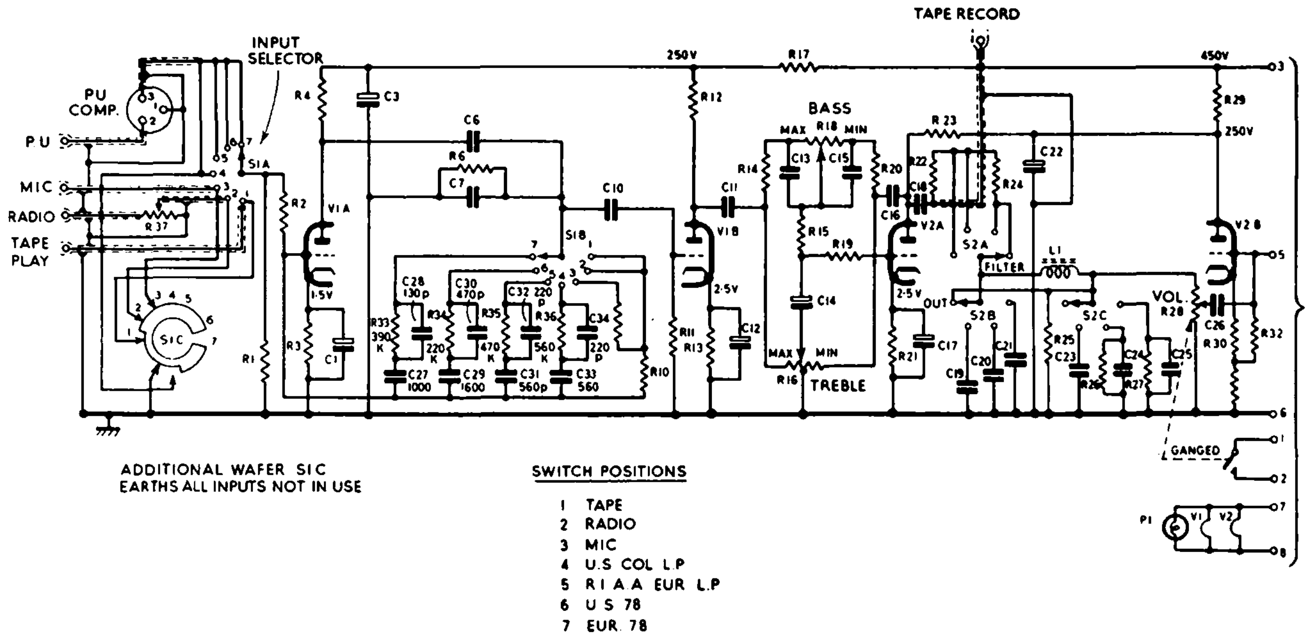


FIG. 2.17. *Complete circuit of the Pye HF25A pre-amplifier. To ensure accurate voltage reading a high-resistance voltmeter (more than 10,000 ohms per volt) must be used.*

the range of the loudness control over ordinary volume levels if this process is followed. Severe bass distortion will result, however, if the loudness control is turned to minimum, and the room-level volume adjusted solely by the volume control.

PRE-AMPLIFIER COMPLETE CIRCUIT

Fig. 2.17 is a complete circuit of the Pye HF25A pre-amplifier, and it will be instructive to look at the various sections in the light of our previous discussion. The required signal source is selected by switch S1A and applied to the grid of the first voltage-amplifier valve V1A. The unused programme sources are short-circuited by switch section S1C to avoid breakthrough, and a variable attenuator is included in the radio channel input so that any high-level signal on this channel can be suitably reduced to prevent overloading of the first valve.

The input selector switch is ganged to the equalizer switch S1B and four positions of equalizing are available on the pick-up channel on settings 4, 5, 6 and 7, negative feedback being used for this purpose and for controlling the gain over the four input channels. The equalized signal is further amplified by V1B, and control of bass and treble is secured by reason of the Baxandall negative-feedback system in conjunction with valve V2A. The signal is then passed into a three-position filter circuit, giving a sharp treble cut at 4 kc/s, 7 kc/s and 12 kc/s. The filter is adjusted by switches S2A, S2B and S2C, which also give a "filter out" position. The filter is of the inductive-capacitive type, with L1 as the inductive element.

The signal at the output of the filter, at the rotor of S2C, is passed through the volume control and on to the final valve V2B, which is arranged as a cathode follower. The signal is finally conveyed across R31 to pin 5 on the octal output plug.

A handy feature, and one which is found on many control units, is the

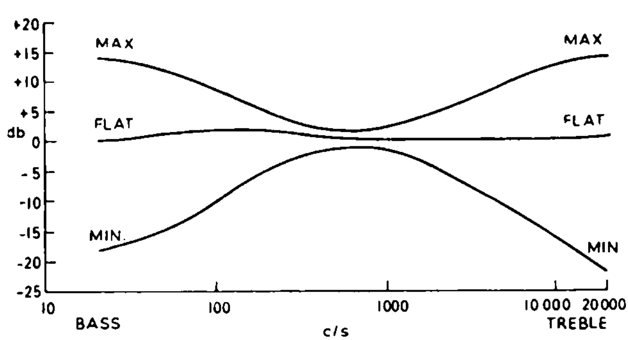


FIG. 2.18. *Tone-control characteristics of the Pye HF25A pre-amplifier.*

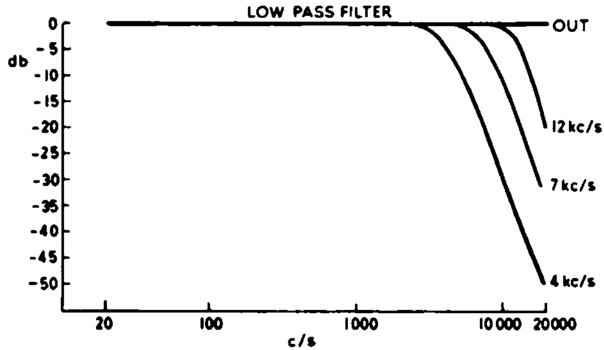


FIG. 2.19. *Filter characteristics of the Pye HF25A pre-amplifier.*

"tape record" output socket. At this point appears the amplified signal voltage exclusive of filter influence. This can be used for feeding another amplifier, if required, or used as an input signal for a tape recorder. Both h.t. and l.t. for the unit is derived from the main power amplifier, which is the subject of the next chapter.

Tone control and filter characteristics of the unit are given in Figs. 2.18 and 2.19 respectively.

To summarize, the pre-amplifier serves to match into the various signal sources so as to secure maximum signal transfer and signal-to-noise ratio, to equalize for the shortcomings of the programme material, to provide control of volume (sometimes loudness) and tone, and to raise the low-level programme signals to a level of about 1·5 volts for application to the power amplifier. To do this the pre-amplifier invariably requires an overall gain of some 60 db.

CHAPTER 3

The Power Amplifier

ELECTRICAL power measured in watts (W) is the product of the voltage (E) and current (I), that is, $W = E \times I$. Thus, for a constant power we can either increase the voltage and decrease the current or decrease the voltage and increase the current. At the output of a pre-amplifier we have in relation to the very small signal current a rather high voltage—because at the pre-amplifier stage we are mainly concerned in obtaining voltage amplification. However, at the output of the power amplifier we need to produce a larger current, since this is what is wanted by the loudspeaker. In fact, a signal current of 1 amp. is required to flow through a 15-ohm speaker to represent a power of 15 watts. This means that 15 volts would be measured across the speaker at this power. This is considerably different from the few millionths of an amp. of signal current which is present in the final load of the pre-amplifier.

The valve is inherently a high-resistance device, which means that the signal *current* in the anode circuit is limited to a few thousandths of an amp., though, in order to secure this condition, the voltage for a given power is relatively high. Clearly, a means is necessary for transforming this high signal voltage and low current so that the current is considerably increased and the voltage proportionately decreased for presentation to the loudspeaker. This transforming process is achieved quite easily by the use of a transformer—in this application it is termed the "output transformer".

The winding connected to the anode circuit of the valve, across which is developed the high signal voltage, is known as the primary winding, and is composed of a large number of turns of fine wire since it is required to pass only a small current. The winding connected to the speaker, through which is passed the high signal current, is known as the secondary winding, and is formed of only a few turns of heavy-gauge wire. The arrangement is represented in Fig. 3.1.

A transformer devoid of losses has a voltage ratio equal to the turns ratio, that is, Voltage across primary: Voltage across secondary = Primary turns: Secondary turns. Thus, a transformer with a 10:1 turns ratio will have

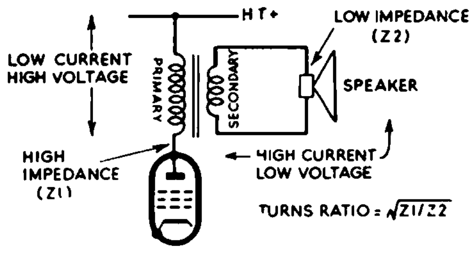


FIG. 3.1. *Matching the speaker to the output stage is an important consideration in hi-fi practice.*

10 volts induced across the secondary when a signal of 100 volts is applied across the primary. The current will be transformed in the same ratio, assuming ideal conditions.

Across the secondary winding of the transformer (Fig. 3.1) is connected the resistance or impedance (impedance is the term used when we are dealing with alternating quantities) of the loudspeaker. This impedance—let us call it $Z2$—is reflected across the primary winding, but is altered in magnitude according to the expression $\left(\frac{N1}{N2}\right)^2 \times Z2$, where $N1$ is the number of turns forming the primary winding, and $N2$ the number of turns forming the secondary winding.

Thus, if the transformer has a turns ratio of 10:1 (step-up from secondary to primary), the factor $\frac{N1}{N2}$ resolves to 10, the square of which is 100. Now, if the impedance of the loudspeaker ($Z2$) is equal to 10 ohms, then the impedance reflected across the primary is $100 \times 10 = 1{,}000$. This is the impedance as "seen" by the valve. It is rather important to get these facts clear, as they have a direct bearing on the performance of the power amplifier.

The transformer, therefore, apart from altering the voltage and current transfer, also serves as an impedance-matching device. In the case cited, the transformer of ratio 10:1 has transformed the 10-ohm loudspeaker impedance to 1,000 ohms. Impedance-matching in the output stage is an extremely important consideration, for unless the impedance of the loudspeaker is transformed to match exactly the optimum working impedance of the valve or valves, the loudspeaker will not receive the full available power.

Re-arranging the expression given above, we can discover the turns ratio required to match the speaker impedance $Z2$ to the high impedance $Z1$.

The turns ratio $= \sqrt{\frac{Z1}{Z2}}$

Most power amplifiers have switching or plug-and-socket facilities available for adjusting the output impedance of the amplifier to suit a wide range of speaker impedances. As the fraction of voltage fed back as feedback is affected by impedance adjustments, it is desirable to alter the value of the feedback resistor (and sometimes the capacitor) when the output impedance is changed. Commercial amplifiers have indicated in their instruction books the value of components required for the various output impedances. Where a positive-feedback loop is incorporated, this also requires adjustment when changing impedance.

A power amplifier should never be operated with the speaker removed unless a resistor of equal value to the speaker impedance is used instead. An unloaded amplifier will develop very high audio peak voltages across the primary of the output transformer, and these can quickly destroy the transformer insulation.

OUTPUT TRANSFORMER

In practice, there is no such thing as a perfect transformer, since losses occur in both the primary and secondary windings and in the core itself. The inductance of the primary governs the relative amplification at low audio frequencies, while leakage inductance in both windings results in a loss of high audio-frequency response. These losses also promote phase shift at the low- and high-frequency sides of the audio spectrum, and sometimes make it difficult to maintain a reasonably high degree of negative feedback without instability.

The windings also possess distributed capacitance, and due to this there is invariably a peak in the response curve at the frequency at which the equivalent leakage reactance resonates with the lumped equivalent capacitance.

In transformers designed for hi-fi work, all these losses are kept at the absolute minimum, resonances are damped and arranged to fall outside the audio spectrum, considerable iron is used for the core so as to maintain adequate low-frequency response, and distributed capacitances are minimized by dividing the windings into sub-sections, as shown in Fig. 3.2. This specialized design is inevitably reflected in the cost of the item, and is the reason why the output transformer is invariably the most costly single component in the whole amplifier. There is much truth in the saying that an amplifier is only as good as its output transformer.

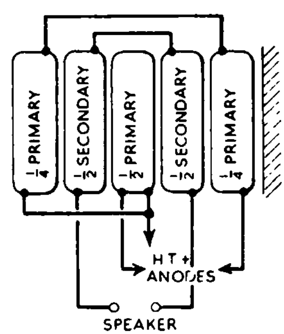


FIG. 3.2. *Distributed capacitances in the output transformer are minimized by dividing the windings into sub-sections, as illustrated.*

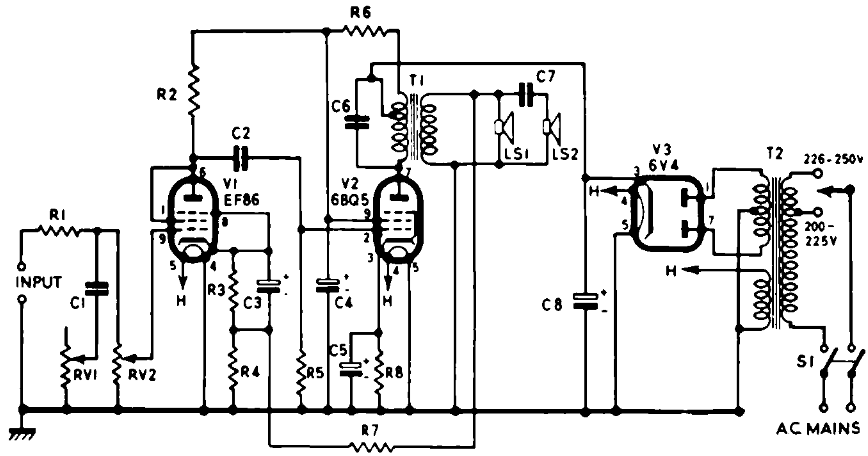


FIG. 3.3. *Circuit diagram of the Cossor 562K amplifier kit. The component values are given below.*

R1	47K	RV1 500K log.	C1	5,000pF	±20%
R2	100K	RV2 500K log. (with S1)	C2	0·05μF	250V
R3	2·2K	*Note:* All resistors ±10%	C3	50μF	12V
R4	100 ohms		C4	50μF	275V
R5	330K		C5	50μF	12V
R6	3·9K		C6	5,000pF	±20%
R7	5·6K		C7	2μF	150V
R8	150 ohms		C8	50μF	275V

Hi-fi transformers need to be bulky devices in terms of iron in order to permit the full rated power of the amplifier to be handled down to about 30 c/s, and to avoid harmonic distortion due to core saturation. The core material affects the primary inductance, which needs to be relatively high for an adequate low-frequency response for the smallest possible amount of wire. Both the cross-sectional area of the core material and its permeability are governing factors on the inductance; increasing either of them results in an increase in inductance.

In recent years a core material having a higher permeability value than the conventional "Stalloy" laminations has been under development. This is a cold-rolled silicon steel, having grain-orientation properties, which is known as "Unisil". For a stated low-frequency performance, a transformer with this new core material is approximately half the weight of a transformer using the older grades of core material. The well-known "C-core" is the result of further developments along these lines and, although somewhat more expensive than "Stalloy" laminations, permits even further reduction in both weight and size.

SINGLE-ENDED OUTPUT STAGE

To illustrate the single-ended output stage, the circuit diagram of the Cossor Amplifier Kit 562K is given in Fig. 3.3. This is an ideal kit, easily made up by the non-technical experimenter, with which to study the rudiments of sound reproduction. It is not claimed to possess hi-fi quality, but in spite of its basic simplicity, the performance may pleasantly surprise even one having long association with hi-fi equipment, especially if used with a first-class loudspeaker.

The first valve, V1, serves as the voltage amplifier, the signal being applied to the control grid by way of the volume control RV2, after correction by the tone control RV1. The signal in amplified form is developed across R2, and from here fed to the control grid of the output valve, V2, through the coupling capacitor C2. The output valve is biased in the ordinary way by the cathode resistor R8. The volts drop across this resistor due to the current in the valve causes the cathode to rise positively with respect to chassis. Because the control grid of V2 is in d.c. connexion with the chassis through the grid resistor R5, the grid is effectively more negative than the cathode and is thus biased, the magnitude of the bias being equal to the volts drop across the cathode resistor R8.

This is the normal function of so-called cathode bias, which is used extensively in radio and television receivers as well as audio equipment.

The valve is biased to class "A" conditions. This means that in the normal condition of operation the anode current is not cut-off for any portion of the cycle of signal applied at the control grid.

The signal in the anode circuit is conveyed to the loudspeakers, LS1 and LS2, by way of the output transformer T1, impedance matching being achieved in the way already described. The tapping on the primary winding of T1 may require explanation. The portion of the winding across which the signal voltage is developed is that across the capacitor C6. The section between the tap and R6 serves as a hum-neutralizing device, h.t. from the cathode of the rectifier valve V3 being applied to V1 through the upper part of the primary and R6. Thus, this part of the primary can, in effect, be considered as a smoothing choke with the added filtering contributed by R6, in association with the reservoir capacitor C8 and smoothing capacitor C4.

Negative-voltage feedback is applied over the two stages from the secondary of the output transformer through R7 to the cathode circuit of V1. The feedback loop is completed from the other side of the secondary winding to chassis. This is one of the most popular ways of applying negative feedback. (Of course, if the feedback and chassis connexions on the secondary winding are reversed, the feedback becomes positive and the amplifier turns into an oscillator. Such a mis-connexion could occur in changing an output transformer of this kind.)

It will be seen that two loudspeakers are used. The one connected directly across the secondary deals with the lower and middle audio frequencies; the one connected by way of C7 handles the higher frequencies, and is often referred to as a "tweeter" or high-frequency reproducer. It is somewhat isolated from the lower audio frequencies owing to the high reactance presented by C7 to these signals. In effect, the arrangement can be considered as a simple cross-over filter, and will be dealt with in more detail in a later chapter.

PENTODE OR TRIODE

In the early days of hi-fi, the triode was invariably held in favour over the pentode (or tetrode) owing to its smaller distortion figure and lower anode impedance as compared with the pentode. For example, at maximum output a power triode in a single-ended circuit may produce something like 5 per cent distortion, mainly second harmonic.

The distortion due to a power pentode operating similarly, however, may rise to some 13 per cent, made up basically of third harmonic and higher order harmonics. The reason for such a high distortion figure is the result of the S-shaped grid voltage/anode current characteristic curve of the pentode. This characteristic is shown in Fig. 3.4, and the general picture is completed by a sine wave corresponding to the signal applied at the control grid, the valve being biased to class A, and the resulting signal across the anode load. It will be seen that the bends at the top and bottom of the curve cause a certain flattening of the peaks of the amplified signal, which promotes a spurious third-harmonic signal. An analysis of the distorted signal would reveal that it is composed of the fundamental component plus a smaller third-harmonic component and higher order harmonics.

Fig. 3.5 shows that the grid voltage/anode current characteristic curve of the triode is considerably more linear than that of the pentode.

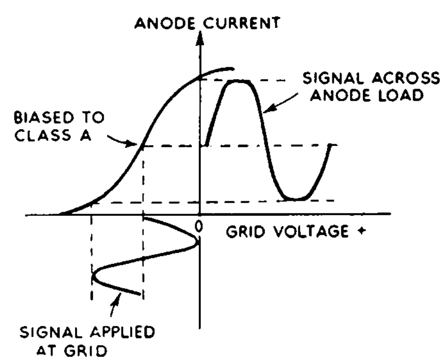


FIG. 3.4. *The S-shaped characteristic curve of the power pentode results in the production of third-harmonic distortion, shown by the flattening of the peaks of the output signal.*

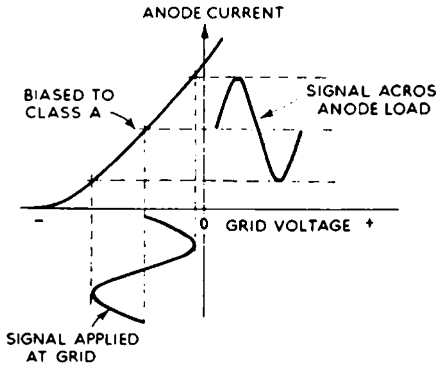


FIG. 3.5. *The triode has a more linear curve, and the resulting distortion is essentially second-harmonic.*

There is a slight inward bend, however, and this results in slightly greater amplification of one half-cycle of the signal than the other. This is representative of second-harmonic distortion. The triode, on the other hand, is less sensitive than the pentode, requiring a greater input signal for a given output power. However, if the drive is exceeded the distortion rises rapidly, and can be very disconcerting if this happens on programme peaks in a poorly designed amplifier, or if the amplifier is insufficiently large for the purpose in hand.

A greater overload margin is available with the pentode, and to a smaller degree with the tetrode, owing to the "cushion" effect created by the gradual bends at the ends of the curve. Unfortunately, the application of negative feedback tends to neutralize this desirable effect, and the valves behave more like triodes with regard to accidental overload.

In order to prevent power being wasted in the output valve, the load impedance should theoretically match the anode impedance of the valve In practice, this does not always follow because the load for maximum power does not coincide with that for minimum distortion. There is, therefore, an "optimum load" value which gives a compromise between the two conflicting factors.

Fig. 3.6 shows curves for power output and percentage distortion against anode-load impedance, and these clearly reveal the need for an impedance compromise. If maximum power output is aimed at, then both second- and third-harmonic distortion will be at a high level. If the load is decreased to 4,000 ohms, third-harmonic distortion is less troublesome but second-harmonic rises, accompanied by a power decrease. Generally speaking, second-harmonic distortion is less disturbing than third-harmonic and, as we shall see later, can be eliminated in a push-pull output stage. Thus, it is desirable to work to the best compromise between third-harmonic distortion and power output. Neither the technician nor the enthusiast need worry about working out optimum loads, since this has already been done by the valve maker, and the figures can readily be obtained by referring to the data supplied with the valve.

Similar curves for a triode power valve are given in Fig. 3.7. Here there

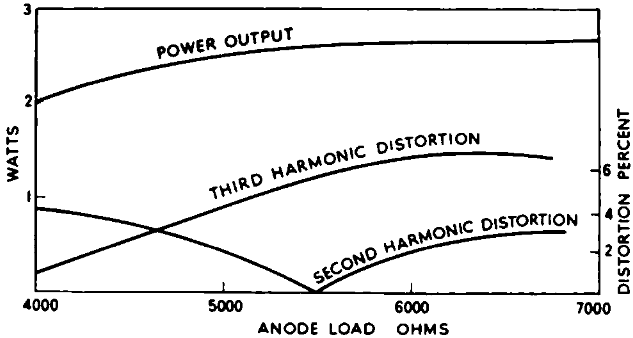


FIG. 3.6. *Curves, related to a pentode output valve, for power output and distortion plotted against anode-load impedance.*

is only second-harmonic distortion to deal with, and it is desirable to use a large-value load if minimum distortion is required.

Account must be taken of the fact that in practice the impedance presented to the output valve changes considerably over the operating frequency range owing to the varying impedance of the loudspeaker. Moreover, as the load is not a pure resistance, but possesses a reactive component, the task of finding the best load compromise is made somewhat more difficult.

PUSH-PULL OUTPUT

The push-pull output stage is usually a feature of hi-fi amplifiers, and the basic circuit is shown in Fig. 3.8. The signal is applied to the two valves in a way that when the control grid of one is swung positive the grid of the other is swung negative, and vice versa. The anode current is thus rising in one valve while it is falling in the other; hence the term "push-pull".

The signal is applied in relation to a fixed "zero signal" point, such as chassis, either from a transformer with a tapped secondary winding or from a phase-splitting valve. In this way, not only is the power output increased but, more important from the hi-fi aspect, all second and even harmonic distortion

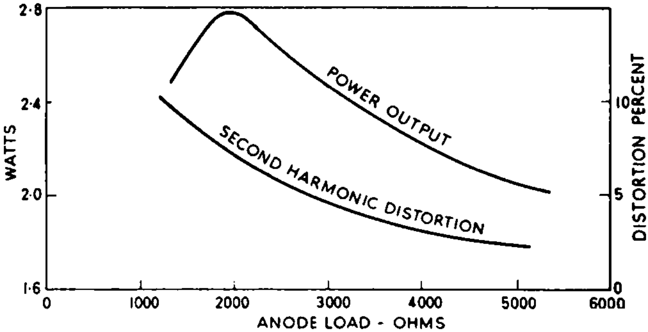


FIG. 3.7. *Curves, related to a triode output valve, for power output and distortion plotted against anode-load impedance.*

is automatically cancelled. Thus, a properly designed and adjusted push-pull stage should produce virtually zero second-harmonic distortion. Third and odd harmonic distortion will still be present, of course, but this can be reduced to a very low figure by the application of negative feedback. Even without feedback, a triode push-pull stage will exhibit a very low distortion figure, bearing in mind that second harmonic is the prominent factor in each valve, and will be eliminated in a push-pull stage. This is one of the reasons why triodes used to be very popular a few years ago, before negative feedback was fully developed.

In hi-fi application, the output valves are arranged to operate in class A, though this is by no means a necessity, for if greater power is called for, and an increase in distortion is permissible, then the valves may be biased towards

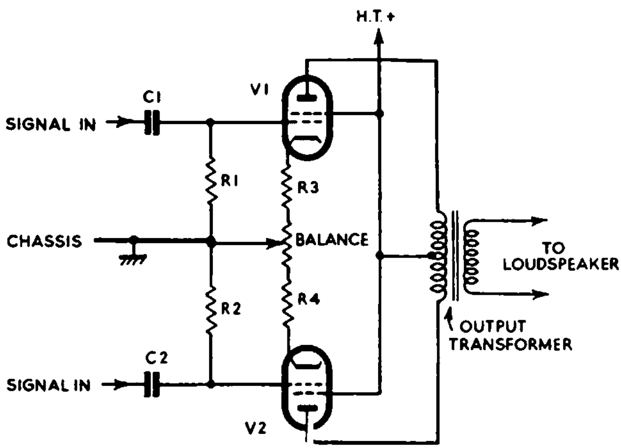


FIG. 3.8. *Basic circuit of push-pull output stage using tetrode valves.*

anode-current cut-off and the signal may be increased to push the valves into grid current on the positive half-cycles. In this way a very much larger signal swing occurs in the anode load, and the output power is correspondingly increased. This scheme is used extensively in public-address and sound-reinforcement amplifiers, where quantity rather than quality is demanded.

In most hi-fi power amplifiers the output valves are biased by reason of the volts drop across their cathode resistors (R3 and R4 in Fig. 3.8). The variable potentiometer serves to adjust the two valves for d.c. balance. In other amplifiers, particularly those whose operating conditions deviate from pure class A, a separate bias line is used, sometimes derived from a separate bias power unit.

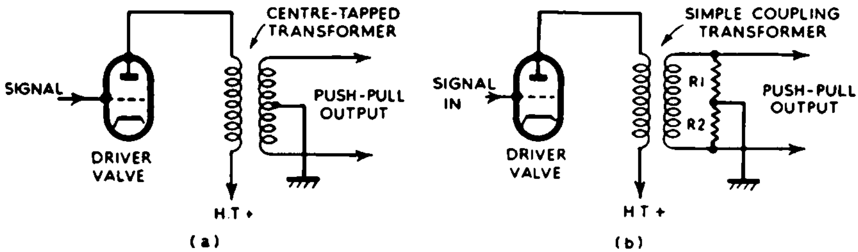


FIG. 3.9. *Phase-splitting* (a) *by means of a centre-tapped transformer, and* (b) *by means of a simple transformer and resistive divider.*

PHASE-SPLITTING

The simplest method of providing two input signals 180 deg. out of phase for a push-pull stage is by means of the centre-tapped transformer (Fig. 3.9*a*). An alternative arrangement, which uses a simple coupling transformer, is shown in Fig. 3.9*b*. In both cases the secondary of the transformer provides two equal anti-phase signals. In the latter case, however, a zero-signal reference point is obtained at the junction of the resistive divider (R1 and R2). If required, the transformer can give a signal step-up to the push-pull valves and, as with the output transformer, it will also serve to match the anode circuit of the driver valve to the grid circuits of the push-pull valves.

In arrangements which deviate from true class A working, the secondary of the transformer is usually placed in the grid circuits of the output valves. The reason for this is to provide a relatively low d.c. resistance in the grid circuits as a means of avoiding severe overload distortion when the output valves are driven into grid current. Transformer phase-splitting is thus usually confined to large public-address and sound-reinforcement amplifiers.

Since a transformer inevitably introduces some degree of distortion, transformer phase-splitting is rarely if ever used in hi-fi amplifiers. Instead, a method of phase-splitting by means of a valve arrangement is always favoured.

A very popular valve phase-splitter circuit is shown in Fig. 3.10. In this circuit, which is designed around a triode or a pentode strapped as a triode, the load resistance is split; half of it is connected in the anode circuit (R1)

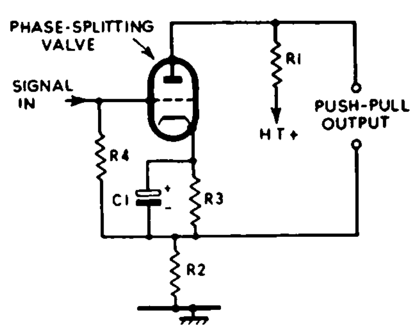
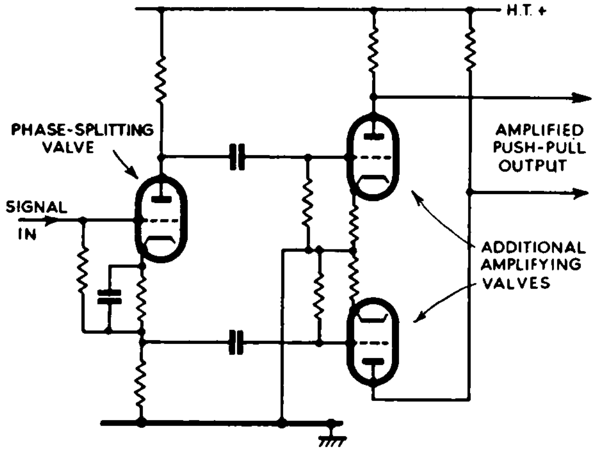


FIG. 3.10. *Circuit diagram of the simple "split-load" phase-splitter stage. The gain of this circuit is less than unity.*

FIG. 3.11. *In order to obtain sufficient drive signal for the push-pull power valves, additional amplifying valves may be necessary, as shown.*



and the other half in the cathode circuit (R2). The input signal from the voltage amplifier is applied between the grid and chassis, and due to the high-value load (R2) in the cathode the resulting large amount of negative feedback gives rise to considerable loss of gain over an ordinary voltage amplifier, in which the net load is in the anode circuit.

In fact, the actual gain between the grid and either side of the push-pull output is around 0·9, giving a total gain between the two output terminals of 1·8. The voltage output at either terminal is half that of a comparable voltage amplifier, which means that if a fairly high drive is required by the push-pull output valves additional amplifying valves have to be introduced between either output terminal and the grids of the output valves (Fig. 3.11). This arrangement will be found in some modern equipment.

The phase-splitting valve is biased by the usual cathode resistor (R3 in Fig. 3.10) and the grid-return resistor, R4, being connected to the junction of the cathode load and the biasing resistor. The heater of the valve is sometimes connected to a positive potential so that it approaches that of the cathode. The reason for this is to reduce hum which may otherwise be reflected from the cathode circuit into the grid circuit.

Fig. 3.12 shows the circuit of a cathode-coupled inverter, used in the Pamphonic Model 2,001. As with a number of modern phase-splitting

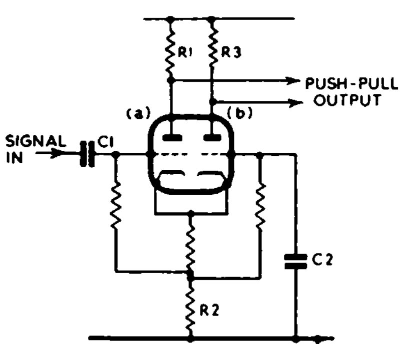


FIG. 3.12. *Circuit diagram of a cathode-coupled phase-splitter stage.*

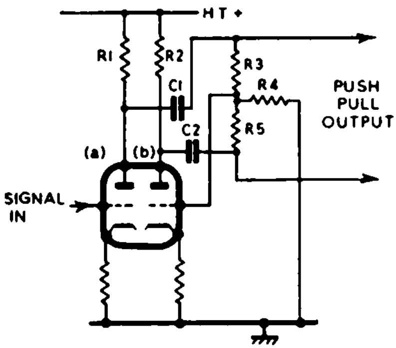


FIG. 3.13. *Circuit diagram of a floating paraphase inverter stage.*

circuits, it calls for the use of two triode valves, a double-triode unit being admirably suited to the function. Section (a) of the valve serves as an ordinary earthed-cathode amplifier, in which the input signal is applied through C1 and delivered across R1. Section (b) receives its signal by way of C2 from that present across the common cathode resistor R2, and delivers its output across R3. Section (b), in effect, is an earthed-grid amplifier, whose output signal is exactly 180 deg. out of phase with the signal at the output of section (a).

The so-called paraphase circuit is also used in various forms for phase-splitting. The version given in Fig. 3.13 is sometimes called the "floating paraphase", in which two triodes (a double-triode valve) are used. The signal applied to triode (a) grid is developed in amplified form across R1 and applied through C1 to one of the push-pull valves. Resistors R3 and R4 in series also form a load on triode (a), and the signal voltage at their junction is applied to the grid of triode (b). Triode (b) thus gives rise to an amplified anti-phase signal across its load R2, which is conveyed by way of C2 to the grid of the other push-pull output valve. Resistors R5 and R4 in series also form a load on triode (b). The circuit is maintained in a good state of balance since opposing voltages are developed across R4 due to the two triode sections, and the signal voltage applied to the grid of triode (b) is maintained at just the right level for optimum balance. In effect, the circuit has a self-balancing action.

There are many variants of the phase-splitting arrangements mentioned, but the information given should be sufficient to assist the technician and enthusiast with any servicing and adjustments which may be required in this section of the amplifier.

To summarize, the requirements of the phase-splitting stage, sometimes known as inverter stage, are to provide a balanced drive signal for the push-pull output valves from a common signal source, such as a voltage amplifier; to provide a signal of sufficient voltage, without distortion, to ensure maximum output of the power amplifier; and to provide signals at each output valve grid differing in phase by exactly 180 deg. throughout the audio spectrum.

The phase-splitting valve itself is usually fed from a voltage-amplifier

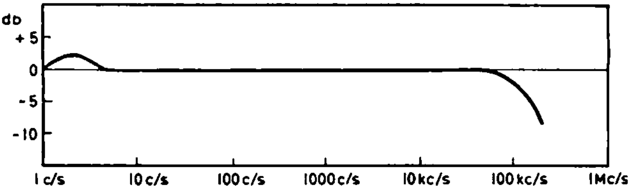


FIG. 3.14. *Frequency response characteristic of the Pye Provost amplifier.*

stage, which invariably forms the first stage of the power amplifier unit, into which are fed the signals from the pre-amplifier. The response characteristics of the power amplifier, therefore, should be sensibly flat over the entire audio range and, with regard to feedback stability and the correct handling of transient signals, should remain reasonably flat without peaks for two or three octaves beyond the highest usable frequency. The excellent response characteristic of the Pye HF25 (Provost) power amplifier shown in Fig. 3.14 illustrates this point.

ULTRA-LINEAR STAGE

The majority of hi-fi amplifiers incorporate an ultra-linear (sometimes called "distributed load") output stage. The arrangement uses tetrode output valves, but instead of their screens being connected direct to the h.t. positive line, they are each connected to a tap on the primary of the output transformer. The basic circuit is given in Fig. 3.15.

This form of connexion gives the output stage a characteristic which in most respects is between that of a tetrode and a triode. The desirable low distortion and good linearity of the triode is maintained, as also is the high output and sensitivity of the tetrode. Comparatively less negative feedback is required for a given result, resulting in a greater margin of stability. The distributed load effect also results in a reduction in total d.c. variations in the output stage at high output levels. Also, since the low capacitances of the tetrode are maintained, there is less reactive shunting at high frequencies and a reduction of phase shift at the high-frequency end of the passband.

The tapping point for the screens is rather important from the distortion aspect. There is an optimum point for different valves, and it usually ranges between 20 and 43 per cent.

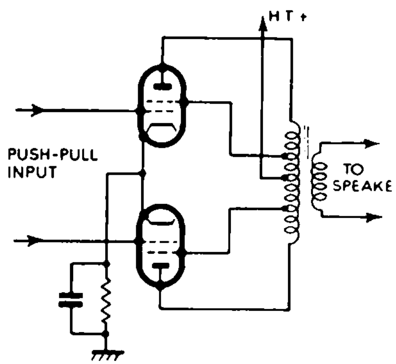


FIG. 3.15. *Basic ultra-linear circuit.*

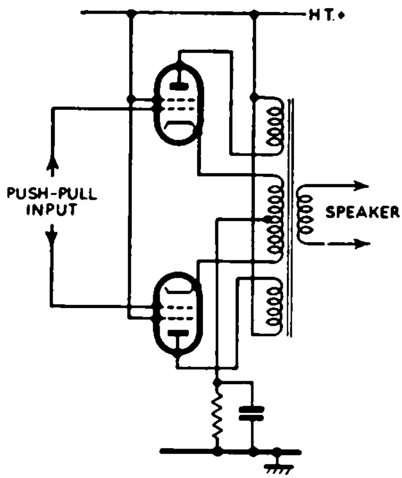


FIG. 3.16. *Partial cathode loading, as used in the Quad II amplifier.*

CATHODE LOADING

Total cathode loading of an output stage (i.e., the load connected in the cathode instead of the anode circuit) results in 100 per cent negative feedback and calls for driver stages capable of supplying some 150 to 200 volts of signal. Whilst this method has received considerable attention, it is rarely used in hi-fi equipment. Partial cathode loading is used successfully, however (for example, in the Acoustical Quad II amplifier); see the circuit diagram in Fig. 3.16.

The cathode winding results in a portion of the output signal being returned to the grids in the form of negative feedback. A conventional negative-feedback loop is also incorporated and the combined effect gives rise to a low output impedance coupled with very low distortion at high power.

Cathode loading combined with ultra-linear operation has been experimented with in the United States (see Fig. 3.17), but the arrangement does not lend itself to class A operation, though quite low distortion figures, it is claimed, are possible by operating at class AB or class B.

SINGLE-ENDED PUSH-PULL

By the connexion of two output valves in series and the application of a push-pull drive signal, the output impedance is somewhat reduced and rendered less critical than that of a more conventional stage. This idea lends itself to transformerless operation, meaning that a loudspeaker (of higher impedance than normal) can be connected direct to the output stage without introducing an output transformer.

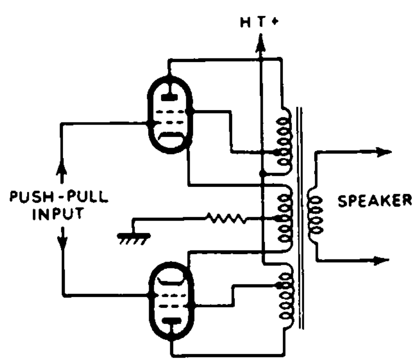
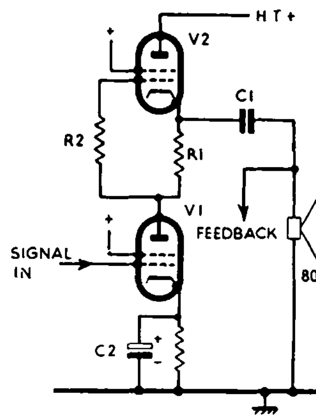


FIG. 3.17. *Combined partial cathode loading and ultra-linear circuit. This is not very suitable for Class A operation, but is of greater use for Class AB and B operation.*

FIG. 3.18 *The Philips version of the single-ended push-pull output stage.*



When it is realized that most of the non-linearity and phase shift (the latter limiting the amount of negative feedback in the power amplifier) are aggravated by the transformer, any device leading to its elimination is well worth consideration. We have already seen that a current of the order of one ampere is required in a 15-ohm loudspeaker to give 15 watts. While such a large current can be obtained easily by the use of an output transformer, very large valves would be required in order to obtain this current without a transformer, though matching could be secured by loading into the cathode circuits.

Experiments have been directed along these lines, but instead of using large valves a number of smaller valves have been tried. In one experiment 16 6AS7G valves were required to obtain 12 watts of power in a 16-ohm load. This is hardly economical, and one might well spend a lot of money on a special transformer.

Philips have solved the problem, however, by the use of two series-connected output valves driven in push-pull and a speaker having an impedance of 800 ohms, against the conventional 15 ohms. The circuit is given in Fig. 3.18. When valves are connected in this way in an output stage, the circuit is often referred to as a single-ended push-pull stage.

Broadly speaking, the valves are biased so that they would pass equal current in the event of zero drive signal. When a signal is applied to the control grid of V1, however, this balance is disturbed at the frequency of the signal. For example, a positive swing at the V1 control grid results in an increase in anode current and a reflected negative-going signal at the control grid of V2 with respect to cathode. In this way a push-pull signal is created and the valves are push-pull driven. The out-of-balance current at audio frequency flows through the loudspeaker coupling (d.c. isolating) capacitor C1 and in and out of the speaker's speech coil, as would be the case if it were connected across the secondary of a conventional output transformer.

In the Philips "Hi-Z" power amplifier, four valves are used in a parallel single-ended push-pull arrangement, and the speaker is fairly heavily damped by the low cathode impedance of V2.

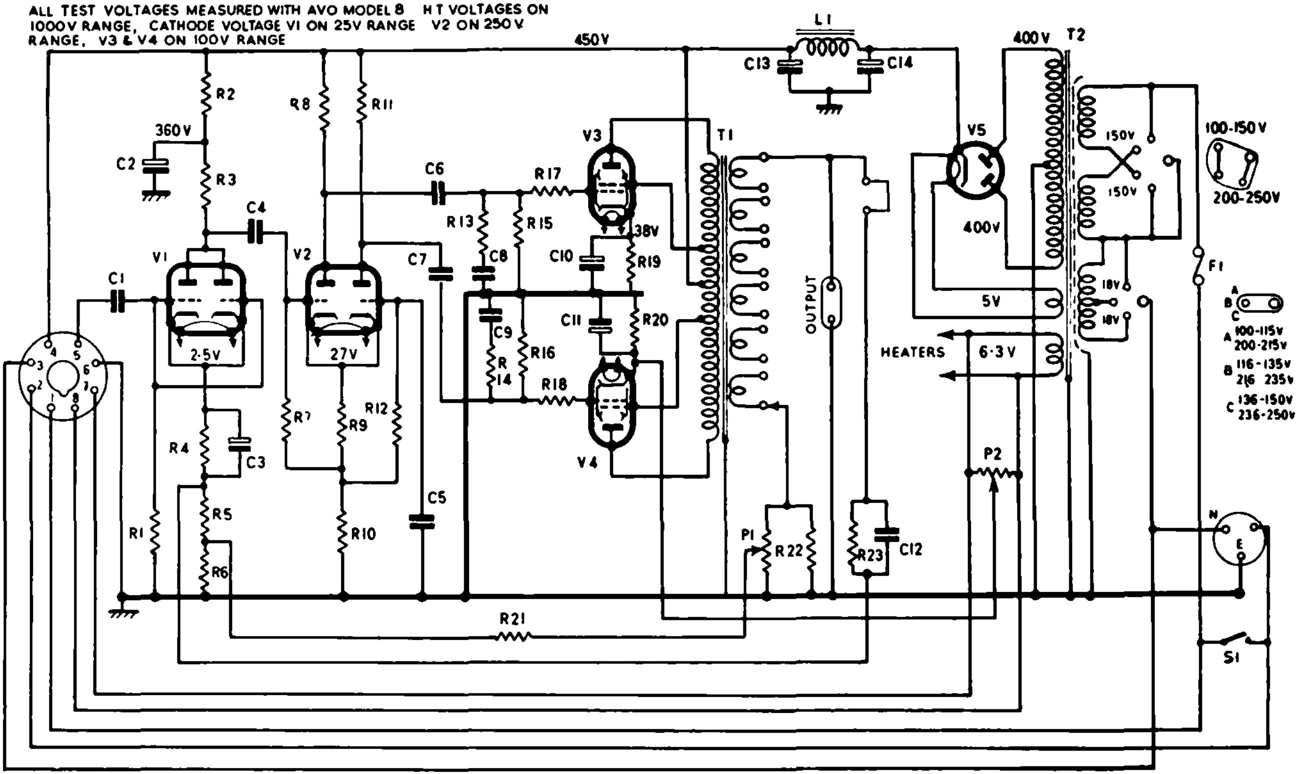


FIG. 3.19. *Circuit diagram of the Pamphonic 2,001 power amplifier. The negative-feedback loop is by way of* R23 *and* C12, *and the positive-feedback loop by way of* P1/R22 *and* R21. P1 *serves as the damping-factor control.*

There are various modifications of this circuit, some of which are driven from a phase-splitter stage which is designed along with the output stage to secure optimum balance of drive signal. In general, the systems cancel second-harmonic distortion by the distortions of each valve appearing in antiphase in the common load, though it is sometimes desirable to unbalance the distortions of the two valves, so that it is less in V1 than in V2. This may be secured by shifting the working point of V2 away from that of minimum distortion or by applying negative feedback to V1 by excluding a cathode bypass capacitor (i.e., removing C2).

APPLICATION OF FEEDBACK

The best way of getting to know the feedback loops is to study them in an actual circuit. In Fig. 3.19 is shown the complete circuit of the Pamphonic 2,001 power amplifier. Here stage V1 serves essentially as the first voltage amplifier. It raises the level of the signals from the pre-amplifier sufficiently to operate the cathode-coupled phase-splitter V2. This in turn drives the ultra-linear push-pull output valves V3 and V4.

Negative feedback is applied over the whole of the power amplifier by feeding back to the cathode circuit of V1 a suitable fraction of the signal voltage developed across the secondary of the output transformer T1. The phasing of the feedback is such that it is negative, while the degree of feedback is governed by resistor R23. C12 is usually known as a phase correction capacitor, whose purpose is to render the feedback loop very slightly frequency-selective (the capacitor is usually of fairly low value) and thus enhance the feedback stability margin. R23 and C12 are required to be altered in value so as to maintain optimum feedback on changing the loudspeaker impedance tapping.

In transferring a signal in opposite phase from the reverse side of the secondary winding of the output transformer also to the cathode circuit of V1, a positive-feedback loop is provided. The positive-feedback signal is developed across R22 and P1 and fed back through R21. P1 is, in fact, a pre-set potentiometer which allows adjustment of the damping factor of the output stage. It will be recalled from the description of negative feedback in Chapter 2 that the output impedance as "seen" by the loudspeaker reduces as the negative voltage feedback is increased, and further reduces as the positive current feedback is increased. It is on this principle that the damping factor control operates.

To recapitulate, the damping factor is equal to the nominal output impedance of the amplifier divided by the impedance as "seen" by the loudspeaker (source impedance); a low source impedance results in a high damping factor and an infinite damping factor results when the source impedance falls to zero ohms.

EFFECT OF DAMPING FACTOR ON LOUDSPEAKER

A low source impedance or high damping factor affects the cone of a loudspeaker in much the same way as the shock absorbers affect the stability of a car. A car with worn or faulty shock absorbers oscillates vigorously up and down on riding rough ground or when a sudden stop is demanded. In the same way a high source impedance causes the cone of the loudspeaker to oscillate beyond the normal pattern of the audio signal when this is of the nature of sharp transients.

The effect is illustrated in Fig. 3.20. At (*a*) is a square wave which may be applied to the input of an amplifier, causing the loudspeaker cone to move rapidly in one direction as represented by A-B. Ideally, at point B the cone should stop dead and remain still until point C, when it should move rapidly in the opposite direction represented by C-D. This ideal will be approached when the amplifier source impedance as "seen" by the loudspeaker is very low (when the damping factor is high).

