HOW TO GET THE BEST OUT OF YOUR TAPE RECORDER

PERCIVAL J. GUY

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Senior Lecturer in the B.B.C. Engineering Training Department

Here in concise form and with the minimum of theory is all you need know to obtain first-class results from the recorder of your choice. The author has been engaged for many years in training beginners in the art of recording and he has developed the facility of explaining technical matters in simple, readily understood terms. His articles on recording and allied subjects have appeared in *Radio Times*, *Hi-Fi News* and *Record News*, etc. Chapter headings are: SIMPLIFIED THEORY OF MAGNETIC SOUND RECORDING; DISTORTIONS; IMPEDANCE AND IMPEDANCE MATCHING; FILTERS AND EQUALIZERS; MIXERS; ACOUSTICS; MICROPHONES; BALANCE AND CONTROL; LOUDSPEAKERS; TESTING AND FAULT TRACING; MECHANICAL MAINTENANCE; PRINTING BETWEEN LAYERS; TAPE JOINTING AND EDITING; and COPYRIGHT AND LEGAL ASPECTS OF HOME RECORDING. There are 115 diagrams; an extensive glossary; and five appendices.

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INTRODUCTION

ANYONE possessing a tape recorder can derive a great deal of enjoyment from it with no other knowledge than that provided by the manufacturer's instruction book. If, however, it is desired to obtain first-class results, a good deal more information is required.

This handbook sets out the basic theory of magnetic recording in such a way that those without scientific training can follow it with comparative ease. Those who have forgotten their elementary algebra can ignore the brief mathematical proofs which are included for completeness.

The reader is first given a clear description of the nature of sound and of elementary magnetic theory as applied to magnetic recording. The requirements of magnetic recorders are then described in simple terms and full details are given of the many points which require attention from time to time. Microphones and loudspeakers are fully discussed and the importance of impedance matching is stressed. The need for equalization is explained and some simple equalizers are described. Attention is paid to the acoustic properties of rooms and halls and to the art of balance and control.

It is shown how, with very little trouble, editing may be accurately effected and the legal aspect of home recording is made plain.

The author expresses his thanks to his colleagues in the BBC Engineering Training Department who gave him invaluable help in the compilation of this handbook.

ACKNOWLEDGEMENTS

The following formulae and figures have been reproduced by kind permission from Frayne and Wolfe, *Elements of Sound Recording*, 1949, John Wiley and Sons Inc., and Chapman and Hall, Ltd.

Formulae (1) and (2) on page 37.

Formula relating to the required loss in dB (Section 10 **Constant Impedance Attenuators)** on page 39.

Formula in the caption to Fig. 70 on page 55.

Figs. 69 and 70 on page 55.

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I

SIMPLIFIED THEORY OF MAGNETIC SOUND RECORDING

1. What Sound is

SOUND is the transfer of energy from one part of a sound-conducting medium to another by means of longitudinal waves. The wave motion gives rise to areas of increased pressure (compressions) and decreased pressure (rarefactions) which travel through the medium in the direction of propagation. An obstacle such as the ear drum or the diaphragm of a microphone placed in the path of the sound wave is therefore subjected to rapidly changing pressure variations. The number of complete vibrations experienced varies from a few to many thousands per second.

The audible frequency range varies with the age and health of the individual. Healthy, young people can hear rates of vibration (called **frequencies**) between 30 and 20,000 cycles per second. (One complete vibration is called **one cycle**). With approaching middle age, it becomes less easy to hear both very high and very low frequencies.

For perfect fidelity of reproduction, all the apparatus used (microphones, amplifiers, recorder, reproducer and loudspeaker) would have to be equally responsive to all audible frequencies of vibration and also to minute changes in pressure over the whole range of pressure variations. The pressure variation range of an orchestra is from 1/145 acoustic bars to about 20 acoustic bars (one acoustic bar is 1/1,000,000 of the barometric bar—the unit in which barometric pressure is normally measured). As will be seen, a number of factors limit both the frequency range and the pressure variation range recordable without distortion.

Properties of Sound distinguishable by the Human Ear

(1) LOUDNESS

This is the degree of sensation, measured in **phons.** (See Appendix B). At the threshold of hearing the loudness is a small fraction of a phon, whilst at the threshold of pain (where the sound becomes loud enough to hurt) the loudness reaches 130-140 phons. The comparative loudness of a given sound depends upon two things: (*i*) the size of the pressure variations at the point of reception; (*ii*) the frequency of the sound—because the human ear is not equally responsive to all frequencies.