If the source impedance is high, however, there will be no electronic shock absorption, and the loudspeaker cone will follow the pattern as shown in Fig. 3.20*b*. Here, from A to B the cone will move rapidly in one direction, but instead of coming to a halt at B it will continue oscillating between B and C. It will change direction at C, but again oscillate about point D. This is known as a "damped oscillation", which may not only develop in the loudspeaker owing to a low damping factor, but may also appear as current oscillations in resonant elements of the amplifier. In this latter connexion, the effect is often referred to as "ringing".

Apart from being damped electronically, the loudspeaker is also acoustically damped by the loading in its enclosure. It is said, therefore, that every speaker has a certain critical damping factor from the electronic aspect, and for this reason a large number of amplifiers are provided with a damping control. The author feels that a loudspeaker can be over-damped as well as under-damped, the effect in this case being like that of a light door coupled to one of those large automatic door-closing devices—the door can be neither opened nor closed sharply. The effect on the loudspeaker is that the cone is unable to follow very sharp transient waveforms.

This electronic damping effect can be demonstrated with a moving-coil milliammeter. With the terminals of the meter open-circuited, vigorous twisting of the meter will cause the pointer to oscillate in a very disturbed

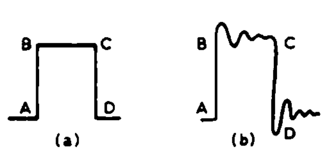


FIG. 3.20. *A square-wave signal, as shown at* (a), *would cause cone oscillation* (b) *as the result of insufficient damping.*

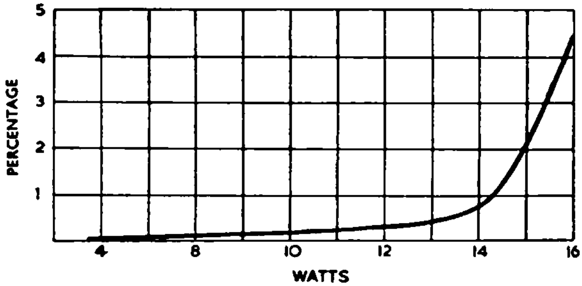


FIG. 3.21. *Typical harmonic distortion curve for an amplifier rated at a nominal* 10 *watts.*

manner on its pivot; by short-circuiting the terminals, however, the pointer oscillation will be found to be considerably curtailed. The moving coil is damped by the low resistance shunt (short-circuit). This is one way of protecting such instruments during transit.

An infinite damping factor is when the source impedance of the amplifier "looks" to the loudspeaker as a dead short. In amplifiers having a damping factor control the setting corresponding to this condition can be established by feeding into the amplifier a 1,000 c/s signal and adjusting the volume control until about 1–2 volts is measured across the loudspeaker terminals (an a.c. voltmeter is necessary). The damping factor control should then be adjusted whilst alternately disconnecting and reconnecting the loudspeaker. The position corresponding to an infinite damping factor is revealed by the output voltage remaining the same whether the speaker is connected or disconnected.

POWER OUTPUT

A hi-fi amplifier should be capable of supplying a speaker with at least 10 watts of power with a total harmonic distortion not exceeding 0·1 per cent. Obviously, the amount of power required will depend upon not only the size of the room, but also the efficiency of the loudspeaker system. A speaker of 10 per cent efficiency (a good figure) will produce only 1 watt of radiated acoustical power when connected to a fully loaded 10-watt amplifier, but this level of sound cannot be endured with any comfort in an average-sized living room or lounge.

Most amplifiers are very conservatively rated, and quite a number of British instruments nominally rated at 10 watts can be pushed up to 14 watts without the distortion rising above 1 per cent. This is illustrated by the typical distortion curve in Fig. 3.21; at 10 watts the distortion is in the region of 0·1 per cent, and is still reasonably low up to 14 watts, but rises very seriously at greater outputs.

It is highly desirable for a high-power amplifier to have a margin of reserve power in hand before distortion becomes serious. For example, an orchestral peak of sound may increase the input signal voltage beyond that specified for maximum output. In the case of a 10-watt amplifier working at nearly full volume, such peaks will push the amplifier into heavy distortion. In the case of a 30-watt amplifier adjusted for the same nominal sound intensity, however, the peaks will still be within the low-distortion power-handling range of the unit, and will be reproduced with equal fidelity as the low-intensity sounds.

FREQUENCY AND POWER RESPONSE

The frequency response is taken as a measure of the deviation of output signal voltage as the input signal voltage is altered in frequency over and beyond the audio spectrum (see Fig. 3.14). The output voltage deviation is expressed in decibels over the frequency range employed. Unfortunately, the frequency response is usually given for a level considerably below the maximum power output of the amplifier. Thus, the response characteristic may be virtually level from, say, 10 c/s to 20,000 c/s at an output of 1 watt, but deviate considerably from this at full output.

An amplifier rated at 10 watts may give this figure at 0·1 per cent distortion at 1,000 c/s (the frequency at which measurement is usually made), but for the same distortion at 100 c/s the maximum power may be only 5 watts. The distortion may well rise to 2 or 3 per cent at 100 c/s unless the input signal at and below 100 c/s is reduced accordingly. This, of course, is a hypothetical consideration and to such a degree may not be common to all amplifiers; nevertheless, it indicates a need for a low-cut filter somewhere in the pre-output stages. To a smaller degree the same reasoning applies to the higher frequency end of the audio spectrum, but here the trouble is not as serious, as the higher-frequency signal components usually comprise low-level harmonics.

POWER SUPPLIES

The mains transformer, rectifier, smoothing choke and filter components are mounted on the power-amplifier chassis, or on the power-amplifier side of a combination chassis. In the case of two-unit models, power for the pre-amplifier is fed from the power amplifier by way of a multi-cored cable and plugs and sockets to suit the individual design.

Referring to Fig. 3.19, the mains transformer T2 has an adjustable primary winding to suit almost any mains supply, and three secondary windings. The h.t. winding is centre-tapped and supplies 400 volts (with respect to chassis) to the two anodes of the h.t. rectifier valve V5. This is a straightforward full-wave rectifier circuit, which needs little comment. H.t. smoothing is provided by the smoothing choke L1 and the electrolytic capacitors C13 and C14, and a full 450 volts d.c. is present on the main h.t.

FIG. 3.22. *The Pamphonic amplifier, Model* 1,004.

line, which is also fed to the pre-amplifier power socket on point 4. The rectifier heater has its own winding supplying 5 volts a.c., and a 6·3-volt winding serves to supply the heaters of all the valves, the l.t. feed for the pre-amplifier valves being taken to points 7 and 8 on the connecting socket.

The potentiometer P2 serves as a humdinger control. Apart from producing an exact balance in the heater chain, it also introduces a small positive bias to the heaters from the cathode of V4. Correct adjustment of this control can reduce the residual mains hum by as much as 20–30 db. Adjustment is best made by connecting a sensitive a.c. voltmeter across the loudspeaker terminals, short-circuiting all the input terminals of the amplifier, and rotating the potentiometer for minimum reading on the voltmeter. The adjustment can also be made by ear for minimum hum level.

Illustrations of three typical equipments are given in Figs. 3.22, 3.23 and 3.24. The Pamphonic Model 1,004 (Fig. 3.22) is a 10-watt combination unit, having a maximum distortion of 0·5 per cent at 1,000 c/s at full output, and a frequency response which is substantially flat from 20 c/s to 50 c/s. It has 20 db of negative feedback, and inputs for microphone, tape, radio and gram. Volume, contour (designed in accordance with the Fletcher-Munson

FIG. 3.23. *The Pamphonic amplifier, Model* 2,001.

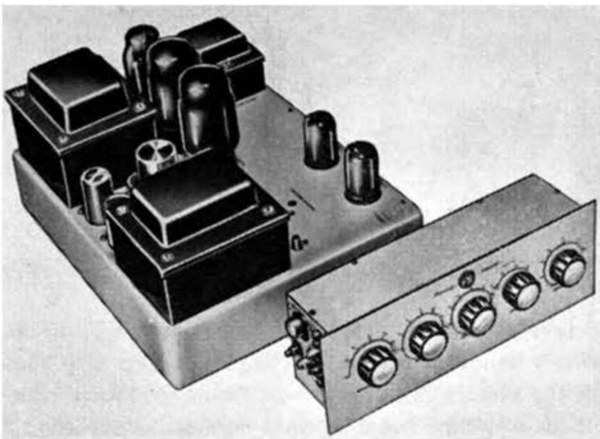


FIG. 3.24. *The Pye Provost power amplifier, Model HF25, and the Proctor remote-control unit, Model HF25A.*

curves), bass and treble controls are featured, the latter employing the Baxandall system. Provision is also made for a plug-in pick-up attenuator so that widely differing types of pick-up can be used. Three types of equalizing are catered for, selected on the main programme-selector switch.

The Pamphonic Model 2,001 is a larger two-unit amplifier (Fig. 3.23), capable of giving a full 25 watts at very low distortion. This model incorporates all the features described in this and the previous chapter.

Fig. 3.24 shows the Pye "Provost" power amplifier and the associated "Proctor" remote control unit. This also has a power output of 25 watts with less than 3 per cent harmonic distortion at 1,000 c/s. At 15 watts the distortion is less than 0·1 per cent, but approaches 3 per cent at 30 watts output. The amplifier has an excellent frequency characteristic (see Fig. 3.14) and is substantially flat from 2 c/s to 160 kc/s. All the requirements of a hi-fi amplifier are catered for, including the Baxandall-type tone control and four recording characteristics, by the use of feedback networks (see Fig. 2.17).

The power units of most hi-fi amplifiers are over-rated so that h.t. and l.t. can be fed to an auxiliary unit, such as a radio tuner. Provision is also usually available for the connexion of a gram motor or tape recorder. A signal output socket is often to be found on the pre-amplifier or control unit. This is usually picked-up from the output of the voltage amplifier, after the tone-control circuits and before the volume control. This feature enables the programme signal to be applied to the input of a tape recorder or additional amplifier.

CHAPTER 4

Tracing and Clearing Faults in Amplifiers

WITH a knowledge of the somewhat critical and specialized circuits used in hi-fi equipment (and also of the peculiarities of temperament of the enthusiastic hi-fi owner!) the service technician will soon find himself as much at home with hi-fi equipment as with radio or television receivers. Basically, the hi-fi chassis is far less complex than a modern television chassis, but because of the delicately balanced circuits in the former and the subjective nature of hi-fi, servicing compromises are never worth adopting.

For example, if an anode-load resistor is found to be open-circuit, and it is a close-tolerance component valued at 50,000 ohms, then replacement should be made with a component of identical characteristics. Whilst a preferred-value 47,000-ohm resistor would restore operation of the amplifier, and the service technician may feel that the performance is then well up to standard, the owner who is highly sensitive to every characteristic of his amplifier will soon sense that something is not quite right.

He will probably possess the circuit diagram of the unit and may eventually find the incorrectly valued resistor. Immediately he will attribute to this the shortcoming in performance. He will obtain and himself fit the correct component, and whilst technically the results may not be improved, the enthusiast will believe that they are and will be quite satisfied that the amplifier is now up to its former standard. Word will soon get around the local hi-fi world about the wicked ways of the unfortunate service technician, and he may have difficulty in regaining the confidence of the local enthusiasts.

COMPLETE FAILURE

This is one of the easiest of faults to locate. The first check would be to establish the connexion of power to the equipment. If the valves and pilot bulb are not alight, the fault is almost certainly in the mains input circuit. The plug and socket connexions at both ends of the mains lead should be examined carefully, and it should be established that power is actually present on the mains socket. There may be a break in one of the conductors of the mains supply lead. This often happens if the equipment is moved around extensively, but the trouble is usually of an intermittent nature.

The next move would be to check the amplifier fuse or fuses for continuity. If these are in order the on/off switch, associated connecting cables, connexions to the primary of the mains transformer and the voltage-selector plug and socket should be carefully examined. Sometimes a poor or intermittent connexion exists on the voltage-selector connector or a dry joint develops on the mains circuit connexions. Testing along these lines will soon reveal the cause of the trouble.

If it is found that the fuse is open-circuit, a check for short-circuits on the h.t., l.t. and mains circuits should be made before a new fuse is fitted and the amplifier switched on. Fuses are fitted to protect these circuits, and a fuse rarely blows without provocation. Test for shorts can be made with a simple ohmmeter. A check on the l.t. circuits should first lead to removal of all the valves and connexion of the ohmmeter across the heater line, bearing in mind that the line is shunted by the heater winding on the mains transformer and possibly a humdinger control. Removal of these components may also be called for if the meter used cannot indicate low ohms.

To check for a h.t. line short, the meter should be connected between the chassis and the h.t. line, bearing in mind the charging and discharging kicks promoted by the electrolytic capacitors. If a reading of some hundreds of ohms, or less, is given, the probe of the instrument should be transferred to the various h.t. feeds until the source of the low resistance or short-circuit is brought to light. Typical faults in this respect are shorting smoothing electrolytics, a short in the h.t. rectifier, a winding-to-core short in the smoothing choke, and valve-holder shorts to chassis.

If the circuits appear completely free from excessive leakage resistance, the fuse should be replaced (one of stipulated value is essential for optimum protection) and the amplifier re-connected to the mains and switched on. While it is warming up the valves, particularly the h.t. rectifier and output valves, should be carefully observed for signs of an internal flashover. A heavy flashover of this nature will immediately cause failure of the replacement fuse. In this event, both the valve and the fuse should be replaced.

If the valves are alight the programme signal should be disconnected or the volume control fully backed-off. At this point it should be made clear that the audio-frequency voltages developed across the primary winding of the output transformer rise to a high level if a signal is conveyed through the amplifier at normal level with the loudspeaker load removed. It is thus essential to establish continuity of the loudspeaker circuit with the signal removed. Many a good and expensive output transformer has been damaged by operating the amplifier without a correct load. The amplifier can be run at full power, of course, by using a suitable load resistor in place of the loudspeaker; a wire-wound component rated at the full output power of the unit should be used in this case.

It is a simple matter to check loudspeaker and connecting-lead continuity and resistance by disconnecting the loudspeaker wires from the speaker terminals on the amplifier and then connecting the leads to the terminals of a battery-operated ohmmeter. A crackle will be heard in the speaker on connexion and disconnexion of the wires.

Once it has been established that the loudspeaker is, in fact, acting as a load and is in good condition, the volume control can be advanced to its normal setting without fear of damaging the output transformer. At this stage, the programme-selector switch can be turned over the various positions, which, provided the appropriate programme signals are available, will indicate whether or not the failure is common to all channels.

Assuming that all channels are dead, tests should be made to find out whether the trouble lies in the pre-amplifier or power amplifier. This is a simple matter with two-unit amplifiers, it being necessary to unplug the pre-amplifier from the power amplifier and apply one of the programme signals direct to the input socket on the power amplifier; the pick-up signal is usually suitable for this test. Whether or not the power amplifier will be fully loaded by this signal will depend upon the overall sensitivity of the amplifier and the level of the pick-up signal—depending upon the type of pick-up used. At this stage, however, we are not interested in the quality or quantity of the sound, and provided we get a reasonable form of reproduction, it is fairly safe to assume that the trouble lies in the pre-amplifier section.

With a single-unit combination amplifier, a similar test can be made by applying the signal between the chassis and control grid of the valve immediately prior to the phase-splitter stage.

POWER AMPLIFIER FAILURE

When making signal tests on certain power amplifiers with the pre-amplifier or control unit disconnected, it may be necessary to short-circuit the points on the control-unit socket corresponding to the mains on/off switch, since this switch is usually situated on the control unit. Fig. 4.1 shows the circuit diagram of the well-known RD Junior power amplifier, in which points 7 and 8 on the control-unit socket are those associated with the mains on/off switch; shorting these will complete the mains input circuit. Referring to the same circuit, the signal would be applied across points 5 and 6 of the same socket, with the earthy side of the signal source to point 6.

A probable cause of the lack of response is failure of the h.t. circuits, and this can be established quite rapidly by testing the temperature of the output valves, V2 and V3, with a finger. The output valves usually operate

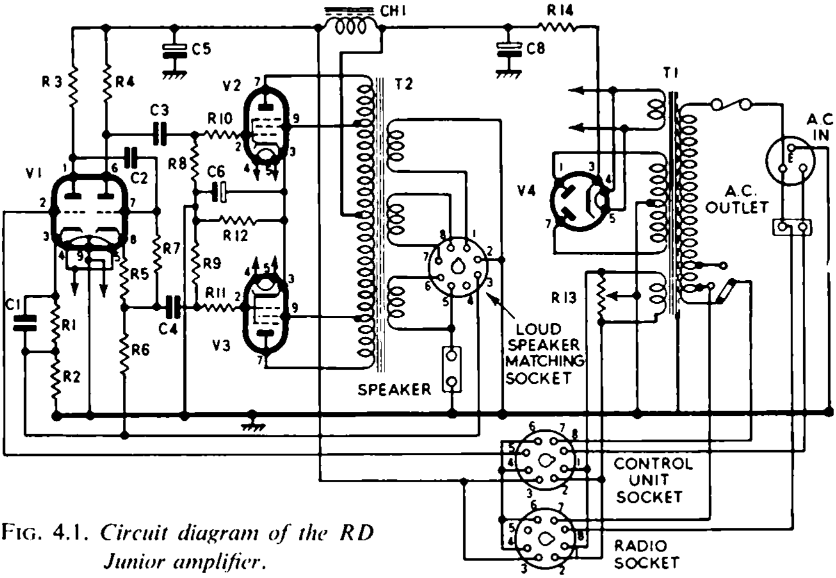


FIG. 4.1. *Circuit diagram of the RD Junior amplifier.*

at a fairly high temperature, and if they are only just warm there is a strong likelihood of a burnt-out h.t. rectifier valve, V4; failing that, the h.t. feed resistor R14 should be subjected to a continuity test. It is surprising how much can be done in the way of simple servicing and diagnosis without instruments —merely by applying a little well-concentrated thought.

It is extremely unlikely that the fault would be caused by simultaneous failure of both output valves; whilst failure of one of the output valves would result in the offending valve losing temperature, the good valve would remain too hot for comfortable touch, and the amplifier would reproduce after a fashion. The same applies with regard to the output-transformer primary windings—it is most unlikely that both sections would go open-circuit at the same time.

There is one more possibility, however, and that is open-circuit of the common cathode resistor R12. This trouble would cause both valves to lose temperature, though it is possible that the bypass capacitor C6—being a low-voltage electrolytic—would leak heavily and give some sort of cathode-circuit continuity. In this case, the amplifier would reproduce, but the distortion would be high. Some amplifiers have separate cathode resistors and, again, both would hardly fail simultaneously, though there is a remote chance of this happening!

If both output valves are working at fairly high temperature, and there is a very slight trace of normal residual mains hum from the loudspeaker, one can be fairly certain that the voltage-amplifier/phase-splitter stage, V1, is defective. To check the phase-splitter section, a signal could be applied to its grid—pin 7 in the circuit of Fig. 4.1; a pick-up signal may not be strong enough at this high-level point (depending upon the output voltage of the pick-up) and it may be necessary to bring into service an audio oscillator. Failing this, however, one side of the heater line could be connected to the grid through an 0·1 mF capacitor. This action will inject into the grid circuit a 50-c/s mains signal (at about 3 volts) and, if the phase-splitter section is operational, will give rise to a very loud mains hum from the loudspeaker. The remaining stage is the first triode section of V1—the voltage amplifier.

Open-circuit of the anode-load resistor R3 or the coupling capacitor C2 represent the most likely causes of the trouble. However, first a valve change and then a check of anode voltage will soon bring to light the trouble.

The same simple tests are all that are necessary if, for instance, the previous tests indicate trouble in the phase-splitter stage.

PRE-AMPLIFIER FAILURE

Fig. 4.2 shows the circuit diagram of the RD Junior control unit (pre-amplifier). If it is found that the power amplifier passes a signal, but some fault is preventing its passage through the pre-amplifier, it is best to make tests with the two units connected together in the normal manner. However, before delving too deeply into the pre-amplifier circuit from the servicing aspect, it often saves considerable time to ensure that the signal-carrying conductors of the multi-core pre-amplifier connecting-cable not only possess continuity, but that they are also in good electrical connexion with the tags on the plugs.

It is best to work back from the tone-control valve, V2b, to the first voltage-amplifier, V1. The signal fed to the power amplifier, by way of point 5 on the octal cable plug, is developed across the volume control P7, being picked-up from the anode of V2b. To check the goodness of stage V2b, the volume control should be advanced about three-quarters of maximum, and pin 7 of V2 touched with the blade of a screwdriver, with the blade making contact with a finger. The other hand should be kept well away from the amplifier, preferably in a pocket to avoid the risk of electric shock. If all is well, a loud hum will emit from the loudspeaker, as the result of the small mains signal being picked up by the body and injected into the grid. This test can be repeated at the grid of V2a (pin 2) and the grid of V1 (pin 9).

If there is a loud hum at pin 7 of V2 and no hum (or a very weak hum) at pin 2 of V2, the trouble lies either in V2a, in R22, R23 or in the coupling capacitor C23. The valve is the most likely cause, and should at least be checked by substitution. If there is a loud hum at the grid of V2a, but no hum at the grid of V1, V1 itself, R8, R7 and C4 should be checked in that order.

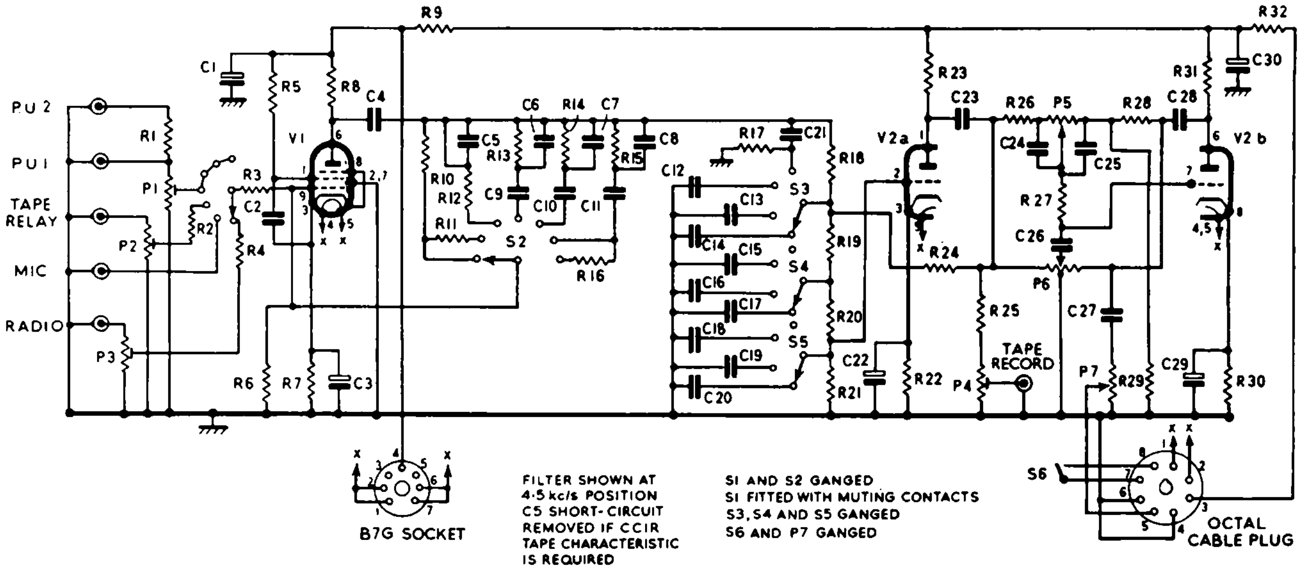


FIG. 4.2. *Circuit diagram of the RD Junior control unit, Mk. II.*

If R5 (the screen-feed resistor) appears to be overheating, suspect a short in C2. A few simple voltage and resistance checks will soon bring to light the component responsible.

H.t. power for the pre-amplifier is applied from point 3 on the octal cable plug. Make sure that h.t. is present here, and that it is getting past R32 and R9 (filter resistors). Overheating of R32 would indicate a short in C30, while the same trouble in C1 would cause R9 to overheat.

There is usually no need to set up elaborate instruments to diagnose for total failure if the tests outlined above are followed logically. Once the defective section has been revealed, the problem is virtually solved, for it is then only a matter of testing a few small components and the voltage at a couple of key points.

Instead of relying on the hum method of testing, the signal from an audio oscillator or generator can be applied to the various stages in turn until the point is reached where the signal is blocked; but generally speaking, the hum method is the quickest, and just as reliable. Alternatively, a pair of headphones, or an ear-piece, can be used to trace the signal through an amplifier up to the stage or component which is preventing it getting any farther. This method of testing calls for a normal input signal from one of the programme sources and average settings of the various controls. The phones can be used to trace the signal from the programme source right up to the point of the trouble. For more complex faults, test instruments are usually required.

DISTORTION

Distortion in one form or other probably accounts for the majority of troubles in hi-fi amplifiers. The symptom ranges from a very low-level distortion, which invariably demands some curious instinct to detect, to a very high-level distortion, whose presence is obvious to any listener.

The reader should understand that there is no such thing as a completely distortionless reproducing channel. Somehow, somewhere, in the electro-acoustic link between the live programme in the studio or concert hall and the ear of the listener at the loudspeaker end, the original sound will be altered slightly in character. It may be "coloured" by the position of the microphones in the studio and by the position of the loudspeaker and room acoustics at the listening end of the link. It will most definitely be modified during its passage in electrical form through the various electronic circuits

If a number of microphones are used close to the instruments of an orchestra, the pick-up of direct sound will be far in excess of the pick-up of reflected sound and the reproduced sound will lack "atmosphere"; it will not sound the same from the loudspeaker as it would in the middle of the concert hall. Little can be done by the enthusiast to correct this trouble, however.

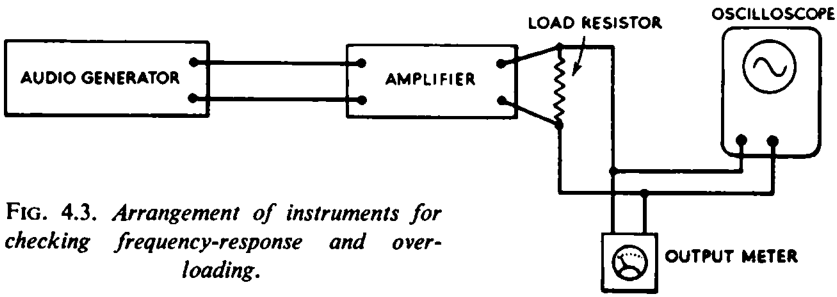


FIG. 4.3. *Arrangement of instruments for checking frequency-response and overloading.*

At the reproducing end, the room acoustics will obviously differ from those at the transmitting end, and even if a desirable degree of "atmosphere" is introduced, the final result will be further coloured by the listening-room acoustics. If "atmosphere" is purposely excluded by the sound engineer, it is most unlikely that the acoustics of the listening room will resemble those expected of a concert hall. A compromise is necessary along these lines, and this is the main reason why hi-fi amplifiers use elaborate tone-control circuits.

FREQUENCY DISTORTION

Frequency distortion is present when the output signal deviates widely in amplitude as a constant-amplitude input signal is altered in frequency over the entire audio spectrum. Almost all hi-fi amplifiers are substantially flat in response over, and beyond, the audio spectrum, as we have already discovered, and they are rarely troubled with this form of distortion. However, at high power outputs, the response may not be quite as flat as suggested by the appropriate response curves.

In Fig. 4.3 is shown an arrangement of instruments which can be used for frequency-response checking and plotting. An audio oscillator or generator is coupled to the input of the amplifier under test, ensuring that it is correctly matched to the input channel selected, a load resistor of suitable value and rating is employed in place of the loudspeaker and the voltage (a.c.) across it is measured by the output meter. The output signal is also monitored on an oscilloscope.

For high-level testing, the amplifier volume control is turned to maximum, the tone controls to the "flat" position, the filters switched out, the generator tuned to 1,000 c/s and the generator gain control adjusted for maximum power of the amplifier as given on the output meter. The waveform is synchronized on the oscilloscope to ensure that it is not highly distorted owing to overloading of the amplifier by too great an input signal.

With the various controls set, the generator should be tuned to about

20–30 c/s, the oscilloscope re-synchronized to that frequency and the waveform checked to ensure that it is still free from distortion. Normally, a pure sine wave will be displayed, depending upon the signal given by the generator, but if the peaks of the wave appear to be flattened, the input signal should be decreased until the distortion disappears, a note being made first of the original setting of the gain control. The output level should be noted at each point as the test is made over the audio spectrum, up to the limit of the generator, and plotted against frequency to give the response curve.

If it was necessary to decrease the input signal at the lower-frequency end, the gain should be advanced progressively up to 1,000 c/s, ensuring each time a test is performed that the signal is not overloading the amplifier.

For low-level testing, the same procedure is adopted, but this time the input signal is adjusted to give about 1 watt power output. In this case, there will be little danger of overloading the amplifier, and an oscilloscope is not essential.

If a proper output meter calibrated in watts of power is used, it will probably incorporate its own load resistor, but it must be ascertained that this represents the correct match to the amplifier; the meter should also have a level response itself over the audio spectrum. It is similarly pointless making such tests with an audio generator whose output voltage varies greatly over the band; if the instrument does not have a voltage-output indicator of its own, then its response should be plotted on a curve, which can later be used to correct the amplifier response curve.

If an output meter is not available, a high-resistance level-response a.c. voltmeter can be used equally well. The power output can be computed by using the expression: $W=E^2/R$, where E is the voltage and R is the resistance of the load in ohms.

The oscilloscope (which is invaluable for many tests on hi-fi equipment) should possess a good low-frequency response in relation to its Y amplifier (preferably from d.c. to 1 Mc/s or above), have a linear timebase and ease of synchronizing the test signal. An instrument highly suitable for this work is the Serviscope, by Telequipment. Among many other refinements, this

FIG. 4.4. *The Serviscope, by Telequipment.*

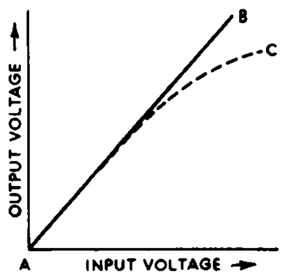


FIG. 4.5. *Input-voltage/output-voltage characteristic: A-B for a distortionless amplifier: A-C for a practical non-linear amplifier.*

has a Y amplifier which is substantially flat from d.c. to 6 Mc/s and a trigger-type timebase which obviates complex synchronizing re-adjustments on altering frequency (see Fig. 4.4).

Incidentally, an audio oscillator having provision for a square-wave output is most desirable, since many tests can be made by injecting a square-wave signal into the input and observing its form after passing through an amplifier.

An amplifier suffering from frequency distortion is characterized by its somehwat "mellow" tone, which is caused by severe attentuation of the higher frequencies in relation to the low frequencies.

NON-LINEAR DISTORTION AND HARMONICS

Owing to the curvature of valve characteristics, deficiencies of the output and coupling transformers, etc., the input voltage/output voltage characteristic of any amplifier deviates from a straight line over the major portion of its range. The effect is shown graphically in Fig. 4.5; here line *A-B* would represent the characteristic of a distortionless amplifier, but in practice the characteristic takes the form of the broken line *A-C*. From this it will be seen that the non-linearity is aggravated as the input voltage is increased. There is no such thing as a distortionless amplifier!

The generation of harmonics of the fundamental frequency of the input signal is one of the by-products of this non-linearity. For example, if the input signal is a pure sine wave of frequency 250 c/s, the output signal will consist of the fundamental 250 c/s signal, a second harmonic at 500 c/s, a third harmonic at 750 c/s, a fourth harmonic at 1,000 c/s, and so on. The magnitude of the spurious harmonic signals will depend upon the extent of the non-linearity, and they are usually expressed in the form of a percentage of the magnitude of the fundamental signal. Hence, if the power of the fundamental signal is 20 watts, and there is a second harmonic power of 1 watt, it could be said that the amplifier has a second harmonic distortion of 5 per cent. Usually, however, it is the *total* harmonic distortion of an amplifier which is given in the form of a percentage. With push-pull amplifiers, as we have already seen, the second and even harmonics are largely precluded by cancellation in the balanced load, and the third and possibly higher-order odd harmonics are the troublesome ones.

Harmonic distortion when present in a large degree is characterized by the harsh, "rough" nature of the reproduction. At lower levels, it is

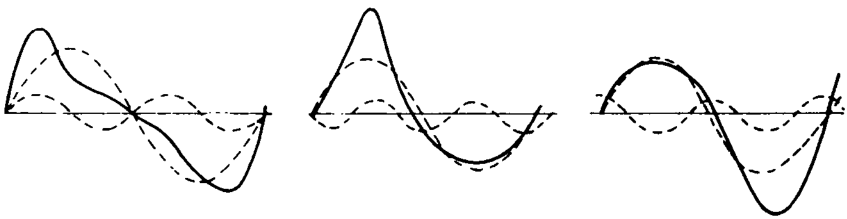


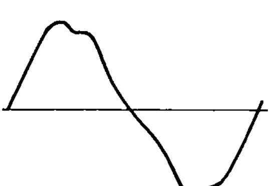
FIG. 4.6. *Various modes of second-harmonic distortion.*

difficult to define objectively, but its presence has a fatiguing effect on the listener; hi-fi enthusiasts are able to sense that something is not quite as it should be, and are glad to get out of audible range! Some harmonics are distinctly unpleasing, to say the least, particularly those which are dissonant with the fundamental frequency, such as the seventh, ninth, eleventh, etc. of a fundamental of 250 c/s.

Conversely, the emphasis or suppression of certain harmonics of certain sounds tends to enhance the original sound, and in some cases makes a displeasing sound more pleasing. This effect may be created by the use of the various tone controls and filter controls on the amplifier.

The deformation of the waveform produced by the harmonic components depends upon the phase of the harmonic relative to the fundamental. In Fig. 4.6*a* is shown, in broken line, a fundamental and second harmonic, which combine to form the distorted wave in full line. The combined wave is obtained by adding or subtracting the instantaneous values of the two waves. In (*b*) the harmonic component is displaced from the fundamental by 45 deg., resulting in a combined wave of somewhat different character, while in (*c*) the harmonic is displaced by 135 deg., which has the effect of inverting the distorted combined wave.

A third-harmonic component has the effect of distorting the waveform like that shown in Fig. 4.7. A characteristic of waves distorted by odd harmonics is that the positive and negative halves of the combined wave are similar, while with even harmonics the positive and negative half-waves are mirror images. In Fig. 4.8 is shown severe harmonic distortion created by iron saturation as the result of overloading of an output transformer.



(Left) FIG. 4.7. *Third-harmonic distortion.* (Right) FIG. 4.8. *Severe harmonic distortion caused by iron saturation in an output transformer due to an overload.*

INTERMODULATION DISTORTION

Another effect of non-linearity is intermodulation distortion. The effect occurs when more than one input frequency is applied to the amplifier, giving rise either to the production of sum or difference frequencies, or to the amplitude modulation of one frequency by the other.

Although in practice there are a host of input frequencies applied simultaneously, an illustration of the first effect is afforded by considering the application of only two frequencies, one at, say, 40 c/s and the other at 1,000 c/s. A whole string of sum and difference frequencies will be formed; for instance, the 40 c/s note will add to and subtract from the 1,000 c/s note, thus producing spurious signals at 1,040 c/s and 960 c/s. If the 40 c/s signal is of greater amplitude than the 1,000 c/s signal (tests are usually made in a ratio of magnitude of 4 to 1), harmonics of the 40 c/s signal will also add to and subtract from the 1,000 c/s signal, thereby giving spurious signals at 1,080 c/s, 920 c/s, 1,120 c/s, 880 c/s, and so on. Harmonics of the 1,000 c/s signal may also come into play if the non-linearity is severe, and make matters even worse! Intermodulation distortion of this type is very unpleasing to the listener, being far more disconcerting than simple harmonic distortion because the spurious sum-and-difference tones are not harmoniously related to the fundamental frequencies. Such distortion is characterized by a "buzz" or "rough harshness" in the reproduction, and is apparent to almost any listener.

Amplitude modulation of one frequency by another has been illustrated admirably by G. A. Briggs in his book, "Sound Reproduction". Instead of being connected to a source of d.c., the field coil of an early-type loudspeaker was inadvertently connected to a source of "raw" a.c. and, to quote the writer: "when a record was played through this equipment, music came out of the loudspeakers but it was almost unrecognizable, sounding as though it had been chopped up in a high-speed slicing machine. One effect of intermodulation is similar to this but not quite so bad."

Another example of intermodulation of this kind is the reproduction of a choir with an organ accompaniment. The author has had occasion to investigate this effect; a tape-recording made during a choir practice was said to have a curious "warbling" characteristic. This was because the choir was amplitude-modulated by the organ!

Harmonic distortion is checked on a wave analyser which is connected to the output of an amplifier. A sine-wave input is applied to the amplifier, and the wave analyser removes the fundamental frequency, and passes only the harmonic components, which are measured as a percentage of the fundamental.

An intermodulation analyser is required for measuring intermodulation distortion. The analyser usually supplies the two input signals over a selected range of test frequencies, and has an attenuator which alters the ratio of the two signal voltages over the range of 1:1 to 10:1. The composite signal is applied to the input of the amplifier under test, and the output of the amplifier is fed through a high-pass filter which eliminates the low-frequency test signal. The remaining signal, consisting of the high-frequency test signal plus the intermodulation, is demodulated. The high-frequency test signal is also eliminated in another filter, and the spurious intermodulation signals are fed to a measuring instrument which usually reads in terms of percentage intermodulation.

PHASE AND TRANSIENT DISTORTION

Phase distortion causes the output waveform to differ from the input waveform, due to alteration of the phase angle between the fundamental frequency and an associated harmonic, and of the phase angle between any two component frequencies of a complex wave. Although phase distortion has an effect on the reproduction of transients, it would, generally speaking, appear to be the least troublesome distortion encountered in audio work, though its presence is clearly visible in television receivers. With a hi-fi amplifier of wide frequency response, phase distortion is usually negligible, but it rises somewhat by the inclusion of filters which serve to limit the frequency range.

Distortion of the transients which, with certain kinds of music, occur at very high level, tends to impair the "attack" performance of the equipment. Transients are representative of sounds of short duration, such as those produced by certain string instruments—the piano, for example—and by percussive instruments. The general effect is that such reproduced sounds tend to "hang-on" after the energizing pulse or waveform has decayed, and where the distortion is severe, the frequency emitted during the period of decay may differ from that of the actual energizing waveform. Reproduction becomes very "slurred" on peaks.

Apart from a good transient response, depending to a large degree on both the electrical and acoustical damping of the loudspeaker system, the amplifier should also possess (1) a wide frequency response, extending beyond the limit of audibility, (2) no phase distortion, (3) a high output damping factor and (4) all resonant circuits, such as tone-control networks, filters and transformers should be sufficiently damped to avoid "ringing".

A circuit which is subject to damped or supersonic oscillations will be triggered into transient distortion by transient pulses, and the spurious oscillation—at frequencies depending upon tuned frequencies of the offending circuits—will become superimposed on the signal waveform.

Testing for transient distortion is performed by injecting a square-wave signal into the input of an amplifier and observing its character on the

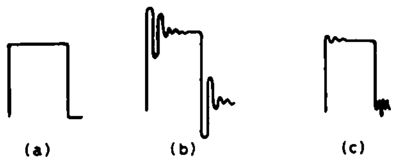


FIG. 4.9. *The input square wave shown at* (a) *promotes severe "ringing", as shown at* (b), *Slight "ringing", as at* (c), *usually has little effect on the transient response.*

screen of an oscilloscope connected across the output terminals; the amplifier should be properly matched at both the input and output terminals for this test. Fig. 4.9*a* shows the input square wave. Diagram (*b*) shows a very high degree of "ringing", which may be of such high amplitude on the peaks as to cause overdriving of one or more stages of the amplifier. The waveform at (*c*) shows a trace of "ringing" which, in practice, may have little significant effect on the reproduction.

OTHER SQUARE-WAVE TESTS

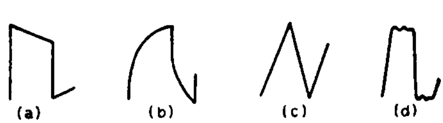
Square waves can tell us other things about an amplifier; they are useful because their formation depends upon the fact that they comprise harmonic components of their fundamental frequency extending well above the limit of audibility.

The low-frequency performance of an amplifier can be checked by applying a square wave having a fundamental frequency of, say, 50 c/s. Fig. 4.10*a* shows the usual resultant waveform on the screen of the oscilloscope. There will be some slope on the top of the waveform, but provided it is no greater than 40–50 per cent of the height of the waveform, the low-frequency performance can be considered satisfactory. It is important to check the oscilloscope on the square wave direct, however, to ensure that the wave is, in fact, square and that the Y amplifier itself (if used) is not responsible for distortion.

Having first ensured that the square wave given by the generator is maintained in accuracy over the whole of the audio spectrum, in terms of generator and oscilloscope performance, the square-wave signal applied to the amplifier can be varied in frequency up to 20,000 c/s, and the display observed at various intermediate frequencies.

With a good amplifier, the square-wave display should remain essentially uniform up to about 5,000 c/s, at which point very slight "ringing" may be

FIG. 4.10. *The frequency response of an amplifier can be checked by square waves;* (a) *reasonable low-frequency response:* (b) *poor high-frequency response:* (c) *very poor high-frequency response:* (d) *good high-frequency response.*



evidenced. If the amplifier has a poor high-frequency response, the waveform will deteriorate to that shown in Fig. 4.10*b*. As the input frequency is increased up to 10,000 c/s, a good amplifier will maintain a reasonable square-wave display, similar to that of Fig. 4.10*d*, but with a poor amplifier the display may deteriorate from waveform (*b*) to waveform (*c*). Increasing the input frequency up to 20,000 c/s really tests the upper-frequency response of the amplifier, but with a good hi-fi unit the waveform should differ little from that shown at (*d*).

PHASE-SHIFT TESTS

If two voltages of the same frequency are applied to the X and Y terminals of an oscilloscope, the result on the screen is either a straight diagonal line or an ellipse. The straight line is produced when the two frequencies are exactly in phase; an ellipse is produced when the signals differ in phase, but the dimensions of the ellipse will depend on the phase angle and the relative amplitudes of the voltages. However, should either of the signals not obey the sine law, the displays will be irregular in appearance.

Here, then, we not only have a method of checking the phase shift between the input and output terminals of an amplifier, but also, if we apply a pure sine wave to the input terminals, we can obtain an idea of the distortion given by the amplifier. The sequence of patterns shown in Fig. 4.11 illustrates such a display of two pure sine-wave signals of equal amplitude and frequency, but differing in phase angle from in-phase to 180 deg. out-of-phase. When the amplitudes of the signals are equal and there is a 90-deg. phase shift a perfect circle will result, and intermediate ellipses will occur either side of this point.

The sine-wave signal can be applied to the input terminals of the amplifier under test in the usual manner and a sample of the signal at this point applied to the Y terminals of the oscilloscope. The oscilloscope's time-base should be switched off and disconnected, and the output signal of the amplifier—that appearing across a correct-value load—applied to the X terminals. An attenuator may be required at this point so that the Y and X signals can be balanced. The degree of phase shift occurring over the pass-band of the amplifier will be revealed on the screen as the sine-wave generator is tuned over the audio spectrum.

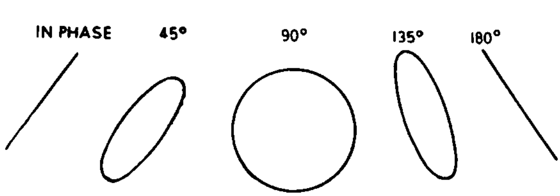


FIG. 4.11. *Sequence of patterns illustrating two pure sine signals of equal amplitude and frequency but differing in phase angle from in-phase to* 180 *deg. out-of-phase.*

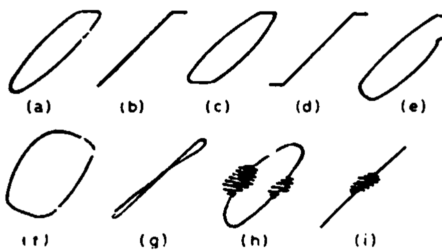


FIG. 4.12. *Phase-shift patterns. The waveforms at* (a) *to* (e) *reveal clipping of the output signal due to overloading or incorrect operating conditions of the valves:* (f) *and* (g) *show the presence of harmonic distortion as the result of excessive non-linearity: waveforms* (h) *and* (i) *show the presence of a spurious signal.*

Deviation in the symmetry of the display will result if the amplifier output signal is distorted on one cycle only, while if both cycles are equally distorted, the trace will remain symmetrical, but distorted in shape. Spurious oscillations in the system will also be shown on the trace. The various effects are illustrated by the waveforms in Fig. 4.12.

CORRECTION OF DISTORTION

Having first established that an amplifier is, in fact, producing distortion, and that the distortion is not present on the actual programme signal or caused by maladjusted controls, steps can be taken to locate and remedy the cause of the trouble. It is a good idea to work from a pure sine wave given by an audio oscillator or generator. and have this signal fed through the amplifier under test and monitored on the screen of an oscilloscope. To ensure correct balance of the circuits, both the input and output terminals should be terminated by the impedance (resistance) specified in the maker's handbook, and an output indicator should be connected across the output load. In this way the output power can be observed in relation to the distortion, and it can be immediately observed whether or not the distortion varies in magnitude as the strength of the input signal is varied.

If it is found that distortion is present only towards the maximum output limit of the amplifier, the most likely cause is overloading of a valve resulting in its being driven into the non-linear portion of its characteristic curve. Low h.t. voltage, due to a low-emission rectifier, or impaired emission of one of the output valves, is a possible cause of the trouble.

If the h.t. voltage is normal, the oscilloscope can be removed from across the output load resistor and the signal at the input and output of the phase-splitter checked for distortion. If there is no distortion at the input of the phase-splitter, but distortion is present at the output, the phase-splitter itself may be responsible. However, there is a possibility that grid current in the output valves is affecting the signal here, and if this is suspected, a test should be made with the phase-splitter coupling components disconnected from the output valves. If the signal from the phase-splitter is free from distortion after this action has been taken, and it remains virtually distortionless when the input signal is increased, there is little doubt that the trouble lies in the output stage.

The valves should be checked for emission and balance, and the cathode and grid resistors should also be checked for balance. If all seems well here, and the valve test is normal, the output transformer should be suspected for shorting turns. Shorting turns or trouble in the output transformer, apart from an open-circuited winding, is not always an easy fault to diagnose, and a suspect usually calls for a substitution test.

The chief symptom in this respect is lack of power, and if the amplifier is opened-up towards full volume, the reproduction becomes progressively more distorted without an apparent increase in output power; also, the faulty transformer tends to overheat. A short-circuit in one half of the primary winding promotes unbalance in the output stage with a resulting increase in second-harmonic distortion.

CHECK FOR OUTPUT STAGE BALANCE

Amplifiers with an adjustment for output-stage balance have an arrangement whereby the bias of one valve can be altered slightly in relation to the bias of the other valve. One method of achieving this is shown in the circuit in Fig. 4.13. When the stage is in balance, the current in either valve is the same, and the current is equal but opposite in each half-section of the primary of the output transformer. Thus, a voltmeter connected across the two anodes, as shown, will indicate zero voltage when the "balance" control is adjusted correctly.

It should be stressed that this adjustment serves only from the d.c. aspect of the circuit, in which case the circuit can be considered as a balanced bridge. In practice, there is little difference between the setting corresponding to optimum d.c. balance and that corresponding to optimum signal balance, but to secure optimum results in the latter case the "balance" control should be adjusted for minimum distortion, as indicated on a suitable distortion meter.

As the "balance" control usually has a very limited range, its inability to balance the circuit should first lead one to suspect low emission of one of the output valves. If both valves show reasonable balance on a valve tester, the

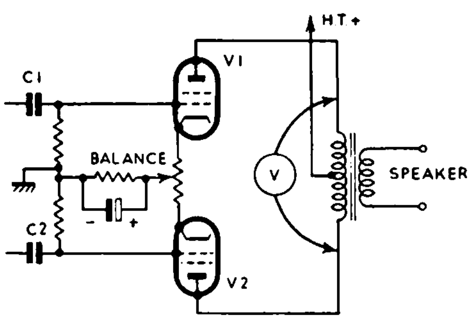


FIG. 4.13. *With a voltmeter connected across the anodes of the valves, as shown, the "balance" control should be adjusted for zero reading.*

remaining circuit elements should be checked for balance. If a valve tester is not available, and it is found that the voltmeter pointer remains one side of zero over the full range of the "balance" control, the position of the output valves should be reversed. A low-emission valve is definitely responsible if this change causes the voltmeter pointer to remain on the other side of zero over the full range of the "balance" control.

If changing the valves in this way does not reverse the movement of the pointer in relation to zero, a change in value of a component should be suspected. Some amplifiers use separate cathode resistors as well as a "balance" control, in which case the trouble may well be caused by a change in value of one of these. There is also a possibility that one half of the primary of the output transformer has a bad short, causing a decrease in resistance of one half with respect to the other half. This trouble would promote severe distortion and lack of power, as already described.

A leak or poor insulation-resistance of one of the coupling capacitors (C1 and C2 in Fig. 4.13) would also seriously affect the balance of the stage. These capacitors are in connexion with a source of d.c. at the phase-splitter side, so that poor insulation would cause the control grid of the associated output valve to go positive. The negative bias given by the cathode circuit would thus be neutralized, and the affected valve would pass considerably more current than the other. Heavy distortion would occur, and it is most likely that the anode of the affected valve would glow a dull red; in any event the temperature of the valve would be considerably higher than normal. The valve would not last very long under this condition, and the resulting abnormally heavy current would most likely cause failure of the h.t. fuse.

Controls available for balancing the output stage should never be used as a means of neutralizing severe unbalance caused by alteration in the characteristic of a valve or value of a component. The control serves essentially to permit a little extra reduction in distortion content which would otherwise not be possible. Such controls are rarely found in equipment for sound reinforcement and public-address use, where the distortion content is in any case greater than that associated with hi-fi equipment.

Optimum balance of the output stage also serves to minimize the residual mains hum. In fact, balance is sometimes made in this respect; a sensitive a.c. voltmeter is connected across the output load, and with the signal input terminals short-circuited the "balance" control is adjusted for minimum hum voltage. It will be remembered that a similar form of adjustment was recommended for the "humdinger" control.

CHECK FOR SIGNAL BALANCE

If the output stage balances correctly from the d.c. aspect, but slight distortion is present across the load in spite of a distortion-free drive signal, the drive signal should be checked for balance at the control grid of each output valve. A square-wave or sine-wave signal should be applied to the input of the amplifier, and the signal amplitude measured at each grid in turn, relative to chassis, on the screen of an oscilloscope. A high-resistance or valve voltmeter can be used if an oscilloscope is not available and, if necessary, a suitable signal can be obtained from the heater supply.

The signals should be almost identical, but opposite in phase. If considerable deviation in amplitude is observed, the phase-splitter valve and associated components should be carefully checked for value and balance. With a signal applied, the coupling capacitors (C1 and C2, Fig. 4.13) can be checked for balance by measuring the voltage across them with an oscilloscope or valve voltmeter. Unbalance of these components may not affect the high- and medium-frequency performance, but may incite harmonic distortion at the low frequencies as the result of the reactance of one capacitor differing considerably from that of the other.

Harmonic distortion may rise above that stipulated for the amplifier by open-circuit or low-value of the bypass capacitor across a common cathode resistor. If such a capacitor is not used even harmonics will be injected into the grid circuit. This does not apply where separate cathode resistors are used.

CHECK FOR NEGATIVE FEEDBACK

If the distortion is sudden and severe, an investigation should be made of the main negative-feedback loop if the tests outlined above have failed to reveal the cause of the trouble. If the negative-feedback loop is in order, there should be a distinct increase in output, with the input signal kept at a constant level, on disconnecting the loop either at the cathode, where it is applied, or at the secondary of the output transformer. No apparent increase, or only a very slight increase, in output would indicate that the feedback loop is either open-circuit completely or that the loop resistor has increased in value. A few simple tests will establish the defective component in this case.

PARASITIC OSCILLATION

Although most amplifiers of hi-fi type have a reasonable margin of feedback stability, an increase in value of the cathode resistor where the feedback loop is connected or a reduction in value of the loop series resistor may increase the feedback above the safety margin and incite parasitic oscillation.

There is a possibility that the frequency of oscillation will be above the audio spectrum, in the supersonic region, where its presence will not be audible as such, but will play havoc with the quality of reproduction. High-frequency parasitic oscillation will immediately be revealed on an oscilloscope test of the output signal, but where such an instrument is not to hand, and

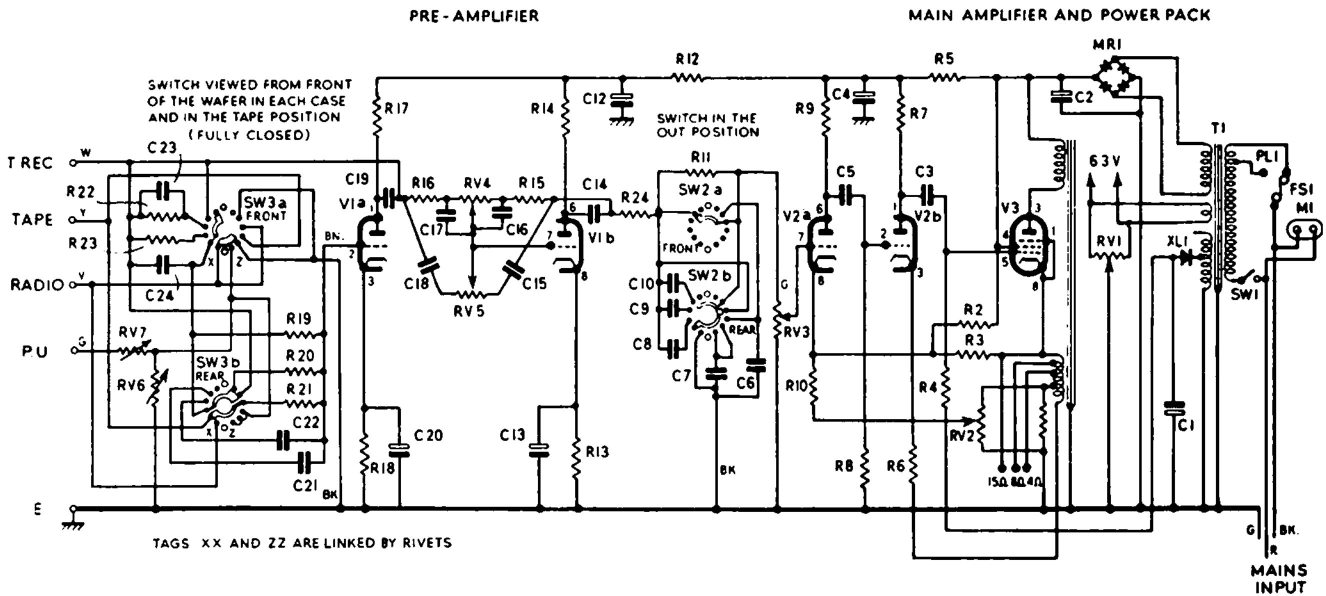


FIG. 4.14. *Circuit diagram of the Pye Mozart combined pre-amplifier and power amplifier.*

FIG. 4.15. *The Pye Mozart amplifier, Model HF10.*

the trouble is suspected, a milliammeter connected in series with the h.t. feed to output valves can be used as to indicate oscillation. A definite drop in current reading when the feedback loop is disconnected is indicative of trouble of this nature.

If the feedback components appear to be of reasonable tolerance, the output valves themselves should come under suspicion, since a severe unbalance of emission has been known to promote oscillation. In certain amplifiers low-value anti-parasitic resistors are sometimes connected in series with the anode and grid circuits of the output valves, and it should be ascertained that these are in good order.

Other expedients for maintaining stability over the very wide frequency range characteristic of modern equipment are (1) a capacitor and resistor in series in the anode circuit of the first valve of the power amplifier, which serve to reduce the gain at the unstable frequency within the amplifier's passband, and (2) a capacitor in parallel with the feedback loop resistor. The latter component promotes a phase shift opposite to that of the output transformer at the high-frequency resonance of this component, and thus prevents the feedback from turning positive at this frequency. Such devices are sometimes adopted in the Williamson amplifier. These components should be checked for value.

If a replacement output transformer introduces parasitic oscillation, then it may be necessary to modify slightly the value of the phase-shift feedback capacitor. The optimum value is best found by trial and error, and if an oscilloscope and a square-wave generator are available, the value giving the least distortion and "ringing" at 20,000 c/s should be chosen. The correct value feedback resistor and phase-shift capacitor must be used for the output impedance selected.

Apart from supersonic oscillations, very low-frequency oscillation may result from a fall in value of an electrolytic decoupling or filter capacitor; this may not be directly associated with the h.t. supply, but serve as a low-pass filter in a voltage amplifier. The effect is usually described as "motor-boating", but in certain instances the oscillation may be less than 10 c/s and inaudible. If all the filter capacitors are up to standard, the output valves should be checked for balance, as also should any push-pull driver valves.

If the feedback connexions on the secondary of the output transformer are reversed, the feedback will be positive instead of negative, and very bad oscillation will occur immediately the amplifier warms up. This trouble will not normally be encountered unless the transformer has been replaced and incorrectly connected.

TRACING DISTORTION THROUGH THE AMPLIFIER

As a basis for our tests we shall now refer to the circuit diagram of a commercial amplifier. Fig. 4.14 shows the circuit diagram of the well-known Pye "Mozart" combined control unit and power amplifier (Model HF10); see also Fig. 4.15. This is a remarkable amplifier with a single-ended output stage, having an output of 10 watts with a total harmonic distortion content of about 0·3 per cent at 9 watts. It has three inputs—"tape", "radio" and "pick-up", and an output for connecting to a tape recorder. In addition, it has a comprehensive tone-control system, a four-position filter and the Pye "Dialomatic" pick-up compensation, which permits easy matching to any pick-up. The single-ended output stage is worthy of note, since the design follows an ultra-linear arrangement centred around a grain-orientated output transformer.

If distortion is well in evidence, and the tests already described eliminate the output stage, there are two general methods which can be adopted to locate the source of the distortion. The procedure, of course, applies to all amplifiers.

The oscilloscope, having been adjusted for distortion tests and aided by a distortion-free input signal from an oscillator or generator, can be moved from the output load to the control grid of each preceding valve in turn, working towards the input signal. For example, if distortion is present across the output load, the oscilloscope should be connected to the control grid of the output valve, the Y-gain adjusted accordingly, and the quality of the waveform noted. If distortion is still present, the signal should be monitored at the grid of V2b, then at the grid of V2a, and so on until a point is reached where the waveform is free from distortion.

Of course, the gain of the Y amplifier will need to be increased as the signal is traced towards the low-level sections of the amplifier. It is also important to avoid overloading the amplifier, and it is best to set the signal level to the point where distortion just occurs—it is assumed that this is well below the maximum power output of the amplifier.

Let us suppose that distortion is present at the grid of V2b, but not at the grid of V2a. It is obvious that the distortion is being produced by misoperation of V2a; a likely cause would be low emission of the valve section itself, though an increase in value of R9 or a leak in C14 would also cause the trouble. Attention should also be paid to the components associated with the cathode circuit, these being related to the feedback network. In this way the signal can be traced back to its source and any deviation in wave-shape observed at each point of test.

If an oscilloscope is not available, an actual programme signal can be applied to the amplifier by way of its appropriate channel, and the signal monitored at each grid in turn from a pair of headphones or earpiece. In order to avoid interference from the loudspeaker, the loudspeaker can be disconnected and its place taken by a suitable resistive load. The point at which the distortion occurs will quickly be traced by this method, and then the circuit section can be analysed in detail.

Unfortunately, low-level distortion cannot usually be traced easily by this method, since headphones are rarely able to detect distortion at a level of, say, 5 per cent. Indeed, one has to be a very critical listener to detect distortion at such low level by way of the loudspeaker—programme material often contains distortion above this figure!

TONE CONTROL AND EQUALIZATION FAULTS

Maladjustment of the various tone controls, equalizers and filters is possibly responsible for the majority of reports of impaired performance at the high or low frequencies. Although the purist may be correct in the assumption that his amplifier has an absolutely flat response only when both the bass and treble controls are adjusted to the centre of their range, the controls should, nevertheless, be varied from this ideal setting as a means of securing a better balance of sound in relation to the listening room and associated equipment. It is surprising how some enthusiasts are extremely reluctant to use tone controls for the purpose for which they were designed.

It is also possible, however, that the overall frequency response may be far from linear at the centre setting of the controls, even though the designer may have intended a centre balance. Slight alteration in value of components associated with the tone-control circuits may shift the "linear" point well towards the end of the range of one (or both) of the tone controls. This possibility should be suspected if there appears to be a boost of bass or treble when the controls are set to the centre of their range. In extreme cases, it may be desirable to plot the frequency response of the amplifier to prove this point.

Maladjustment of the loudness control (if fitted) will incite excessive bass boost and possibly low-frequency distortion. If this trouble is suspected the loudness control should be switched out of circuit, or turned right off, and the volume and tone controls then adjusted in the ordinary manner. Finally, the loudness control can be brought back into circuit and adjusted for the correct level of sound, which will automatically give the correct degree of bass lift.

Some loudspeakers like more bass and/or treble than others, and the same applies to the listening room, as governed by the acoustics. It is quite in order to swing the tone controls over the whole of their range to get the "feeling" of the acoustics of the room and the response of the loudspeaker, after which the controls can be re-set more critically to give the results most pleasing to the listener, and most suitable for the programme material. Newcomers to hi-fi may be tempted to turn on too much bass or treble; this should be avoided. As G. A. Briggs points out, "if you notice the bass in the reproduction, or if the extreme 'top' is prominent, then there is something wrong because you do not notice bass and treble emphasis at a concert".

Too little bass is sometimes caused by misphasing of the loudspeakers when two separate units are used on the same amplifier. Usually, the speakers are marked at their terminals with a blob of red paint or a positive and negative sign so that they can easily be connected together in correct phase. When in parallel, the red terminals should be connected together (positive to positive and negative to negative); when in series, a positive terminal should be connected to a negative terminal, as when connecting batteries. If in doubt, a small cycle-lamp battery should be connected across the loudspeaker terminals and the resulting movement of the cone observed. The terminal of the loudspeaker which is connected to the positive tag on the battery to cause the cone to move, say, forward, should be clearly indicated with a blob of red paint.

Distortion of bass is invariably caused by core saturation of the output transformer. Such trouble is promoted by unbalanced output valves, causing a greater current in one half of the primary winding than in the other half. The output stage should be checked for balance by the method already described.

Another cause of this trouble is low value of one of the output-valve coupling capacitors. Here the capacitive reactance of the defective component will be considerably below that of the good component at low frequencies, thus resulting in overdriving of one output valve with respect to the other. In addition, the phase of the signal on the grid of one valve at the lower frequencies will deviate from that provided by the phase-splitter, and the phase difference will not be maintained at the ideal 180 deg. This will result in insufficient cancellation of harmonic distortion in the output load.

If the programme material possesses an abundance of bass, the amplifier itself may be overloaded at the lower frequencies. Some amplifiers have a fixed high-pass filter to preclude this trouble, while others have a switched "rumble" filter to cut the bass at the extreme end of the spectrum, essentially to obviate transmission of gram motor rumble through the amplifier.

Troubles in the treble may be caused by faulty components in the tone-control circuits, and this should first be suspected if the tone controls themselves appear not to be operating as they should. It should also be ensured that the equalization control is adjusted to suit the record being played. Matching of the pick-up and the various programme signals to the amplifier is most important if the correct response is to be maintained throughout the system. (Simple pick-up equalizers are considered in a later chapter.)

It should be remembered that the response of certain loudspeakers is affected somewhat by the damping applied to them from the amplifier. Insufficient damping—for example, by maladjustment of the damping control (if fitted)—will sometimes lead to a rise in the low-frequency resonance of the loudspeaker and an accompanying increase in the bass response. The bass in this case is of a purely synthetic nature.

HUM TROUBLES

Audio equipment is subject to two kinds of hum. There is the residual hum which is injected into the h.t. feed circuits as the result of a defective component associated with the smoothing and filter networks—this being synonymous with the hum experienced in radio receivers due to a breakdown of one of the electrolytic smoothing or filter capacitors. Then there is the hum caused by an alternating mains field being in proximity to the low-level stages of the amplifier. Here the radiated hum signals are picked up by the highly sensitive signal circuits, amplified by the equipment along with the required signal, and emitted by the loudspeaker in the characteristic manner.

Hum which is carried by the h.t. circuits usually presents little difficulty in remedying. The trouble is invariably caused by a reduction in value or open-circuit of one or more of the electrolytic filter capacitors. If the effect is present on a two-unit amplifier, the pre-amplifier should be disconnected from the power amplifier, and the residual hum level of the power amplifier noted. If the hum level is little different from that given when both units are connected, the power amplifier should receive attention.

If the hum is fairly loud, all the large-value capacitors associated with the h.t. supply should be checked either on a capacitor bridge or by substituting with good components. The connexions on the capacitors should be examined and re-made if necessary, and if an electrolytic unit relies for negative connexion upon clamp-contact with its case a check should be made to ensure that there is, in fact, a good low-resistance contact between the two points concerned.

As almost all hi-fi amplifiers use a full-wave h.t. rectifying circuit, residual h.t. hum will have a frequency twice that of the mains supply (100 c/s in Great Britain and 120 c/s in America), and it will also probably contain several harmonics of this frequency, thereby distinguishing it from normal 50 c/s to 60 c/s ripple.

Hum on the h.t. line can be traced with an oscilloscope or a.c. voltmeter isolated from the d.c. component by a paper capacitor. The hum level at the output of the h.t. rectifier should be noted, and then compared with the hum level at the other side of the smoothing choke, and so on through the filter chain. The hum reading should diminish considerably from section to section. There is the possibility of a shorting turn in the smoothing choke in cases where the hum persists. If the main filter capacitors are in order, the a.c. reading across the choke should be approximately equal to that across the output of the rectifier; a lesser voltage should lead one to suspect choke trouble. Smaller amplifiers of the 10-watt rating often use a wire-wound resistor in place of a choke, and a test should be made to ascertain that this part is of the stipulated value.

Unbalance of the rectifier valve can also lead to excessive hum, as can a shorting turn in one half of the h.t. winding on the mains transformer; in the latter event, the transformer will overheat and emit wax or pitch.

If the hum is just about audible with the signal input socket shorted, connect a sensitive a.c. voltmeter or output meter across the loudspeaker to register the hum level and adjust the humdinger control for minimum reading. If this does not reduce the level sufficiently, try adjusting the "balance" control, as unbalance of the output valves is another cause of high residual-hum level.

If the hum becomes prominent only with the pre-amplifier connected to the power amplifier, impaired h.t. filtering in the pre-amplifier is a most likely cause, particularly if the hum is present with the volume control at zero. Electrolytic capacitors should be checked as before and if the pre-amplifier has a separate humdinger control, this should be adjusted for minimum hum, as already described.

If the hum is not reduced, poor heater-to-cathode insulation in the final pre-amplifier valve may well be responsible. The best check is by valve replacement. The possibility of a hum voltage being induced into the pre-amplifier/power amplifier connecting lead should also be considered, especially if the lead has been increased in length for any reason and if the output of the pre-amplifier is at high impedance. A low-impedance cathode-follower output circuit is far less susceptible to such spurious pick-up.

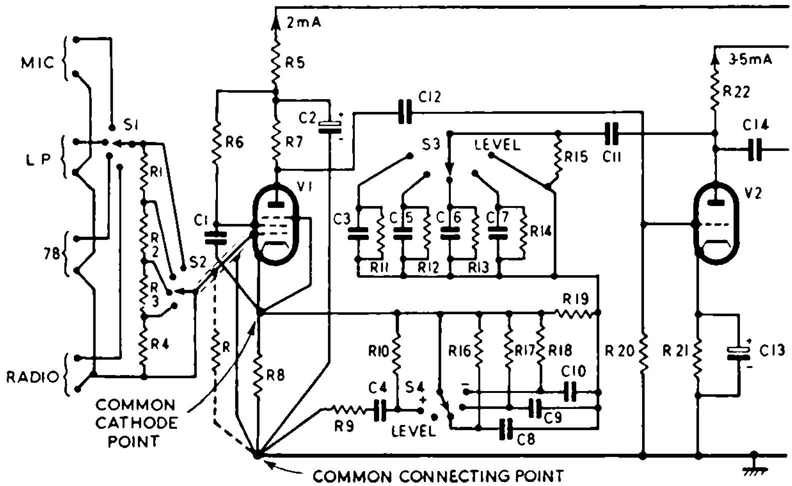


FIG. 4.16. *Circuit of input stage, showing common connecting points.*

If the hum level increases as the volume control is advanced, one can be certain that the hum is getting into the stages preceding the volume control. Make sure that it is not being induced into an open-circuit signal input socket by shorting the socket appropriate to the setting of the selector switch. If the hum is still present, check all electrolytic capacitors, and all valves for heater-to-cathode insulation. Suspect hum pick-up from stray fields.

Induced hum has been dealt with in Chapter 2, but there are one or two additional points which are worthy of note. Having first ascertained that the programme material is free from hum, and that hum is not being picked up on the programme-source connecting leads, attention should be paid to such things as high-resistance connexions between "earthing tags" and chassis, magnetic and electrostatic screens (including valve screens), misplaced grid or heater leads (particularly if the wiring has been disturbed during a previous servicing operation), the proximity of mains cables to grid circuits, etc.

It is surprising how much hum can be induced into an amplifier if it happens to be standing on the floor with the base cover removed, and if there is a mains cable running beneath the floor at this point! Even if the amplifier is lifted on to a table in similar proximity to the mains cable, the hum level may still be well above normal. Never run high-gain amplifiers with the screens removed, for it is remarkable how much a.c. mains field exists under domestic conditions. One can prod for hours trying to clear a slight hum which suddenly disappears on re-orientating the amplifier!

To avoid hum voltages being introduced into a low-level stage from the "earthy" points of the circuit, a chassis connexion common to the associated

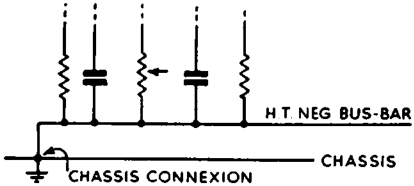


FIG. 4.17. *A bus-bar taking all the "return" circuits minimizes hum, providing it is connected at one point only to the chassis.*

circuits is often adopted in commercial and home-built equipment. The idea is shown in the circuit in Fig. 4.16; apart from a common chassis point, it will be seen that there is also a common cathode point.

Owing to circulating alternating currents in the chassis itself, particularly if it carries the power transformer and smoothing choke, the common-"earthing" device is often taken a stage farther. A heavy-gauge h.t. negative bus-bar is used for all the earth-return connexions (Fig. 4.17), and this is "tied" to the chassis at one point only. In this way, there is no danger of the difference in a.c. mains potential, which may exist between two points on the chassis when circulating currents are present, being reflected back into the grid circuits of the low-level stages.

Hum troubles may also be experienced if the pre-amplifier and power amplifier are earthed separately as well as being connected together electrically. As before, this results in a circulating alternating current, but this time in the loop between the two earth points and the conductor connexion between the two units. The disturbance can be reduced considerably by disconnecting the earth from the pre-amplifier.

Whilst this condition may not be incited purposely, it may exist in slightly different form between, say, a pick-up and pre-amplifier, due to earthing at the motor as well as the amplifier, or between a f.m. tuner and pre-amplifier, in which case the tuner may be efficiently earthed in the normal manner while the earth point of the amplifier is connected to the earth tag of a three-pin power plug. In both cases a common impedance, in which is circulating small alternating currents at mains frequency, is developed between the signal source and the amplifier input. This presents to the amplifier a spurious 50 c/s (60 c/s in America) signal along with the programme signal.

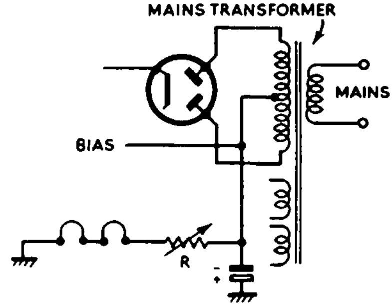


FIG. 4.18. *The first-stage heaters can be energized by the rectified h.t. current to reduce hum injection.*

In very high-gain low-level stages the valve heaters are sometimes energized by direct current as a means of reducing the hum level. If the required d.c. is not obtained from a small l.t. rectifier in association with an l.t. winding on the mains transformer, the h.t. current is suitably adjusted and allowed to pass through the heater chain.

The basic idea is illustrated in Fig. 4.18. Instead of the centre-tap of the h.t. winding on the mains transformer being connected direct to the chassis, as is usually the case, it is first passed through a variable resistance *R* and the valve heaters. It is assumed that the total h.t. current is more or less equal to the current required by the heaters, and any small discrepancy is corrected by the variable resistor. If the total amplifier h.t. current is, say, 90 mA, and the pre-amplifier valve heaters are rated at 100 mA, a h.t. bleeder resistor is connected across the h.t. circuit to pass the additional 10 mA, so that the total current flowing from chassis through the heaters into the centre-tap matches the 100-mA valve rating.

Capacitor *C* serves to hold the circuit down to chassis at low frequencies, and also acts as a part of a filter when the negative voltage, relative to chassis, at the centre-tap is used as a bias for the output valves.

This arrangement avoids having to run a.c. heater leads into the pre-amplifier section where either capacitively or inductively they may inject a hum signal into the grid circuits. Normally, however, if all the basic precautions are observed, and the heater leads are twisted together to cancel hum fields, there is little need for d.c. operation these days, particularly with modern valves such as the EF86.

The practice of providing a slightly positive potential on the heater line, either from a decoupled potential-divider across the h.t. circuit or from the cathode of one of the output valves, is frequently adopted in modern equipment. This prevents the a.c.-modulated electrons emitted from the heater. at the point where it enters and leaves the cathode, from reaching the anode and causing hum. Since the heater is made more positive than the cathode, random emission of electrons from the heater section which is outside the cover of the cathode are attracted back to the heater, and thus do not contribute to the normal electron stream.

The hum and noise levels are usually given as a composite figure in terms of decibels relative to the full output of the amplifier. Figures range from -60 db to -90 db; for example, the GEC BCS2317/8 is approximately -66 db relative to full output (12 watts), while the Pye HF25 is given as -90 db on 25 watts. In neither case can the hum be heard.

The general "hiss" that a high-gain amplifier gives at full volume represents the noise output. As already mentioned, this is often referred to as "white noise", since it is not confined to any particular frequency, and is contributed mainly by the valves and resistors in the low-level stages.

CHAPTER 5

Loudspeakers and Enclosures

THE electrodynamic or moving-coil loudspeaker in hi-fi use today has exactly the same fundamental principles of operation as the first moving coil unit ever manufactured, and it remains as popular as ever. However, whereas in early units the reproduction was severely "coloured" by the limited frequency response and by disturbing resonances in the centre of the audio spectrum, development over the years has resulted in the elimination of most of these shortcomings, at least in the better class of speaker.

There has been a reduction in all forms of distortion, the weight of the unit has been reduced and the overall efficiency increased by the use of new magnetic materials, while the main cone resonance has been pushed well down to the low-frequency end of the spectrum by improved methods of cone suspension. Although the response curve is by no means as sleek as that attributable to hi-fi amplifiers, there are twin-cone units which have a frequency range from 20 c/s to 20,000 c/s.

The moving-coil speaker has a strong magnetic field, produced by a permanent magnet, in which is placed a free-moving speech coil loaded by a cone. The speech coil moves axially either into or out of the magnet as the result of the signal current in the coil setting up a field which interacts with the magnetic field. The cone thus acts as a piston on the surrounding air and gives rise to pressure waves.

Apart from being proportional to the length of the conductor (the speech coil) and the signal current, the driving force—and hence the acoustic output—is also proportional to the strength of the field produced by the permanent magnet. The *gauss* is the measure of the field or flux density; 17,000 gauss being a typical value for a 10-inch unit. The total flux, however, is measured in *maxwells*, the gauss unit being equal to one maxwell per square centimetre. The strong magnets and large pole pieces of modern units create a total flux value approaching 200,000 maxwells.

The developments which have taken place in the production of permanent magnets have not only improved the electro-acoustic efficiency of modern loudspeakers, but have resulted in improved transient response and extended high-frequency performance. Smoothness of response over the audio spectrum is a very desirable characteristic of hi-fi units. Even though the overall response curve still has its ups and downs, the undulations of a good modern speaker are mild compared with those attributable to a general-service unit of a few years back.

Fig. 5.1 shows the response curve of the Model HF816 8-inch unit by Whiteley Electrical (WB Stentorian). A unique feature of speakers made by this firm is the universal-impedance speech coil, providing instant matching to transformers of 3 ohms, 7·5 ohms and 15 ohms. The response curve is made up of major and minor resonances over the frequency range. The major resonances can be dealt with from the loading aspect, but the minor resonances tend to give small coloration to the reproduction, and since these vary from speaker to speaker it follows that each speaker contributes its own

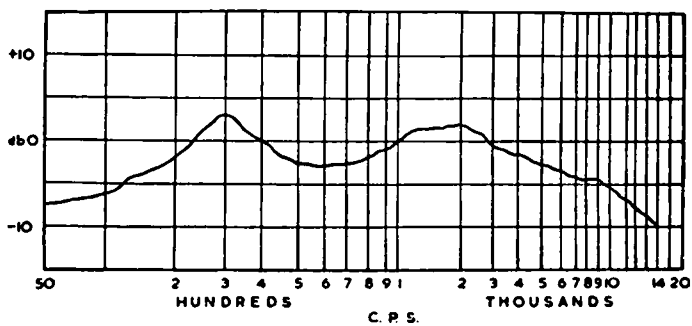


FIG. 5.1. *Response curve of the WB 8-inch loudspeaker, Model HF*816.

characteristic sound to the reproduction. Normally, this coloration is not very noticeable when speakers are compared on a complex signal such as orchestral music, but if a white-noise signal is used, the difference in rendering of the wide-band signal is remarkable.

CONE AND SUSPENSION

Both the character of the cone and its method of suspension on the loudspeaker chassis have a bearing on the various resonances evoked. The new plastic-foam method of cone suspension has brought about a much flatter overall response. The old method made use of a moulded corrugated surround of cone material glued to the speaker chassis; the new method does away with the moulded surround, and the periphery of the cone is suspended to the chassis by a soft plastic-foam material (see Fig. 5.2). This lighter suspension tends to damp the cone movement somewhat, and hence improve

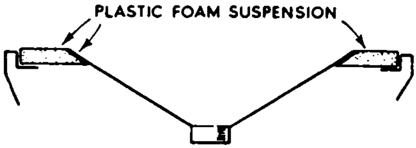


FIG. 5.2. *The modern method of plastic-foam suspension for loudspeakers.*

the transient response. It also reduces the main resonance to a very low frequency, and has the effect of alleviating cone break-up problems.

Cone break-up is essentially responsible for the irregular response towards the middle frequencies. At low frequencies, the whole cone moves up and down after the style of a piston, but as the frequency is raised its movement becomes so rapid that the direct energy of the actuating force is not simultaneously conveyed to the whole area of the cone. The movement towards the speech coil is reasonably faithful, but towards the periphery the amplitude of the movement decreases, and whole sections of the cone break up and vibrate in their own mode and phase, as governed by the driving frequency and inherent resonances. The effect is rather like the waves formed on a length of rope which is vigorously shaken up and down at one end.

Another approach to this problem is that of the General Electric Co., who have introduced a metal cone. The cone is made of Duralumin, and is thus light and rigid. There are shaped deformations over its area which tend to neutralize break-up and smooth the frequency response. Further irregularities in the middle-frequency range are reduced by a special "bung" which is secured to the pole piece. The effective working range of the speaker is from 30 c/s to 20,000 c/s. It has a maximum instantaneous power rating of 12 watts and a continuous power rating of 6 watts.

TWIN-CONE AND MULTIPLE UNITS

As a means of obtaining a wide overall response from a single unit, a twin-cone assembly is sometimes used. The main low- and middle-frequency range cone is so designed that above a certain frequency (known as the mechanical cross-over frequency) the speech coil effectively becomes decoupled from the cone and very little sound energy is produced by the main cone. However, tightly coupled to the speech coil is a very light and small cone which takes over at this point and maintains the response to the high-frequency end of the spectrum. The Axiom range of Goodman's loudspeakers uses this principle. The Axiom 80, a 9½-inch unit, has a frequency range from 20 c/s to 20,000 c/s, while the 12-inch units have a range from 30 c/s to 15,000 c/s.

Instead of using a single loudspeaker to cover the whole range of audible frequencies, two or even three units may be used, each being designed to provide optimum results over a limited band of frequencies. This method eases the design problems associated with full-range units, eliminates undesirable compromises in design and brings about a marked reduction in intermodulation distortion.

Sounds occupying the low- and middle-frequency portions of the spectrum are served by an ordinary moving-coil unit of fairly large dimensions, while the top frequencies are catered for by a much smaller unit designed for a smooth response up to 20,000 c/s. Some loudspeaker systems include an additional unit for the reproduction of the middle range of frequencies, in which case the low-frequency unit operates over the range of about 1,000 c/s, at which point the middle-frequency unit takes over and responds up to about 5,000 c/s. The high-frequency unit then gradually takes over and caters for the remainder of the spectrum. The names of "woofer", "squawker" and "tweeter" are sometimes given to the low-frequency, middle-frequency and high-frequency units respectively.

A frequency-dividing network (cross-over unit) is used to split the output between the loudspeakers so that there is no call upon any unit to handle large amplitudes of frequencies beyond its range. Since the change-over from one speaker to the next is gradual, and occurs at the cross-over frequency of the dividing network, each unit should be capable of handling at least half an octave beyond the cross-over frequency at full power, and two to three octaves beyond the cross-over frequency at reduced power. The type of dividing network utilized has some bearing on this.

Apart from improvement in the directivity characteristics and response to transients, multiple-speaker systems have a distinct advantage in the reduction of distortion arising from the Doppler effect. Briefly, the Doppler effect is that of an increase in frequency when the source of the signal is advancing and a decrease when it is receding. We have all heard the change of pitch of a whistle of a railway engine when it passes through a station at considerable speed; this is an example of the Doppler effect from the aspect of sound waves—the greater the velocity of the source of the sound (or radio signal), the greater the change in frequency.

So far as the loudspeaker is concerned, the movement of the cone at low and middle frequencies is often of the order of plus and minus $\frac{1}{8}$ in. and, depending upon the actual frequency, the *velocity* of the cone is of considerable value. If, while the cone is being actuated by a relatively low-frequency sound, there is introduced a sound of greater frequency, then this will suffer a change in pitch since the source of the sound is moving at high velocity, and a Doppler discord will result. During the time that the cone is moving forward under the control of the low frequency, the pitch of the high-frequency sound will appear to increase at the front of the speaker, and decrease at the rear, thus doubling the net effect of the discord between the two surfaces of the cone. If the frequencies are segregated over a number of loudspeakers, the distortion is considerably reduced.

MIDDLE- AND HIGH-FREQUENCY UNITS

Ordinary 8-inch or 10-inch moving-coil units are often used to cater for the middle frequencies, and these are quite successful provided a good cross-over unit is adopted. There are, however, more specialized pressure-driven horn-loaded units, such as the Goodman's "Midax". This has a frequency range from 400 c/s to 8,000 c/s, the recommended cross-over frequencies being 750 c/s and 8,000 c/s, and it can handle something like 25 watts (British rating).

Pressure units are often used for the reproduction of the middle and high frequencies in conjunction with a horn for the purpose of increasing the acoustic loading on the cone or diaphragm. The pressure unit is, in effect, a moving-coil loudspeaker, but employing an aluminium-alloy diaphragm instead of a conventional cone. This reduces the mass, which is desirable for high-frequency work, while permitting easy pressure loading to the horn. The horn is usually of exponential nature, though other shapes, such as conical or parabolic, are sometimes used. The overall length of the horn determines the lowest frequency at which it will load adequately. For a reasonable low-frequency response the horn may have to be several feet in length. For the middle range of frequencies, the length is less of a problem (the overall length of the Midax is $18\frac{9}{16}$ in.), while for the high frequencies, a horn length of a few inches is all that is required.

Since horn loading increases the efficiency considerably above that of direct-radiator moving-coil units, variable attenuators are usually required in the lines feeding the horn-loaded units, so that the sensitivities can be easily matched.

Apart from horn-loaded pressure units using a diaphragm, ribbon loudspeakers are sometimes used, also in conjunction with a horn, for the faithful reproduction of high frequencies. This type of unit has a very thin corrugated aluminium ribbon suspended in a powerful magnetic field. The ribbon itself serves as the conductor, being analogous to the speech coil of conventional moving-coil units, and through this are passed the audio-frequency currents from the output transformer. The ribbon vibrates in sympathy with the original sound, and is pressure-loaded to a small horn of dimensions to match the higher audio frequencies.

PLESSEY IONOPHONE

A remarkable high-frequency reproducer which has no moving parts at all is the Plessey ionophone, an invention of Mr. S. Klein of Paris. The functional portion of the unit is a small quartz-glass tube. One end of this is open and the other drawn down to a small hole in which is inserted an electrode known as the "Kanthal" (see Fig. 5.3).

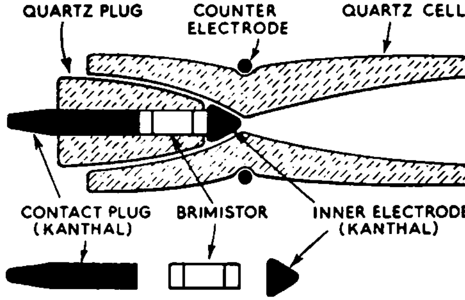


FIG. 5.3. *The functional portion of the Plessey ionophone.*

A glow discharge is arranged to take place in the air within the open end of the tube by applying a high voltage at radio-frequency across the Kanthal contact plug and the outside counter electrode. The r.f. oscillator providing the discharge voltage is modulated by the amplifier's a.f. signal, and since the intensity of the discharge at any instant is proportional to the instantaneous value of the applied r.f. voltage, the glow discharge will vary in sympathy with the a.f. signal. Because one end of the quartz tube is closed, pressures varying in sympathy with the a.f. signal will be set up in the open end of the tube, and these are conveyed directly to the mouth of an exponential horn for resolution into sound waves, of frequency range governed by the dimensions of the horn.

The Plessey unit is designed to work into a cross-over frequency of some 2,000 c/s, and from this frequency the response is perfectly smooth up to about 17,000 c/s. Since the arrangement uses no moving parts, the response to transients approaches the theoretical ideal.

The oscillator unit is built into the rear of the horn mounting, and is arranged to operate at 27 Mc/s, while the power-pack and modulation transformer are carried separately. Fig. 5.4 shows the unit in more detail, and its connexion to the output circuit of the oscillator, while Fig. 5.5 gives a suggested arrangement for the cabinet mounting of the ionophone, power-unit and Plessey 15-inch loudspeaker for reproduction of the frequencies below the 2,000 c/s cross-over.

ELECTROSTATIC UNITS

Of recent years considerable attention has been focused on the electrostatic type of loudspeaker and its potentialities as a

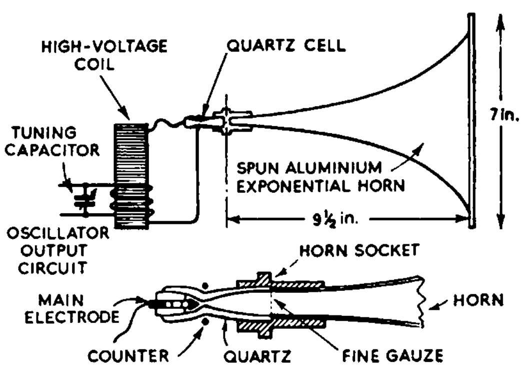


FIG. 5.4. *Connexion of the Ionophone to the output circuit of the r.f. oscillator.*

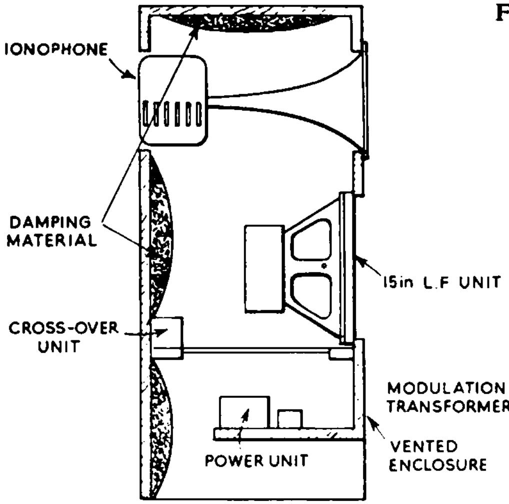


FIG. 5.5. *Suggested cabinet and mounting for the ionophone.*

full-range unit. High-frequency electrostatic units have been available for a number of years, and are adopted in a number of commercial radio sets, as well as in hi-fi loudspeaker systems. The principles involved for the reproduction of sound by this means are not new by any means, but the development of sound reproduction has revived interest in electrostatic units, formerly known as "condenser loudspeakers".

As will be seen from the basic arrangement of a simple unit in Fig. 5.6, two plates are used, as in a capacitor. One plate is fixed and perforated, while the other is very thin and movable, and mounted in such a way that it can vibrate without touching the fixed plate. In some designs the loudspeaker is curved to widen the angle of sound radiation.

The signal from the output stage of an amplifier (at high impedance) is applied across the two plates, sometimes through an isolating capacitor (as in Fig. 5.6). A fairly high voltage varying at the signal frequency is thus present across the plates, and this gives rise to mechanical forces which cause the thin movable plate to vibrate in sympathy with the sound signal. This is the normal electrostatic action. Under this condition, however, the movable plate will be attracted towards the fixed plate on every half-cycle of the signal, and two vibrations will occur on each full cycle. To avoid this useless function a polarizing voltage is applied along with the signal, across the

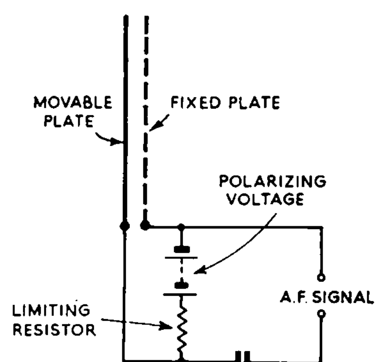


FIG. 5.6. *Basic arrangement of a simple electrostatic loudspeaker.*

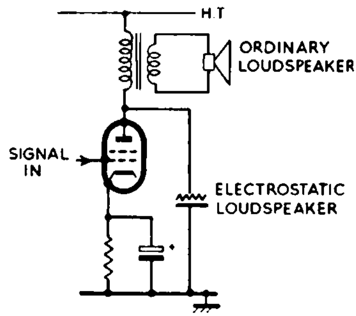


FIG. 5.7. *One method by which an electrostatic unit can be coupled to the output stage.*

plates, as shown in Fig. 5.6. The polarizing voltage causes an initial attraction of the movable plate towards the fixed plate, and provided the peak signal voltage does not exceed the polarizing voltage, the plate will vibrate in correct unison with the signal pattern. The limiting resistor in the circuit prevents the low-impedance polarizing source from short-circuiting the signal.

For the simple application of an electrostatic unit, the h.t. voltage at the anode of the output valve can be used as a polarizing voltage, as shown in Fig. 5.7. This voltage is varying at the signal frequency, and is fairly high (and at high impedance)—which is just what is required.

More elaborate electrostatic units employ a push-pull system (sometimes known as bilateral) in which the movable plate is situated between two fixed plates (Fig. 5.8). This idea is extended to full-range units, such as the Quad electrostatic loudspeaker (developed by P. J. Walker, of the Acoustical Manufacturing Co.). The two fixed plates are energized by equal and opposite signal voltages from the secondary of a push-pull transformer, the primary being connected to the signal source. The polarizing voltage is applied, along with the signal as already described, from a small e.h.t. generator. As current is not required for this operation, a simple generator similar to that used in certain television receivers is all that is required.

Electrostatic units can be loaded either by a horn or by an ordinary reflex cabinet, but certain designs provide automatic loading and do not usually require artificial means.

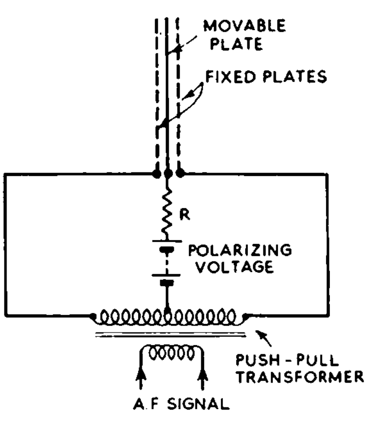
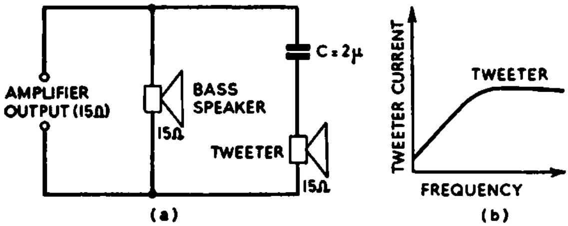


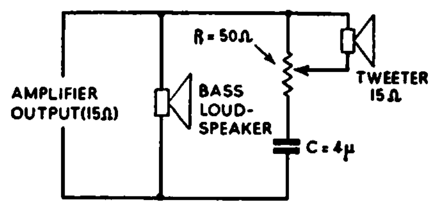
FIG. 5.8. *Push-pull electrostatic system.*

CROSS-OVER NETWORKS

A high-frequency unit or tweeter can be added easily to any loudspeaker system to augment the high-frequency response, as shown in Fig. 5.9 (*a*). In effect, the tweeter is supplied by way of capacitor *C*, the reactance of which decreases with rise in frequency, and thus limits the current in the speech coil



(Left) FIG. 5.9. *Simple method of connecting a tweeter* (a), *showing its operating curve* (b). (Below) FIG. 5.10. *Showing how a tweeter volume control can be introduced.*



at low frequencies. The curve in Fig. 5.9 (*b*) illustrates the action graphically.

If the tweeter is required to start taking over at about 5,000 c/s, then at this frequency the reactance of *C* should equal the impedance of the tweeter. Excluding any inductive effects of the tweeter's speech coil, and assuming an impedance of 15 ohms, a capacitor of 2 mF will serve the purpose (a 2-mF capacitor has an impedance of 15 ohms at 5,000 c/s). Under this condition, the current in the tweeter's speech coil at 5,000 c/s will be approximately 30 per cent below that in the speech coil of the bass unit: 5,000 c/s can thus be considered as the cross-over frequency (or more accurately, the take-over frequency).

If the speech coil impedance is, say, 7·5 ohms, or if a tweeter volume control is used, as shown in Fig. 5.10, and the "effective" impedance of a 15-ohm tweeter is reduced approximately to 7·5 ohms, then the value of *C* should be doubled. The same applies if it is required to lower the "take-over" frequency one octave when using a 15-ohm unit.

In the circuit in Fig. 5.11 (*a*) a choke *L* has been interposed in series with the bass speaker, the treble section remaining as in Fig. 5.9 (*a*). An inductor or choke has a reactance which is opposite to that of a capacitor; it rises with increase in frequency, and thus limits the amount of current in the

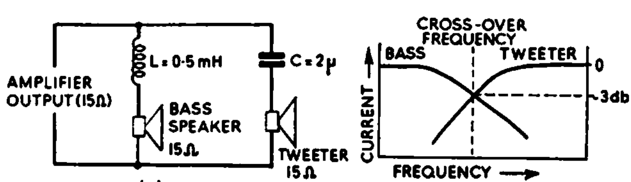
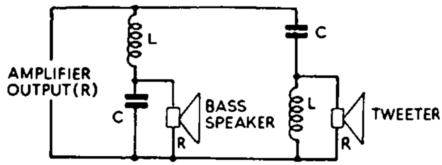
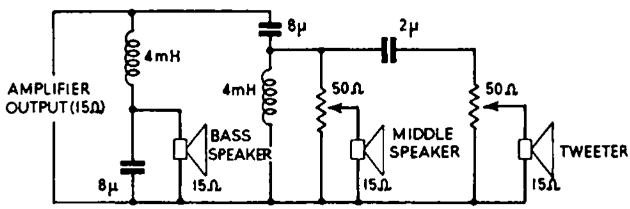


FIG. 5.11. *Simple quarter-section cross-over filter* (a), *and its operating curve* (b).

(Right) FIG. 5.12. *Circuit of a half-section cross-over filter.* (Below) FIG. 5.13. *A more elaborate arrangement for three speakers.*





speech coil of the bass unit at high frequencies. In this way a simple quarter-section parallel cross-over filter results. Provided all the impedances are 15 ohms and essentially resistive (which, unfortunately, is not always the case in practice), a 5,000 c/s cross-over can be secured by using an inductor valued at 0·5 mH and a 2-mF capacitor. The curves in Fig. 5.11 (*b*) illustrate the operation of the filter. These show that the current in the speech coils of both units at the cross-over frequency is 3 db below full current. This has little effect on the acoustical power, since both units will be contributing towards the total sound more or less equally.

This simple quarter-section filter is not always favoured owing to each section having an attenuation of only 6 db per octave at the cross-over frequency, and the greater attenuation of a half-section arrangement is often preferred. The circuit of such a network is given in Fig. 5.12. It will be seen that it is very similar to the quarter-section filter already given, but it has the addition of an inductor across the tweeter and a capacitor across the bass unit. The result of having these is to speed up the rate of attenuation at the cross-over region by a further 6 db per octave, thereby giving a total rate of attenuation of 12 db per octave.

As this kind of filter is extensively adopted, the following expression relating the component values with the cross-over frequency will be useful:

$$l(\text{mH}) = \frac{R\ 10^3}{\pi\ fc\ \sqrt{2}}$$

$$C(\text{mF}) = \frac{10^6}{2\pi fc\ R\sqrt{2}}$$

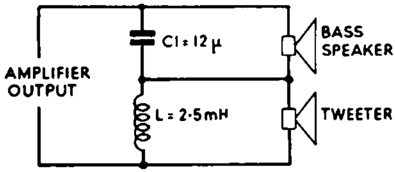


FIG. 5.14. *Quarter-section series cross-over network. With 15-ohm speakers, the component values given provide a 1,000 c/s cross-over.*

where R is the common impedance of the amplifier output and speakers in ohms and fc is the cross-over frequency in c/s.

Fig. 5.13 shows a more elaborate network suitable for three loudspeakers, in which the bass and middle units cross-over with a 12-db per octave slope, while the tweeter has a 6-db per octave take-over.

There are a host of cross-over networks, ranging from the simple high-pass/low-pass arrangement forming the quarter-section parallel system already discussed to a complex full-section arrangement incorporating three inductors and three capacitors. So-called "series" and "parallel" formations are used in practice. A quarter-section series network is given in Fig. 5.14, which makes interesting comparison with the quarter-section parallel circuit in Fig. 5.11. Similarly, the half-section series network in Fig. 5.15 can be compared with the half-section parallel network in Fig. 5.12.

The values given to the components in the circuits in Figs. 5.14 and 5.15 provide a cross-over in the region of 1,000 c/s when used with 15-ohm speakers. Since the component values are inversely proportional to the cross-over frequency, the cross-over is lifted an octave by halving the values, and dropped an octave by doubling the values. If the speaker impedance is halved, the capacitor values should be doubled and the capacitor values halved, to maintain the cross-over frequency for 15-ohm units.

All the networks given are of the constant-resistance type, since these are preferred by most equipment manufacturers and users. It should be noted, however, that all cross-over networks are much of a compromise—even though they perform well in practice—since the impedance of the speech coils of the loudspeakers is not maintained at a constant 15 ohms throughout the audible range. The theoretical ideal is to employ a power amplifier for each speaker, the high-frequency amplifier and speaker being fed from the pre-amplifier by way of a high-pass coupling, and the low-frequency amplifier

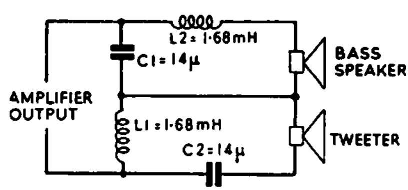


FIG. 5.15. *Half-section series cross-over network. With 15-ohm speakers, the component values given provide a 1,000-c/s cross-over.*

and speaker through a low-pass coupling, each coupling being arranged to have the required characteristic shape and attenuation rate.

SPEAKER PHASING

Two or more speakers are in phase when their cones move together in perfect unison under the control of the same signal. If one cone moves in while the other moves out, then the speakers are exactly 180 deg. out of phase. Intermediate phase displacements occur between these two extremes. For example, at the cross-over frequency, a quarter-section cross-over network usually introduces a phase displacement of 90 deg. For this reason, it is often desirable to position the tweeter one-quarter of a wavelength behind the plane of the bass unit, so that the sounds from the two units reach the ears of a listener at the same time (provided the speakers are phased correctly from the d.c. point of view). A cycle-lamp battery can be used to check this, as already described.

Incorrect phasing often gives the effect of "emptiness", "disembodied treble", lack of bass or lack of middle register, depending upon the frequency range fed to the units and the acoustics of the listening room.

MATCHING

An amplifier can only give its maximum undistorted output when the loudspeaker system represents a perfect match to its output terminals. If two 15-ohm units are connected in parallel across the output terminals, then the speaker impedance as "seen" by the amplifier is 7·5 ohms. Similarly, two 15-ohm units connected in series add up to a total impedance of 30 ohms, and in both instances the impedance adjustment on the amplifier must be altered to correspond.

Generally speaking, it is not a good idea to connect two hi-fi speakers in series, even though the amplifier output is adjusted to match the sum of the two impedances. The reason for this is that the actual resistances of the speech coils are also added in series, which has an adverse effect on the damping mechanism.

From Chapter 2 it will be recalled that the resistance of the speech coil represents the dominant impedance at high damping factors. When the two speakers are connected in series, each speaker "sees" something like 30 ohms—its own resistance in series with the resistance of the other speaker across the low source resistance of the amplifier. When they are in parallel, however, each speaker "sees" only its own resistance of 15 ohms or thereabouts. In other words, the introduction of another series-connected speaker is as futile as putting a resistor in the speech-coil circuit, and then trying to damp the actual speaker by reducing the source impedance of the amplifier.

Speakers connected in series tend to have a very poor transient response, and disconcerting "hangover" effects will develop. The same applies with series-parallel networks. These arrangements may suit public-address systems, where ease of connexion of speaker units is demanded, but they should not be used with hi-fi or sound-reinforcement equipment.

BAFFLES AND VENTED ENCLOSURES

Sound is emitted from both sides of the cone of a loudspeaker, and as a compression wave is formed on one side, a rarefaction wave is formed on the other side. At the lower frequencies, where the wavelengths of the sound waves approach the dimensions of the cone, sound waves from the rear tend to cancel sound waves from the front and the acoustical response of the unit diminishes almost to zero as the frequency is lowered. This undesirable effect can be avoided by mounting the unit on a baffle board so that the rear and front sound waves are isolated. Since the baffle cannot have infinite area, front-to-back cancellation will occur at some low frequency—governed by the dimensions of the baffle used. So that the cancellation will not be common in all dimensions, as in the case of a circular baffle on which the speaker is mounted in the exact centre, a rectangular baffle should be used and the speaker displaced from the centre.

A near approach to the infinite baffle is secured by mounting the speaker on a wall dividing two rooms. This idea is used by many enthusiasts, and will be encountered from time to time by the technician. Disconcerting irregularity of response can be evoked by this method, however, unless the hole in the wall is bevelled so as to avoid the cylinder formed by the wall thickness from resonating. Some enthusiasts make an extra large hole and introduce a small sub-baffle on which the speaker is mounted.

Baffles for treble and middle units need not be as large as their counterpart for the bass unit, though they should be large enough to respond down to an octave below the cross-over frequency.

A synthetic method of providing infinite baffle characteristics is to enclose the rear of the loudspeaker in a substantial air-tight box. The air trapped in the box acts as a cushion on the cone and thus provides acoustical damping, whilst also reducing the main cone resonance, depending upon the volume of the box.

A vented enclosure encloses the rear of the loudspeaker, is formed of a substantial box and has a vent hole cut in one of its walls, usually near the speaker aperture. The enclosure thus acts as a resonator (see Chapter 1), the resonant frequency being governed by the volume of air within the enclosure and the area of the vent. The effect that this has on the loudspeaker and acoustical response is rather interesting. At the resonant frequency of the enclosure most of the sound is radiated by the vent and the speaker cone is subjected to maximum acoustical damping. For this reason it is often

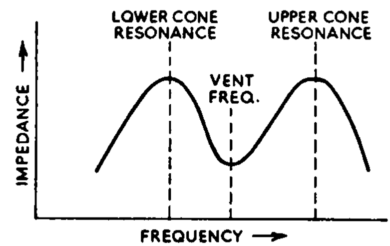


FIG. 5.16. *Impedance characteristics of a speaker correctly loaded by a reflex enclosure.*

advantageous if the cone and enclosure resonances coincide in frequency. An increase in low-frequency radiation is secured whilst the disturbing effect of the speaker resonance is alleviated. This, however, represents an ideal situation.

As the frequency decreases below the resonance of the enclosure, the output from the vent falls off, but the output from the speaker rises due to a lowering of its resonant frequency as the result of the air-pressure build-up between cone and vent. However, since the sounds from the two sources approach the anti-phase condition, the resultant sound output diminishes fairly quickly, and no useful increase in bass response is obtained by lowering the resonance of the loudspeaker.

As the frequency rises above the resonance of the enclosure, the loading on the speaker cone is lifted and its resonant frequency rises. Output from the vent tails off and the normal middle- and high-frequency characteristics of the speaker take over.

From the foregoing it will be realized that two auxiliary resonances of the cone occur on either side of the enclosure resonance. These are often illustrated in the form of a curve as shown in Fig. 5.16.

MEASURING SPEAKER IMPEDANCE

As an impedance curve of this kind provides useful information with regard to the loading of a speaker by its enclosure, it will be instructive at this stage to consider a method by which speaker impedances can be taken. Precise results require the use of an impedance bridge, but as such an instrument is not generally available to the enthusiast or technician other methods less exacting but practicable, will be described.

Fig. 5.17 shows a set-up which utilizes the Ohm's Law method. The voltage across the speech coil and the current in it are measured throughout the frequency range by the a.c. voltmeter V and the a.c. ammeter A. The test signal is provided by an audio oscillator, and the necessary power produced by the amplifier (any good hi-fi amplifier will serve). The impedance value at

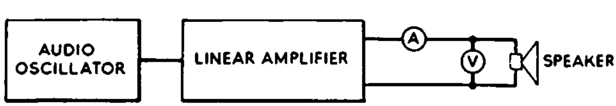


FIG. 5.17. *Set-up for measuring speaker impedance.*

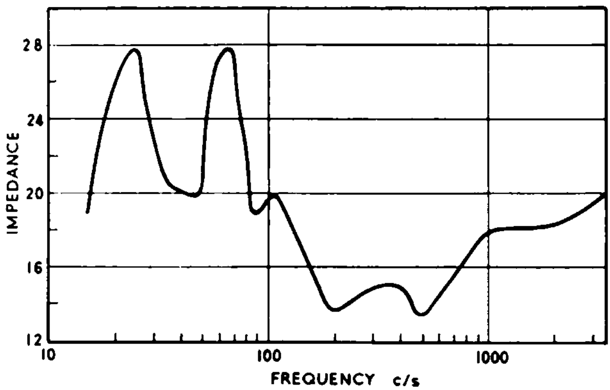


FIG. 5.18. *Experimental impedance curve of Goodman's Axiom 12-inch unit, Type 150, Mk. II, in a reflex enclosure which has a 1½-inch slate front.*

any frequency is computed by dividing the voltage reading by the current reading (in volts and amps). Thus, if 3 volts at 0·2 amp. are indicated, the speaker has an approximate impedance of 15 ohms at the frequency of the test.

Fig. 5.18 shows an experimental impedance curve using the above method of measurement. The speaker unit was a Goodman's Axiom 150 Mk. II, and was mounted in a reflex enclosure having a slate front of 1½ in. thickness. It will be seen that the two resonance peaks either side of the enclosure's resonance are nicely spaced and of equal amplitude, indicating that the speaker is ideally loaded by the enclosure.

Another method of checking impedances (see *Hi-fi News*, February, 1957) is shown in Fig. 5.19. Here a 30-ohm 1-watt resistor is interposed in the speech-coil circuit and the voltage across it measured by a Model 40 Avometer set to the 2·5-volts a.c. range. The speaker terminals are short-circuited and the oscillator and amplifier controls adjusted to give full-scale deflection —making sure that the oscillator is not overloading the first stages of the amplifier. The short-circuit is then removed, and the impedance of the speaker in ohms, at any particular frequency, is given on the "ohms" scale divided by 100. Thus, if the meter reads 2,000 ohms on the scale, the impedance is 20 ohms.

The signal voltage from the amplifier must remain constant throughout the frequency test range, but this can be monitored, if

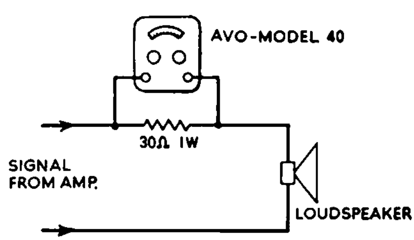
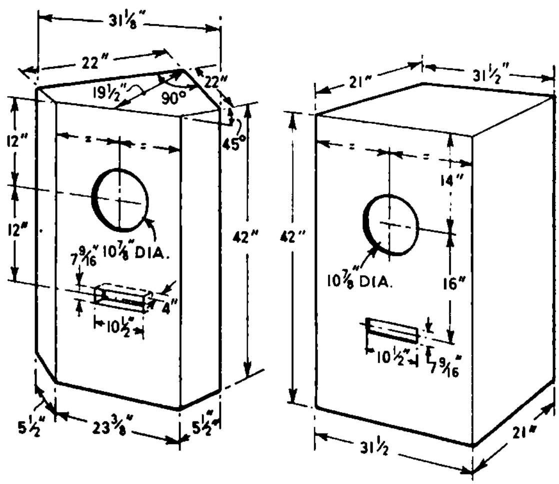


FIG. 5.19. *Another method for impedance measurement, using a Model 40 Avometer.*

FIG. 5.20. *Suggested enclosures for the TSL Lorenz LP 312-2 wide-range speaker system.*



necessary, by an a.c. voltmeter or on an oscilloscope. The output voltage from most hi-fi amplifiers is reasonably consistent over the frequency spectrum, provided the oscillator voltage applied is constant.

A few words of warning: the impedance curve is *not* an indication of acoustical output from the speaker. The only way that this can be assessed is by measuring the sound field produced by the speaker system. Since quite a lot of sound comes out of the loudspeaker during impedance tests, first make sure that the neighbours are out or that the test room is sound-proof! Even though hi-fi enthusiasts experiment late into the night, be absolutely certain that the neighbours are not in bed, for it is surprising how the bed springs can resonate when the oscillator is tuned to the critical frequency and plenty of watts are emitting from the speaker—even though the springs are heavily damped by the bedding.

ENCLOSURE CONSTRUCTION AND DIMENSIONS

Reverting to enclosures, the cabinets should be of solid and rigid construction. A ⅞-in. wall thickness is desirable, though double walls of thinner material can be used if the space between the walls is filled with dry sand. At least three walls, at right-angles to each other, should be lined to a depth in excess of 1 in. with sound-absorbing felt or plastic foam, as a means of reducing standing waves within the enclosure. Acoustic curtains are hung in the enclosure for the same reason.

The size of a reflex cabinet, for a given low-frequency response, can be reduced by the inclusion of an acoustical filter over part of the vent. Such filters take various forms, one type being marketed by Goodmans under the name "acoustical resistance unit". The filter reduces spurious resonances, while also minimizing the magnitude of the upper resonance which, as we have seen, is a characteristic of conventional reflex enclosures.

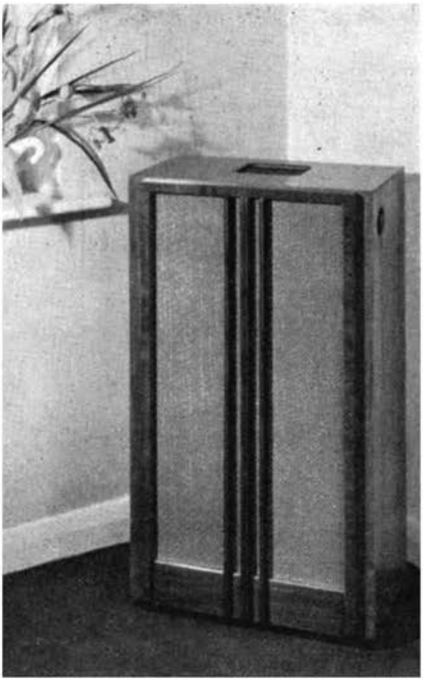


FIG. 5.21. *The Pamphonic Victor speaker system. This has a treble unit at the top, the volume of which can be controlled from the side of the cabinet. The preferred position for this system is in the corner of a room.*

In Fig. 5.20 is given the dimensions for suggested enclosures for the TSL Lorenz Type LP 312-2 loudspeaker system. This is a 12-inch unit, which also incorporates two Lorenz Type LPH 65 treble units mounted in front of the cone. Its overall frequency response is from 20 c/s to 17,000 c/s.

Fig. 5.21 shows the Pamphonic "Victor" speaker system. Apart from the large bass unit, this has a cone-type treble unit at the top of the enclosure. A cross-over network is contained in the enclosure, and the treble unit is fed by way of an attenuator which can be adjusted to give the correct degree of "presence". The treble unit is fully enclosed so that there is no interaction between the two sound sources, and it is mounted at an angle so that the sound is directed towards the corner of a room, whence it is deflected into the room at large.

OTHER ENCLOSURES

There are numerous other enclosures and speaker-loading devices used by enthusiasts, full constructional details of which are given elsewhere (i.e., in manufacturers' literature and in "Sound Reproduction" by G. A. Briggs). It is not intended to study them all here, but a few words on the more popular arrangements will not be amiss.

The exponential horn in relation to treble and middle-range speakers has already been mentioned. Whilst this system represents the most efficient way of loading a speaker, its large size at the lower frequencies is a disadvantage. Nevertheless, horn loading in the bass is adopted, usually by folding the horn in various ways within an enclosure.

The tuned-pipe arrangement is worth consideration. The pipe has one closed end and is critically dimensioned so that a fundamental anti-resonance occurs at the frequency of the major low-frequency resonance of the speaker. This gives the system a characteristic similar to that of the reflex enclosure, whilst being less difficult to manufacture. The principle of operation is rather like that of an organ pipe, though in some cases the pipe is tapered and the speaker mounted one-third of the overall length away from the closed end. At resonance the sound radiated from the open end is out of phase with that radiated from the cone.

The acoustical labyrinth is a type of enclosure which avoids resonance effects, but has a slight falling off at the low-frequency end of the spectrum. It consists essentially of a very long pipe (about 11 ft.), heavily lined with a thick felt or other acoustic damping material, and has the effect of completely absorbing radiations from the rear of the cone.

Another method of securing adequate low-frequency performance is by the use of a battery of some nine speaker units mounted on a wood baffle with short sides, which is left open at the rear. This idea is sometimes used by American enthusiasts.

LINE-SOURCE SPEAKER

One type of loudspeaker system, used particularly for sound-reinforcement applications rather than domestic hi-fi work, consists of several speaker units mounted close together one above the other in a wood or metal enclosure, depending on whether it is to be used indoors or outdoors. In the latter case the enclosure is weather-proofed. These loudspeaker systems are sometimes called "sound columns", and by Pamphonic Reproducers, Ltd., "line-source loudspeakers". Fig. 5.22 shows the Pamphonic system, with the units mounted one above the other. All the units are connected in phase, and the small units are connected by way of a suitable cross-over network to cater for the high frequencies.

The reason for mounting the units in

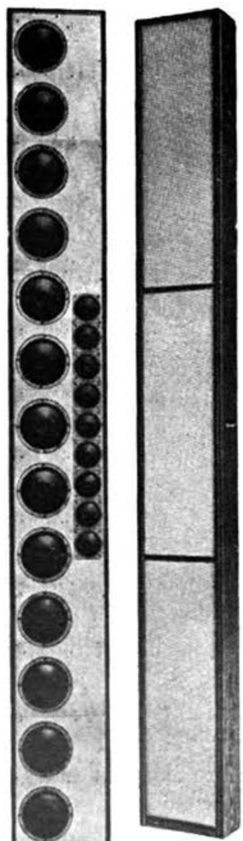


FIG. 5.22. *The Pamphonic line-source loudspeaker system, showing how the speaker units are mounted in a straight line one above the other.*

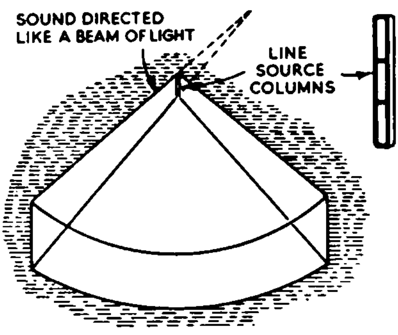


FIG. 5.23. *The sound is radiated from a line-source speaker system rather like the flat broad beam of light emitted from car fog lamps.*

a straight line is to secure a directional effect in the radiated sound rather like the flat, broad beam of light emitted from certain car fog lamps (see Fig. 5.23). Whilst the length of the column does not affect the horizontal distribution, which approximates 120 deg., it does affect the vertical distribution—the longer the column the smaller the vertical angle of distribution and the more concentrated the vertical distribution of the sound.

The greatest advantage of this concentrated distribution is that very little sound energy is wasted unnecessarily outside the range of the listeners. This leads not only to greater efficiency, but also avoids difficult reverberation problems which always arise in lofty buildings when fairly high-level sound is provided by single-source speaker systems. Service technicians having experience of public-address installations in swimming baths and churches will appreciate how the hard reflecting surfaces of such buildings evoke many indirect reflected sounds as well as the directly radiated sound. In extreme cases, speech is made almost unintelligible. Line-source speaker systems alleviate this trouble to a large degree owing to the reduced radiation of superfluous sound.

During tests with Pamphonic line-source equipment in St. Paul's Cathedral, the improved efficiency of line-source speakers was demonstrated by the fact that a power-input of only one watt provided sufficient sound from one 11-ft. line-source speaker standing beside the pulpit to cover the whole dome seating area of 9,000 sq. ft.

Another advantage of the system is that a listener situated close by a column is not disturbed by a very high sound level as he would be close by a high-level single source unit. Because the intensity of sound at a short distance from the column is less than the sum of the sound intensities of all the units, acoustical feedback between the microphone and line-source speaker represents less of a problem than when single-source units are adopted. The microphone can, in fact, be brought remarkably close to a line-source unit before feedback takes place, even when the amplifier's gain control is set for a normal level of reinforced sound.

For a large number of installations a single line-source unit is all that is required, for both indoor and outdoor use, and from the practical aspect

FIG. 5.24. *Two Pamphonic line-source speakers mounted in the rear of the dome of Rhodes House, Oxford, in connexion with the Duke of Edinburgh's Study Conference held there. The installation was carried out by Lowe & Oliver Ltd.*

this saves considerable time with regard to the wiring of the installation, as compared with the wiring of a large number of single-source units to provide the same coverage of sound.

Fig. 5.24 shows two Pamphonic line-source units mounted in the rear of the dome of Rhodes House (Oxford) during the course of the Duke of Edinburgh's Study Conference held there in 1956. Fig. 5.25 shows the neat installation of an outdoor unit, mounted on a specially-designed pole and stand.

GEC PERIPHONIC LOUDSPEAKER SYSTEM

The periphonic loudspeaker system evolved by the General Electric Co. is based on the GEC metal-cone speaker units in conjunction with a system of GEC presence units and a cabinet specially designed to eliminate all structural resonances, whilst also exploiting the enhanced bass response of the periphonic principle. In effect, there are two metal-cone units mounted in a V-shaped enclosure, the units being powered in such a way that the cones are driven in push-pull (from the mechanical aspect). This results in cancellation of any residual harmonic distortions produced in the cones themselves, and reduces the total second- and third-harmonic distortion to some 2·7 per cent at 40 c/s. This is remarkable when it is compared with the 40-per-cent distortion of this nature produced in a paper cone unit at the same frequency and power.

Four presence units are installed at the front and sides of the cabinet, which can be switched if required so as to alter the apparent nearness of the orchestra to individual listening requirements and conditions. The presence units are fed through cross-over networks.

SOUND DISTRIBUTION

Although up to the frequencies of about 1,000 c/s a loudspeaker functions as a spherical radiator and has a radiation angle of almost 180 deg., at high frequencies the whole area of the cone or diaphragm is unable to serve as a "piston" on the surrounding air. Radiation of the sound thus becomes confined to a narrow beam, whose diameter reduces as the frequency is raised. This is because the area of the cone responsible for the high-

FIG. 5.25. *Outdoor version of Pamphonic line-source loudspeaker unit.*

frequency radiation progressively diminishes towards the speech coil with increase in frequency.

Various devices are available for diffusing the high frequencies and thus preventing this beaming effect. A simple, though effective, method takes the form of directing the sound from the treble unit into the corner of the room, whence it is scattered into the listening area. It will be recalled that this idea is adopted in the case of the Pamphonic Victor loudspeaker system.

Another method which appears to be gaining popularity is the use of reflectors of various shapes positioned in front of the cone of the treble unit. The Burne-Jones tweeter unit incorporates this idea. The tweeter unit proper is housed in an attractively finished wood cylinder having three wide-spaced feet so that it can be unobtrusively positioned on top of the normal enclosure. The sound is emitted from the top of the cylinder by way of a horn-loading system of very small dimensions, and full omni-directional radiation in the horizontal plane is secured by a cone-shaped reflector, styled in the form of a mushroom, mounted on top of the cylinder. The unit carries a cross-over network and balance control, and has a frequency response from 2,000 c/s to 18,000 c/s.

Apart from a cone, other shapes can diffuse the high frequencies to any particular pattern, though it should be remembered that simple geometric reflection of sound occurs only when its wavelength is small compared with the dimensions of the reflector. The Lowther Type PM6 high-frequency unit has a diffuser actually installed at the centre of the cone.

Another method of evoking dispersion of the high-frequency beam is by means of a so-called acoustic lens mounted in front of the treble or middle-range unit. Theoretically, an acoustical lens serves to bend sound waves just as an optical lens bends light rays and a radio lens bends radio waves. Since we have a narrow sound beam to start with, however, and it is required to radiate this over a greater angle, the acoustical lens is arranged to be of the divergent type.

The lens is formed of an array of slant plates through which the sound has to pass. The distance between the plates and their angle of slant determine the frequency of operation of the lens and its comparable refractive index. Since the slant has the effect of extending the path of the incident wave in relation to the normal path of the wave, the waves undergo an effective change in velocity on passing through the lens, and on leaving thus tend to either converge or diverge from normal, depending upon how the lens is designed.

The high-frequency limit of the device is governed by the spacing of the plates in terms of half a wavelength, while the refractive index is given by the reciprocal of the cosine of the angle of slant.

Since it is necessary for the lens to receive as near an approximation of a plane wave as possible for correct operation, acoustical lenses are usually associated with horn-loading, which satisfies this condition. The Westrex high-frequency horn-loaded unit "Acoustilens" very successfully combines horn-loading and the acoustical lens principle.

ADJUSTMENTS TO SPEAKER SYSTEMS

Properly used, it is very rare these days for the loudspeaker unit to require detailed attention. However, should a definite fault develop, it is often best to return hi-fi units to the maker for reconditioning. This procedure is not usually necessary with the less expensive types, and with those units falling outside the accepted hi-fi definition.

Complete failure is an almost certain indication of an open speech-coil circuit. Before getting the cone and speech-coil assembly replaced, however, careful attention should be given to the flexible leads connecting the speech coil to the terminal block or tags for, apart from a definite burn-out caused by a severe overload, these are the most vulnerable trouble points. If the speech coil is definitely open-circuit, as can be determined by making a continuity check at the points of connexion of the flexible leads on the cone, either a replacement can be obtained from the manufacturer, complete with cone, and fitted in the workshop, or the unit can be sent to the maker or a firm specializing in speaker repairs.

Probably the most common of all speaker troubles is an out-of-centre speech coil, resulting in its fouling the pole pieces of the magnet. This trouble can easily be established by grasping the cone at diametrically opposite points and gently moving it in and out. If a scraping noise is heard when this action is performed, the centring screw or screws should be released a half a turn or so and the cone manipulated until the speech coil moves freely in and out of the gap without any scraping or rubbing. If necessary, feeler gauges can be inserted, and the centring screws carefully tightened without imposing too much pressure on the spider or centring disk.

Extreme caution must be taken to avoid particles of metal being attracted to the pole pieces. Speaker units are designed to prevent this happening, but if it has been necessary to break the dust seal to recentre the speech coil and particles of metal have been let in, the cone will almost certainly have to be taken from the chassis in order to clear the gap. A thin slip of modelling clay is useful for this purpose.

Excessive buzzing should lead to examination of the fixing of the cone to the chassis and the speech coil to the cone. A good-quality cement should be used to re-fasten these items, if necessary. A damaged cone must be replaced, even if only a temporary repair is made with adhesive plastic tape. Similar trouble may be caused by odd resonances of the enclosure or baffle

A check over the audible range with a variable-frequency oscillator soon reveals trouble of this nature, and steps can then be taken to avoid the resonance.

The inside walls of enclosures can have their natural resonances broken by glueing across their width stout pieces of wood at odd intervals. When this is done it is as well to lag the inside of the enclosure with acoustical damping material so that it actually covers the wood struts.

If the bass performance appears to be lacking when an amplifier known to be in good order is used with a certain loudspeaker system, there is a possibility that the enclosure is not providing optimum match to the speaker unit. If a bass reflex-type enclosure is employed, a measurement of the speaker's impedance curve will prove or disprove this.

If the upper resonance gives a response of greater amplitude than the lower resonance (see Fig. 5.16), the trouble may be caused by the vent being too small, thus evoking resonance of the enclosure at too low a frequency. This often results if a speaker unit having a relatively high main resonance is used in an enclosure which is designed for a speaker with a very low main resonance. Improved results can be secured by increasing the vent area, but it is better to use a speaker whose cone resonance matches the resonance of the enclosure.

If the balance is disturbed in the opposite way, and the response of the lower resonance is of greater amplitude than the upper, the trouble is likely to be caused by either too small an enclosure or too large a vent. In this case the area of the vent can easily be reduced until tests indicate a reasonable match. If the enclosure is much too small, however, the required reduction in area of the vent may promote too great a drop in bass output. This can be overcome either by extending the vent aperture inwards by means of a duct (Fig. 5.26*a*) or by inserting an acoustically lagged partition between the speaker and vent (Fig. 5.26*b*). The partition should extend about three-quarters of the way into the cabinet and should very snugly join the front and sides. When either of these procedures is adopted, the original vent area can remain, and in consequence there is considerably less loss in bass input.

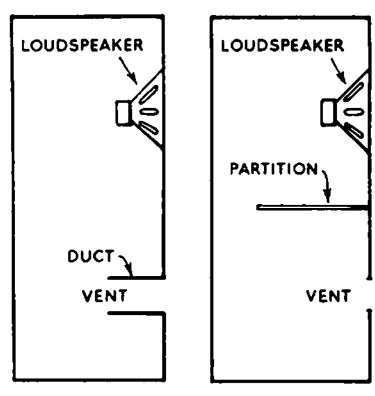


FIG. 5.26. *Methods of reducing the enclosure resonance;* (a) *by means of a duct:* (b) *by fitting a partition.*

SPEAKER PLACING

The performance of a speaker system can be greatly affected by its position in the listening room, while the room furnishings are also a governing factor on how the reproduction "sounds" to a listener. A room without heavy furnishings, such as armchairs and carpets, is prone to be very "live"; little sound is absorbed and reflections occur between the walls and hard objects. There is a rising top response and a possibility of "ringing" at certain frequencies. Careful adjustments to the tone controls can often effect a compromise under these conditions, but such adjustments should be made in relation to the placing of the loudspeaker.

Standing waves, which give rise to undesirable acoustics, can be modified in amplitude and frequency simply by moving the speaker system from one point to another. In a room which is subject to standing waves it is often desirable to situate the speaker in one corner, as distinct from having it in the centre of a wall. However, the best position can be found only on a trial-and-error basis, and it is almost useless to try to formulate any hard-and-fast rules about this. The speaker should be situated where it sounds best and not where it looks best, but, unfortunately, this is not always acceptable to the lady of the house!

CHAPTER 6

Disk Recording

DESPITE many recent developments in magnetic tape recording, the ordinary gramophone record remains at present the most popular medium for the storing of musical programme material. Whilst there are available a few instruments designed essentially for the playing of tape records, as distinct from tape recorders, their number falls very far short of the millions of disk-record reproducers of various types that are in current use.

The advent of the stereophonic technique has not altered this general trend, as was expected in some quarters when stereophonic tape records first made their appearance, and such records now have to compete with stereophonic disk records. Although magnetic tape is extensively used as a hi-fi programme medium, it is principally associated with tape recorders, with which a private library of musical works can be built up by way of a V.H.F. radio channel.

To facilitate editing, save expense and avoid undue retakes, disk records usually begin as tape records. After a work has been recorded, and both the artist and technician are satisfied with regard to their own particular interests in the material, it is transferred from the tape to a wax or lacquer disk. From this original is made a master disk by a process of copper plating. The ridge of the master represents the modulated groove of the original, so the master can be used as a direct working matrix for producing the final pressings. Usually, however—particularly where a large number of pressings is required—there are two more stages in the process, resulting in two more intermediate disks, a positive "mother" and a negative "stamper". In this case the stamper is used as the actual working matrix.

Most commercial 78 r.p.m. records are produced basically of shellac, while their microgroove counterparts of the so-called unbreakable type are made of a soft plastic called vinylite.

The basic theory of disk recording has already been briefly discussed at the end of Chapter 1, and enlargement on the subject is hardly warranted in a book of this nature. However, there are one or two rather important points about the care of records which should be mentioned.

Records are rapidly ruined by the use of an incorrect stylus, a too-heavy pick-up, by dust and binding of the tone arm on its bearing, and, of course, by rough handling. Service technicians should bear these points in mind when using a customer's records for test purposes. Certain hi-fi enthusiasts lavish as much care on their records as does a mother on her new baby, and the technician should always handle records with care.

Microgroove records require a stylus of dimensions between 0·0008 in. and 0·001 in. (one thou' or 25 microns), while a tip radius of between 0·0025 in. and 0·003 in. is satisfactory for 78 r.p.m. records. If a microgroove stylus is used on a 78 r.p.m. record the tip will skate along the base of the groove and give rise to a very high noise level, besides ruining the stylus if not the record. A 78 r.p.m. stylus used on a microgroove record will have great difficulty in remaining in the groove; it may intermittently ride on the "horns" between the junction of the top of the groove wall and the "land" between the groove. There will be a considerable loss of high-frequency response. Indeed, if persistent use is made of the incorrect stylus, this response will disappear from the record permanently.

Dust is one of the worst enemies of disk records. Simply wiping the record with a dry cloth does little to alleviate the problem, in fact it aggravates it by imparting a high static charge to the disk which makes it act as a magnet for dust particles. There are several devices on the market for the elimination of dust, and fluids with which the records can be polished whilst also being rendered temporarily anti-static. G. A. Briggs recommends a detergent such as Stergene or Quix, diluted by adding 95 per cent distilled water. He states that a medicine bottle of the liquid can be filled at a cost of less than sixpence, and that in the dustiest of districts this should last a year.

Unless the tone arm is perfectly free to rotate in its bearing the groove wall will be subjected to extreme pressure, the impressed modulation pattern will become distorted and in bad cases the "land" between the groove may collapse. There is usually an adjustment on the tone arm in relation to the bearing which can be set to provide adequate freedom of movement whilst avoiding undue up-and-down play. A spot of very light machine oil on the bearing often helps.

The downward pressure of the stylus on the record is much of a compromise and depends upon the type of pick-up and record: 78 r.p.m. records usually require a greater downward pressure than microgroove records in order to hold the stylus in the groove. This applies particularly to older pick-ups. The downward pressure of modern pick-ups may be as low as 4 grams. This has been brought about by progressive attention to such points as the total compliance and total effective mass in relation to the stylus tip. Whilst a reduction in the downward pressure, or playing weight as it is sometimes called, will increase the life of the record and stylus, it is never a

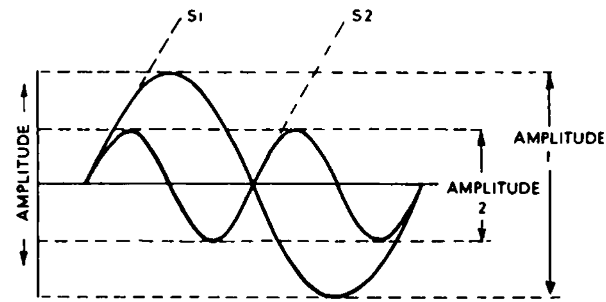


FIG. 6.1. *Diagram of constant-velocity recording for two frequencies.*

good idea to reduce the playing weight below that stipulated by the maker as this may result in "groove jumping", which may be more detrimental to the record and stylus than a gram or two of extra weight.

RECORDING CHARACTERISTICS

The recording head transduces the modulation pattern of the a.f. signal at the output of an amplifier into lateral movement of the recording stylus. In this way the pattern corresponding to the modulation is impressed upon the groove as it is being cut during the recording process. The actual lateral oscillations (low-frequency) of the recording stylus can readily be felt by lightly placing a finger on the stylus while the amplifier is receiving a programme signal.

The two basic factors associated with the lateral oscillation of the stylus are amplitude and velocity. If the velocity of the stylus is to be maintained constant over the whole of the audible frequency range, as is usually required, it is clear that the amplitude of the stylus will increase with decrease in frequency. At high frequencies the amplitude will be very small and at low frequencies it will be very large. This simple rule can be expressed in terms of velocity as $2\pi fA$, where f is the frequency in c/s and A is the peak amplitude. Thus, in order to maintain the velocity at a constant value, A increases as f reduces.

The idea is not clearly understood by all service technicians, and since it is rather important in hi-fi work, Fig. 6.1 is given to illustrate it better. Waves A and B represent the modulation pattern imparted upon the groove of a record owing to the lateral oscillation of the stylus. The velocity of the stylus is represented by the slopes $S1$ and $S2$ of the waves. Wave B has twice the frequency of wave A, but in order for the slopes to remain equal (representing constant velocity) wave B is half the amplitude of wave A. For a constant stylus velocity, this means that the amplitude will increase by 2:1 for every 2:1 decrease in frequency; or, expressed more technically, the amplitude will decrease 6 db per octave.

Fig. 6.2 shows the effect for constant amplitude. As before, wave B is

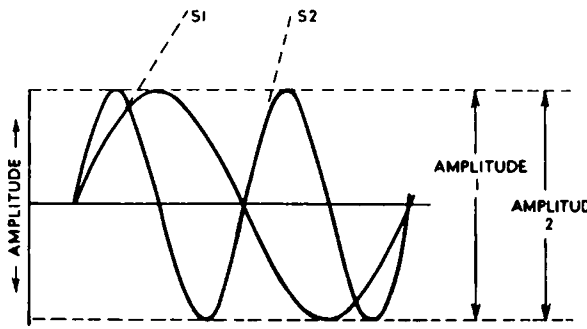


FIG. 6.2. *Diagram of constant-amplitude recording for two frequencies.*

twice the frequency of wave *A*, but since the amplitudes are constant and equal, the slope (*S*2) of wave *B* is steeper than the slope (*S*1) of wave *A*—indicating that the velocity of the stylus required to impart the higher-frequency wave is greater than that required to impart the lower-frequency wave. For constant amplitude, the velocity, in fact, increases at the rate of 6 db per octave.

Since the amplitude of the recording stylus would be excessive at low frequencies, and result in break-through from one groove to the next, if the constant-velocity principle were applied to disk recording, the constant-amplitude idea is adopted for the low frequencies and the constant-velocity idea for the higher frequencies. This works well in practice, since constant-amplitude recording would never do for the high frequencies owing to distortion which would result from the excessive velocity of the stylus tip. Pick-ups would never track and the modulation pattern would soon collapse, even if it were possible to impress it at high velocity on the groove.

The point at which the constant-amplitude recording changes over to constant-velocity recording is known as the "turnover" or "crossover". The response of the recording head is usually equalized in such a way as to provide constant-amplitude recording up to the crossover point, which is positioned somewhere in the region of 500 c/s. The change-over from constant amplitude to constant velocity is not "sharp", but occurs gradually as governed by the equalizing network. A representative curve is given in Fig. 6.3, which is typical of a 78 r.p.m. recording characteristic.

If a sliding-frequency record is cut to this characteristic, and is played back with a moving-coil or moving-iron type pick-up, whose output voltage is equal to the velocity, the output voltage from the pick-up will follow the curve very closely. Thus, in order to maintain a constant output at the lower frequencies, the pick-up circuit will have to be equalized to provide a bass lift, and the equalization curve must be the inverse of that at Fig. 6.3. A crystal-type pick-up will not require the same degree of equalization, since its output is proportional to displacement and not to velocity, but more will be said about this later.

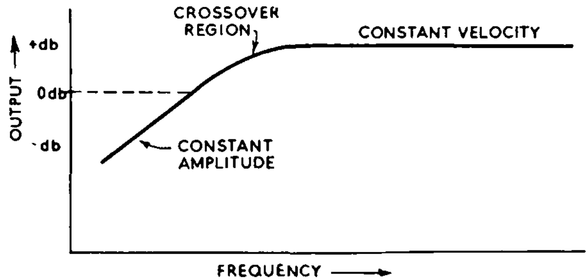


FIG. 6.3. *Typical recording characteristic of 78-r.p.m. record.*

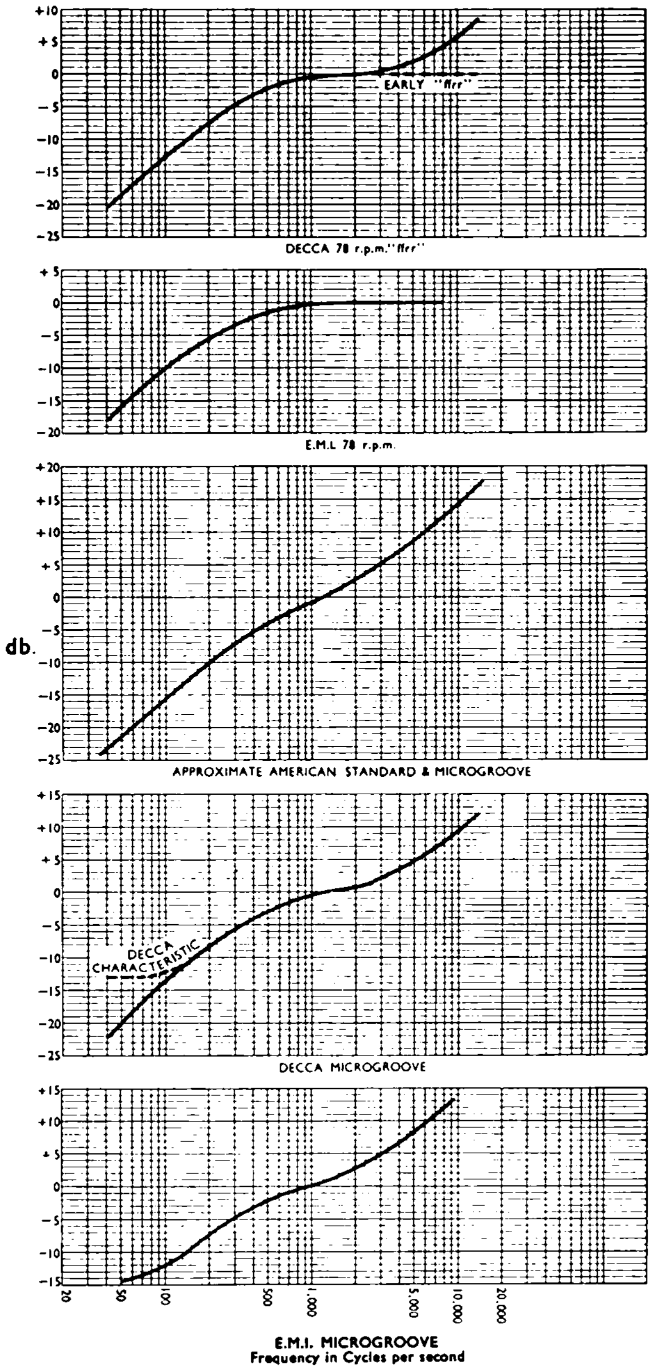
The recorded level is usually given in terms of velocity (r.m.s.) at 1,000 c/s or in decibels relative to 1 cm/sec (zero db). Maximum recorded level may lie between 15 db and 26 db, depending upon the type of record and recording characteristics. In accordance with this practice, the output voltages of pick-ups are given in terms of mV per cm/sec.

Since the microgroove record is recorded at a lower level than the 78 r.p.m. record, the noise actually generated through the playback stylus tracking in the groove assumes proportions approaching the modulation level of soft passages of music. This disconcerting background noise (usually referred to as "record hiss") is reduced by the vinyl-base material of the record itself, and additionally by the application of a progressive boost to the higher frequencies of the recording signal.

This recorded emphasis of the higher frequencies is of no consequence from the "quality" aspect, since on playback a de-emphasis network can be used to linearize the response. However, it has some bearing on record hiss because the noise frequencies which are most troublesome are also reduced considerably in level by the de-emphasis or equalizing network. A similar idea is adopted in f.m. receivers and adaptors (see the author's "F.M. Radio Servicing Handbook").

This noise-reducing arrangement has led to a large number of recording characteristics, each requiring its own particular de-emphasis curve to provide the correct degree of equalizing. Modern hi-fi control units have three or four record-equalizing positions on the selector switch, and whilst these do not cater completely for the many recording characteristics, they do permit a fairly close compromise between characteristics which have much in common. Slight deviations can be compensated for by the use of the tone controls.

A formidable array of recording characteristics which have been adopted over the years by leading gramophone record manufacturers is given in Fig. 6.4. Whilst these curves are not necessarily identical with those published by the record manufacturers, they do represent close approximations which, when correctly equalized, have been found to give the best quality on a perfectly linear amplifier. It will be seen that, apart from the E.M.I. 78 r.p.m


EARLY "ffrr"
DECCA 78 r.p.m. "ffrr"
E.M.I. 78 r.p.m.
db.
APPROXIMATE AMERICAN STANDARD & MICROGROOVE
DECCA CHARACTERISTIC
DECCA MICROGROOVE
20
50
100
500
1,000
5,000
10,000
20,000
E.M.I. MICROGROOVE
Frequency in Cycles per second

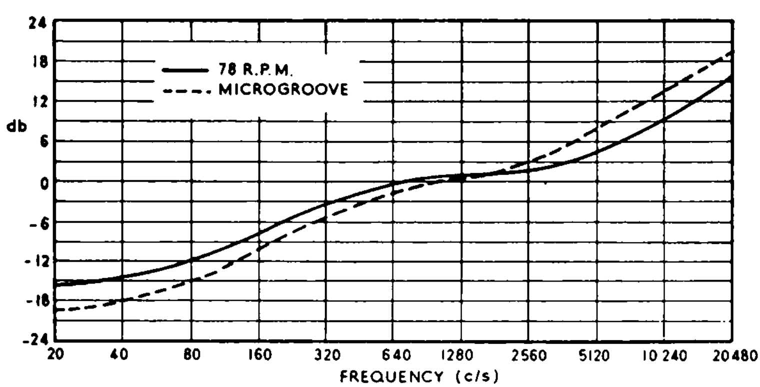
characteristic, and the early Decca "ffrr", the curves indicate the use of varying degrees of high-frequency emphasis; this being made possible in later years by improvements in the design of pick-ups. The American curve is one of almost constant amplitude, apart from the slight fall-off in slope in the region of 1,000 c/s.

Fortunately, there now appears to be a move to standardize recording characteristics throughout the world. In 1955 a curve sponsored by the Radio Industries Association of America (R.I.A.A.) was accepted by the major recording companies, and in this country was embodied in British Standard 1928:1955. This curve is given in Fig. 6.5 (see also Fig. 2.6 for the equalization curve). It is to be hoped that there will be an early move to standardize stereophonic records before the situation there gets out of hand.

DISK RECORDING PROBLEMS

There are one or two further points with regard to disk recording which concern the service technician. As the recording stylus cuts a modulated groove, the speed of the record at the point of contact with the stylus decreases linearly towards the inner diameter of the disk. With a 78 r.p.m. 12-inch disk, the speed at which the groove is cut falls progressively from about 47 in. per second to 17 in. per second. This means that towards the inner diameter the modulation waves are cut at a far steeper angle than are the waves of the same frequency towards the outer diameter.

Owing to this, there is a progressive falling-off in the high-frequency response towards the centre of the record. With a 78 r.p.m. 12-inch record, the actual wavelength of the impressed modulation at 10,000 c/s drops from about 0·0047 in. to 0·0017 in. from the outer to the inner diameters of the



(Opposite page) FIG. 6.4. *Recording characteristics approximating those used by leading record manufacturers.* (Above) FIG. 6.5. *Recording characteristic equal to B.S.* 1928: 1955.

record. There is little that can be done on playback to alleviate this falling response, apart from ensuring that the stylus is in good condition. The smaller diameter of the tip of a microgroove stylus greatly reduces this trouble on the overall slower speed of microgroove records, but reducing the diameter of the tip on standard 78 r.p.m. records would not likewise enhance the response.

During the recording process the higher frequencies are sometimes given a boost, in addition to normal pre-emphasis, which increases progressively as the recording stylus approaches the innermost groove. This idea, known as "radius compensation", is used by home recordists as well as by certain commercial recording organizations.

An optical method, devised by Buchmann and Meyer, of determining the level of recording on a disk over the whole of the audio spectrum is shown in Fig. 6.6. Various spot frequencies spaced over the spectrum are recorded at intervals throughout the record, and by manipulating the record in relation to a light source the bands of modulation pattern can be clearly observed. Since the breadth of the bands corresponds to the velocity of the recordings, such a record produced from a constant-input signal will readily reveal the response deficiencies of the recording amplifier and head. The record is also of value for checking the frequency response of pick-ups. The photograph of the Buchmann and Meyer pattern shows that the recording response is maintained reasonably constant at the higher frequencies, and also shows the constant amplitude (falling velocity) characteristic at the lower frequencies, towards the inner diameter of the record.

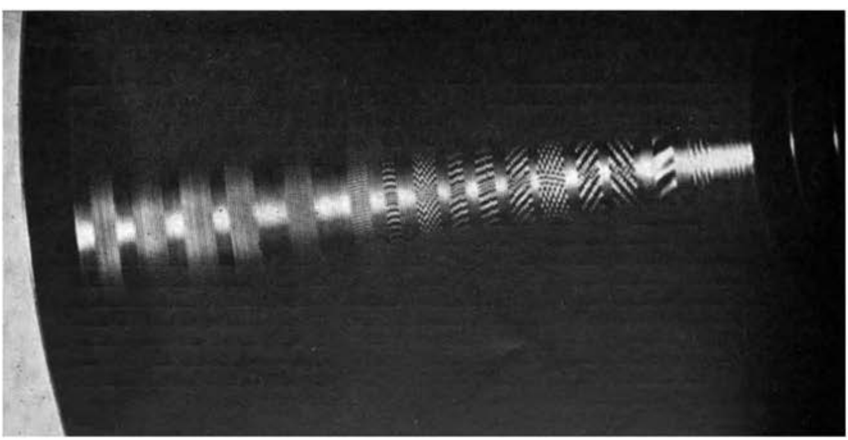


FIG. 6.6. *Showing how a source of light reflected from a recording of spot frequencies over the audio spectrum indicates the recorded level in terms of breadth of the bands. (The Buchmann and Meyer test.)*

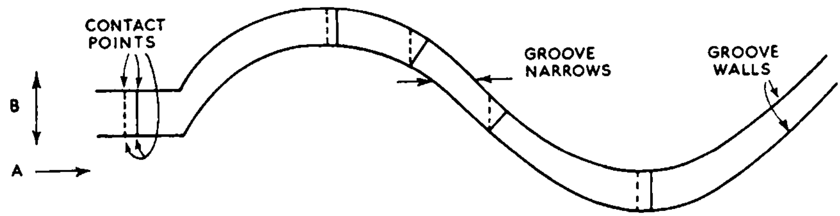


FIG. 6.7. *Tracing distortion is caused because the line (full lines on diagram) between the points of contact of the reproducing stylus progressively deviates from the line (dotted lines on diagram) between the points of contact made by the recording stylus on the sloping parts of the wave. The lines coincide only at the peaks of the wave and when the groove is unmodulated. This results from the fact that a spherical-tipped stylus is used to reproduce a waveform imparted by a chisel-edged recording stylus. The pinch effect is the narrowing of the groove on the sloping parts of the wave, causing a vertical movement at twice the recorded frequency of the reproducing stylus.*

In order to reduce the high-frequency loss during the recording process as the result of the increasing impedance offered to the recording stylus, the "hot stylus" technique is sometimes adopted. A few turns of resistance wire are wound around the recording stylus, and a controlled current is passed through the coil, thus raising the temperature of the stylus sufficiently to soften the cellulose lacquer. This reduces the impedance presented to the oscillating stylus whilst recording, and not only enhances the "top" response, but also reduces surface noise on playback.

TRACING DISTORTION AND THE PINCH EFFECT

Inherent in disk recording are other distortions which, although not of direct concern to the hi-fi service technician and enthusiast, at least warrant consideration. Both tracing distortion—not to be confused with "tracking distortion", which is a direct function of replay—and the pinch effect arise from the fact that the reproducing stylus has a spherical tip while the recording stylus has a chisel edge. The facets of the recording stylus, which is usually of sapphire, are set to produce a low-noise polished groove, of depth between 0·0015 and 0·0025 in.

Fig. 6.7 shows a waveform of modulation which may be impressed upon a groove of a record. The cut is in the direction of arrow *A*, while the amplitude characteristics are imparted by reason of the lateral movement of the stylus as at *B*. The cutting face of the stylus is always maintained in the same plane, at right angles to the direction of the cut, irrespective of the modulation pattern. This is shown by the dotted lines across the groove. On playback, however, the reproducing stylus maintains contact at the sides of the groove at points directly opposite, thus giving the full lines shown on the diagram which are always at right angles to the direction of the groove.

It will thus be seen that the points of contact of the recording stylus on the sides of the groove coincide with those of the reproducing stylus only at the peaks of the wave and when the groove is unmodulated. On the sloping parts of the wave, the line between the points of contact of the reproducing stylus progressively deviates from the line between the points of contact made by the recording stylus. Now, since the lateral movement of the recording stylus was in the direction of the dotted lines and the output signal of the pick-up is promoted by the movement of the reproducing stylus in the direction of the full lines. it will be apparent that the reproduced waveform differs from the waveform of the signal applied to the recording head.

Briefly, harmonic and intermodulation distortions are introduced by tracing distortion because a spherical-tipped stylus is used to reproduce a waveform imparted by a chisel-edged recording stylus, which results in the curve traced by the centre of the tip of the reproducing stylus not being an exact replica of the modulated groove. It will be evident that this kind of distortion increases with amplitude and frequency of the recorded wave, and also towards the inner diameter of the disk, since in all these cases there will be an increase in the slope of the waveforms recorded.

The "pinch effect" is also caused by the difference between the shapes of the recording and reproducing styli. As is shown in Fig. 6.7, the groove is equal to the full width of the recording stylus only when it is unmodulated and at the peaks of the modulation. At other times during modulation, the cutting edge is at an angle to the direction of the groove, and at these times the groove width decreases. Modulation of a simple sine waveform will, therefore, cause the reproducing stylus to oscillate vertically at twice the frequency of the modulation. This is because the reproducing point is truly spherical and thus rides up the groove when it narrows. The same effect occurs in a much more complex way when ordinary sounds and music are recorded.

Whilst a perfect pick-up would not give an output from a vertical movement, there is a small unwanted signal generated from this cause in most pick-ups, and because of its method of generation it is usually rich in second harmonics. Modern pick-ups which adopt the cantilever-stylus principle are less prone to this trouble than their older counterparts.

Owing to the reduced width of the groove of microgroove records and the reduced diameter of the tip of the reproducing stylus, these records show a distinct improvement in terms of distortion over the standard 78 r.p.m. record. Nevertheless, as the innermost grooves are approached the distortion rises rapidly, and at 5,000 c/s it may be as high as 20 per cent at a recorded diameter of 4 in. This is where the microgroove record scores again, since the minimum recorded diameter is $4\frac{3}{4}$ in., compared with the $3\frac{3}{4}$ in. for standard 78 r.p.m. records. Home recordists should never be tempted to cut too far towards the centre of a disk if the quality of reproduction is to be maintained.

CHAPTER 7

Pick-ups and Record Playing Equipment

THE duty of the pick-up is to change the lateral vibrations of the reproducing stylus, as it traces the modulated groove, into an electrical equivalent of the modulation pattern which was imparted during the recording process. It is the aim of the designer of pick-ups to produce a unit whose output signal is an exact replica of the signal which was applied to the recording head when the record was made. Since there are so many mechanical factors involved between the change-over from the electrical to the mechanical at the recording end and the change-over from the mechanical to the electrical at the reproducing end, the designer is presented with a great number of complex problems which, from the hi-fi aspect, are aggravated by the extra pre-emphasis and consequent increased velocity of the reproducing stylus.

Pick-ups can be considered as small generators of electricity. There are two basic types, those whose output voltage is proportional to the *velocity* of the movement of the stylus, and those whose output is proportional to the *force* applied to the stylus. The former adopt the electromagnetic principle for the generation of their output voltage and may be likened to the bicycle dynamo whose output voltage is governed by the speed (velocity) of the driving wheel. The latter type rely upon the property of Rochelle salt crystals of producing a potential difference when subjected to changes of pressure.

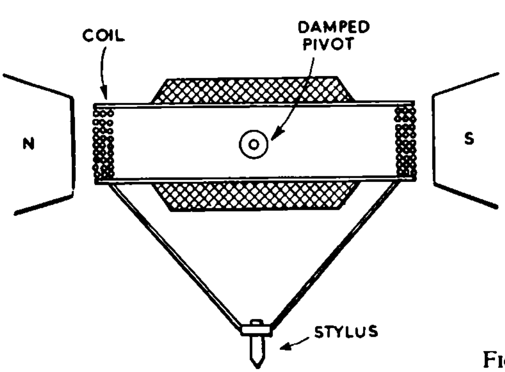


FIG. 7.1. *Moving-coil pick-up unit.*

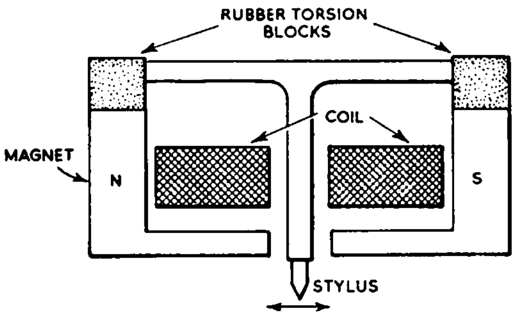


FIG. 7.2. *Moving-armature pick-up unit.*

Electromagnetic pick-ups employ a coil or conductor together with a small permanent magnet; a voltage is set up in the coil or conductor as a result of the variations of magnetic flux caused by the side-to-side movement of the stylus. The stylus is coupled to one or other of the two elements either directly or magnetically.

With the moving-coil pick-up (Fig. 7.1), the coil is directly coupled to the stylus, and the same applies to the ribbon pick-up, which is really a moving-coil unit having a single-turn coil. With the moving-armature pick-up (Fig. 7.2), the stylus is coupled directly to the armature which vibrates in the gap of a magnet supplying the steady field, and the coil is wound over the magnetic circuit. There are several variants of this type of unit, from the older moving-iron and needle armature units to the tiny micro-armature types and the so-called variable-reluctance pick-ups, the latter probably being the most popular of the electromagnetic range.

Whilst all electromagnetic pick-ups, with the exception of the moving-coil unit, are essentially variable-reluctance types, it is the pick-up illustrated in Fig. 7.3 which is usually known by this designation. Its operation is quite straightforward; there are two coils wound over the magnetic circuit, which is provided by the small cylindrical permanent magnet, the pole pieces and the ferrous (magnetic material) stylus arm. The stylus arm thus vibrates within the gap between the pole pieces, which causes the flux alternately to increase in one pole and decrease in the other, thus resulting in increasing and decreasing e.m.f's in the associated coils, which are phased so that the voltages are series-aiding.

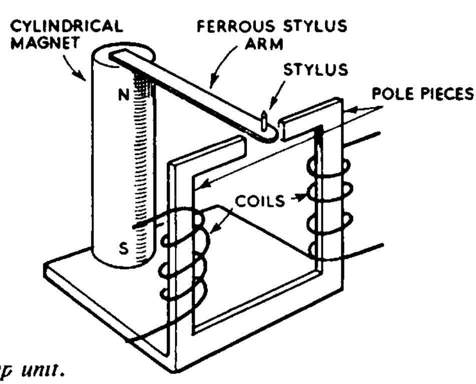


FIG. 7.3. *Variable-reluctance pick-up unit.*

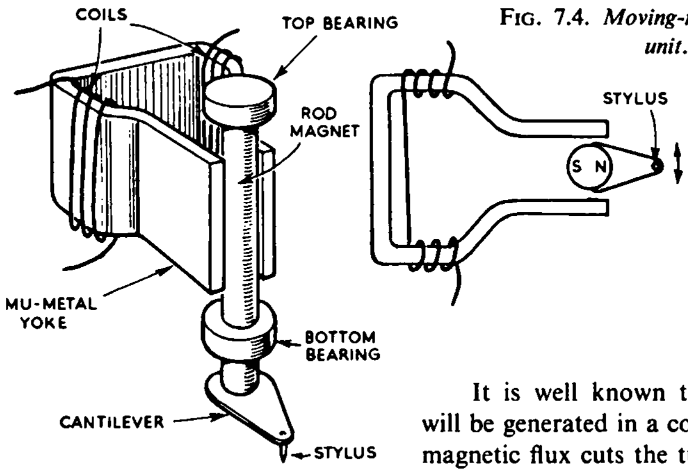


FIG. 7.4. *Moving-magnet pick-up unit.*

It is well known that a voltage will be generated in a coil if a varying magnetic flux cuts the turns either by the coil moving in the flux or by the movement of a part of the magnetic circuit. This follows the basic law of electromagnetic induction which was first postulated by Faraday.

Exactly the same effect will be achieved, and a voltage will be developed in an associated coil, if the magnet moves while the other element remains stationary. This principle, which is used extensively in some bicycle dynamos, is adopted in the Philips Novosonic pick-up. The magnet consists of a thin ferrite rod which is magnetized across its diameter, and to which is coupled a cantilever stylus. The magnet thus undergoes a slight twisting movement, due to the lateral movement of the stylus, within the gap of the mumetal yoke. As with the variable-reluctance unit, the flux will alternately increase in one side and decrease in the other, resulting in increasing and decreasing e.m.f's in the associated coils. The idea is shown in Fig. 7.4.

The Rochelle salt crystal used in crystal pick-ups is cut from a big crystal at a critical angle and placed between two metallic plates so arranged that the pressure on the crystal is varied in sympathy with the vibrations of the stylus. The crystal element is very fragile and easily broken or damaged by excessive mechanical pressure on the stylus and by high temperature. Extreme care should, therefore, be taken when replacing the stylus in one of these units. Instead of Rochelle salt crystal, poly-crystalline barium titanate has been used in some recent units. This material is less affected by temperature and humidity, but unfortunately does not lend to the extended frequency range of the Rochelle salt crystal desirable in pick-ups.

PICK-UP MECHANICS

Modern recording technique, resulting in extended high-frequency response, pre-emphasis as a noise-reducing agent, and the long-play microgroove record, has introduced records with very sharp changes in modulation pattern. Thus, in order to reproduce the modulation impressed upon the grooves of these records with true fidelity and without causing damage to either the pick-up stylus or record grooves, a pick-up whose stylus is able to respond immediately to the rapid accelerations and decelerations is essential.

At high frequencies and recorded levels the accelerations encountered by the stylus may approach some thousand gravities (1,000*g*). Acceleration involves a variation of either velocity or direction, or both, and the pick-up stylus is subjected to both of these factors when it is tracing the modulated groove of a record. The stylus and associated linkage also possess mass (which is the quantity of the matter contained in the items referred to). As the result of the accelerations of the mass, the stylus is subjected to a force, as we have seen. The magnitude of this force is compared with the force due to the pull of gravity—this being denoted by the letter *g*. Thus, the force acting on a stylus when it is under the control of a heavily-recorded sound of high frequency may rise to some 1,000 times the pull of gravity—hence the term 1,000*g* given above.

In order to prevent the record groove from being torn to pieces, the pick-up stylus must be free to move very easily and rapidly from side to side. If the pick-up is light and the stylus stiff, the stylus will be unable to respond to sudden large changes in modulation and it will ride up on the groove wall and skate across the record. This is what happens when an old-type pick-up is used on a modern record.

As a means of reducing the mass of the moving parts of a pick-up and ensuring that the effective inertia at the high acceleration rates encountered on modern records is kept as low as possible, very lightweight stylus mounting materials are essential; micro-armatures are often made hollow for this purpose. The modern pick-up rates the mass of the moving parts in milligrams, which is a considerable achievement when it is considered that only a few years ago these parts were rated in ounces, and remembering also the much greater stress imposed upon these parts due to the high recorded velocities of modern records. In this respect, it is interesting to note that the effective mass at the stylus tip of the Goldring variable-reluctance cartridge Type 500 is only 3·5 milligrams!

Another factor affecting the free movement of the stylus is the damping arrangement needed in the pick-up to return the stylus and armature to the point of balance in the centre of its lateral movement. In the pre-war heavy type of pick-up in which a needle chuck was employed the iron armature was clamped in the centre of its movement by hard rubber pads, which invariably perished after a short time. In those days, considerable force was required to promote a lateral displacement of the needle. There was little trouble in tracing the grooves of the old-type records though, since the vertical force on the needle due to the great weight of the pick-up (two or three ounces) left no alternative; the recorded accelerations were also far less than they are today. Nevertheless, neither the needles nor the records lasted very long, and after several playings by this "brute force" method the quality on playback deteriorated sadly. Today, pick-ups track admirably at a pressure of seven to ten grams.

The tracking pressure required is related to the lateral stiffness of the stylus, of lateral compliance as it is known, compliance being the reciprocal of stiffness and rated in centimetre/dynes. Thus, if the compliance is low the movement is stiff and a greater vertical force is required to hold the stylus in the groove. The compliance of hi-fi pick-ups is rarely less than $3{\cdot}5 \times 10^{-6}$ cm/dyne.

In addition to the lateral compliance, a vertical compliance is necessary to combat the pinch effect which was described in the previous chapter. If the pick-up lacked vertical compliance, the whole of the pick-up and the arm would have to follow the vertical movements as the result of the pinch effect. The inertia of such a large mass would cause the stylus to plough along the groove at high modulation levels and also evoke groove jumping. In such cases the vertical movement of the stylus may be as much as one-tenth of the lateral movement, and for this reason the vertical compliance is generally set in the region of 5×10^{-7} cm/dyne, but it is not always given in the data supplied by the makers.

The cantilever-type stylus, which is formed of a straight or trailing shank on which the sapphire point is mounted, provides by virtue of its design a degree of vertical compliance, which usually satisfies the above condition.

Pick-ups suffer from resonances just as much as other items of audio equipment. The high-frequency resonance is a function of the mass of the moving parts; the smaller the mass the higher the resonant frequency. With the small mass demanded of hi-fi pick-ups the upper resonance usually falls outside the audible range, around 17–18 kc/s. The low-frequency resonance, on the other hand, is usually governed by the tone arm and characteristics of the pick-up head and mounting arrangements. An average arm gives a low-frequency resonance in the region of 20 c/s, which is of little moment since the response of the amplifier is pretty well limited at this frequency to avoid motor rumble and low-frequency overloading troubles. With regard to the pick-up proper, a high compliance in the lateral sense damps low-frequency resonances and keeps them well down the low end of the scale, out of harm's way.

STYLI

There are only two types of stylus used these days, namely, sapphire and diamond. Pick-ups which use metal or fibre needles and needle chucks are useless for hi-fi work and should be discarded. There are dozens of different kinds of styli for use in the many patterns and makes of pick-ups produced over the last few years. It is impossible to go into detail, and in any case there would be little purpose served in so doing. Pick-ups designed for serious hi-fi work usually have available styli in sapphire or diamond. From the actual playing point of view there is no difference between the two, but from the aspect of longevity there is considerable difference.

The rate of wear of any stylus will be governed by a number of factors, including tracking weight and vertical and lateral compliance of the pick-up, the condition of the records, the amount of dust in their grooves and how the equipment is handled. Generally speaking, the life of a diamond stylus is some 20 times that of a sapphire, and under ideal conditions it is often possible to get some 60 hours of playing time from a sapphire. The diamond is much more expensive than the sapphire, however.

The great advantage of the diamond is that it avoids that gradual deterioration in quality which is characteristic of a sapphire as it slides down its life/efficiency curve. A point is reached where replacement becomes necessary from the quality point of view, but not from the aspect of economy. As a stylus nears the end of its useful life so the rate of wear of the records increases. The quality of the reproduction also suffers considerably not only by loss of the higher frequencies, but also by the introduction of harmonic and intermodulation distortion as the result of the asymmetrical tracing motion of the worn stylus.

Unless a microscope is available to assess the wear of the tip of the stylus, it is good policy to replace the stylus regularly as governed by the type of pick-up employed, and by the maker's recommendations in this respect. Some dealers make use of the Philips "Needle Clinic" microscope, and such an instrument is well worth acquiring if considerable hi-fi work is contemplated.

STYLUS REPLACEMENT

Extreme care should be observed during the operation of replacing a worn stylus. It is often necessary first to remove the cartridge or head from the pick-up arm. With the Acos SA and SB series, the stylus holder should be lightly gripped with a pair of long-nose pliers or tweezers, while the worn stylus is extracted from the holder with a second pair of pliers or tweezers arranged to lever against the first pair. If undue pressure is applied to the stylus holder, or if this item is not held rigidly without movement while the stylus is being extracted, there is a strong possibility that the internal assembly will be damaged, necessitating replacement of the cartridge as well as the stylus.

Generally speaking, the cantilever-type of stylus can be withdrawn with little difficulty in the way described above, and on replacing care should be taken to ensure that the shank section of the new stylus is pushed well home into the holder or "pocket".

The Collaro Studio crystal cartridges use a small machine screw to secure the cantilever-type stylus to the body of the cartridge. Near the jewel end of the cantilever the generating system is mechanically coupled by means of a small pad resting on the cantilever arm. The styli for the types "O" and "P" cartridges have phosphor-bronze shafts, while aluminium shafts are used on the styli for the transcription cartridge, the long-play type having a series of holes drilled along its length.

These various styli have an effect on the overall frequency response of the pick-up, whilst also affecting the output voltage. They are, in effect, tuned to fit in with the various responses of the pick-up itself. On no account should the shape of the shank or cantilever be altered. If, after replacing a cantilever-type stylus, it appears that the tracing angle of the jewel is incorrect, the trouble will invariably be caused by incorrect fitment and not by distortion of the shape.

Styli are usually colour-coded, red indicating the microgroove variety, and green the 78 r.p.m. type. They should never be interchanged, since the tip radius for microgroove is 0·001 in. and for 78 r.p.m. 0·0025 in.

The cantilever-type stylus, by virtue of its operation as two simultaneous levers operating on the generating coupling, evokes four rather complex resonances. Additional resonances are produced by the formation of the cantilever section proper, as the result of twisting and torsional effects. However, these are smoothed out to provide the required response of the modern hi-fi unit.

There are a number of single pick-up units which can be used to play either 78 r.p.m. or microgroove records, the cartridge having two styli, one for 78 r.p.m. and the other for microgroove, mechanically coupled to either a single generating system or independent generating systems. These are usually called "turnover units" and operate quite successfully. Before this idea was adopted it was necessary to change the head on changing from 78 r.p.m. records to microgroove records.

Another arrangement along these lines is the so-called "turnaround" cartridge. Here the composite cantilever-type stylus is pivoted in its centre so that it can be rotated through 180 deg. to bring the required stylus into mechanical coupling with the common generating system. This is an admirable idea, since the requisite damping can easily be given to the styli to maintain optimum results on both types of record.

One problem associated with pick-ups is "needle talk". This is the acoustic rattle that vaguely resembles the modulation on the record which is emitted from the mechanical function of the pick-up itself. This used to be extremely troublesome in the days when heavy pick-ups were used. In fact, the noise used to be so great that it was necessary to use acoustically-treated lids on radiograms to prevent the noise affecting the reproduction from the loudspeaker. (Hence the notice "Please close the lid when playing".)

The actual sound radiated from the pick-up and the record (the record contributes towards it) was caused by the rise and fall of the complete pick-up assembly and tone arm due to the pinch effect; the sound was, in fact, second harmonic of that recorded. The trouble these days has been eased by the smaller masses and dimensions of pick-ups and associated parts. Nevertheless, it is still present to some degree and is evoked by the mechanical backlash of the stylus on the record, as the stylus rises and falls at twice the recorded frequency due to the pinch effect. The record in this case serves as a sounding board.

OUTPUT FROM PICK-UPS

The output from pick-ups in terms of voltage is a function of their sensitivity, and is measured in terms of r.m.s. volts for a certain stylus velocity. As an example, the Goldring variable-reluctance cartridge Type 500 is said to have an average output of 3·2 mV per cm/sec. This means that for each cm/sec of stylus velocity the output is 3·2 mV. The recorded velocity is always changing with the programme material and pattern so the output voltage changes accordingly. At a recorded level of 5 cm/sec the output from the pick-up would thus be 5 times 3·2 mV, or 16 mV. This is still very low, and requires a high degree of amplification.

Crystal pick-ups provide a greater output voltage and so require less pre-amplification. Usually, however, the more hi-fi the pick-up the less its output voltage. This should not be taken as a general rule, but as an approximate guide. It must also be remembered that if equalization is required, the relating network itself absorbs some of the signal power from the pick-up, thus reducing the mV per cm/sec across the output terminals.

SIMPLE EQUALIZING NETWORKS

Let us suppose that we have an amplifier with a perfectly linear response within the workable limits of the audio spectrum, to which it is required to connect a pick-up. The record, it will be remembered from the chapter on disk recording, has a certain recording characteristic. In terms of velocity, the bass is attenuated and the top frequencies accentuated. If we use a pick-up whose output is proportional to the velocity of the recording (electro-magnetic type), the reproduction will be most disappointing; there will be a marked lack of bass and a marked increase in "top". This is because the output voltage over the frequency range follows the recorded curve.

What is wanted is a network which lifts the bass and attenuates the high frequencies to the extent that the bass was attenuated and the high frequencies accentuated during the recording process. The equalizing network should in fact have a response/frequency curve which is the exact inverse of the recording curve. This will correct the output signal from the pick-up and linearize it so that it suits the linear characteristic of the amplifier. The reproduction will then sound much more "natural".

Although it is impossible to match inversely all the various recording characteristics that have been adopted, a fair compromise is possible in practice and final adjustment can be made with the amplifier's tone controls. With this in mind, it would appear that future amplifiers will need to have provision for only two recording characteristics—those approved by British Standard 1928:1955 (R.I.A.A.) which was referred to in the previous chapter.

EQUALIZING CIRCUITS

An equalizing circuit is frequency-selective in a sense which corrects the recording characteristic, whilst also taking into consideration the pick-up and amplifier loading and deficiencies of the response of the pick-up itself. Assuming that the pick-up does not introduce its own response coloration and that it has a truly constant velocity output, then the equalizing network given in Fig. 7.5 can be used, either between the pick-up and the input of the amplifier (or radio receiver) or after the first voltage-amplifier in the control unit.

If we study the recording characteristic in Fig. 6.5, it becomes obvious that the equalizing network has to perform three functions. It has to lift the bass from about 600 c/s; to cut the lower bass at around 50 c/s; and to cut the treble around 1,000 c/s. It has to do these things to secure an output linear with frequency from the pick-up circuit.

So that these various functions can be understood, the circuit in Fig. 7.5 has been broken down into three basic sections which are given in Fig. 7.6. The circuit at (*a*) serves to cut the lower bass, that at (*b*) provides an overall bass boost, and that at (*c*) cuts the treble. When the circuit is in the composite form given in Fig. 7.5, then it performs all these functions simultaneously.

The actual affect that these frequency-selective circuits have on the overall recording characteristic depends upon the values given to the capacitors *C* and resistors *R*. It will be remembered that a resistor and capacitor associated in a circuit form a time-constant *T* whose value in

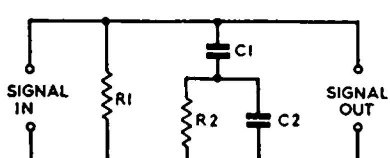


FIG. 7.5. *An equalizing network of this kind is often used for modern recordings in conjunction with high-impedance electro-magnetic pick-ups.*

seconds is equal to the capacitance C in farads multiplied by the resistance R in ohms. The time-constant is extensively used in equalizing networks, its value being given in practice by the following expression: CR = time-constant (T) = $10^6/2\pi f$, where C is in microfarads, R is in ohms, T is in microseconds and f is in c/s. The term f is known as the turnover frequency, and it is by reference to this that the curves are calculated so that they merge inversely with the recording characteristic curve to produce a linear response.

The turnover frequency is reckoned to be that frequency at which the response—as indicated by the curve—is 3 db below or 3 db above the reference response or datum line. With reference to Fig. 6.5: At about 320 c/s the 78 r.p.m. curve is 3 db below the 0 db datum line. Thus, to equalize, a bass boost circuit (Fig. 7.6*b*) will have to be used whose time-constant suits this frequency. Similarly, in the region of 3,000 c/s the same curve is 3 db above the datum, indicating the necessity of a top-cut circuit (Fig. 7.6*c*) having a time-constant related to this frequency. The lower bass requires cutting at about 40 c/s, calling for the use of circuit Fig. 7.6*a*.

It will be obvious that the same answer for the time-constant T can be secured from a host of C and R combinations. Generally speaking, however, the value for the resistor $R1$ (Fig. 7.5) is affected by considerations relating to the matching of the pick-up to the amplifier input circuit. It is assumed that the load presented across the output terminals of this kind of circuit will be of high impedance, represented, for instance, by the control-grid circuit of a voltage-amplifier valve. If the network is fed from a source of low impedance, then a resistor should be placed in series with the signal whose value is high in relation to $R1$. Having fixed the value for $R1$, in terms of matching, $R2$ should be arranged to be approximately 12½ times below $R1$.

The time-constant elements associated with Fig. 7.5 are $C1$, $R1$ and $C2$, $R2$, whose time-constants respectively should be 2,940 microseconds and 81·2 microseconds for microgroove records, and 2,780 microseconds and 57·3 microseconds for 78 r.p.m. records.

All equalizing circuits of the nature described give a signal across their

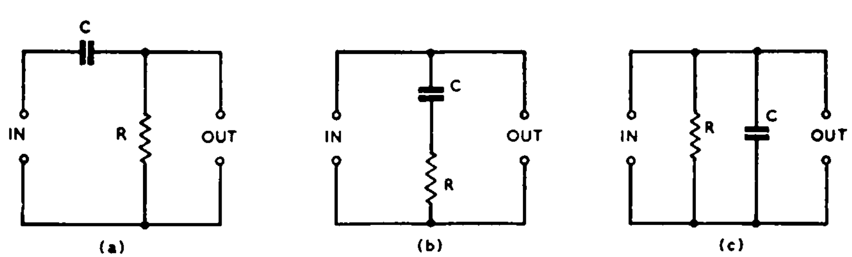


FIG. 7.6. *Frequency-selective networks;* (a) *for lower bass cut:* (b) *for bass lift: and* (c) *for top cut.*

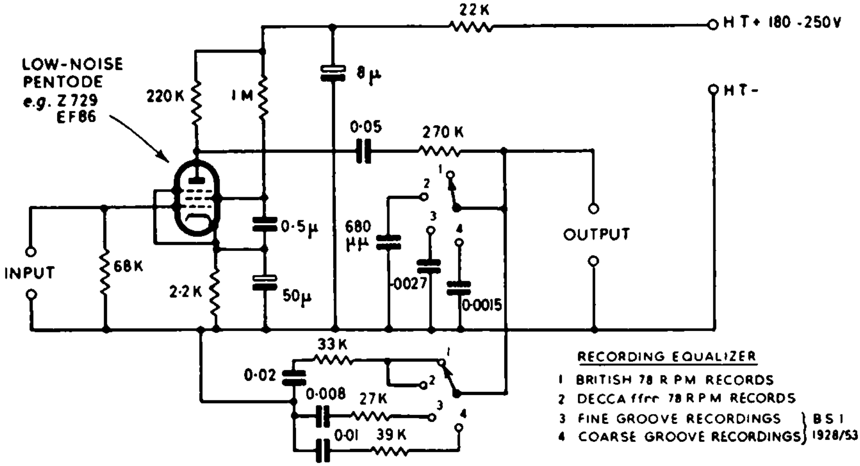


FIG. 7.7. *Single-stage amplifier recommended by Goldring for use with their Type* 600 *pick-up. The output is* 80 *millivolts (equalized) at* 1,000 *c/s at a recorded level of* 3·16 *cm/sec.*

output terminals considerably below the level of the signal applied across their input terminals, and thus attenuate the signal. In certain cases, particularly where the output voltage from the pick-up is low to start with and the amplifier is insufficiently sensitive, or where it is required to connect a low-output equalized pick-up signal to the pick-up terminals of a radio receiver or radiogram, a pre-amplifier will be required to make good this attenuation and provide the amplifier with a signal of sufficiently high level to load it fully.

Two such pre-amplifiers recommended by the Goldring Manufacturing Co. for use with their Type 600 variable-reluctance cartridge are shown in Figs. 7.7 and 7.8. Both circuits provide four degrees of equalization, and the equalizing circuits in both cases follow the low-noise pentode valve. The single-valve circuit provides an output of 80 millivolts at 1,000 c/s at a recorded velocity of 3·16 cm/sec (from the Decca test record LXT5346). Without the amplifier and using the equalizing circuit recommended (see Fig. 7.9 *a* and *b*), the output is only about 4 millivolts at a recorded level of some 10–12 cm/sec. Thus, the single-valve circuit permits this excellent pick-up to be used in conjunction with most makes of hi-fi amplifier.

The two-valve circuit, in which the output is by way of a cathode-follower, is recommended for the Williamson amplifier or similar types requiring a very low-impedance input. The Goldring 600 pick-up is intended for operation into a resistive load of approximately 68,000 ohms, and is not intended for use with a transformer.

As the output voltage over the frequency range is dependent not only

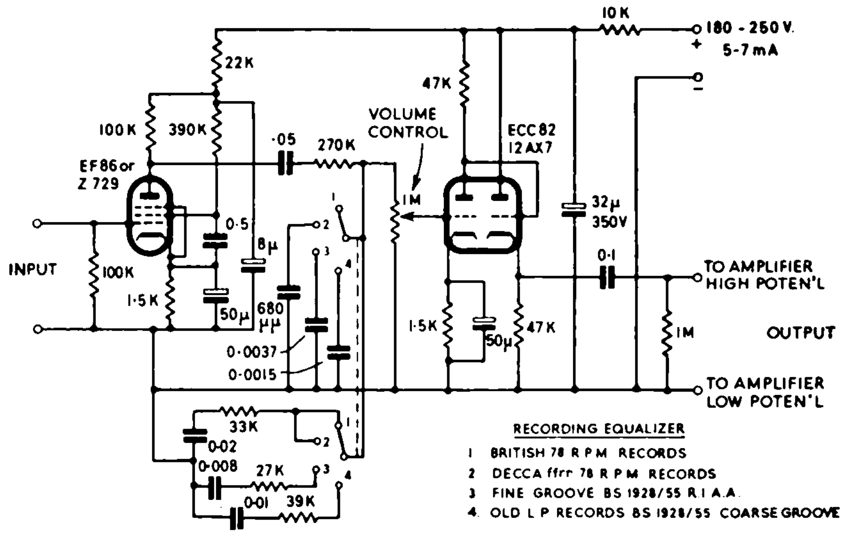


FIG. 7.8. *A two-stage Goldring circuit. This is recommended for the Williamson amplifier and similar units.*

on the type of equalization network employed, but also on the frequency response of the pick-up, the actual design of an equalizing network is not always as simple as it may first appear. Fortunately, it is rarely necessary for the enthusiast or technician to have to work out networks and component values for specific pick-ups as these are nearly always given in the maker's instructions.

CRYSTAL UNITS

The output voltage of a crystal unit is proportional to the force to which the stylus tip is subjected when it is tracing a record. Thus, excluding resonance effects, the open-circuit generated voltage is approximately linear with respect to frequency with reference to the recorded amplitude. Since modern recordings have a characteristic approaching constant-amplitude rather than constant-velocity (due essentially to the bass cut and treble lift), some crystal units, such as the Collaro Studio "O", certain Acos units and others, have an output/frequency curve which is almost the inverse of the recording curve. The replay characteristic is thus automatically secured, and additional equalizing networks are not required. In such cases, the pick-ups can be connected direct to the input terminals of a linear amplifier and acceptable reproduction is obtained.

When using crystal pick-ups of this kind care should be taken to see that they are not connected to an *equalized* input of an amplifier, otherwise the

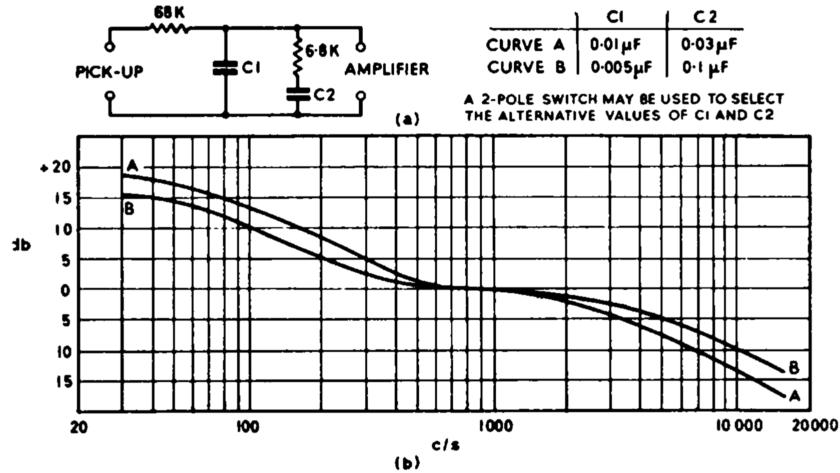


FIG. 7.9. (a) *Equalizing network recommended for the Goldring Type 600 pick-up:* (b) *associated equalization curves.*

bass will be overpowering and the top considerably muted, as the result of an effective double-equalizing function. The same trouble would result if a record player whose pick-up signal passes by way of an internal equalizing network is connected to an equalized pre-amplifier input socket. Some record players, such as the Decca, employ internal equalizing so that they can be connected direct to the pick-up sockets of a radio set. Hi-fi players, however, rarely adopt this idea, the pick-up wires coming direct from the pick-up and, unless the crystal types mentioned above are used, they require an equalizing network either externally or in the amplifier. We have seen in the previous chapters that switched equalizing circuits are usually incorporated in the pre-amplifier or control unit of hi-fi systems.

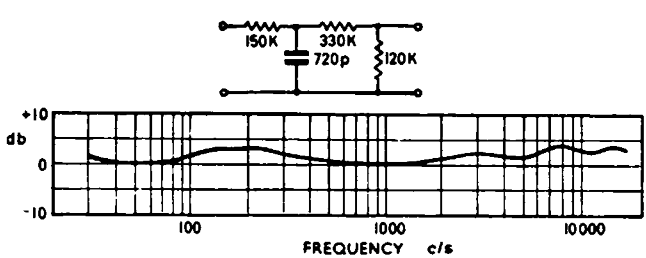


FIG. 7.10. *The network recommended for the Acos Black Shadow pick-up to provide the substantially linear output shown by the curve for records cut to B.S.I./R.I.A.A. standards.*

An essential element in the crystal pick-up is a capacitance formed by the crystal as the dielectric and the plates either side. If this capacitance is loaded by a resistance, then the signal is differentiated and the output will approach the velocity characteristic. The extent of this modification in characteristic depends to a large degree on the response of the pick-up unit itself and the value of the load resistor. This makes it most important that the value stipulated for the resistor by the makers is always used.

Fig. 7.10 shows the network recommended for the Acos Black Shadow pick-up to give the substantially linear output from records cut to the B.S.I./R.I.A.A. standards. The network in Fig. 7.11 should be used to modify the output to the velocity characteristic so that the pick-up can be connected direct to the magnetic input terminals of the control unit. The inclusion of this simple resistor-capacitor network avoids the necessity of having to alter the existing equalizing circuits to cater for the pick-up. Both networks work into an amplifier impedance of 100,000 ohms, while the equalized signal at the output of Fig. 7.10 is approximately 20 mV per cm/sec and 30 mV per cm/sec at the output of Fig. 7.11, thus requiring amplifier sensitivities in the region of 60 mV and 100 mV respectively.

MATCHING

Matching should not be looked upon lightly if best results are to be secured. Fig. 7.12 shows a selection of frequency-response curves of the Goldring variable-reluctance cartridge Type 500 taken under different loading conditions. This shows clearly how the high-frequency response may be considerably impaired by operation of the cartridge with resistive loads other than the optimum value, or with capacitive or combination resistive and capacitive loads.

The effect of capacitance on the circuit is important when it is

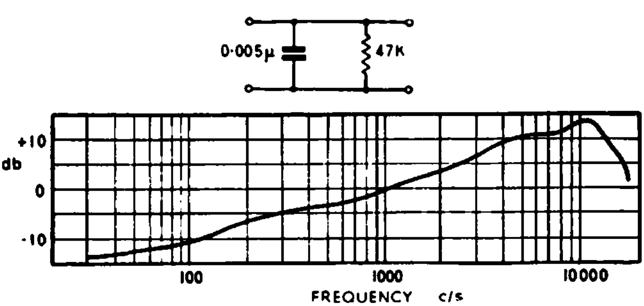
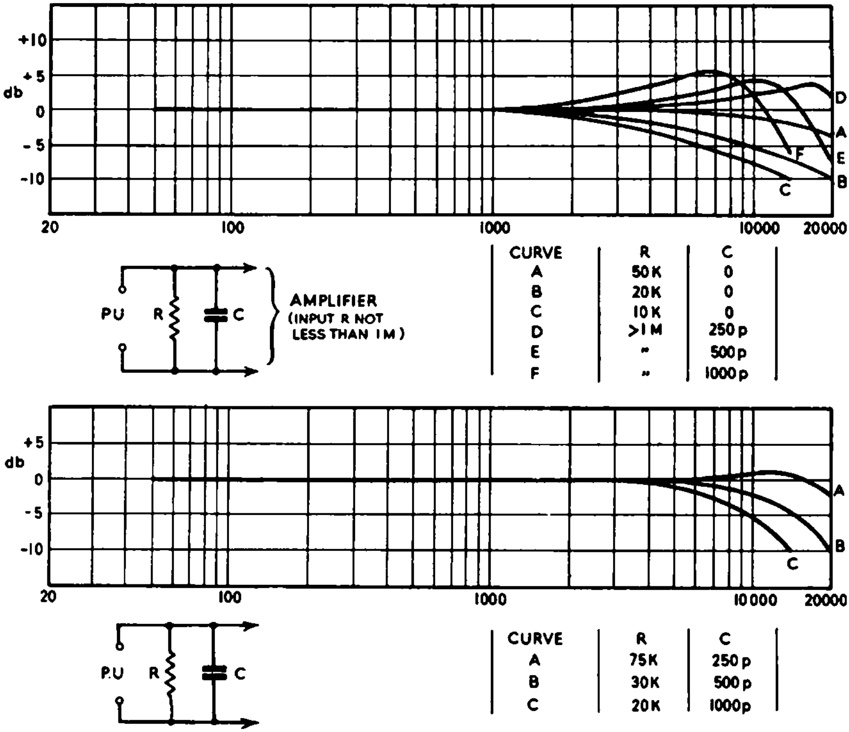


FIG. 7.11. *The network recommended for the Acos Black Shadow pick-up to provide a velocity characteristic as shown by the curve. This enables the pick-up to be connected by way of the magnetic pick-up terminals of the control unit.*



CURVE	R	C
A	50K	0
B	20K	0
C	10K	0
D	>1M	250p
E	"	500p
F	"	1000p

CURVE	R	C
A	75K	250p
B	30K	500p
C	20K	1000p

FIG. 7.12. *Frequency-response curves of the Goldring Type 500 variable reluctance cartridge taken under different loading conditions of resistance and capacitance.*

considered that pick-ups are often connected (of necessity) to the pre-amplifier by way of screened cable whose capacitance value may be in the region of 50 pF per foot of length. Then there are the stray circuit capacitances and the capacitance reflected into the control-grid circuit of the first valve because of the Miller effect. All these capacitances contribute to the general loading and matching of the pick-up, though they cause little trouble (provided the pick-up lead is of reasonable length) if the equalizing network stipulated by the maker is adopted.

Low impedance pick-ups, such as ribbon and moving-coil types, require the use of an impedance-matching transformer, so that the pick-up impedance can be stepped-up to match that of the first valve. Any odd transformer is not suitable; the transformer designed specifically for the pick-up should always be used. Transformers modify the overall pick-up circuit frequency-response, and equalizing circuits designed to suit the pick-up and transformer are always given by the makers.

Low-impedance pick-ups and matching transformers bring with them hum troubles if the correct transformer is not used or if it is situated in a hum field due to a smoothing choke or mains transformer. The transformer should have a good-quality mumetal screen, and screened leads should be used throughout (see also the sections on hum in Chapters 2 and 4). In some cases mumetal screens are fitted to low-output magnetic pick-ups so as to avoid hum voltages being induced into the pick-up windings from stray alternating magnetic fields occurring in the neighbourhood of the pick-up (e.g., from the turntable motor). The Goldring Type 600 unit uses a light-weight mumetal case for this reason and as an added precaution, since the push-pull arrangement of the pick-up coils also serves as a hum-cancelling device.

TURNTABLE UNITS

It is not here intended to delve deeply into the mechanics and principles of turntable units and record changers since full information of this nature can be obtained free of charge on application to the various manufacturers. However, there are one or two points of interest to consider.

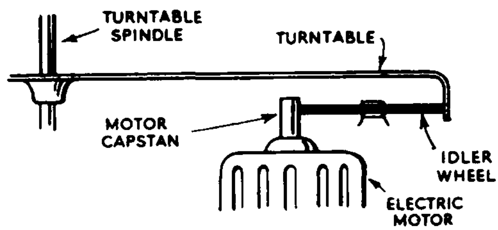
Generally speaking, hi-fi enthusiasts are not lovers of record changers; they usually prefer a good-quality four-speed transcription unit. Apart from the skidding of records, one on top of the other, and the resulting wear and speed variation, a lot of the fun of being a hi-fi enthusiast is lost when the need for record changing is eliminated throughout a session. (This view may not be shared by all, but it is the author's personal view and that of his associates!)

The turntable unit, whether automatic or not, has to rotate the record at a constant velocity whilst maintaining perfect balance to avoid such effects as wow, flutter and rumble. This is achieved by the use of a carefully designed and balanced turntable having a large proportion of its mass concentrated at its periphery, well-engineered bearings and a good-quality constant-speed driving motor.

As distinct from the centrifugal-governor type of motor which was popular in the very early days of hi-fi, modern turntable units, including autos, invariably use an induction-type motor to energize the turntable. Although its speed is governed to a large degree by the frequency of the a.c. mains supply, the induction motor is not truly synchronous, as is an electric-clock motor for example. The motor is usually of the four-pole variety, giving a loaded rotor speed of 1,320 r.p.m. or thereabouts.

Some transcription units incorporate a speed adjustment in the form of an eddy-current brake which applies an even load to the motor and thus permits control of the speed over a range of plus and minus 2½ per cent. The Garrard transcription unit has such an arrangement. The load is applied

FIG. 7.13. *The modern method of turntable drive.*



by means of an aluminium disk attached to the rotor shaft passing between the poles of a permanent magnet. The magnet is pivoted in such a way that operation of the speed control increases or decreases the field over the disk.

Another form of speed variation, used in the Goldring (Lenco) transcription unit, takes the form of a speed-cone coupling between the motor and the turntable; adjustment of the speed control alters the position of the coupling wheel on the cone and hence the ratio of the coupling between the motor and the turntable.

In most units, the drive to the turntable is by means of a rubber idler wheel which engages with both a capstan on the motor spindle and the inner edge of the turntable rim (see Fig. 7.13). The turntable speed is controlled by the ratio of the diameters of the motor capstan and the turntable rim. The idler wheel has no direct bearing on the speed of the turntable, a fact which is not always realized. The idler wheel, or jockey pulley, is held under slight pressure between the two drives by means of a spring.

There are various methods in use for obtaining the three or four speeds (16⅔, 33⅓, 45 and 78 r.p.m.). A popular method, employed in B.S.R. changers and players, makes use of a four-speed pulley on the motor spindle and an adjustable jockey assembly. Speed changing is effected by raising or lowering the jockey wheel by a set of levers actuated by a control knob. This arrangement is shown in Fig. 7.14. The jockey wheel is made of hard rubber, and should therefore be kept absolutely free from oil or grease.

A frequent fault on this type of unit, preventing correct selection of the various speeds, is caused by the four-speed pulley slipping down the motor spindle as the result of the

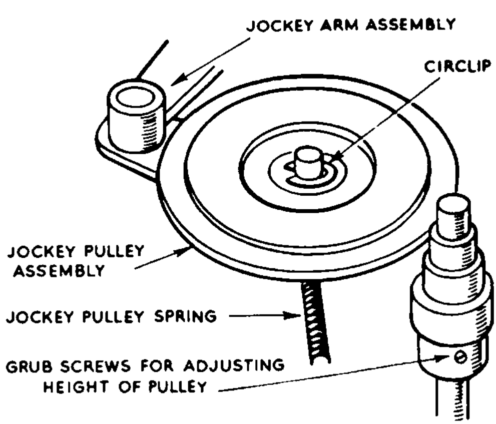


FIG. 7.14. *Speed change is effected by a four-speed pulley.*

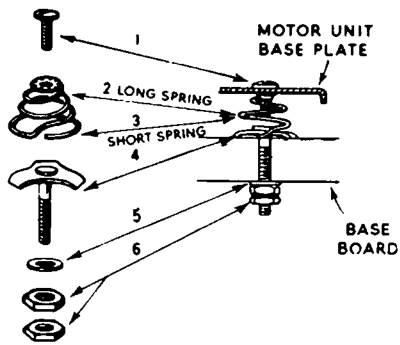


FIG. 7.15. *The Goldring-Lenco method of player-unit suspension.*

retaining grub-screws loosening. Resetting the pulley is a simple matter, as will be seen from Fig. 7.14.

To avoid a "flat" appearing on the rubber idler wheel, the wheel is automatically retracted on some units when the motor switches off at the end of a record or series of records, or when the speed control is set to the "neutral" position.

WOW, FLUTTER AND RUMBLE

Speed variations of the turntable below a frequency of about 20 c/s are referred to as *wow*, since this is the effect given to the reproduction as the result of this fault. The general causes are unbalance in the turntable, slip due to oil on the drive pulleys, a distorted or off-centre record, slip of one record upon another on an auto unit, and non-concentricity of the idler wheel.

Flutter refers to speed variations of the turntable above a frequency of about 20 c/s. The causes are untrue motor bearings (due to wear in the bearings), unbalance of the rotor, slight eccentricity of the idler wheel or in the mechanical coupling between the motor and turntable, and the transmission of vibrations from the motor affecting the speed of the turntable.

A low-frequency rumble is sometimes superimposed upon the reproduction also as the result of slight unbalance in the motor and drive assemblies, or motor vibrations transmitted to the pick-up way of the idler wheel and turntable. In this respect, resilience of the motor mounting is important, as also is the method of mounting the unit on the motor board. Fig. 7.15 shows the spring mounting arrangement adopted on the Goldring-Lenco transcription units.

Crystal pick-ups are usually more sensitive to motor rumble than their magnetic counterparts, which suffer more from hum pick-up. This is because crystal units are more sensitive to amplitude changes at low frequency than are magnetic types. Sometimes, due to this cause, an aucostical feedback path is promoted between the pick-up and loudspeaker by way of the stylus, record, motor unit and loudspeaker mounting. When this happens a low-frequency howl is emitted from the loudspeaker which can be stopped only by turning down the volume control or removing the pick-up from the record. The trouble is characteristic especially of the portable record player or radiogram in which the loudspeaker is mounted in the same case as the record playing unit.

A microphone valve in the amplifier may evoke a similar symptom, but this trouble can easily be determined by gently tapping each valve in turn with the end of a pencil. Persistent acoustical feedback of this nature should lead to investigation of the spring mounting of the motor unit on the motor board. It should be ascertained that the clamping screws, which secure the motor unit during transit, are either fully released or removed, depending upon their nature, so that the unit is freely suspended upon the springs. Sometimes it may be necessary to replace the crystal cartridge if acoustical feedback cannot be cleared by other means.

RECORD CHANGERS

When servicing record changers it is always desirable to support the unit at working level on two small boxes. The boxes or blocks should be situated so that they are clear of the mechanism and permit normal operation of the unit. The various operating cycles can be observed by placing a flat mirror on the table or bench beneath the changer. Generally speaking, once the sequence of operations has been carefully observed over several cycles, the cause of the trouble becomes apparent and it requires only a logical approach to apply a remedy. However, if the fault appears complex, or if the action of the sections cannot be understood, the service sheet appropriate to the unit should be studied.

With modern units, the turntable drive is similar to that associated with single players. The intermediate idler wheel is used, and speed change is effected by means of the stepped motor pulley, as already described. There is, however, a secondary drive and cycling mechanism which serves to change the record. This is brought into action by rapid movement of the pick-up when tracing the play-out groove or by operation of the "start" control.

Several things happen as the result of the operation of the cycling mechanism. The next record in the pile on the turntable spindle drops on to the turntable, the pick-up lifts and moves to its setting-down position—this often being determined by the operation of a "feeler" as the record drops—and the cycling action ceases until the pick-up is again moved rapidly by the play-out groove or until the "reject" control is turned. When the last record is played the unit switches off automatically, since then the control arm which rests on the pile of records is at its lowest level. There are a number of variations of these actions, but the results are always the same.

The automatic trip which is brought into operation by the play-out groove on the record is common to both auto units and single players, though transcription players do not always employ it. There are two general types operated by either a velocity trip or ratchet. The former type is the most popular and this takes the form of a quadrant which carries a trip lever by way of a friction coupling. The quadrant is coupled to the pick-up arm and moves with it as it traverses the record. At the end of the trip lever is a small felt pad; on the turntable spindle is a striker which makes contact with the felt pad when the pick-up nears the centre of the record.

As the record is played the trip lever moves very slowly inward, and at the point where the felt pad makes contact with the striker the trip lever is pushed back against the friction coupling on the quadrant. This action continues at each revolution of the turntable until the trip lever is accelerated by the pick-up stylus tracing the widely spaced play-out groove; the striker then positively engages with the felt pad and pushes the trip lever in such a way that the motor on/off switch is actuated. In some units a turntable break is also brought into action.

The ratchet arrangement includes a lever connected to the pick-up arm to which is attached a lightly loaded pawl. As the arm moves towards the centre of the record the pawl rides over a lever-type ratchet which is mechanically coupled to the on/off switch. When the stylus traces the play-out groove the lever which is attached to the pick-up arm oscillates to and fro owing to the eccentric nature of the groove spacing. The pawl thus engages with the ratchet when the movement of the pick-up arm reverses, the associated lever comes into play and the switch disconnects the motor from the mains supply. With auto units, the trip mechanism starts the cycling action of the record-changing mechanism.

These automatic functions are much of a compromise between the mechanics of the unit and the mechanical and electrical nature of the pick-up. Whilst modern units present little load on the stylus, the fact that a load is presented to operate the mechanism puts the idea out of favour with hi-fi types. There is, of course, the possibility of impaired tracking and groove jumping as the result of the additional load of the auto-stop mechanism, and a soft, though definite, knock is heard in some units at each revolution of the turntable as the striker makes contact with the felt pad on the trip lever. For these reasons transcription units use a manual on/off switch.

When servicing auto units, care should be taken to avoid the application of too much oil as several of the functions rely upon friction. The motor bearings themselves are usually of the self-oiling type, though occasionally a few spots of very light oil are required on the felt pads at the seat of the bearings.

With transcription units, the heavy turntable (the weight of the turntable on the Collaro Model 4T200 unit is 8½ lb.) is fitted with a highly machined shaft which runs in a bearing (sometimes self-lubricating) with a steel ball pressed into its lower end. This ball takes the total thrust of the turntable, reduces friction to a remarkably low level and eases wow and rumble problems.

FIG. 7.16. *The Collaro Conquest record-changer.*

FIG. 7.17. *The Collaro 4T200 transcription unit.*

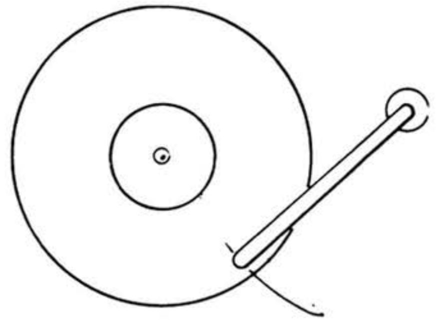


FIG. 7.18. *Illustrating tracking error. The broken line shows the path traced by the recording stylus and the full line the path traced by the reproducing stylus.*

Other features sometimes found are a groove-location device (Goldring-Lenco Models GL55 and GL56) which introduces the stylus to the start of the groove, a stroboscopically-marked turntable to determine the accuracy of the speed of the turntable, and a spring- or weight-loaded device on the pick-up arm which can be adjusted to provide a range of stylus pressures.

Figs. 7.16 and 7.17 show the Collaro Conquest record changer and Model 4T200 transcription unit respectively.

TRACKING PROBLEMS

During the recording process the recording head is arranged to move radially across the disk, as shown by the broken line in Fig. 7.18. On replay, however, as the result of the pick-up swinging in an arc about the pivot of the pick-up arm, the stylus follows the full-line curve in Fig. 7.18. This means that over most of the record the oscillations of the reproducing stylus, as the result of the impressed modulation pattern, occur at an angle to those of the recording stylus. The reproducing stylus deviates from the true lateral movement by an amount depending upon this angle of error, usually referred to as "tracking error".

Whilst this leads to even harmonic distortion, the stylus has a reduced compliance away from the true lateral movement and greater energy is required to cause it to vibrate, resulting also in greater wear on the sides of the groove where the tracking error is greatest. The tracking error can be kept at a very low figure by mounting the arm so that the tip of the stylus overhangs the centre of the turntable by a small amount, by offsetting the pick-up head on the arm at a slight angle, inclined towards the record centre (this is catered for during manufacture), and by the use of an extra-long pick-up arm—there is a limit to this, of course, though 16-in. arms are often used for studio work.

The recommended position for the pick-up, giving the optimum overhang, is stipulated by the manufacturers, often in the form of a template for the initial setting-up of the equipment. There are critical values for overhang and pick-up head displacement which result in two positions of zero error. For example, an 8-in. arm can be set so that the maximum error at two points is only 2½ deg., while by the use of a 16-in. arm the maximum error can be reduced to 1½ deg. On the other hand, a poorly mounted 8-in. arm and pick-up can lead to an error of some 18 deg., producing considerable distortion at high recorded levels.

A method which virtually cancels the tracking error makes use of a special type of pick-up arm on which the pick-up head carrier is pivoted. As the stylus traces the groove over the radius of the record so the pick-up swivels on its pivot in a way that counteracts the error. This idea is embodied in the Burne-Jones (B-J) arm.

As already mentioned, the low-frequency resonance of the pick-up system is somewhat governed by the pick-up arm. Hi-fi arms usually have a very low resonance—well below the troublesome level. The Goldring Jubilee arms, for instance, have a resonance in the region of 9 c/s. An average arm has a resonance slightly above this figure—15 c/s to 30 c/s. Other spurious resonances sometimes occur in less exacting arms due to torsional effects.

The bearing friction of pick-up arms should also be at a low level. This friction is usually measured in terms of the weight required to overcome the friction. A good arm may have a value equivalent to ½-gram, while 4 or 5 grams may represent the value if the bearing is tight or if the arm is poorly designed. The average value is something like 1·5 grams. A high value of bearing friction may promote groove-jumping, particularly if a low tracking pressure is used. Other causes of this symptom are (1) a chipped or worn stylus, (2) the use of a pick-up with too small a value of lateral compliance, particularly when used with modern records of extended frequency range, and (3) a faulty record—this should be suspected if groove-jumping persistently occurs at a certain spot on the record.

CHAPTER 8

Microphones and Mixers

A MICROPHONE performs two essential functions; it first transforms the sound energy it receives into vibrations, and then transforms the vibrations into a voltage whose pattern matches the original sound. A microphone is, in fact, a generator of electricity, but is driven by sound energy instead of mechanically. In the days when the author first began to study microphones. he conceived an idea that he thought would help reduce the cost of electricity. He proposed the installation of giant exponential horns on noisy railway stations which would "capture" the wasted sound energy and turn it into electricity through the medium of huge microphones loaded to the throats of the horns! His tutor was pleased that the basic idea of microphones had been grasped, even though he didn't seem very impressed with the scheme!

Excluding the "lo-fi" carbon microphone, there are four basic types which are employed in hi-fi work: (1) crystal, (2) moving-coil, (3) ribbon, (4) condenser. With all types there is a diaphragm, or a moving member, which is caused to vibrate by the sound energy and which actuates the generating system.

There are two methods by which the sound energy can evoke sympathetic vibrations of the diaphragm. Except in the case of the ribbon microphone, which has a thin ribbon instead of a diaphragm, the sound is applied to one side of the diaphragm only and the pressure component of the sound radiations is utilized. The rear of the diaphragm is cut off from the pressure wave by the microphone housing and is thus at normal atmospheric pressure, which results in the diaphragm being deflected inwards and outwards by the pressure variations in accordance with the pattern of the sound radiations. Microphones adopting this principle are said to be *pressure operated.*

In addition to the pressure component of a sound wave, there is another component called the particle velocity (see Chapter 1). Since it is not possible to produce a ribbon of sufficiently slender dimensions to couple with high efficiency to the velocity component of a sound wave, there can be no such thing as a purely velocity-operated microphone, though this term is sometimes used in regard to ribbon microphones. Ribbon microphones represent an approximation to velocity operation, since both sides of the ribbon are exposed to the sound field, and movement of the ribbon is caused by the sound pressure *difference* between the two sides. Such microphones are known as *pressure gradient* types, and since the pressure gradient of a sound wave is proportional to the particle velocity, there is some excuse for the term "velocity operated".

CRYSTAL MICROPHONES

The generating system of a crystal microphone, as of the crystal pick-up, is Rochelle salt crystal which produces an electrical potential difference when subjected to changes of pressure. The crystal is cut from a big crystal at a critical angle and placed between metallic plates arranged so that the pressure on the crystal is varied in sympathy with the vibrations of the diaphragm. The idea is shown in Fig. 8.1. The voltage thus generated varies in accordance with the sound vibrations.

There is another type of crystal microphone, generally referred to as the sound-cell microphone, in which the sound pressure operates the crystal directly without a conventional diaphragm. This type is considerably less sensitive than the diaphragm-actuated type, but its frequency response is much better since there is no coloration from the diaphragm.

Crystal microphones are high-impedance devices and can, therefore, be connected direct to the grid circuit of a valve. They are not affected by magnetic hum fields, are fairly light in weight and, in the case of the diaphragm type, provide a fairly high output voltage, thereby avoiding the necessity of high-gain microphone amplifiers. They are used extensively by home tape-recordists, and are useful for a number of applications which do not need long connecting cables—these, because of the necessary screening, would be likely to attenuate the higher frequencies.

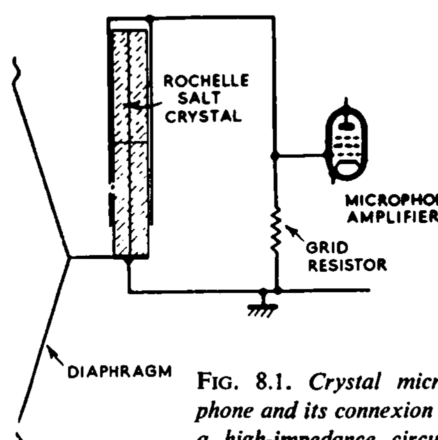


FIG. 8.1. *Crystal microphone and its connexion to a high-impedance circuit.*

MOVING-COIL MICROPHONES

This microphone is basically the same as the moving-coil loudspeaker (see Fig. 8.2). The diaphragm is mechanically coupled to the coil which operates in the air gap formed by the poles of a permanent magnet. The winding thus cuts

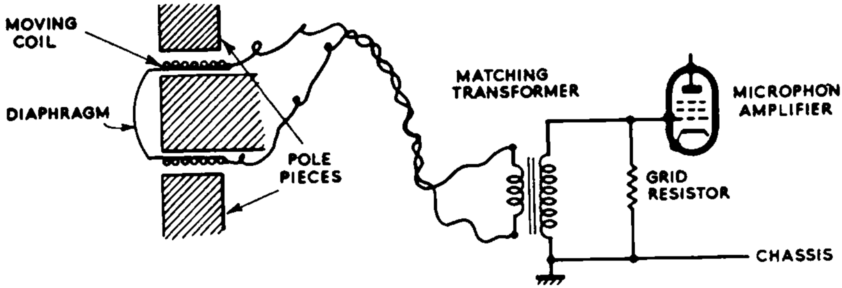


FIG. 8.2. *The moving-coil (dynamic) microphone. Its low impedance requires the use of a matching transformer.*

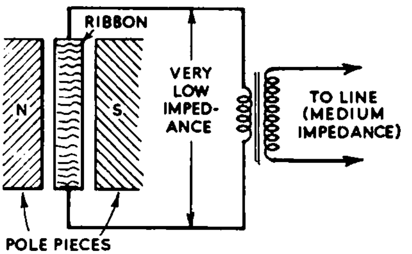
the lines of force of the magnet when it is actuated by the diaphragm. A voltage is, therefore, produced whose magnitude is proportional to the *rate* of cutting of the lines of force, or the velocity of the coil. Moving-coil units are of low impedance (around 30 ohms), and usually give a smaller output than the crystal unit. However, they lend themselves to operation at greater distances from the amplifier than crystal types, since a low-impedance line is less subject to losses of all kinds. A matching transformer is required, and this is usually mounted in or near the amplifier.

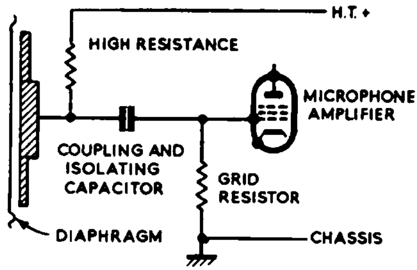
RIBBON MICROPHONES

Fig. 8.3 shows the basic construction of the ribbon microphone. It operates in the same way as the moving-coil unit, in that the ribbon represents the moving conductor. Both the output voltage and the impedance are very low, and to bring the impedance up to a reasonable figure a small transformer is often incorporated in the stem of the microphone. The Reslo unit embodies this feature.

CONDENSER MICROPHONES

As with the electrostatic loudspeaker, the condenser microphone (the term electrostatic microphone is rarely used) is essentially a condenser formed of two plates separated by the air as a dielectric. One plate is fixed, while the other serves as the diaphragm and is caused to vibrate by the incident sound pressure. The capacitance across the two terminals thus varies in accordance with the sound pattern. A polarizing voltage is required, and is connected in series with the microphone by way of a high-value resistor which, irrespective of diaphragm movement, holds the charge on the microphone at a fairly constant value. Thus, as the capacitance alters in value due to sound pressure, the potential across the capacitor varies accordingly, since this is equal to the charge divided by the capacitance. This varying potential is applied to the grid circuit of the microphone amplifier valve, as shown in Fig. 8.4.





(Left) FIG. 8.3. *The ribbon microphone. Its very low output impedance usually requires the use of a microphone transformer in the housing to produce a reasonable line impedance.* (Right) FIG. 8.4. *The condenser microphone requires a polarizing voltage, which is obtained from the h.t. line through a resistor.*

This type of microphone is used with certain Continental tape recorders, and also for laboratory tests and studio applications. Since the output impedance is extremely high, and the output voltage is affected by cable capacitance, a small pre-amplifier is sometimes built into the microphone housing so that the line impedance can be reduced to a workable value.

POLAR RESPONSES

Within the limitations of frequency, the pressure-operated microphone is essentially omnidirectional, i.e., it is responsive to sounds arriving from any direction within its range. Its polar response thus has a spherical distribution. However, at frequencies where the wavelength of the sound becomes comparable with the size of the housing, it tends to become unidirectional, and will have greater sensitivity to sound arriving at the front. This is illustrated in Fig. 8.5.

The pressure-gradient microphone, on the other hand, has a figure-of-eight polar response, as shown in Fig. 8.6. This kind of microphone does not respond at all to sounds arriving at the sides but has a usable response over about 100 deg. both at the front and rear. The polar response can, however,

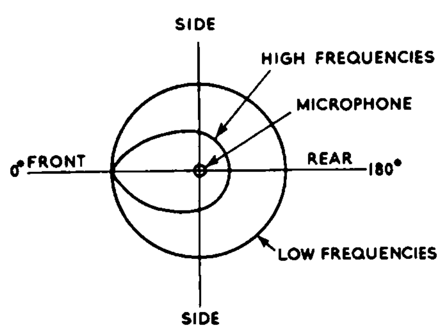


FIG. 8.5. *The omnidirectional response of pressure microphones tends to become unidirectional when the wavelength of the sound is comparable with the microphone dimensions.*

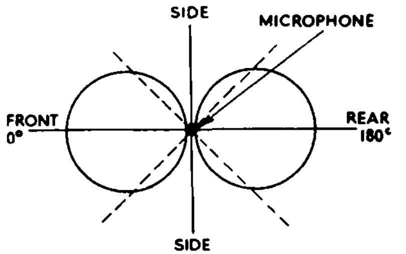


FIG. 8.6. *The figure-of-eight response of pressure-gradient microphones.*

be modified to suit the prevailing acoustical conditions by partially closing the rear of the microphone by means of small acoustic filters (pads).

A cardioid (heart-shaped) polar response can be obtained from a microphone which combines the output from a pressure-operated unit with the output from a pressure-gradient unit. Combined microphones of this kind, known as cardioid microphones, are used extensively for broadcasting work. The cardioid response diagram is given in Fig. 8.7. Because they combine a high-quality pressure-operated unit with a pressure-gradient unit, whilst maintaining a sensitivity and acoustical balance over the greater part of the sound spectrum, true cardioid microphones are rather costly instruments, and are usually too expensive for the average enthusiast. However, they are sometimes employed by the "serious" amateur tape and disk recordist, and by organizations operating sound-reinforcement services.

It is interesting to note that a semi-cardioid response can be obtained from a pressure-gradient microphone by closing the back half of the ribbon with an acoustical filter. Two responses are thus obtained, circular and figure-of-eight, which combine to give the cardioid response. A cardioid response can also be obtained from specially constructed condenser microphones, in which two diaphragms are used, separated by a perforated electrode.

MICROPHONE SENSITIVITY

The sensitivity of a microphone is usually expressed in decibels relative to a fixed reference level. The reference level chosen is invariably 1 volt (equals 0 db) with a sound pressure of 1 dyne per square centimetre (1 dyne/cm^2). Thus, a microphone quoted as having an output level 60 db below 1 volt/dyne/cm^2 would generate about 1 millivolt when subjected to a sound pressure of 1 dyne/cm^2. A sound pressure or sound intensity of twice the

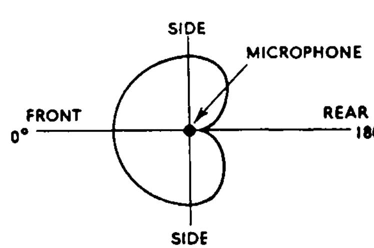


FIG. 8.7. *A cardioid or heart-shaped response is obtained by combining the principles of the pressure and pressure-gradient units.*

value would increase the output voltage by a factor of 2, while a sound pressure of half the value would decrease the output voltage by a factor of 0·5.

The overall sensitivity is somewhat governed by the output impedance. For instance, the Lustraphone Full-Vision microphone is quoted as having an output of — 88 db at 25 ohms, and an output of — 54 db at 50,000 ohms. In the latter case an impedance step-up transformer is used, which also increases the output voltage.

CHOICE OF MICROPHONE

No hard-and-fast rules can be given in this connexion, since the final choice depends not only upon the particular application, but also upon economic factors. Nevertheless, no one microphone does everything equally well; the diversity of situations for which microphones are required calls for different types if results of the highest order are desired. In this case, considerable knowledge of microphone techniques is essential; the amateur may well obtain better results from the use of one versatile ribbon unit than by the unskilled arrangement of an array of more specialized instruments.

Indeed, the ribbon microphone of modern design can be used for almost all applications if reasonable thought is given to its positioning; the response pattern can easily be varied to suit special conditions by the inclusion of small acoustical filters, as already described. The modern unit has a good sensitivity, and can thus be connected direct to most amplifiers and tape recorders without the need for pre-amplification. The majority of commercial units employ inbuilt transformers providing an output impedance sufficiently low for connexion to long lines, whilst also providing a good match to the input impedance of most amplifiers. The Reslo Type RB miniature ribbon microphone can be obtained with output impedances ranging from 30 ohms to several thousands of ohms (i.e., high impedance). The Lustraphone range of ribbon microphones are also available with various impedance values.

Ribbon microphones have excellent frequency-response characteristics, often maintained substantially level up to 14–15 kc/s. This type of microphone is therefore usually ideal for the recording or reproduction of music in all its aspects. It is not always suitable for outdoor work, where the delicate ribbon may be affected by wind pressure, though it is possible to employ so-called windshields to minimize this disturbance which manifests itself in the form of a roar from the loudspeaker.

Owing to the bi-directional characteristic, the orientation of the ribbon microphone can be adjusted so as to discriminate against unwanted pick-up off the main axis. This feature can be used to advantage to provide a fair degree of balance when a single unit is employed for the reinforcement or recording of an orchestra.

With its general freedom from response peaks, which are inherent in less exacting instruments, the ribbon microphone can also be used for sound-reinforcement applications in rooms which are liable to produce acoustic feedback (the howl effect when the amplifier gain-control is advanced). In spite of the inherently lower sensitivity as compared with, say, moving-coil microphones, improved acoustical efficiency is often possible by the use of a ribbon unit.

The recording operator or sound-reinforcement engineer should always make a special point of instructing the artist or speaker in the use of the microphone. A few minutes spent in the serious consideration of this point is well worth while. Incidentally, considerable accentuation of the lower frequencies, resulting in a "boominess" of reproduction, occurs if a ribbon microphone is used too close to the sound source. If this type of microphone is used closer than about 3 ft. a suitable degree of bass cut should be applied at the amplifier. Perhaps this is the reason why crooners favour the ribbon microphone!

The moving-coil or dynamic microphone is more sensitive than the ribbon unit; it is also more robust, less expensive, and suitable for outdoor as well as indoor functions. It is a popular unit with tape recordists generally and with public-address operators (it should be observed that the term "sound-reinforcement" has been taken throughout this book to mean "hi-fi public address"). The average frequency response of this type of microphone usually falls short of that of the ribbon unit and ranges about 8 kc/s. Its omnidirectional characteristic makes it difficult to avoid acoustic feedback effects in some applications.

The crystal microphone is also used by tape recordists, though it is losing favour with public-address and sound-reinforcement operators because of its high output impedance. It is usually less expensive than the other types considered. The output voltage is a little higher than for moving-coil units, and both its frequency response and response characteristics are rather like those associated with moving-coil units, though they vary widely in different designs. This microphone is also employed in office dictating machines.

The condenser microphone is rarely seen in amateur circles, but, as already mentioned, is sometimes employed with Continental (i.e. Grundig) tape recorders. It has an excellent frequency response, and certain specialized types have been produced which respond to frequencies up to 100 kc/s!

There is a great diversity of designs of the three basic units. There are microphone heads of various types for screwing to a floor or table stand, microphones complete with table stand, hand microphones, so-called full-vision microphones designed to avoid hiding the artist (these are often seen on television), lapel microphones, noise-cancelling microphones and others. It is outside the scope of this book to describe the merits and demerits of all these types, but in all cases the functional units are similar to those described.

MICROPHONE MIXERS

There always comes a time when it is necessary to use more than one microphone. Microphones can be connected in parallel and then to a common microphone input socket on the amplifier, but this practice is not to be recommended. It is far better to use a microphone mixer so as to maintain optimum matching of the microphones to the input impedance, whilst at the same time having full control over the gain setting of each microphone channel.

A circuit of a microphone mixer (Pamphonic Sound Equipment) is given in Fig. 8.8. It will be seen that each microphone is fed into its own pre-amplifier valve, and the outputs are combined, at a level determined by the setting of the appropriate "gain" or volume controls, and then fed to a common voltage amplifier, and thence to the common output transformer. The five input transformers and the output transformer ensure that the correct load is presented to the microphones and the microphone input channel of the main amplifier or pre-amplifier; which in turn results in the maximum transfer of signal with the minimum generation of noise, whilst exploiting the frequency-response characteristics of the equipment to the full.

The 330k resistors connected to the sliders of the volume controls avoid heavy loading on the grid circuit of the output triode when only one channel is in operation; i.e., when four of the controls are backed right off. A degree of frequency correction is also applied to this stage through frequency-selective feedback being given by the 680k resistor and the 0·1 mF capacitor connected between grid and anode. Further correction is applied across the primary of the output transformer T6.

The mixer has its own power supply, which uses a Mullard EB91 (usually employed as a signal detector) as the h.t. rectifier. In order to keep the valve within its limits of operation, the circuit is arranged in the form of a voltage-doubler, and the potential between the heater and cathode of the valve is reduced by the heater being connected to a point of positive potential. The author has had frequent occasion to use this instrument, and it has always proved reliable and has given virtually no trouble at all.

Another neat little four-channel mixer is the Grundig Type GMU3. This is designed essentially for use with Grundig tape recorders, and two of the channels cater for the Grundig condenser microphone by having the necessary 100-volt polarizing voltage available to these circuits. Of the other two channels, one is suitable for a low-impedance microphone, such as the Grundig ribbon unit, and the other is intended to accept a fairly high-level (approximately 300-mV) signal, such as that given by a radio receiver, amplifier control unit or another tape recorder. A magic-eye signal-level indicator is also included on the front panel.

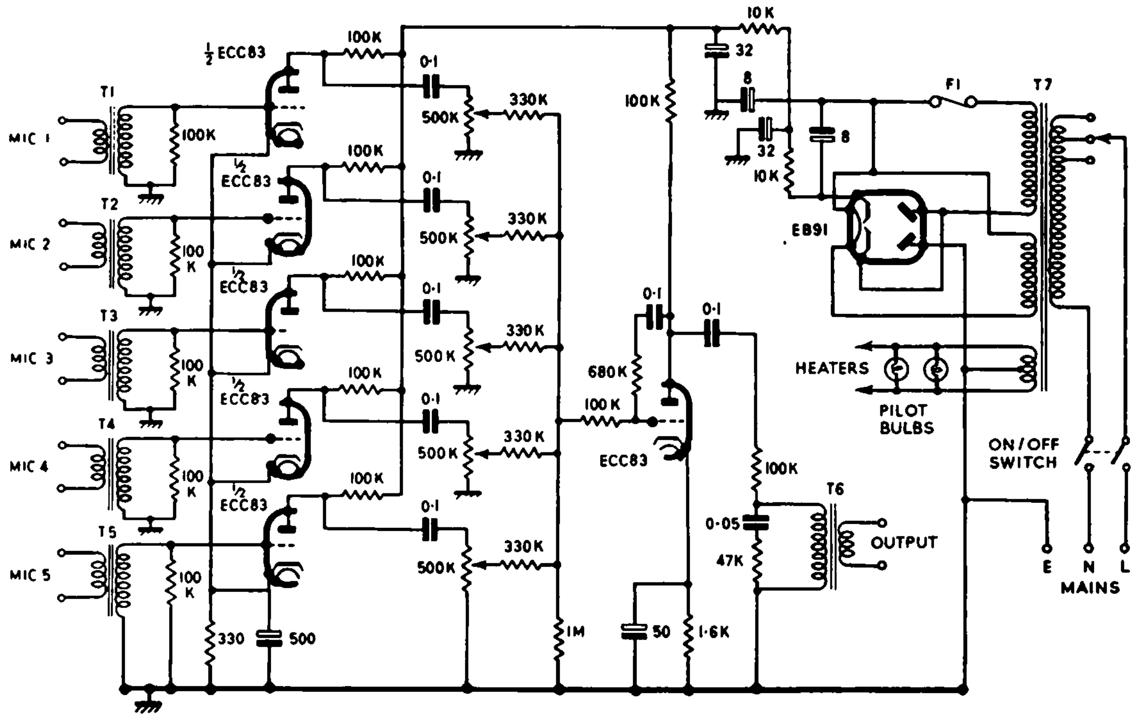


FIG. 8.8. *Circuit diagram of the Pamphonic mixer unit, Model 5W/600.*

TRANSISTOR UNITS

Although power transistors are now used in the output stages of public-address amplifiers of the semi-portable variety, they have not yet found their way into the comparable stages of hi-fi equipment. Whilst transistors are capable of delivering some 10-20 watts or more of audio, the distortion content is above hi-fi acceptance (it is difficult to keep harmonic distortion below about 5 per cent). However, in low-level audio stages the transistor is now beginning to be exploited. One application is in microphone amplifiers and mixers. Recent introductions in this field are the transistorized pre-amplifier units by Lowther and the transistorized mixer unit by Lustraphone.

The mixer unit has four channels, two of high impedance and two of low impedance. The output circuit is suitable for direct connexion to the high-impedance socket of almost all hi-fi amplifiers or control units. The power is provided by a single miniature mercury cell, which has an estimated life of some 1,000 hours. The frequency response is substantially flat from 50 c/s to 14,000 c/s.

Since transistors are inherently free from hum and microphony trouble that are always present to some degree with high-gain valve amplifiers, they are ideally suited to high-gain "front-ends". Small transistor amplifiers can easily be built into the housing of low-level microphones, including the battery power supply. Since a signal of high level can thus be distributed from the microphone circuit, the need for high-gain settings on the main amplifier is precluded, and the danger of hum and noise pick-up on the microphone cable is considerably alleviated.

Another advantage of the transistor is that its input impedance can be arranged to match the low impedances of high-quality electromagnetic pick-ups and microphones, without the need for a matching transformer. The transistor thus serves admirably as an impedance-matching device.

A circuit of a pre-amplifier suitable for electromagnetic pick-ups or microphones of from 100 to 1,000 ohms impedance is given in Fig. 8.9. A transistor can be looked upon as two crystal diodes formed between the emitter and base and the collector and base. In the circuit the letters *B*, *C* and *E* around the transistor symbol represent base, collector and emitter respectively. These three points are often likened to the electrodes of a triode valve as follows: collector = anode, base = grid, and emitter = cathode.

The transistor is biased in the forward direction in the emitter/base circuit and in the reverse direction in the collector/base circuit. With the emitter/base bias disconnected there is theoretically no current (a very small amount in practice) in the collector/base circuit. When bias is applied to the base/emitter circuit, however, current flows in the base/collector circuit, and when the current in the base/emitter circuit increases, the current in the base/collector circuit also increases, but to a greater extent. This action is promoted by the emission of so-called positive holes from the emitter to the collector circuit with a consequent lowering of the resistance of the base/collector circuit.

The signal is applied to the transistor so as to cause variation of the negative current in the base/emitter circuit, which, depending upon how the circuit is arranged, results in an equal or greater variation of current in the base/collector circuit. Generally speaking, there occurs a power gain because the applied signal promotes a current change in a low-resistance circuit (base/emitter) while reflecting a similar or greater current change in a high-resistance circuit, represented by the base/collector junction.

There are a number of methods by which the transistor can be connected into the circuit, as with the triode valve. The arrangement in Fig. 8·9 is usually referred to as a "common base" circuit (the 8mF capacitor connected to the base makes the base common to the input and output circuits, or earthed base circuit, and it corresponds roughly to an earthed-grid valve circuit. The input signal is fed into the emitter by way of the 25mF coupling capacitor (such a large value being necessary to maintain a low-frequency response in a low-impedance circuit), and the output signal is taken by way of the 0·1mF capacitor from between the collector and positive line (chassis). The circuit is thus given a low input and a high output impedance, which is ideal for feeding a signal from a low-impedance microphone or pick-up to a high-impedance input circuit of an amplifier. There is no reversal of phase between the input and output signal voltages.

The circuit is capable of delivering approximately 1 volt r.m.s. of signal for an input of 16mV r.m.s., and thus has a voltage gain of some 62 times. Power is derived from a 6-volt battery, and due to the very low current drain (400 microamps quiescent) a very small battery is all that is required. The response is about 3 db down at 100 c/s and 20,000 c/s relative to 1,000 c/s.

The 6·8k resistor connected to the collector can be considered as the output load, while the resistor connected to the emitter and the two resistors whose junction is connected to the base serve to stabilize the circuit from the d.c. aspect.

One or two points regarding the servicing of transistorized equipment will be useful. Transistors are very sensitive to heat, and

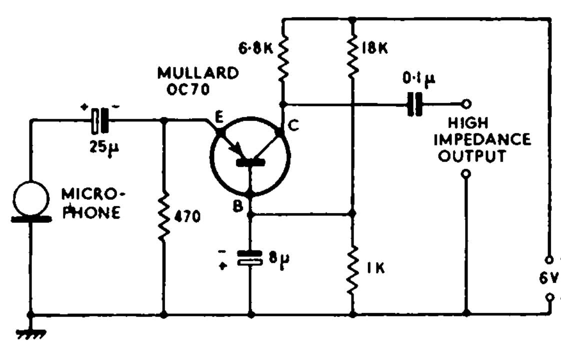


FIG. 8.9. *Circuit diagram of a transistorized micro-phone pre-amplifier.*

if overheated when operating are liable to be destroyed in a very short time. They should therefore be kept clear of soldering irons and heat-producing devices such as valves and resistors of the main amplifier. Soldering in and out of the circuit should be performed as rapidly as possible. A miniature low-power soldering iron is desirable, but even then a heat "sink" should be produced by holding the transistor wires with long-nose pliers while soldering is being done. The transistor wires should never be bent close to the seal on the transistor.

Reversing the polarity of the supply voltage will almost certainly result in immediate failure of the transistor. This must be borne in mind when performing *in situ* voltage, current and resistance checks. It is worth remembering that the negative terminal of most multi-range meters is usually in connexion with the internal battery positive connector when used as ohmmeters.

So as to avoid disturbing the balance of voltage in a transistor circuit, voltage measurements are best made on a high-resistance instrument of at least 1,000 ohms-per-volt. It is not a good thing to make or break a connexion in a transistor circuit while the supply voltage is connected. If it is required to perform a current test, the supply voltage should be disconnected and then the milliammeter inserted where required. The power can then be re-connected and the measurement taken. The same applies on removing the milliammeter.

SERVICING MICROPHONES

Generally speaking, it pays to let the maker have the microphone back if the need for servicing arises. Replacement diaphragms and ribbon elements can, however, be obtained for most quality microphones, as can crystal inserts for sound cell units. It is only a little more expensive to let the maker replace the faulty parts, whilst at the same time ensuring that the performance of the equipment will be up to the normal standard. In a number of cases, the microphone fixing screws are sealed so as to avoid unnecessary tampering, and if these seals are broken the manufacturer may charge for the correction of a fault, even during the guarantee period.

With moving-coil and ribbon units, continuity should be registered across the terminals, and a low-value resistance reading is obtained if the test is directed across the moving coil or ribbon element. A higher resistance reading will, of course, be obtained if the test is made across the secondary of an internal transformer. A crackling is usually heard from the microphone when this test is made, due to the battery in the testmeter causing the microphone to act as a loudspeaker. Crackling noises are also heard from crystal units when subjected to this test, in spite of the normal lack of continuity as indicated on the ohmmeter.

Microphone switches are a constant source of trouble on certain microphones, but this is usually fairly easy to remedy. Broken conductors in microphone cables also represent a frequent source of trouble, particularly if the microphone is in constant employment and moved around a lot. These faults are quickly located by means of simple continuity or resistance checks.

MICROPHONE BALANCE

As we have already seen, the choice of microphone is somewhat governed by the polar characteristic and the frequency response, with due regard also to such things as sensitivity, the type of material to be amplified or recorded and (most important) the depth of one's pocket. The beginner invariably commences operations with a relatively cheap crystal microphone—very often the one supplied with the tape recorder or amplifier, though some manufacturers are now wisely leaving the choice of microphone to the user.

With a little experience, the beginner soon realizes that something more leaborate in the way of a microphone is desirable, if only to cut out the squeak of the door, the tick of the clock or the crackle of the fire. It is amazing how such noises assume prominence on a tape recording. While the ear is able to discriminate against unwanted noises, since there are two of these organs (stereo helps in this respect), the microphone responds to every noise and brings both wanted and unwanted sounds into focus at the loudspeaker. If one microphone is used at some distance from the sound source, then the ambient sounds are going to be recorded at almost equal intensity.

The novice gradually discovers such things for himself, and possibly experiments with various microphones, combinations and orientations. This is a good thing, because it provides the necessary experience in microphone technique for which words can never be used as a substitute.

When using more than one microphone, particularly if the microphones are connected in parallel across the amplifier's common microphone input socket, care must be taken to ensure that they are not placed equidistant from the sound source. This is because there is the possibility of the microphones being out-of-phase (there is an analogy with out-of-phase loudspeakers), in which case serious distortion would occur as the result of cancellation effects at certain frequencies. If this trouble is suspected, the connexions on one of the microphones should be reversed, or one of the microphones should be turned through 180 deg. if it is of the ribbon type with a figure-of-eight response.

In general, however, even if the microphones are phased correctly, it is not good policy to use them close together, because interference effects of the nature described may result at certain frequencies. If the response characteristics of the microphones are known (they can usually be estimated fairly accurately), they should be orientated with regard to each other so that their polar responses do not overlap to any large degree.

When acoustic feedback is troublesome (with, for example, sound-reinforcement work), the placing of the microphone is of great importance. If it is found that insufficient audio power can be obtained before the feedback point on the volume control, and it is impossible to re-position the microphone, other microphones should be tried, such as the ribbon or cardioid. Just before reaching the setting of the volume control which evokes the characteristic howl, a slight ringing sound may be heard when the microphone is being used. At this point distortion may also be at a high level, and for this reason the volume-control setting must be retarded.

Intelligent use of the treble and bass controls may allow a greater volume setting to be used, since the acoustic feedback is a function of the room acoustics. A "live" room, for example, will reflect the higher-frequency sounds and possibly promote feedback conditions, while a "dead" room will tend to absorb certain frequencies of the sound and thus prevent it bouncing back into the microphone. Line-source loudspeakers assist in this respect also, as we have already seen, by concentrating the sound over the required area of coverage and leaving little for spilling into the microphone.

The reverberation of a room has an appreciable effect on a recording. If the microphone is placed a reasonable distance away from the sound source, then it is going to pick up not only direct sound, but also quite a lot of reflected sound; the recording will be coloured by the room acoustics. If the microphone, on the other hand, is placed fairly close to the sound source, the room acoustics will have less influence since most of the sound will be picked up direct, and only a small proportion will be reflected sound. It is, in fact, possible to arrange the position of the microphones to secure almost any required degree of recorded reverberation effect.

Too close a position of the microphone in relation to the sound source should be avoided for most applications, however, since this tends to promote a "bass heavy" effect, but is possibly useful for the recording of dance bands and rhythm groups, where plenty of bass may be required. It should also be remembered that the sound radiation from musical instruments varies considerably with frequency. With a piano, for example, the maximum treble occurs to the right-hand side of the keyboard, and diminishes progressively towards maximum bass in an arc towards the rear of the instrument. With string instruments, the maximum treble is confined to a narrow angle from the major dimension of the instrument.

It is obviously impossible to explore microphone balance in relation to all musical instruments, and from the point of view of the home recordist and enthusiast it often comes down to a matter of trial and error, aiming for overall balance without introducing undue coloration.

CHAPTER 9

The Use of Tape

ONE major advantage that tape has over the disk is that it is a recording medium which can be exploited with almost equally good results by the amateur enthusiast as well as by the professional recording engineer. A high-quality disk-recording outfit necessarily costs much more than the average enthusiast can afford, whereas a hi-fi tape recorder complete is little more expensive than, for example, a hi-fi amplifier and loudspeaker system or a television set.

Interest in tape recording is growing rapidly, and its popularity has undoubtedly been boosted by recent developments in stereophonic reproduction; one of its most attractive attributes is the comparative ease with which a library of tape records can be built up by recording radio programmes, preferably those broadcast on the VHF–FM system (see the author's "F.M. Radio Servicing Handbook"). The quality of reproduction of a tape record made by this means can be as good as that obtainable from long-playing disk records. Tape records, like disk records, are also available commercially and are made in both single-channel and dual-channel (stereophonic) versions. Not only music lovers, but amateur dramatic societies, educational authorities, business, science and industry generally are finding more and more uses for tape recording.

Magnetic tape is also used for the recording of television pictures; as with sound, pictures are broken down into component parts to form electrical impulses, and the impulses are recorded on the tape. The pictures are thus "stored" for future use, and can be reproduced whenever required by playing the tape back through the same machine, which then serves to change the impulses back into the original pictures.

There is little doubt that the enthusiast of the future will have available a vision tape player which can be plugged into the video channel of a television receiver for the showing of pre-recorded tapes. Vision tape records, also carrying a sound accompaniment, will be available, and the enthusiast will eventually be able to make his own vision records with the aid of a small television camera and a tape recorder; immediate replay facilities will be available by way of the television screen.

THE USE OF TAPE

Tape will also store most of the programme material to be transmitted over television networks. Hybrid s.h.f. and wired television networks, will be extensively used in the future, and video tape machines will be connected into such networks to facilitate the showing of items of local interest, such as news and advertising, sporting events and even films. Films, as we know them today, will disappear in favour of magnetic tape, and the cinema industry will tie up closely with the electronics industry; cinemas will close, and every household with a television set will be a potential box office for the new cinema industry. The insertion of a coin into a small box by the side of the television set will bring on to the domestic screen the very latest film through the medium of magnetic tape. We have, in fact, only just begun to exploit the potentialities of tape recording.

THE TAPE RECORDING PROCESS

Magnetic tape is a thin plastic material ¼ in. in width and coated with oxide of iron on one side; this is the "sensitive" side on which the recording is magnetically impressed. With the highly developed processing of modern tapes, it is often difficult to see which side of the tape actually contains the oxide of iron. From both the mechanical and electronic points of view, this is a good thing, because the smoothing of the coated surface as the result of the "buffing" process during manufacture reduces wear on the recording, playback and erase heads and ensures consistent mechanical contact of the tape with the heads, which is desirable for extended high-frequency response.

The recording head magnetizes each section of the tape as it passes over the gap between the poles of the head. The recording head is, in fact, an electromagnet which is energized by the signal current at the output of an amplifier. Thus, if a person is speaking into a microphone connected at the front of the amplifier, the recording head connected to the output is energized by the current which is varying in unison with the sound waves. This means that the magnetic field across the poles of the recording head is also varying both in polarity and in magnitude to the same pattern as the sound waves. It is this varying magnetic field which is used to impress a magnetic pattern on the coating of the tape as it passes over the gap of the recording head.

The idea is illustrated in Fig. 9.1. Here the signal current is represented by a sine wave which causes the polarity of the field across the recording gap to reverse each half-cycle. Thus, as the tape passes steadily over the gap small magnets are formed on the coating of the tape, as shown. The effect would be the same if a more complex signal waveform, such as speech or music, were used, but then the amplitude and the wavelength of the recorded

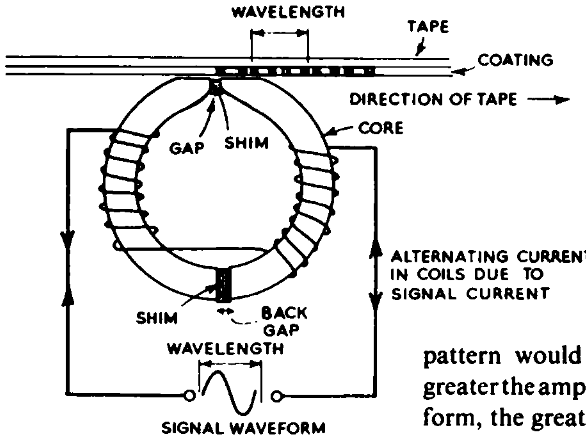


FIG. 9.1. *As the recording tape passes steadily over the gap of the recording head, small magnets are formed on the coating, the character of which conforms to the signal waveform.*

pattern would also be complex. The greater the amplitude of the signal waveform, the greater will be the current in the recording coils and the greater the strength of the magnetic field across the gap. Therefore the strength of the magnets formed on the tape will also be greater.

The actual length of the magnets formed depends upon the wavelength of the signal waveform. Two magnets go to make up one wavelength as the result of the positive and negative half-cycles of the signal waveform (i.e., each half-cycle of signal produces one magnet). The length of the magnets is also governed by the speed at which the tape passes over the gap—the wavelength is equal to the speed of the tape divided by the frequency of the signal current. Thus, at 7·5 in. per second, the wavelength representative of a 7·5 kc/s signal is 0·001 in., so the length of each magnet formed on the tape is approximately half this value, or 0·0005 in. This is a very short magnet indeed, but it is even shorter at higher frequencies.

We must now consider the hysteresis loop, sometimes referred to as the cycle of magnetization. When an electromagnet is energized progressively by the building-up of electric current in its winding there is built-up a magnetizing force around the coil or between the pole pieces of the magnet. This magnetizing force is denoted by the capital letter H, and if a ferrous material (one which is affected by magnetism, such as iron or steel) is brought within range of the field of this force it will have induced into it a magnetic flux; i.e., the material will also be magnetized. Flux is denoted by the capital letter B.

It follows that the greater the magnetizing force, the greater will be the flux induced into the material and the stronger will be the magnetic symptom. This action can be studied in more detail from the so-called hysteresis loop, shown in Fig. 9.2. Here the magnetizing force, both positive and negative, is represented by the horizontal line, while the flux is represented by the vertical line. Let us assume that the ferrous material is in a magnetically

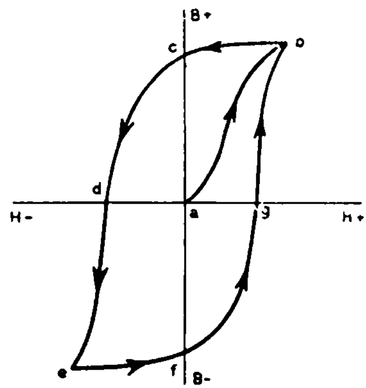


FIG. 9.2. *The hysteresis loop.*

neutral condition, and that the force H is increased from zero (point a) in a positive direction. This can be brought about by increasing the current in the electromagnet; if it is increased in the opposite direction, then the value of H will rise negatively.

Thus, as H increases so will the value of B, and will trace the curve ab. If now the force H is decreased (by decreasing the current in the coil), the value of B will not fall to zero, but line bc will be traced and at zero H there will be an appreciable value of B in the material as represented by point c on the $B+$ line. This value of B is representative of the residual magnetism held by the material. In other words, the material has been magnetized by the magnetizing force which was applied.

However, if force H is now increased from zero in a negative direction, the value of B will fall and line cd will be traced. At point d the material is once again in a demagnetized state. The value of H which is required to demagnetize the material is a measure of the *coercivity* of the material. As H is further increased negatively, the line de is traced, and at e the material is once again magnetized, but in reversed polarity. At point f H is again at zero, and then as H is increased positively once again, the remainder of the loop fg and gb is traced. It should be noted that the initial path ab is traced only when a material in a magnetically neutral condition is subjected to force H.

From the foregoing, it should be clear how a tape is magnetized by the changing magnetizing force across the gap of the recording head—bearing in mind that the tape is passing the gap all the time.

Whilst the hysteresis loop shows the relationship of the flux induced into a material to the force producing it over a complete cycle, it is not wholly representative of the residual magnetism retained by the material owing to the applied magnetizing force for a number of different loops, working from zero H up to *saturation*, both in positive and negative values. (Saturation indicates the point where an increase in H does not result in an increase in B.)

The residual magnetism retained by a material (usually referred to as *remanent induction* and still denoted by B or sometimes Br) and the magnetizing force H required to produce it are plotted to form a transfer characteristic or remanence curve. Such a curve is shown in Fig. 9.3, from which will be seen that the remanent induction or intensity of magnetism impressed upon

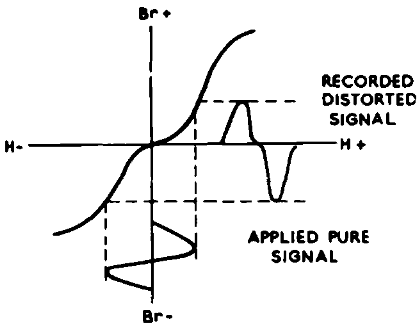


FIG. 9.3. *The transfer characteristic of the remanence curve, showing how the kink in the curve distorts the recorded signal.*

the tape will not be proportional to the force *H* or the current in the recording head owing to the pronounced curvature of the characteristic.

This "transfer distortion" can be avoided by applying a bias in the form of a high-frequency sine wave to the recording head along with the actual signal. The frequency of the bias is usually in the region of 30–100 kc/s, and therefore in the supersonic region and outside the range of hearing. The bias signal is superimposed on the signal to be recorded and has the effect of combating the kink in the centre of the transfer characteristic. How this happens is somewhat complex, and there are several theories on the matter. However, a working knowledge can be obtained from Fig. 9.4, which shows the applied signal on which is superimposed the bias signal, and the resulting distortion-free recorded signal. It appears that the applied signal is alternated either side of the transfer characteristic because of the bias, and that an exact replica of the applied signal is impressed in magnetic form upon the tape as the result of an average of the remanent induction being "sampled" over the two branches of the characteristic curve.

PLAYBACK

Having recorded a magnetic pattern on the tape as described, it is a simple matter to re-convert the pattern back to electrical signals which can be amplified and fed to a loudspeaker system, by passing the tape over a replay head. The replay head is virtually the same as the recording head, and in most portable-type recorders the one head serves for both recording and playback; the tape speed must

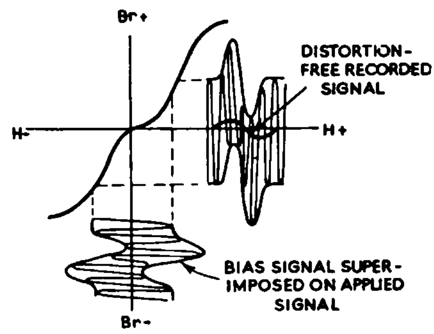


FIG. 9.4. *Showing how the application of a high-frequency bias on the applied signal has the effect of combating the kink in the centre of the transfer curve and gives a distortion-free recorded signal.*

be the same for both recording and playback if distortion is to be avoided.

As the tape passes over the replay head the magnetic pattern produces corresponding variations of flux in the core, which in turn induces e.m.f. variations in the windings matching the pattern of the original signal; these variations in e.m.f. are applied to the grid of an amplifier, in the same way as are the variations of e.m.f. produced by a pick-up or microphone.

The gap between the pole pieces of the replay head has a considerable bearing on the response of the head to high frequencies, and in order to secure optimum output voltage at high frequencies the length of the magnets impressed upon the tape should not be much smaller than the size of the gap. We have seen that two magnets represent one wavelength and that the speed of the tape past the head as well as the frequency of the signal governs the length of the recorded magnets. For example, if the tape speed is 7·5 in. per second and the frequency of the recorded signal is 7,500 c/s, the wavelength of the recorded signal will be 1/1000 in. (0·001 in.). This means that the length of each recorded magnet is half this value, which is 0·0005 in. From theoretical considerations, therefore, it would seem that in order to secure optimum response at 7,500 c/s the gap size should be 0·0005 in.

In practice this is not strictly true, because there are other factors involved. One is that the magnetic field across the gap due to the magnetized tape tends to bulge outwards and cover a greater distance than that represented by the gap itself. This effect is known as "fringing", and tends to reduce the high-frequency response for a gap size as computed above. The effect is more apparent, and loss in high-frequency response becomes very marked, if the tape is not making intimate contact with the head pole pieces.

Generally speaking, the maximum output with a gap size of 0·0005 in. occurs in the region of 4–5 kc/s at a tape speed of 7·5 in. per second. This is not the highest frequency which can be reproduced, however, since as the frequency is increased above that representing optimum output, the output does not suddenly fall to zero but falls off over several octaves, and when the gap size is equal to the wavelength of the recorded signal (i.e., two magnets in the gap) the output falls to zero owing to flux cancellation in the head. At frequencies below that representing optimum output the output falls at the rate of about 6 db per octave.

As a means of securing an improved high-frequency response, without too great a tape speed, manufacturers are investigating the problems involved in producing heads with very small gaps. Not so long ago, a gap size of 0·0005 in. was considered to be about the practical minimum. These days, however, gaps smaller than 0·0002 in. are becoming commonplace. The BBC's Vision Electronic Recording Apparatus (VERA) uses heads with gaps of the order of 0·00002 in., and at a tape speed of 200 in. per second there is only a 3 db fall in response at a frequency of 2·5 Mc/s. As a rough guide, the effective gap size should be approximately 0·5 of the shortest half-wavelength to be reproduced.

Mumetal, Permalloy and other high-permeability alloys are used for the head pole pieces, and the gap is closed by a non-magnetic shim as a means of keeping it clear of magnetic coating from the tape and of concentrating the flux towards the tape. An additional gap is usually introduced diametrically opposite the functional gap, whose purpose is to maintain a linear relationship between the recording current and the flux at the gap. The rear gap is also closed with a non-magnetic shim, and has a width approximately ten times that of the functional gap.

It has already been mentioned that a common head is usually employed for both recording and replay. Where a separate recording head is used, however, the gap size is not so important as that on the replay head. Since the recording does not occur until the tape section reaches the end of the gap, a gap size of the order of 0·0001–0·0015 in. is adequate.

ERASURE

One of the most attractive features of tape recording is that the material recorded can be erased from the tape if it is not suitable or no longer required, and the tape can be used over and over again almost indefinitely. The problem is simply that of demagnetizing the tape. This is best done by passing the tape through a strong alternating field; the field is produced across the gap of an erase head and the winding is energized from the high-frequency oscillator supplying the bias current. The construction of the erase head is slightly different from that of the recording or replay head. The core material has a higher saturation value, and a fairly wide gap in the region of 0·015 in. is usually adopted. Some erase heads have twin gaps, or two separate erase heads may be employed, as a means of ensuring complete erasure.

For successful erasure a high value of alternating current in the erase head winding is necessary, and in order to help satisfy this condition the head is often arranged to resonate at the frequency of the applied signal.

At one time home recorders employed a small permanent magnet to erase an unrequired recording by magnetizing the tape fully in one direction only. Although this simple method, by which the magnet is brought into contact with the tape before it is presented to the recording head, does result in complete elimination of the recording, distortion and a poor signal-to-noise ratio are inevitable when the

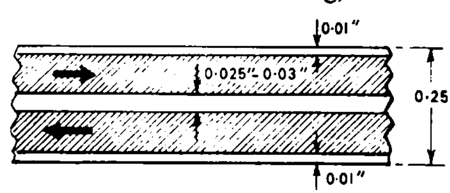


FIG. 9.5. *Dimensions of tape and tracks.*

tape is next brought into service, owing to the standing magnetism on the tape.

There are devices available which are capable of bulk erasure of a tape. The whole tape is set up on the instrument, and the tape is subjected to a very strong magnetic field for several seconds or more, while the spool is rotated on a spindle. This is a very rapid and efficient method of erasure, is convenient when it is required to remove a recording before the tape is re-issued, and also ensures a very desirable high signal-to-noise ratio. A bulk erasure device of this kind is marketed by Leevers-Rich Equipment, Ltd.

RECORDING TRACKS

In early models of home tape recorders almost the full width of the tape was subjected to the recording flux and magnetized. This arrangement is still adopted in professional equipment and machines employed for disk-dubbing because of the ease of editing and the slightly improved quality obtained by using the full width of the tape. With modern domestic tape recorders, however, dual-track facilities are incorporated. Two separate recordings can therefore be impressed side by side on one tape. The track dimensions for this method are given in Fig. 9.5. Whilst these dimensions are almost standardized, one or two very slight deviations are sometimes found, but these have little or no adverse effect on the interchangeability of recorded tapes, provided the direction of scanning of the tracks is as indicated by the arrows in Fig. 9.5, that is with the coated side of the tape away from the observer.

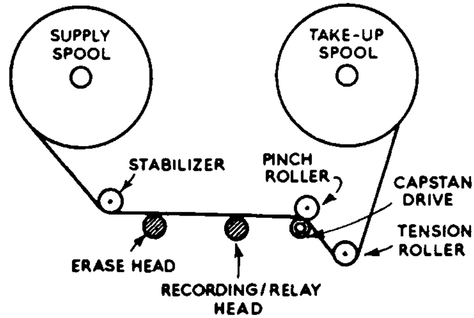
Some machines employ two recording/replay heads, one for each track, and the mechanism is arranged so that a button or switch changes from one track to the other, whilst also reversing the direction of the tape travel. The Simon Model SP4 tape recorder incorporates automatic tape reversal foı continuous recording or replay, without button-pressing or transposition of spools. When one track runs out, the tape automatically reverses and the other track is switched in.

With machines having a single recording/replay head, track change is accomplished by reversing the spools and turning them over, so that the unrecorded bottom half of the tape is scanned by the head.

TAPE DECKS

Generally speaking, the tape deck is a complete unit in itself, as a record player is a complete unit in relation to a radiogram. There are one or two variations in this respect, however, particularly with Continental equipment, where the tape mechanism is an integral part of the recorder. Several makes of British recorder have decks made by organizations specializing in this field, such as Collaro, Truvox and Wearite. The recorders show many variations in the style of cabinet, facilities provided, design of amplifier, type of loudspeaker used, etc. However, to give some idea of the principles

FIG. 9.6. *General layout of tape deck.*



involved, the layout of a typical tape deck is illustrated in Fig. 9.6.

The tape deck must arrange for a constant drive of the tape past the various heads, whilst maintaining control of the tape on the spools. The actual drive is obtained by means of a very accurately machined capstan which is driven at a constant speed against a pinch roller, the tape being pressed between the two, as shown. The capstan is either driven by the motor through a friction coupling or, in less elaborate machines, is mounted direct on to the spindle of the motor. Whatever the method adopted, a weighty flywheel is always employed on the capstan shaft to reduce any irregularity in the bearings or motor, and thus assist in maintaining constant speed.

In some cases the tape speed is controlled by interchangeable capstans. This is the case in the Truvox deck, where speeds of 3¾ in. per second and 7½ in. per second are available. The slow speed gives a playing time of 60 minutes per track for standard tape and 90 minutes per track for long-playing tape, while the playing time is reduced by 50 per cent at 7½ in. per second. (That is for 7 in. diameter spools containing 1,200 ft. of standard tape and 1,800 ft. of long playing tape, which is thinner than the standard.)

Various other arrangements are in use for speed change, including stepped drive wheels, of similar arrangement to those incorporated in record players, and control by switching the field windings in the motor (Grundig, for example). Apart from the two speeds mentioned above, a speed of 15 in. per second is also available on some models as a means of securing a really first-class high-frequency response. A speed of 1·875 in. per second is sometimes used for the recording of material where quality of reproduction is not a prime factor.

Whilst the tape is being driven past the heads the tape take-up spool is slightly under load—sometimes through a mechanical or electromagnetic clutch—to ensure that the recorded (or reproduced) tape is adequately controlled. In some cases, the supply spool is also loaded lightly to avoid spilling of the tape from the spool due to overdrive.

Facilities are available for fast rewind or wind-on. A separate motor or motors are sometimes used in the more expensive equipment, while very successful results are possible by employing for these functions the capstan drive motor. The Truvox deck can re-spool a 1,200 ft. reel of tape in less than

FIG. 9.7. *Collaro Mk. III tape transcriptor.*

one minute, using three B.T.H. shaded-pole motors. Two motors are embodied in the Collaro deck, while the Grundig works very successfully with one motor.

There are other refinements, and various combinations of switching and mechanical functions, which need not be described here. It is now becoming common practice to incorporate such things as "pause control" (for stopping the transit of the tape past the heads and applying the breaks to the spools whilst leaving all the switches and their mechanical functions in the selected positions), "playing time indicator" (facilitating the location of any recorded passage), and "instant track change", as already described.

The well-known Collaro Mk. III tape transcriptor is illustrated in Fig. 9.7, while the extensively used Truvox Mk. IV tape deck is shown in Fig. 9.8. Note the three digit counters on both decks, which serve admirably as place locators. The Collaro deck has facilities for instantaneous changes of track, while the same facility is available on the Truvox deck by spool transposition,

FIG. 9.8. *The Truvox Mk. IV tape deck with place locator.*

FREQUENCY RESPONSE

If a tape record is made with an amplifier possessing a flat response, and it is also replayed on a linear amplifier, the frequency/output curve of the recording will be similar to that shown in Fig. 9.9 (*a*). Up to the point of maximum response, the output rises steadily at the rate of about 6 db per octave. This is because the voltage across the recording head, which is essentially an inductive element fed through a resistor to provide a recording current which is constant irrespective of frequency, doubles every time the frequency is doubled.

Towards the peak of the response, however, the rate of rise tails off slightly, and beyond maximum response it falls fairly quickly. This deviation from the "natural" 6 db per octave rise is promoted by high-frequency losses inherent in the system generally. For example, there is the effect of the gap, the speed of the tape past the head, the coercivity of the tape and capacitive losses in the amplifier proper. Also, the very short magnets which correspond to high recorded frequencies tend to demagnetize themselves, though high-coercivity tape is less prone to this trouble. A reduction of high-frequency loss owing to this cause of some 8 db may be expected by the use of a good-quality high-coercivity tape (it will be remembered that coercivity is a measure of the ability of the tape to resist demagnetization).

It will be clear that the response relative to frequency is governed by the speed at which the tape passes the heads. The curve given in Fig. 9.9 (*a*) is for a speed of 7½ in. per second. From this curve clearly emerges the fact that some form of equalization is called for on replay. In practice, equalization is applied during the recording process as well as during replay. Whilst recording, the recording head is invariably fed through a filter which provides a treble lift as a means of combating some of the high-frequency losses. This filter may be either a tuned device, including an inductance, or a relatively simple resistance-capacitance network.

On replay there is a circuit which gives a rise in bass response and a further circuit which gives a high-frequency lift to provide additional compensation for losses in the replay head. Fortunately, there is some standardization with regard to recording and reproducing characteristics. These are dealt with in British Standard 2478:1954, which accepts the recommendations of the Comité Consultatif International des Radio-communications (C.C.I.R.) for programme interchange on tape. For tape speeds of 15 in. per second and 30 in. per second, the specified frequency-response curve represents a bass rise equivalent to that secured by a series-connected resistor and capacitor having a time-constant of 35 microseconds. For a tape speed of 7½ in. per second a time-constant of 100 microseconds is recommended.

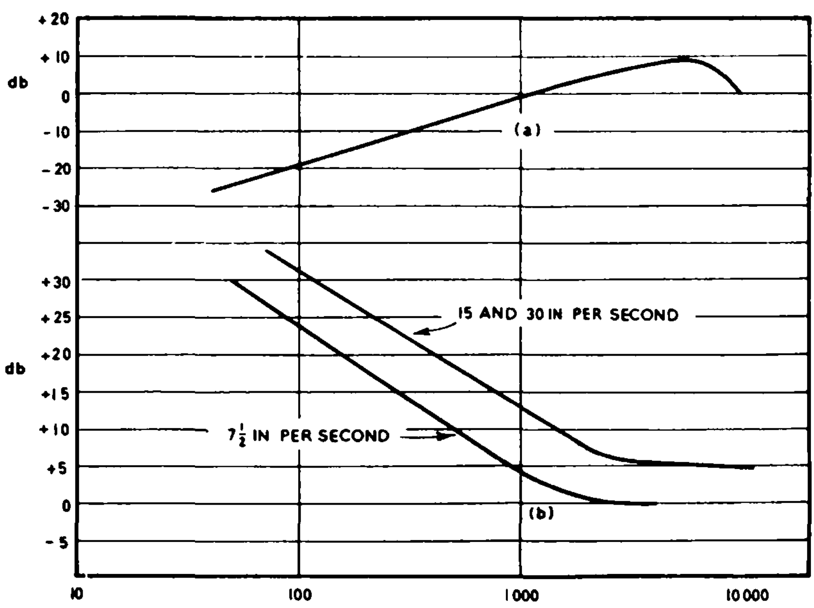


FIG. 9.9. *Frequency characteristics of tape recording.*

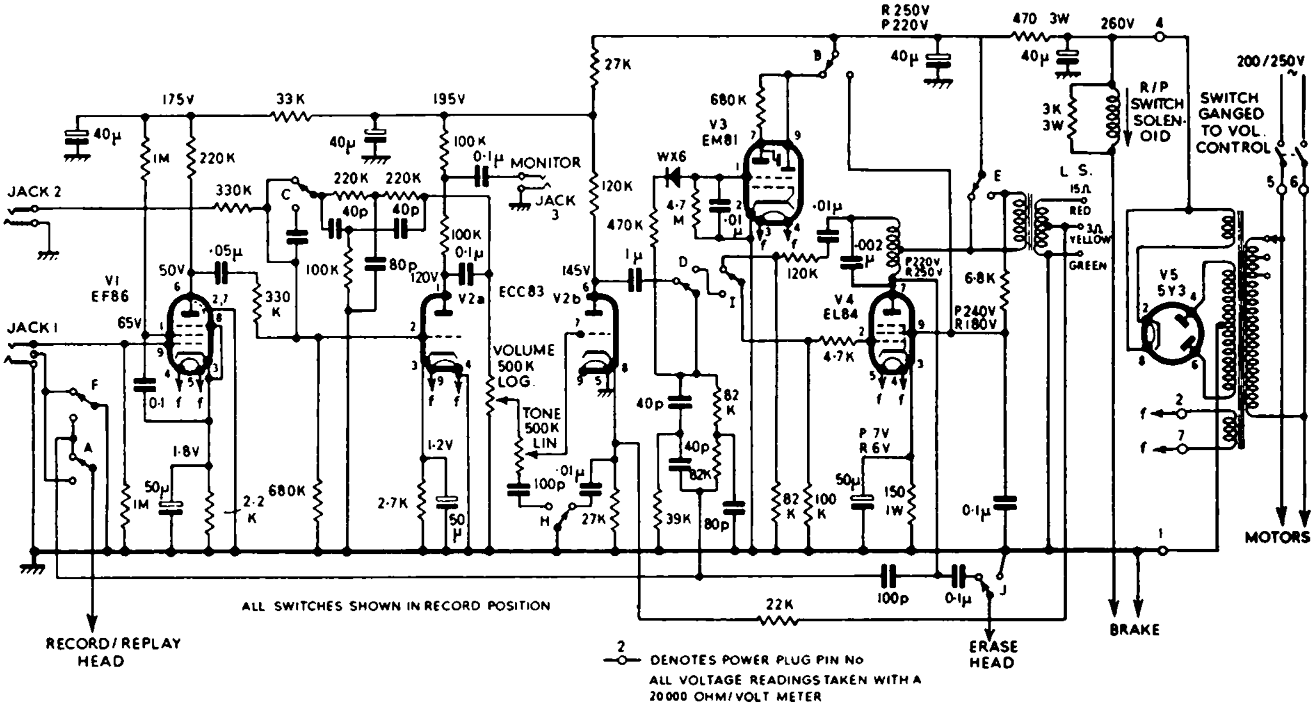


FIG. 9.10. *Circuit diagram of the Truvox Type K tape-recorder amplifier.*

The curves obtained by these time-constants, assuming an "ideal" reproducing head, are shown in Fig. 9.9 (*b*). In practice, these time-constants may have to be altered slightly to suit the particular head in use. It is usually necessary to consider the equalized pre-amplifier and reproducing head as a complete unit with regard to shaping the overall response. All tape records, including "Stereosonic" versions, are recorded to suit the C.C.I.R. characteristics, so it is as well for the recorder to maintain this replay response within the tolerance of plus and minus 3 db, at least.

AMPLIFIERS

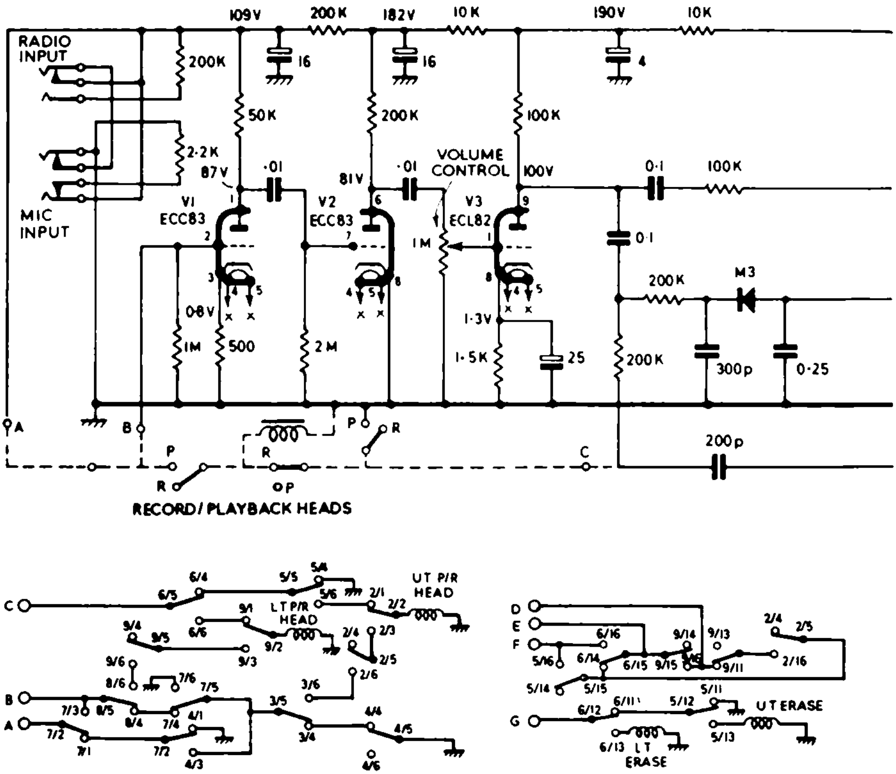
Fig. 9·10 shows the circuit of the Truvox Type K tape-recorder amplifier which, apart from being used in certain Truvox tape recorders, can be obtained as a separate unit if an enthusiast requires to make up his own recorder, or if it is required to add another channel for the playing of "Stereosonic" tape records.

It should first be noted that the various switches, representing record/replay, are shown in the "record" position. The first stage, V1, serves on the "record" position as a high-gain microphone amplifier. The microphone signal is applied to the control grid of the valve, which is a low-noise pentode, by way of jack 1. It will be noted that this is a high-impedance input and suitable for a crystal microphone or an electromagnetic type embodying an impedance-matching transformer.

The amplified signal is fed to the grid of V2A (half of Mullard ECC83) through the 0·05-mF coupling capacitor and 330k resistor. The resistor-capacitor network between the anode and grid of V2A forms a parallel-T feedback system which, being frequency-selective, gives a top lift to compensate for high-frequency losses, as already described. The frequency-compensated output from V2A is passed by way of the volume control—the tone control being out of circuit on "record"—to a further amplifier V2B (the other section of the ECC83). This section also functions as the recording output valve, the signal being fed through switch D, through a further resistor-capacitor network for additional head compensation, to the record/replay head through switch A.

The signal at the anode of V2B is also applied to a rectifier WX6, and the d.c. voltage developed across the 4·7-megohm load resistor is applied to the recording-level indicator valve V3, being brought into operation by switch B.

During the recording process, valve V4 (EL84) serves as the high-frequency oscillator. L1 is the oscillator coil, which is resonated by the parallel 0·002-mF capacitor. Switch I brings the oscillator into circuit by completing the coupling between the grid and the anode. It will be seen that the oscillator voltage is taken from the anode of the valve and is fed through a

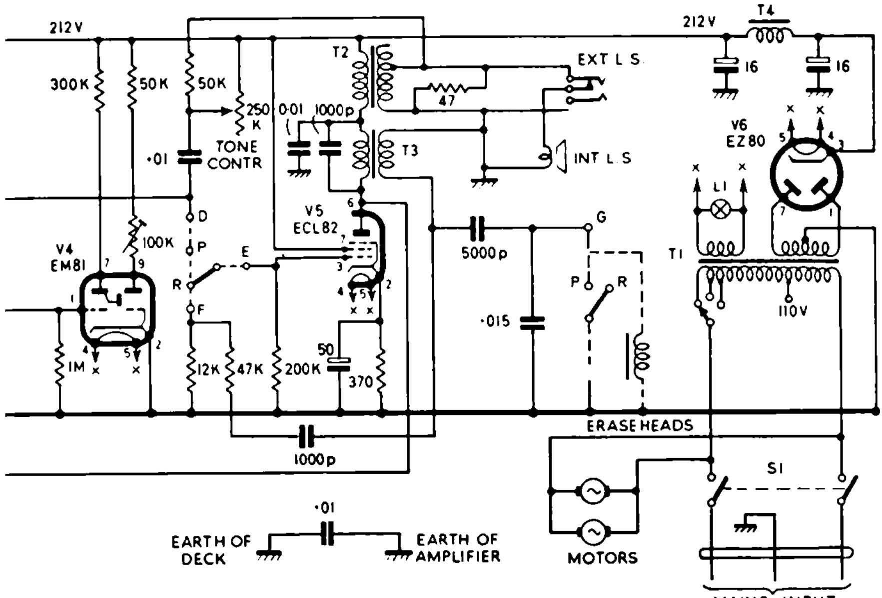


SWITCH CIRCUIT THE CONTACTS ARE SHOWN IN THE RELAXED (i.e STOP) POSITIONS

FIG. 9.11. *Circuit diagram of the Sound Sales A20 tape recorder. Connexions shown as dotted lines in the circuit diagram are simplifications of switch connexions, the true connexions being given in the lower diagram.*

capacitor of 100 pF to the recording/replay head by way of switch A. The erase head is also energized by the oscillator current through the 0·01-mF coupling capacitor and switch J. The bias voltage is fixed at 48 kc/s, and the level of the signal is of the order of 150 volts at the erase head and 80 volts at the recording head. Both heads are of the high-impedance type and thus match directly into the circuits without transformers.

The amplifier is powered by means of a full-wave h.t. rectifier valve, V5 (5Y3). The mains transformer carries two l.t. windings, one for the heaters of the valves and the other for the heater of the rectifier. Although there is no smoothing choke, a remarkably low level of hum (45 db down at 4 watts) is achieved by the use of four 40-mF electrolytic capacitors and associated filter resistors.



CIRCUIT SHOWN IN RECORD POSITION

The "record/replay" switch is actuated by a relay whose winding —denoted R/P switch solenoid on the diagram—is energized by the instantaneous shorting together of the two wires labelled "brake" on the diagram. This action trips the relay switch so that it returns to the "replay" position. It is necessary to re-set the switch to the "record" position by pressing the appropriate button on the amplifier control panel.

On the "replay" position the record/replay head is switched into the grid circuit of V1, which now acts as the reproducing-head amplifier, by the changing-over of switches F and A. Switch C alters the coupling response so as to give a further lift of treble, switch H brings in the tone control and removes the 0·01-mF capacitor from across the cathode resistor of V2B. The oscillator is switched off by switch I and switch D disconnects the head equalizer and couples the signal at the anode of V2B to the control grid of V4 (V4 now acts as the sound output valve). Switch B disconnects the level indicator valve, switch E brings in the output transformer and switch J short-circuits the erase head which, of course, is not required on replay.

It will be seen that negative-voltage feedback is applied over two stages from the output transformer to the cathode of V2B. Since the 0·01-mF capacitor is disconnected from across the cathode resistor, the feedback is not frequency-selective as it would be if the capacitor was left in circuit.

The output transformer provides for loudspeakers of either 3 ohms or

15 ohms impedance. The signal outlet at the anode of V2A by way of jack 3 permits monitoring of the recording or recorded signal. If required, the signal at this point can be applied to the input circuits of a separate hi-fi amplifier. The signal level at this point is approximately 0·5 volt at 500 ohms impedance.

Whilst it will usually be found necessary to employ V1 as the microphone amplifier during the recording process, the sensitivity at jack 1 being in the region of 1–2 mV, a higher signal level should be applied by way of jack 2, where the sensitivity is something like 500 mV; for example, a gramophone pick-up signal or a radio signal can be introduced here. The Truvox radio jack is useful for the recording of radio programmes, but this should be inserted into jack 1 owing to the low output signal from the crystal detector.

SOUND TYPE A20 TAPE RECORDER

The circuit diagram in Fig. 9.11 of this recorder shows that it has much in common with the Truvox circuit. Instead of a low-noise pentode, however, two cascade-connected triodes V1 and V2 (Mullard ECC83) provide the necessary high gain for use as a microphone amplifier during recording and a replay-head amplifier during replay, these functions being governed by the setting of the changeover switches, as before.

The ECL80 valve is arranged so that the triode serves as recording amplifier and the pentode as oscillator during the recording process, and as voltage amplifier and output valve respectively during replay. The EM81 acts as a recording-level indicator, the controlling signal being obtained from the recording signal (in the anode of V3) and rectified by the metal rectifier M3. There are two inputs, but these operate in the grid circuit of the first triode.

ADDING A TAPE DECK

Many enthusiasts, already in possession of a good hi-fi amplifier, may wish to add a tape deck without going to the expense of purchasing a complete tape recorder. This can be done fairly easily by utilizing the existing amplifier. However, it is not usually possible to connect the replay head direct to one of the inputs of the amplifier, even though one may be marked "tape input", owing to the lack of gain and need for special equalization. From the replay point of view, an equalized-head amplifier is required, whose equalization response can be switched to suit the various tape speeds in accordance with the C.C.I.R. characteristics.

A single low-noise pentode, such as the Mullard EF86, can usually be arranged to provide sufficient rise of signal from the replay head to drive a control unit of most makes. The stage will incorporate an equalizing network, preferably of the selective-feedback type already dealt with. Special attention should be given to the layout of the components and to the common-negative

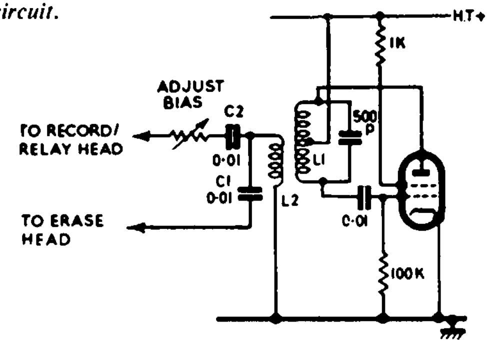


FIG. 9.12. *Typical bias oscillator circuit.*

connexions on the chassis, so as to avoid circulating earth currents which become manifested as hum from the loudspeaker.

The precise characteristics of the equalization will depend to a large degree on the type of head adopted, but manufacturers of tape decks and heads invariably issue a suitable circuit. Such an arrangement will permit the reproduction of tape records, but if a facility for recording is required as well, a recording amplifier and oscillator will also be necessary.

The reproducing-head amplifier can serve as the microphone amplifier for recording. This will have sufficient gain, since the output from an average reproducing head is little more than 1–3 mV, which is more or less the same as the output from a microphone.

A recording current of the order of 100 microamps is usually required and this can be obtained with little trouble from a simple triode stage, which may be fed from the microphone amplifier. It will be necessary to have provision on the amplifier to switch out the bass-lift equalizing and to introduce a form of pre-emphasis. Alternatively, the amplifier can be switched to have a linear response, and the pre-emphasis introduced in the network from the recording output valve to the recording head. Additional pre-emphasis can, if required, be introduced between the coupling from the microphone amplifier and the recording output valve; this can take the form of a parallel-T network.

The considerable amount of high-frequency power required for efficient erasing can only be obtained from a fairly large power pentode or tetrode, such as the 6V6 or EL84. Oscillators in general use employ either the series-fed Hartley or the Colpitts circuit. A typical Hartley circuit is given in Fig. 9.12. Here L1 is the tapped oscillator winding, which is tuned within the frequency range of 30–80 kc/s by the 500-pF shunt capacitor. L2 is the coupling coil which feeds the oscillator current into the erase head through C1 and into the record/replay head through C2 and R1. R1 is variable so that the optimum value of high-frequency bias can be selected for the type of tape and the signal voltage employed.

For the purpose of erase the value of the bias in the erase head is not critical, provided that the bias current is sufficient to secure complete erasure. The current in the recording head is, however, much more critical. As the bias current is increased so the response falls, but as the recording current is increased so a greater level of bias current is required to alleviate distortion. It is usually necessary to perform a series of tests in order to establish the best setting for the bias level control. The aim should be to achieve best signal-to-noise ratio consistent with the minimum of distortion and the best high-frequency response.

A distorted bias signal will tend to impair the signal-to-noise ratio to a large degree. The bias oscillator should be capable of producing a pure sine wave at full output, and to assist in this respect professional equipment often uses a push-pull oscillator circuit.

SERVICING TAPE EQUIPMENT

Faults likely to develop in tape recorders are of two distinct types: there are mechanical faults relating to the tape deck proper, and electrical faults associated with the amplifiers, oscillator and various equalizing networks. Faults of amplifiers—particularly those resulting in complete failure of recording, reproducing or both—can be located with reasonable ease by adopting the techniques described in Chapter 4.

It helps considerably if a circuit diagram of the defective instrument is available, for then a study of the various stages will reveal the valves and components which are common to both recording and reproducing, and if the cause of failure of both services is being sought the components most likely to be responsible can be examined in more detail.

For example, failure of V5 in Fig. 9.11 will affect both recording and reproducing, since this valve serves both as sound output on replay and bias oscillator on record. The replay section will be totally dead, assuming that V5 has failed completely, but on the "record" position the signal-level indicator will function normally. This may lead to a suspicion of trouble in the sound amplifier or associated networks. If a trial on another machine of the tape that has supposedly been recorded results in severe distortion owing to the lack of recording bias, the diagnosis will be proved accurate. In practice, it is usually the faults of a more subtle nature which cause difficulty, and these can be of both electrical and mechanical origin.

A poor high-frequency response, particularly from tapes recorded on another machine or from a tape record, is often caused by maladjustment of the recording/replay head. It is essential that the angle that the gap of the head makes with the direction of motion of the tape is the same both on record and replay. Whilst this angle will obviously remain consistent when the tape is recorded and reproduced on the same machine, it is desirable to know that a recording is going to be satisfactory when played on a different machine.

For this reason the head is best adjusted for optimum high-frequency response with a standard test tape or a tape record having plenty of "top". Test records giving a tone of 10 kc/s or 12 kc/s are available for this purpose. The azimuth adjusting screw in proximity to the head or head securing bracket should be slowly turned first one way and then the other whilst the output from the test tape is being observed on an output meter or a.c. voltmeter. The point giving maximum output should be selected.

If the recording/replay head becomes slightly magnetized a marked increase in background "hiss" on replay results. Theoretically, the head should not become magnetized since a large-value time-constant is used in the h.t. feed to the oscillator valve which causes a gradual decay of oscillator current in the heads, and thus avoids surges which are likely to magnetize the heads. However, in practice small current surges occur which promote the trouble; there is also the possibility of the heads coming into contact with a magnetized screwdriver or a magnetic field. Again, there is the possibility that the head may be checked for continuity with an ohmmeter.

Whatever the cause, it is necessary to demagnetize the head to reduce the "hiss" and to prevent a constant flux being impressed upon the tape. The best way of securing this is by the use of a "defluxer". An ideal instrument of this kind is the Wearite Defluxer (Fig. 9.13). This has a special shaped pole piece which can easily be brought into contact with the head pole pieces. The instrument is mains operated, and at the end removed from the pole piece is a press-button control switch.

In operation, the pole piece is brought into contact with the head and the button depressed. The pole piece is then gradually taken away from the head, but the button should not be released until the head is outside the influence of the field. In this way complete removal of residual magnetism is achieved. The instrument is also useful for removing residual magnetism from tape guides, pulleys, etc., which may fall in the path of the tape.

If the tape is not making close contact with the head pole pieces, the effect is the same as an increase in the size of the gap, and a loss of the higher frequencies results. If this trouble is suspected, the heads should be examined for wear, and if very badly worn should be replaced. All traces of accumulated particles of oxide coating from the tape should be very carefully removed from the head pole pieces. A soft brush can be of considerable assistance in this, but cleaning fluid should not be used unless specifically indicated

FIG. 9.13. *The Wearite defluxer.*

in the manufacturer's instructions. The tape guides and pulleys should be examined for adjustment to ensure that the tape is being evenly transported past the heads.

Excessive wow should lead to checking the pressure between the capstan and the pressure roller, drag on the spools due to an incorrect clutch adjustment, the regularity of the various drive faces (particularly the rubber rollers and jockey pulleys), the lubrication of the bearings and the alignment of the flywheel and motor spindle. Worn rubber or composition friction-drives and oil on the surfaces of the drives are common causes of the trouble, especially of the lower-frequency wow. Flutter, at a higher frequency, may also be caused by an eccentric drive wheel, or by "snatching" of the tape owing to irregular binding of the supply-spool bearing.

If a double-beam oscilloscope, an audio oscillator and a 1,000-c/s test tape are available a useful test for wow can be made. The test tape should be played on the recorder and the output signal monitored on one beam. The other beam should be connected to the audio oscillator, which should be adjusted for 1,000 c/s. Adjustments should be made to the controls of the oscillator and oscilloscope until the two waveforms occur approximately in step. Wow in the recorder will cause a regular movement back and forth of the waveform from the recorder, while the waveform from the oscillator will remain constant. The amount of shift relative to the stationary trace can easily be observed and the percentage in speed variation ascertained. For example, a shift of 1 c/s represents a wow content of 0·1 per cent.

RECORDING FROM RADIO

It will often be required to make a recording from a radio signal. One way of doing this is to simply stand the recorder's microphone in front of the receiver's loudspeaker. This is not a desirable method owing to the introduction of distortion from the receiver's output stage and loudspeaker; the recording will also be coloured by the acoustics of the room.

Another method is to connect an input channel on the recorder to the extension-loudspeaker sockets on the receiver. This also is undesirable because the distortion in the receiver's output stage will be recorded. By far

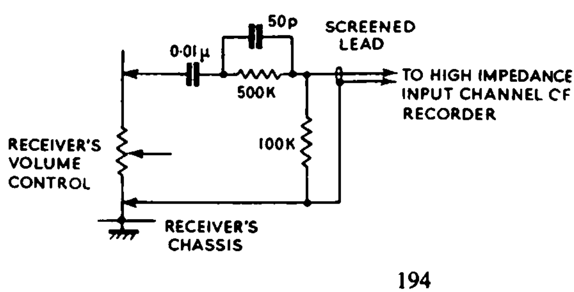


FIG. 9.14. *Method of extracting a high-impedance low-level signal from a receiver. (Not suitable for a.c./d.c. receivers, or where the chassis is connected to one side of the mains supply.)*

FIG. 9.15. *Tape recorders and sound equipment installed for use at the Duke of Edinburgh's Study Conference at Rhodes House, Oxford* (1956).

the best method is to take a connexion from the detector circuit of the receiver and feed the signal at this point into an input channel on the recorder. The idea is shown in circuit form in Fig. 9.14.

To conclude this chapter a photograph is reproduced of a galaxy of tape recorders and sound equipment (Fig. 9.15) which was employed at the Duke of Edinburgh's Study Conference at Rhodes House, Oxford, in 1956. This shows two Grundig Stenorette recorder dictating machines—a Grundig TK820 and a Grundig TK819, both being high-quality home recorders—two Pamphonic amplifiers, a B.S.R. amplifier, a Trix amplifier, an M.S.S. disk recorder, a bulk tape erasure, and sundry other items of equipment. In the centre, on the upper shelf, will be seen a microphone mixer unit which was designed by the author to control the various signal inputs from the conference microphones and BBC sound channels.

Tape recordings of every word of the conference were produced and facilities were available for the immediate production of disk recordings from the master tapes—which in overall length represented some 20 miles! The Stenorette dictation recorders aided with the transcription of all the principal speeches. All these facilities were provided by the Electronics Division of Lowe and Oliver, Ltd., of Oxford, to whom the author is indebted for permission to publish this photograph and also Fig. 5.24.

CHAPTER 10

Stereophony

THE endeavours of hi-fi enthusiasts, designers and manufacturers of hi-fi equipment have resulted in a single-channel, or monaural, reproducing system of extremely high standard. Harmonic distortion has been cut to less than 1 per cent, the frequency response has been extended well beyond the audible range as a means of improving the transient performance, while the dynamic range and power-handling capabilities of modern hi-fi equipment leave little to be desired. Nevertheless, there is something lacking!

With a monaural system, the sound which is picked up by the microphone in a concert hall, for example, represents a sample of the sound pressure which exists solely at the position of the microphone, and this sound eventually arrives at the two ears of the listener at the reproducing end by way of the loudspeaker. In other words, there is a complete, single channel between the sound source and the listener, and this is so irrespective of whether the connexion between the microphone and the loudspeaker consists of a radio link, directly-connected cables or a recording system of some kind. The idea is illustrated in Fig. 10.1.

This arrangement does not fit in with things as they are in reality, for at a concert performance the sound from the various instruments embraces a wide area in front of the listener and, even without the use of his eyes, he finds no difficulty in establishing the position of the instrumentalists. The reason for this is that both ears collect the sounds coming from the orchestra at various angles. The brain compares several factors of the sounds

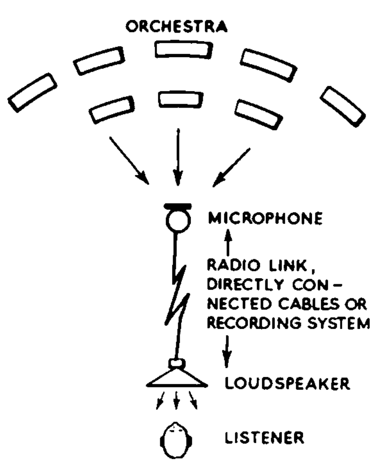


FIG. 10.1. *The monaural system of sound reproduction.*

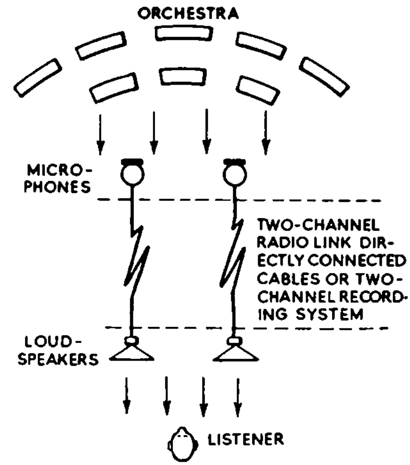


FIG. 10.2. *The stereophonic system of sound reproduction.*

picked up by the two ears, and from the information thus abstracted brings into operation a form of sound location sense. Apart from the relative intensity of the sounds, this sound location function is dependent also upon the relative phase of the sounds at the two ears. It will be appreciated that there is a time difference between the sounds, and this may or may not be the same thing as a phase difference; the former term, however, applies essentially to sounds of a random and transient nature, while the latter applies to repetitive sound waves.

Sounds originating at the extreme right and left of a listener are dealt with by the right and left ears respectively, while sounds originating between these two points have their positions fixed (so far as the brain is concerned) by one ear picking up more sound than the other and by comparison of phase. It will be clear from this that a single-channel system cannot promote operation of the sound location sense, and that as a result of this the illusion of "perspective" and "spaciousness" of an orchestra is lost.

The problem is not solved by the use of two or more microphones connected in series or parallel at the studio end and two or more loudspeakers similarly connected at the reproducing end, since the individual signals, though they may differ in phase and amplitude, lose their separate identity in the single-channel network. With such a system, the sound simply appears to come from the loudspeaker nearest to the listener or from the loudspeaker which is radiating the greatest volume of sound; true perspective is not given because all the loudspeakers reproduce in proportion the same sound. Two or more loudspeakers in a single-channel system can help considerably, however, in spreading the sound over a large area, provided that due attention is given to their phasing.

THE BINAURAL SYSTEM

Correction of the loss of spaciousness and perspective is possible by the use of two isolated channels, as shown in Fig. 10.2. The greater the number of channels, the better the stereophonic effect, but economic considerations limit the number of channels for domestic use to two (the cinema industry is able to afford multi-channel systems). Fig. 10.2 shows that, in effect, the listener's two ears have been extended into the studio or concert hall where they are represented by microphones. Because the differences in the character of the sound picked up by the microphones are conveyed to the loudspeakers, the sound-location sense of the listener is brought into action, and he is given the impression of the spaciousness and perspective of the orchestra as if he were actually in the concert hall.

Consider a two-channel system with the microphones placed about 6 ft. apart, in front of which are placed about six people in line. If the loudspeakers are similarly placed in the listening room with regard to separation and channel, and the people in line at the front of the microphones are instructed to clap their hands in turn, starting with, say, the person nearest the right-hand microphone, the listener at the loudspeakers will obtain the impression of the sound starting at the right-hand loudspeaker, moving across between the loudspeakers and finishing at the left-hand loudspeaker.

This is a stereophonic trick which can be used with great success in demonstrating stereophonic sound reproduction; usually, however, something more elaborate than hand-clapping is featured—an express train rushing through a station, a game of table tennis or something which illustrates vivid movement. The sound source will appear to be in the centre between the two loudspeakers when the outputs are balanced. An announcer standing in the centre between the two microphones would give this effect. If he moved towards the left-hand microphone the sound would increase from the left-hand loudspeaker and decrease from the right-hand one; the converse effect would occur if he moved towards the other microphone.

This impression of movement, coupled with phase-comparison of the sound from the loudspeakers, produces a two-channel stereophonic effect of remarkably high standard. The illusion is somewhat enhanced by the reverberant sounds also being put into their correct perspective, so that they do not appear to come from exactly the same place as the original sound, as is the case with a single-channel system.

Once stereophony has been experienced, it is extremely difficult to settle down again to single-channel listening—a big attribute of stereophony is the marked reduction in listening fatigue. The placing of the loudspeakers, their phasing and the balance of sound are important factors which have to be considered with the greatest care if the best results are to be secured.

MICROPHONE PLACING

The use of fairly widely spaced microphones, as shown in Fig. 10.2, is not always to be recommended. This is because the space between the two loudspeakers may not be adequately balanced in relation to the level of sound from them. The sound may appear to be concentrated either side of the listener, with a distinct weakness in the centre. This may prove distressing when the sound source is situated approximately in the centre between the two microphones, since then the sound may appear to jump from one loudspeaker to the other at the listening end, particularly if the sound source happens to be someone moving about.

One method of overcoming this effect, developed by Philips, made use of an artificial head with microphones instead of ears. The masking effect of the head was relied upon to provide the amplitude and phase differences of the sound signals passed over the two channels. This scheme provides a stereophonic illusion at the higher frequencies only, since the masking given by the head reduces in efficiency as the frequency is decreased and as the wavelength of the sound becomes comparable with the head dimensions.

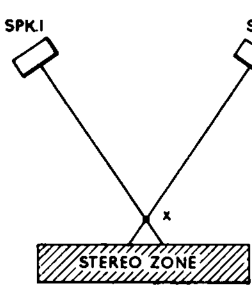
A system which is in current operation by E.M.I. for the production of their "Stereosonic" tape records calls for the microphones to be situated as close together as possible, one above the other. Use is made of pressure-gradient microphones orientated so that their directions of maximum pick-up are at right-angles, and positioned in relation to the centre of the sound source so that the maximum pick-up axis of each microphone falls at an angle of 45 deg.

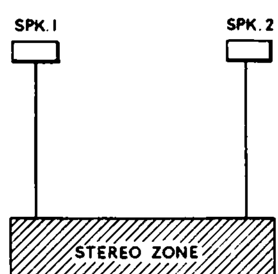
Another scheme which has received some attention both in America and Great Britain is the placing of two microphones, one above the other, over the orchestra. The lower microphone is arranged to respond to the direct sounds, while the upper one is shielded from the direct sounds, but responds to reverberant and indirect sounds. It has been claimed that this method enhances the "presence" and atmosphere of an orchestral reproduction.

LOUDSPEAKER PLACING

The subject of loudspeaker positioning is a controversial one, and also rather a problem if the most desirable listening position technically is not to conflict with day-to-day household functions.

The arrangement shown in Fig. 10.3 has much to commend it, even





(Left) FIG. 10.3. *This method of loudspeaker positioning has much to commend it.* (Right) FIG. 10.4. *Another form of loudspeaker set-up.*

though the area of stereophonic effect may be somewhat limited. The point of optimum effect is shown at X, but movement from this ideal position disturbs the balance of sound and gives the impression of movement of the sound source. Within the stereo zone one can walk around without unduly disturbing the effect of balance. This is because if one moves from, say, the centre of the stereo zone to the left the resulting increase in distance from loudspeaker 2 is offset by moving into its beam of greater sound level, while the decrease in distance from loudspeaker 1 is corrected by moving away from its beam of greater sound level. This effect was investigated in some detail by G.E.C. engineers during the course of their experiments in stereophony. (G.E.C. gave its first public demonstration of stereophonic sound reproduction at the National Radio Show in 1951, magnetic tape being the recording medium used at that time.)

The precedence effect also has some bearing on the balance and apparent location of the sound source, relating sound intensity with relative time of arrival of the sound at the two ears. For example, if the same sound is radiated from two sources, the brain senses that the sound comes from the source whose sound reaches the ears first. However, this effect can be balanced by increasing the intensity of the delayed sound accordingly.

The set-up shown in Fig. 10.4 also serves reasonably well, particularly in smaller rooms. Reflections from walls and furniture also disturb the stereo balance when listening under domestic conditions. In order to combat this trouble it is often necessary to experiment with the placing of the loudspeakers, and best results may be secured with the loudspeakers arranged in some odd pattern which appears to have no theoretical validity. There should be no obstructions between the loudspeakers and the listener, however, and in order to satisfy this condition it sometimes helps to mount the loudspeakers above the level of the furniture.

LOUDSPEAKERS FOR STEREO

Experiments have been carried out with several makes of loudspeaker and enclosure in different combinations, and in all cases the stereophonic effect was secured once the sound levels were correctly balanced. In one test a small battery portable receiver tuned to the BBC Home Service served as the left-hand channel and a table television receiver as the right-hand channel in connexion with the BBC's stereophonic experiments. The effect was truly astounding; the inherent distortion of the non-hi-fi receivers appeared to disappear, and the reproduction resulting from the two unmatched loudspeakers was definitely of hi-fi quality.

This would appear to bear out the opinions of other workers in the field; in fact, it has been suggested that a two-channel stereo system with a response good to 6 kc/s sounds better than a single-channel system with a response extended to 15 kc/s. However, the best results are undoubtedly gained by the use of two hi-fi channels with matched loudspeakers and amplifiers. The stereo effect is most marked in the middle and higher range of frequencies, and at these frequencies it is desirable for the loudspeakers to have omni-directional characteristics. Ralph West ("Hi-Fi Year Book", 1958) suggests the use of reflecting cones serving as high-frequency diffusers, with the loud-speaker enclosure preferably lying on its back. This idea has been tried, and works admirably.

Loudspeaker phasing is not as critical in the author's opinion as is made out by some authorities. Anti-phase conditions undoubtedly result in a loss of heavy bass, as would be expected, but the overall stereo effect is not severely affected, though there is a slight unevenness of the movement of the virtual sound source between the loudspeakers. Ralph West says "out-of-phase usually produces three sources—ahead, and one at each loudspeaker".

What can be most disconcerting is unbalance of hum levels from the loudspeakers. During the course of recording the BBC's stereo experiments by way of V.H.F.-F.M. channels, one channel suffered badly from modulation hum due to the close proximity of e.h.t. cables. On playback (the stereo was recorded remarkably well) the hum appeared to be removed from the actual programme material and to exist close to one ear of the listener, that nearest the humming loudspeaker.

EQUIPMENT REQUIRED FOR STEREO

In order to create the stereo effect in the home it is necessary to employ two completely separate hi-fi systems. Starting from the listener and working backwards towards the signal source, we require two loudspeaker systems, two power amplifiers to drive them and two pre-amplifiers or control units to feed the power amplifiers. Then there is the most important component —the two separate signals representative of the programme material.

For use in the home, the programme signals will be derived from either tape or disk records, and it will be shown later how the two signals are maintained separately on a single length of tape and on a common groove of a disk record. It is also possible to utilize two radio channels, one to carry the right-hand signal and the other to carry the left-hand signal. This method has been used on several occasions by the BBC in their experimental stereo-phonic broadcasts. The television sound transmitters as well as the medium-wave and V.H.F.-F.M. transmitters have been used on these tests in various combinations so as to allow the maximum number of listeners to participate.

It is not the intention of the BBC to make a permanent feature of stereophonic broadcasts at the present time, but radio-derived stereophonic sound is available to listeners in New York by switching on both radio and television receivers at the same time.

FIG. 10.5. *Pamphonic Model S1 loudspeaker, specially designed for use in pairs with the Pamphonic stereo amplifier Model 3,000.*

With regard to loudspeakers for stereophonic applications, we have already seen that these need not be accurately matched, or even of the same type, to secure good-quality stereophony. Obviously, the better the loudspeaker systems, the better the quality of reproduction, but since difficulty may be experienced in the housing of two large reflex enclosures in an average living-room, it is likely that miniature reflex enclosures suitable for table mounting or small console versions will become popular as stereophonic systems become more widely used, and several manufacturers are concentrating along this line. Fig. 10.5 shows the Pamphonic Model S1 stereo loudspeaker, which uses a concentric cone unit which is adequately loaded by the cabinet, and is specially designed for use in pairs with the Pamphonic Model 3,000 stereo amplifier.

Dual-channel, or stereophonic, amplifiers complete are now available, and these represent an ideal choice for the enthusiast just entering the field of hi-fi. Those already in possession of a single-channel hi-fi system are catered for, however, as stereo control units are also available for handling the two programme signals, derived from radio, tape or disk, and passing them on to two (preferably matched) power amplifiers.

STEREO FROM DISK

There have been several attempts at producing stereo disk records, but the system which is now in use dates back to A. D. Blumlein's experiments around 1929, when a scheme for the simultaneous recording of two separate sound tracks in a common groove was evolved. The idea was patented in 1931, but at that time it did not represent a commercial proposition and was shelved. It was not until 1957 that the scheme was again brought to the notice of the public: it was demonstrated at the London Audio Fair and the B.S.R.A. exhibition by A. R. Sugden and in America by London Records (a Decca associate) and Westrex. Its re-introduction coincided with the growing interest in stereo reproduction, fostered by stereo tape records, and the advanced state of development of the gramophone industry as a whole.

The system of stereophonic disk recording as first expounded by

Blumlein and now in current use requires only a single recording stylus and recording head to produce recordings of both stereo channels in a common groove, and a single pick-up with only one stylus to reproduce both channels of the recording.

It will be remembered that a disk record can be made in two ways: the conventional method whereby the modulation is imparted due to the recording stylus oscillating to-and-fro in sympathy with the applied modulation, and the hill-and-dale process whereby the modulation affects the depth of the cut and the groove remains straight with no side-to-side curves. Basically, stereo records are produced by combining both of these methods, so that one channel modulates the groove laterally, as with an ordinary record, and the other channel modulates the same groove vertically. The stereo recording head is therefore arranged to have two electromagnetic systems so that when energized by the applied signal one causes the recording stylus to move laterally, while the other causes it to move vertically; the movement of the stylus is thus in two planes, and corresponds to the complex modulation pattern of the two stereo channels.

Similarly, the pick-up has two generating systems which respond individually to the vertical undulations and the lateral displacement of the recorded groove. The voltages generated by the two systems correspond to the signals in the two channels and, since the pick-up is so designed that there is very little interaction between the two systems, connexion can be made direct to the appropriate amplifiers in the usual manner.

The patents of A. D. Blumlein allowed not only for the lateral-vertical system of recording two channels in a common groove, but also for an arrangement known as the 45/45 system. Whilst the modulation is imparted into the groove by a slightly different method to that already described, the 45/45 system gives almost comparable results. With due attention given to the phasing of the two electromagnetic systems in the recording head and of the generating systems in the pick-up, either system can be used for recording or reproduction on the same type of equipment. With the 45/45 system, which is the one adopted, the two stereo channels are recorded at an angle of 45 deg. to the surface of the disk, and each channel is cut towards the opposite wall of the groove, thus producing a single complex groove.

Let us investigate how a stereo record is cut, so as to secure a better understanding of the 45/45 system. In Fig. 10.6 is shown, very much simplified, the essential elements of an electromagnetic stereo recording head. This may be of the moving-iron, moving-coil or even the crystal type, but it would appear that the moving-coil type has much in its favour. The electromagnetic type has two coils, one for each of the two stereo channels, which are denoted A and B in Fig. 10.6. Moving in the coils (or coupled to them in the case of a moving-coil system) are two armature links C and D which are coupled

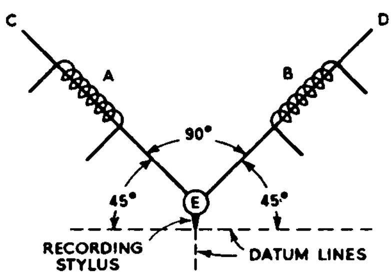


FIG. 10.6. *Simplified diagram of the essential elements of an electromagnetic stereo recording head.*

to the stylus mounting shown at *E*. It will be seen that the armature links diverge from the stylus mounting at an angle of 90 deg. and that each coil when energized would promote a 45-deg. diagonal thrust on the armature links in relation to the stylus tip or surface of the record. Hence the term 45/45.

In Fig. 10.6 the stylus is shown at rest in relation to the dotted datum lines, that is when neither coil is energized. The diagrams in Fig. 10.7 illustrate how the stylus moves or oscillates under various conditions. Let us take the case first of coil *A* (Fig. 10.6) being energized by an alternating current of, say, pure waveform. Armature link *C* will oscillate up and down in the coil in sympathy with the applied signal, rather like an engine piston, but very much faster, and the stylus tip will follow the arrowed line as shown in Fig. 10.7 (*a*). Similarly, if the signal is removed from coil *A* and applied to coil *B*, the movement of the stylus will follow the arrowed line in diagram (*b*), this being opposite to that in (*a*).

However, when both of the coils are energized simultaneously the stylus will follow a path governed by the relative phase of the two signals. For example, if the signals are in phase, then both armature links will move up and down in the coils together and, provided the signals are equal and the coils are balanced, an up-and-down movement of the stylus will result, as shown by the arrowed line in diagram (*c*), which is comparable to hill-and-dale recording. If the signals are in anti-phase, one armature link will move up while the other moves down. This will result in the stylus following the arrowed line in diagram (*d*), which is comparable to ordinary lateral recording.

Now, if it is borne in mind that while the stylus is oscillating in the various modes described a groove is being cut, it will be realized that the impressed modulation will be a combination of lateral and hill-and-dale, the same as that produced by the lateral-vertical system of stereo recording. It will be understood, therefore, that the two systems are basically identical. It comes down to a matter of phasing of the two signals; the inclusion of a simple sum and difference network in the recording head or pick-up circuit will permit the pick-up or recording head of one system to operate with a recording made by the other. It should be noted, however, that the 45/45 system has been accepted as an international standard.

The recommendations in this connexion were withheld from publication

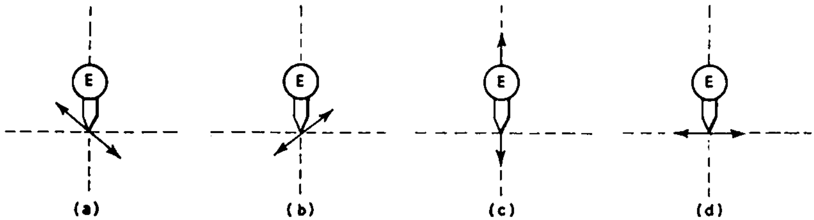


FIG. 10.7. *The arrowed lines indicate how the recording stylus oscillates under various conditions:* (a) *coil* A (*Fig.* 10.6) *energized;* (b) *coil* B *energized;* (c) *both coils energized with in-phase signals;* (d) *both coils energized with anti-phase signals.*

for a time by the European record manufacturers, who were represented at the Zürich Conference in November, 1957, until it was perfectly clear that these harmonized with the views and recommendations of the Record Industry Association of America. There has been close co-operation between the recording companies regarding the standards for stereo disk records, and no doubt this has helped in the rapid graduation of the stereo disk from the laboratory, where it has been available for quite a long time, to the open market.

With regard to other record parameters, such as the diameter of the disk, speed and recording characteristics, the recommendations were that they should follow wherever possible the existing standards for microgroove records laid down by the International Electro-Technical Commission. Thus it seems unlikely that a state of chaos will reign in relation to stereo disks as it did at one time with regard to the parameters of monaural disks.

With regard to right and left channels—as defined by the loudspeaker which supplies the sound to a listener in front of the loudspeakers—the standard is that modulation normal to the surface of the groove wall which faces the axis of the disk should supply the right-hand channel, while the modulation on the groove wall which faces away from the axis of the disk should supply the left-hand channel. This is really not very important from the user's point of view, since it is usually a simple matter to change the position of the loudspeakers or change over the pick-up connexions to ensure that the right-hand modulation actuates the right-hand loudspeaker.

Further, a maximum radius of 5 microns is recommended for the bottom of the groove, but the included angle remains the same as that of single-channel microgroove disks. However, in order to reduce pinch effect and tracing distortion, the radius of the tip of the reproducing stylus requires to be reduced to the region of 0·0005 in.; the recommended limits are 12·5 microns minimum and 15 microns maximum.

COMPATIBILITY

There is another important standard relating to the phasing of the two channels: it is stated that lateral movement of the reproducing stylus shall provide equal, *in-phase*, acoustical signals from the loudspeakers. From this it follows that pick-ups designed for stereo reproduction are suitable also for the playing of normal single-channel microgroove records. This mode of phasing also results in a theoretically "compatible" record, which will thus provide a fully balanced single-channel signal when played with a single-channel pick-up.

However, at the time of writing there is no monaural pick-up available with sufficiently small mechanical impedance to the vertical movement of the stylus to permit this concept of compatibility being exploited. Attempted playing of a stereo disk with even the best of pick-ups currently available will, without a doubt, wipe the stereo from the groove.

Apart from the high mechanical impedance of the vertical movement of the stylus, it must also be remembered that single-channel pick-ups are fitted with a larger stylus tip than is recommended for stereo disks.

This question of "phasing" the two stereo channels to secure a "compatible" record is worth further investigation. Let us look again at Fig. 10.6. Here we see that when the signals applied to coils A and B are in phase, the stylus moves up-and-down as shown at Fig. 10.7 (c). This does not follow the recommended standard which, for the in-phase condition, requires the stylus to move parallel with the surface of the disk (i.e., laterally). This, however, is easily remedied by changing the phase of the signal in one of the channels so that it is shifted by 180 deg. Recording companies do, in fact, phase the two channels in this way for reasons of compatibility, as explained above.

Assuming in-phase signals from the stereo microphone system at the studio, which, for example, would be obtained by an announcer speaking in a position such as to provide sound waves of equal pressure and phase at the microphones, the signals at the stereo recording head are applied to the two electromagnetic systems in the manner which causes lateral modulation of the groove.

On playing the record, the phasing of the generating systems of the pick-up must also match, so that the corresponding lateral vibrations of the reproducing stylus cause in-phase signals at the two loudspeakers. This particular condition is, in effect, the same as that obtained by running two correctly phased loudspeakers on a single channel, which would be expected in view of the lateral modulation of the groove.

This illustrates the importance of maintaining correct phasing throughout the system, from the microphones to the loudspeakers. Only when this condition exists will the true stereophonic illusion be fully realized. The actual phasing of the two channels, from the aspect of the recording head serves essentially to modify the mechanical operation of the stylus to provide a compatible record—the phase of one channel is purposely reversed here, it will be remembered, to provide a lateral movement of the stylus for equal in-phase signals. This method of phasing also helps to reduce the tracing distortion and decreases the demands in relation to the vertical movement of the reproducing stylus.

STEREO PICK-UPS

While two stereo signals are required to cause complex vibrations of the recording stylus in accord with the waveforms of the signals, the vibrations of the reproducing stylus as it traces the complex groove produce signal voltages in the two generating systems of the pick-up of similar relative phase and level as those applied to the recording head during the recording process.

Clearly, if the construction of the pick-up is similar to that of the recording head then the movement of the armatures or coils in the pick-up will follow the same movement as the armatures or coils in the recording head, for a given mode of vibration of the stylus. Thus, the voltage produced in each generating system of the pick-up will, within limits, duplicate the voltage applied to each section of the recording head, through the medium of the complex stereo groove.

In practice there is some slight interaction, or cross-talk, between channels, caused by undesirable resonances in the pick-up and limitations of the recording head and the recording itself. The desirable minimum of cross-talk has been reckoned at 25 db: however, this may be rather an optimistic value, and it has been suggested that the *total* value of cross-talk may be as high as 10 db at high frequencies (the effect is more troublesome as the frequency is raised). While there are a number of unknown factors, tests indicate that a cross-talk value as high as 10 db has little adverse effect on the stereo illusion.

Stereo pick-ups range from crystal and moving-coil to balanced-armature and variable-reluctance types. With its relatively high output voltage in relation to tracking weight, the crystal-cartridge has much to commend it, even though its top response may not quite reach the standard of low-level electromagnetic types.

The stereo crystal-cartridge element is formed of two slabs of crystal, each with its own electrodes, cemented together to form a "bimorph". The crystals are cut in such a manner that outputs are generated when one is subjected to a torque and the other to a flexure, and they are known as the "twister" and "bender" respectively. The vertical and lateral modulations of the groove are transmitted to the two crystals through a form of mechanical coupling so that an output is obtained from the bender crystal as the result of vertical movement, and an output is obtained from the twister crystal as the result of a lateral movement.

There is another arrangement whereby the crystals, instead of being cemented together, are mounted so that their faces form a 90-deg. vee shape. Stylus movement is mechanically coupled to the crystals, but in this case it is possible to use two twister crystals, which require less stylus energy to provide an output—which is desirable in view of the improved high-frequency response.

As with crystal pick-ups, electromagnetic pick-ups for stereo must also have two sensing elements operated from a common stylus. There are always two coils, therefore, and these are invariably wound on a common magnetic circuit. The moving armature is arranged in such a manner that stylus movement in one direction induces an e.m.f. in one coil, while movement in the other direction induces an e.m.f. in the other coil. Two output voltages are thus obtained and, provided the phasing of the coils is correct, the relative phase of the signals will be as required for true stereo reproduction.

STYLUS

Owing to the very small tip radius of the stereo reproducing stylus, the rate of wear of the point is faster than for the 0·001-in. microgroove stylus. For this reason, most stereo pick-ups are fitted with diamond styli as standard. This puts up the cost, but it is really essential when it is considered that a 0·0005-in. sapphire stylus tracking at 7 grammes does not last much longer than about five hours! Most manufacturers aim at about 3–4 grammes tracking weight, which increases the life of the sapphire stylus to about 70 hours; the diamond lasts some 20 to 30 times longer than this.

The reader should note that both pick-up channels require equalization to conform to the standards given in Chapter 7.

STEREO FROM TAPE

Stereo tape records have been available for a number of years both in Britain and America, having been introduced in this country by E.M.I., Ltd. This company's "Stereosonic" tape records are becoming more popular as more people are buying tape recorders and converting them for stereo operation. Unlike the disk, however, which enjoys a considerably greater degree of popularity, stereo tape records are still mainly of interest to the specialist.

There is undoubtedly a great future for magnetic tape in stereophony, but from the point of view of the general public it seems likely that the disk will hold the field for some time to come. Of course, the disk has long been established, whereas tape is a relative newcomer. There is also the question of cost. The enthusiast already in possession of a single-channel record-playing system must buy a second amplifier and loudspeaker and also a stereo pick-up in order to exploit stereo disks to the full. If he decides to start on tape, however, he will require a complete set of equipment, including a stereo tape deck (his original loudspeaker will do for one channel, of course).

FIG. 10.8. *The Truvox stereo head.*

With tape records, the two stereo channels are recorded one above the other on standard $\frac{1}{4}$-in. tape. They are recorded in conformity with the C.C.I.R. recommended characteristic at a speed of $7\frac{1}{2}$ in. per second. E.M.I. "Stereosonic" tapes, which are the only stereo tapes available in this country at the time of writing, are recorded with the recording heads in line across the tape. This makes it necessary to employ a stereophonic head for replay whose gaps are in line.

A stereophonic head of this kind is illustrated in Fig. 10.8. This is the well-known Truvox head, which can easily be fitted to the Truvox Mark III and Mark IV tape decks with very little trouble. By the inclusion of a Truvox Type K amplifier to provide a second channel, a Truvox R2 recorder or any Truvox tape deck can be used for the playing of "Stereosonic" tape records. It is also possible to produce one's own stereophonic tape recordings, but a high level of studio technique is necessary for good results, "a good half-track recording being preferable to a poor stereophonic recording", according to Truvox.

Other machines could be adapted to cater for the Truvox stereo head, and to help experimenters in this respect, details of the head are given. The gap is 0·00025 in. beryllium copper; output voltage 1–3 mV; impedance approximately 50,000 ohms at 10 kc/s; frequency response attainable with a suitable amplifier 50 to 15,000 c/s; cross-talk better than 45 db; bias for recording 120 V r.m.s. approximately; recording current 0·1 mA approximately.

If it is required to play stereo tape records and one is already in possession of a tape deck and a single-channel hi-fi outfit, there is usually no need to go to the expense of adding a second complete reproducing/recording amplifier. One scheme which can be adopted is to double-up on the hi-fi amplifier and loudspeaker (or if just starting in the field to obtain a stereo amplifier complete), replace the existing recording/replay head with a stereo model and feed the second channel from the head to the second hi-fi amplifier, or to the second channel of the stereo amplifier either direct or by way of a head amplifier.

A head amplifier will not be required if the amplifier has a low-level (1–3 mV) tape input channel, but a large number of amplifiers, although making provision for a tape input channel, require an input of some 200–300 mV of tape signal, and need a head amplifier to raise the low-level signal accordingly. The circuit of such an amplifier is given in Fig. 10.9. This is based upon the Mullard Type C tape amplifier, with equalization fixed for a tape speed of 7½ in. per second. The power requirements are quite modest, and can be obtained with ease from almost any hi-fi power amplifier.

There are various commercial head, or sub-, amplifiers on the market for those who do not wish to construct their own. A versatile unit is that by Cape Electrophonics of Southampton. In basic form this uses a single-valve circuit with a gain in the region of 100, but it can easily be modified for particular applications. For example, Model B is already set-up for use with a tape deck, and embodies equalization to C.C.I.R. recommendations.

Now that stereo amplifiers are becoming popular, more manufacturers are adding tape input channels with sufficient sensitivity to operate direct from the replay head of a tape deck. This means that if one is interested simply in playing stereo tapes, the head circuits can be connected direct to the stereo amplifier, in the same way as can a pick-up. Once stereo amplifiers have become well established, stereo tapes could very much be popularized by a go-ahead manufacturer producing a reasonably-priced stereo tape player on the lines of a record player, and suitable for playing monaural as

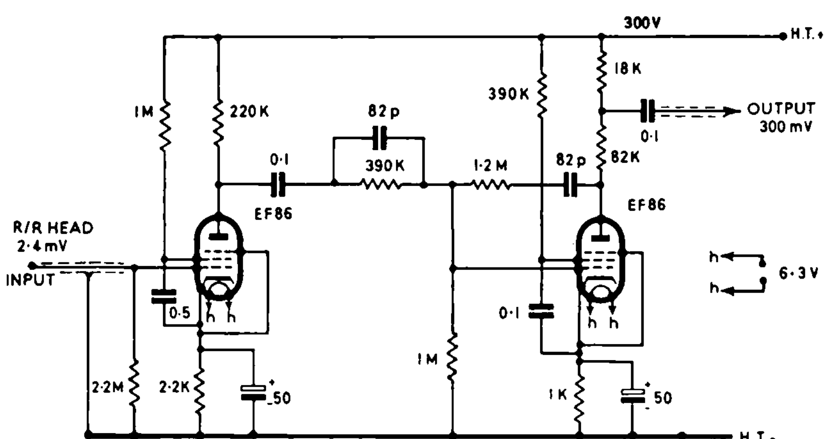


FIG. 10.9. *Circuit diagram of a tape-head amplifier. This is based upon the Mullard Type C tape amplifier, with equalization for a tape speed of 7½ in. per second.*

FIG. 10.10. *Making a stereo tape recording of the BBC's stereo broadcast.*

well as stereo tapes. Such an instrument would find a ready market among hi-fi enthusiasts who are not interested in making tape recordings. Instruments providing for both recording and reproduction are necessarily rather expensive, and this may be the reason why tape records have not made more rapid progress.

If a stereo head is added to a standard tape recorder or tape deck, single-channel work can be carried on as usual, but if stereo recording is to be attempted, it would pay to obtain either a full-track erase head or a bulk erasure. It is very disconcerting to find that only one half of the tape has been erased after expending considerable effort in the creation of a stereo tape record.

STEREO FROM RADIO

This aspect of stereophonic reproduction is as yet in the experimental stage, at least in Great Britain. There have been a number of experiments in stereo radio broadcasts by the BBC, and these have proved remarkably interesting. Whilst television, medium frequencies and V.H.F.-F.M. channels have been utilized in these experiments, there is little doubt that the V.H.F.-F.M. channels represent the ultimate solution to the problem of obtaining stereo from radio signals. With this in mind, several manufacturers of stereo

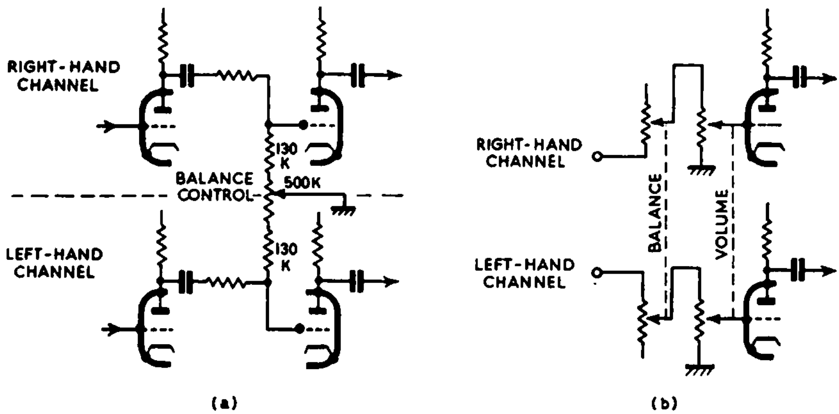


FIG. 10.11. *Two balance-control systems:* (a) *circuit with variable loading;* (b) *with balance control associated with ganged volume control.*

amplifiers have incorporated two radio input channels. Further stereo experiments by the BBC will most likely use two V.H.F.-F.M. channels, as hitherto, so the enthusiast with two V.H.F.-F.M. tuners and a stereo amplifier will really be able to make the best of the experiments.

Two V.H.F.-F.M. tuners (E.A.R. switched units) were used by the author during the course of the experiments in May, 1958. Fig. 10.10 shows the set-up employed, with the author observing the operation of the tape deck. The two E.A.R. tuners are seen situated on the floor beneath the trolley—a handy device for tests of this nature. V.H.F.-F.M. signals were fed to these from a common aerial through a distribution system, and the stereo a.f. signals from the Third and Home programmes were channelled to the amplifier in a Truvox tape recorder and to a Truvox Type K recording amplifier, seen on the second shelf of the trolley.

The recorder was fitted with a stereo head so as to record the stereo signals obtained from the two channels. In addition, two hi-fi amplifiers (a Pamphonic and a Pilot) received signals from the appropriate outlet sockets of the recording amplifiers so as to provide immediate reproduction from the two E.A.R. loudspeakers situated either side of the trolley. The set-up thus permitted the immediate reproduction of the stereo sound as well as the recording of the signals. The tests proved very successful from both aspects; excellent live stereo was obtained during the transmission and, on playing the tape recording afterwards, the performance was almost equal to the original. The tests were performed some 60 miles from the V.H.F.-F.M. transmitters, showing that stereo via radio is feasible in fringe areas and also that it can be recorded with remarkable ease.

When making stereo tape records, the accepted practice is to record the left-hand channel on the top track, that is when the tape is travelling from left to right with the active side away from the observer.

BALANCE CONTROL

If two separate control units or hi-fi amplifiers are used for the playing of stereophonic records or for reproducing stereo from the radio, it is essential to maintain optimum balance by very carefully adjusting the two volume controls. Tonal balance between the two channels is also important. Stereo control units or amplifiers use ganged controls, so that operation of the common control knob will vary the volume or tone of the two channels in step. In addition to ganged controls, a balance control is often desirable so that the volume of one channel in relation to the other can be altered slightly to compensate for a difference between the sensitivities of the loudspeakers, for example, or to counteract acoustical shortcomings of the listening room.

Two methods for obtaining balance control are given in Fig. 10.11. Diagram (*a*) is a straightforward arrangement which varies the loading, and thus the applied signal, at the grids of the two triodes. An ordinary 500k linear volume control serves as the balance control, and the two 130 k resistors in series with it avoid excessive loading which would result in a fall of top response.

A more commendable arrangement is given in Fig. 10.11 (*b*). Here the balance control is directly associated with the ganged volume control. The balance control is also a ganged component, but is connected differentially, so that as it is rotated the signal input is increased in one channel and decreased in the other.

These controls can be installed in existing amplifiers if required, but

FIG. 10.12. *The Pamphonic Type* 3,000 *stereophonic amplifier.*

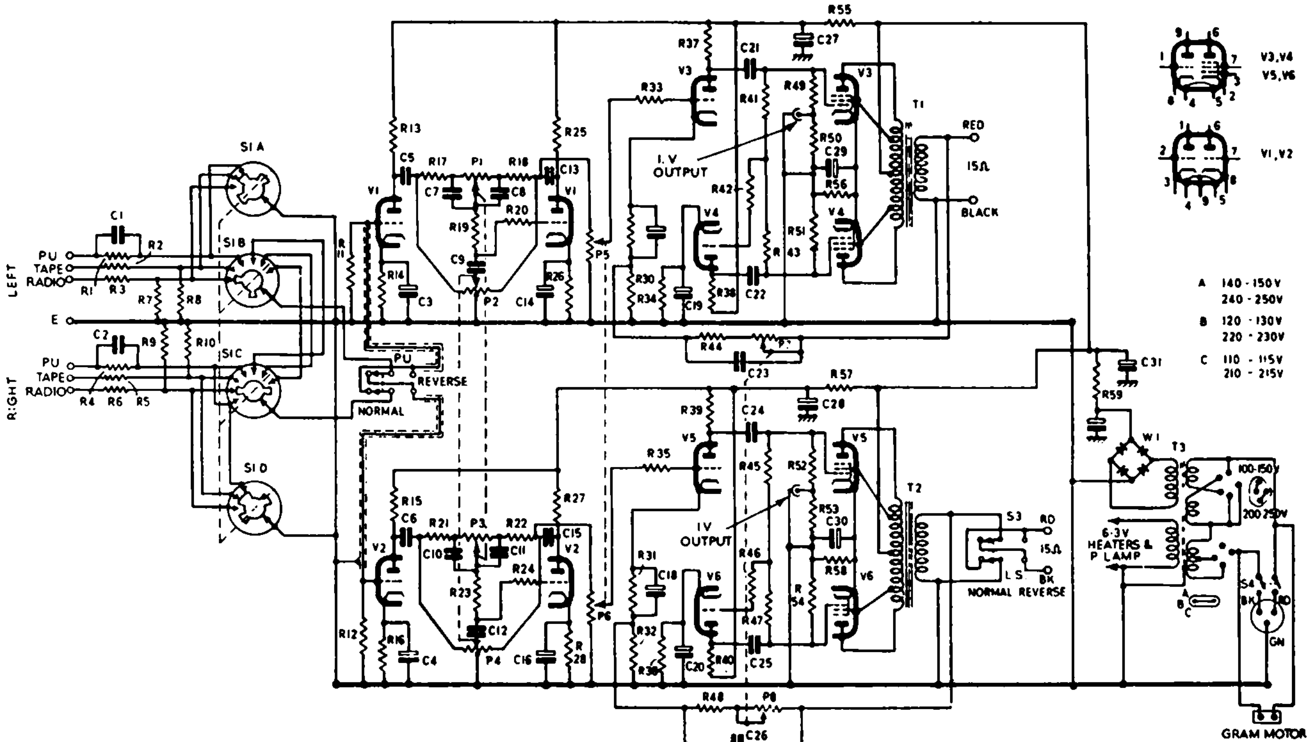


FIG. 10.13. *Circuit diagram of the Pamphonic Type 3,000 stereophonic amplifier. The component values are given on the opposite page.*

Resistors		*Resistors*		*Resistors*		*Resistors*	
R 1	1M	R16	2·2K	R31	2·2K	R46	22K
R 2	100K	R17	100K	R32	100Ω	R47	470K
R 3	100K	R18	100K	R33	22K	R48	680Ω
R 4	1M	R19	470K	R34	2·2K	R49	470K
R 5	100K	R20	22K	R35	22K	R50	27K
R 6	100K	R21	100K	R36	2·2K	R51	470K
R 7	3·3K	R22	100K	R37	100K	R52	470K
R 8	6·8K	R23	470K	R38	100K	R53	27K
R 9	3·3K	R24	22K	R39	100K	R54	470K
R10	6·8K	R25	220K	R40	100K	R55	22K (½W)
R11	100K	R26	2·2K	R41	470K	R56	400Ω (3W)
R12	100K	R27	220K	R42	22K	R57	22K (½W)
R13	220K	R28	2·2K	R43	470K	R58	400Ω (3W)
R14	2·2K	R29	2·2K	R44	680Ω	R59	220Ω (6W)
R15	220K	R30	100Ω	R45	470K		

Note: All resistors ¼W unless otherwise stated.

Capacitors			*Capacitors*		
C 1	50pF	5%	C17	50μF	25V
C 2	50pF	5%	C18	50μF	25V
C 3	50μF	25V	C19	50μF	25V
C 4	50μF	25V	C20	50μF	25V
C 5	0·1μF	350V	C21	0·1μF	350V
C 6	0·1μF	35V	C22	0·1μF	350V
C 7	4,700pF	10%	C23	3,000pF	10%
C 8	4,700pF	10%	C24	0·1μF	350V
C 9	100pF	20%	C25	0·1μF	350V
C10	4,700pF	10%	C26	3,000pF	10%
C11	4,700pF	10%	C27	32μF	450V
C12	100pF	20%	C28	32μF	450V
C13	0·1μF	350V	C29	50μF	25V
C14	50μF	25V	C30	50μF	25V
C15	0·1μF	350V	C31	32μF	450V
C16	50μF	25V	C32	32μF	450V

Potentiometers		
P1	1M	lin.
P2	½M	lin.
P3	1M	lin.
P4	½M	lin.
P5	½M	log.
P6	½M	log.
P7	1K	lin.
P8	1K	lin.
Valves, Rectifier		
V1	12AX7 or ECC83	
V2	12AX7 or ECC83	
V3	ECL82	
V4	ECL82	
V5	ECL82	
V6	ECL82	
W1	B250 C150	

Component values for Pamphonic Type 3,000 stereophonic amplifier (see Fig. 10.13).

care should be taken to position the circuit so that it does not affect the frequency response or incite overloading of the low-level stages.

STEREO EQUIPMENT

Because of the enormous difference between single-channel hi-fi and stereophony, there is little doubt that stereophony is destined to become the hi-fi of the future. It will be available from tape, disk and radio; the gradual development of the propagation of television and radio signals by way of cables instead of the ether will lend itself admirably to regular stereo broadcasts on ordinary radio as well as television sound.

The change from single-channel sound to stereo is now taking place; stereo attachments are available for existing equipment and complete stereo amplifiers and control units. Stereo will not be adopted only by hi-fi enthusiasts, as add-on units are already becoming available for the popular record player and radiogram. B.S.R. have adapted two of their "Monarch" record changers (Models UA8 and UA12) to take stereo cartridges for stereo reproduction, and these are being supplied to leading manufacturers. Combined stereo-monaural pick-ups are also being developed. During the London Audio Fair in 1958, R.G.D. demonstrated that the "Victoria" radiogram is easily convertible for stereo operation by the fitting of a stereo pick-up cartridge and an additional loudspeaker-amplifier unit. The "Victoria" radiogram comes within the hi-fi category, of course; however, a number of ordinary record-players and radiograms of reasonable quality which are now on the market can readily be adapted to stereo operation.

Nevertheless, the hi-fi enthusiast will still wish to employ hi-fi amplifiers, pick-ups and loudspeaker systems, as hitherto. For really high-quality stereo reproduction, special attention will have to be paid to such things as turntables and pick-up arms. Motor rumble will be found to be more troublesome with a stereo system than with a monaural system, since the stereo pick-up responds almost equally in all directions of movement of the stylus. Consequently, a motor which is virtually rumble-free on a monaural system may exhibit a most disconcerting rumble when a stereo pick-up is incorporated.

There is also the question of the pick-up arm. If the pick-up is tracked at the ideally low pressure of 2–3 grams, this may be found to have too much friction in the main bearing. There may be a temptation to use a greater tracking pressure and a stylus point radius exceeding the recommended 0·0005 in. as a means of keeping the stylus in the groove. Such expedients should be avoided if possible, for a record once played under these conditions will never give its best when later it is played under the correct conditions.

The Pamphonic Type 3,000 stereo amplifier (Fig. 10.12) is of interest. As may be seen from the circuit in Fig. 10.13, it employs two independent channels with ganged controls. There are three switched inputs catering for pick-up, tape and radio. These are selected by switch S1, which has six positions, giving three stereo positions and three monaural positions. The pick-up channel is compensated for R.I.A.A. (British Standard 1928:1955 Fine Groove) recording characteristic and is adjusted for use with a stereo crystal cartridge, while the radio and tape channels are substantially flat from 50 c/s to 15 kc/s and require signal inputs of 1 volt and 0·5 volt respectively for 5 watts output. Each channel is rated at 7·5 watts, and on the monaural positions the signal input is applied to both channels simultaneously, thus giving a total power output of 15 watts.

Baxandall tone-control circuits are incorporated around stages V1 and V2 and provide 15 db variations at 50 c/s and 10 kc/s. There is also an interesting negative-feedback balance control, P7/P8, which serves to vary the gain of each channel differentially by 6 db from the level position. A

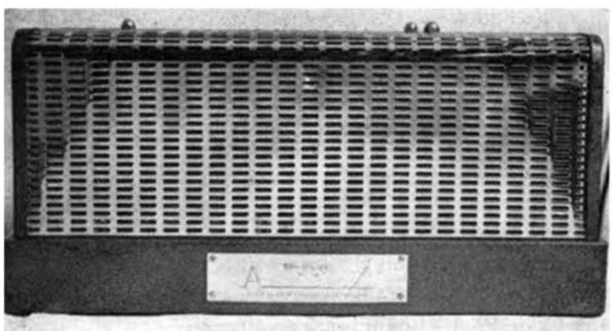


FIG. 10.14. *The Sound Sales Tri-channel Mk. IV power amplifier.*

channel-reversing switch, S2, is provided at the input of the amplifier, and is useful in cases where the pick-up is connected wrong way round, or if the pick-up wires are not known. There is also a phasing switch, S3, in the right-hand channel loudspeaker circuit. This avoids having to fiddle with the loudspeaker connexions.

The remainder of the amplifier follows conventional lines, details of which have already been given. It is, however, interesting to see an ECL82 connected in the ultra-linear mode. A metal bridge rectifier supplies h.t. for both of the channels, with excellent regulation.

Stereo tape equipment is marketed by E.M.I. for the playing of their Stereosonic tape records, and the G.E.C. also make a stereo tape-playing outfit. The latter incorporates the G.E.C. metal-cone loudspeaker system

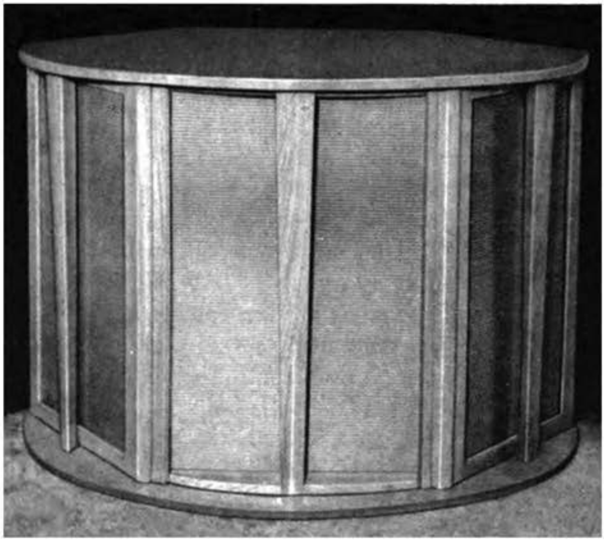


FIG. 10.15. *The Sound Sales Tri-channel loudspeaker system.*

and presence unit, two BCS2417A/18A 12-watt amplifiers and the Truvox Type TR2112 tape deck.

Sound Sales, Ltd. market an excellent stereophonic version of their "Tri-Channel" equipment. Each channel uses, in effect, three separate amplifiers each covering limited sections of the frequency range and fed from an infinitely variable electronic cross-over system. Three loudspeaker systems are used on each channel, all being contained in a single cabinet of special design.

This arrangement avoids the necessity of reactive cross-over units at the loudspeakers themselves. The stereo Tri-Channel outfit comprises two Tri-Channel main amplifiers, each rated at 50 watts, two Tri-Channel loudspeaker systems and a Tri-Channel stereo control unit. The control unit has a stereo/monaural changeover switch, giving parallel (100 watts) or stereo connexion. Sound Sales also market a transistorized stereo control unit, which has provision for direct replay from a low-level pick-up or tape head. The Sound Sales Mk. IV power amplifier is illustrated in Fig. 10.14 and the loudspeaker system in Fig. 10.15.

Especially worthy of mention is the stereo equipment by Cape Electrophonics, Ltd., comprising a stereo control unit of high standard, including a facility for the direct connexion of a tape head, and associated power amplifiers. For enthusiasts not interested in building their own equipment, the control unit and amplifiers are available in ready-made and tested form, Cape Electrophonics also manufacture and market the "Cape Audio System", which includes facilities for recording and reproducing tape, as well as control units and very high-quality power amplifiers. The equipment is designed to very rigid specifications, thus ensuring results of professional standard.

At the time of writing stereo pick-ups in ever-increasing numbers are being made available which are suitable for playing the Nixa (Pye) stereo records. Fig. 10.16 illustrates the "Acostereo" crystal cartridge (Cosmocord, Ltd.), whose essential details are as follows. At a recorded level of 1·5 cm/sec

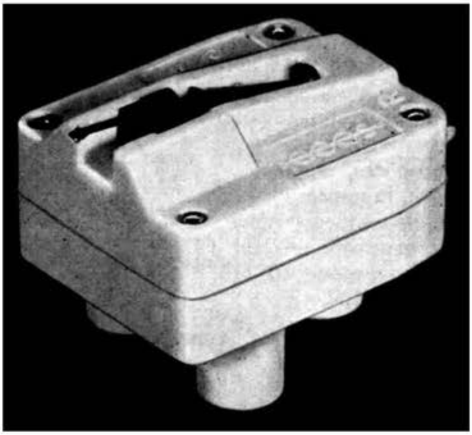


FIG. 10.16. *The Acostereo crystal cartridge by Cosmochord, Ltd.*

an output of 200 mV is available; the frequency response is from 40–12,000 c/s and the separation between channels at 1 kc/s is better than 15 db. The cartridge will track at a minimum of 2 grams, but since this very small tracking pressure is dependent upon the lateral freedom of the tone arm, it is suggested that pressures in excess of 2 grams may be required, especially with autochangers. The recommended load is 2 megohms, and the radius of the stylus tip is approximately 0·0005 in. to conform with the stereo standard for disk records.

In order to avoid undue wear on the stereo record and stylus, the lowest possible tracking weight should be used. A tracking weight in excess of 6–7 grams will ruin a 0·0005-in. stereo stylus after a few hours' playing, while the use of a 0·001-in. stylus, particularly if it is tracking fairly heavily, will ruin a stereo disk almost immediately.

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