(2) PITCH

This is a musical term which expresses the comparative frequency of the lowest tone (called the **fundamental frequency**) in a complex sound. [Pure (single frequency) tones rarely occur on their own. Most musical sounds, for instance, consist of a mixture of several pure tones superimposed upon each other, the frequency relationship being usually a harmonic series, *i.e.* the overtones have the ratios 2 : 3 : 4 : 5 : 6: etc. with respect to the fundamental (also called the first harmonic). Unmusical sounds are also made up of a mixture of numerous pure tones but the component vibrations are not necessarily in harmonic sequence and many of them are highly damped, *i.e.* start with a large amplitude and then decay rapidly. These are called **transients.** Percussion instruments (drums, cymbals, pianos and the like) produce sound waves that are full of transients. These often produce distortion when recorded, because of their shock effect on electrical and mechanical resonant circuits].

(3) TIMBRE

This is the tone quality of the sound, governed partly by the number and relative peak pressures of the component overtones and partly by their rate of rise and decay. One of the experiments you may be able to make with your tape recorder is to record the notes of the piano and then to reproduce them in reverse. The reversed notes will sound like organ tones. This is because the characteristic of a piano note is a quick build-up and a slow decay whilst that of the organ is a slow build-up and a slow decay.

2. How Sound is Recorded

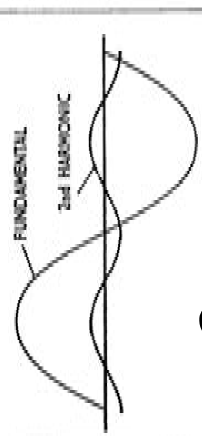
Sound is recorded in terms of the size and frequency of the pressure variations at the microphone and it is reproduced by setting up similar pressure variations in the air in the vicinity of the loudspeaker. In magnetic recording the tape is coated with a magnetic substance and is magnetized along its length to a degree determined by the loudness of the sound at the source, the induced magnetism being made to vary in strength in sympathy with the variations in pressure at the microphone. The pressure in a pure tone varies sinusoidally, *i.e.* the instantaneous value of the pressure in the path of the sound varies according to a **sine law** (see Appendix A).

The waveshape of a complex sound is determined by the number and relative amplitudes of the component frequencies—it is a simple addition of the various sine wave components (Figs. 1, 2, 3(a) and (b)).

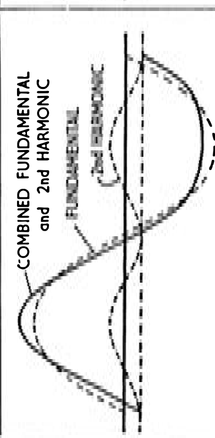
However complex a sound may be and however varied the frequencies arriving at the microphone, the instantaneous pressure at the diaphragm will always have some definite value depending upon the instantaneous sum of the pressures of the various pure tones which make up the given complex sound.

The first reproduceable recording of sound was in the form of a trace engraved on a cylinder, the ramifications of the trace being determined by the vibration imparted by the sound wave to a tightly stretched diaphragm. (Thomas Edison's " Phonograph ").

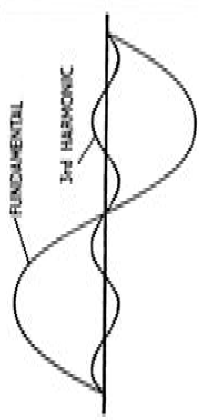
Another principle is followed in the magnetic recorder. To understand that, it is necessary to have an elementary knowledge of magnetism.



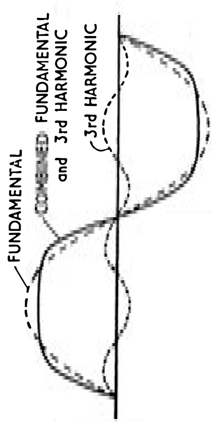
(a) FIG. 1(a). THE TWO COMPONENTS OF THE COMPLEX WAVEFORM SHOWN BELOW IN FIG. 1(b).



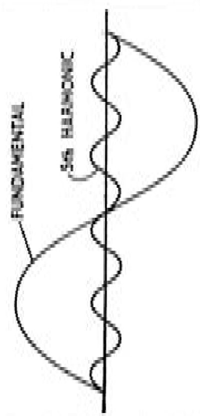
(b) FIG. 1(b) WHEN THE TWO COMPONENTS SHOWN IN FIG. 1(a) ARE ADDED THE RESULTANT WAVESHAPE IS NO LONGER SINUSOIDAL. EACH POSITIVE HALF CYCLE IS MORE POINTED AND EACH NEGATIVE HALF CYCLE IS FLATTENED AS SHOWN IN THE SOLID CURVE.



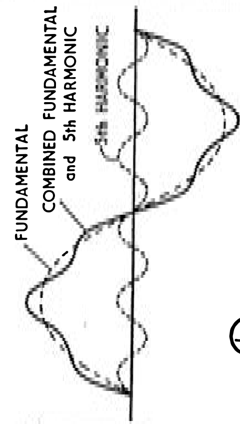
(a) FIG. 2(a). COMPARATIVE WAVESHAPES OF A PURE TONE AND ITS 3RD HARMONIC.



(b) FIG. 2(b). THE ADDITION OF THE TWO TONES SHOWN IN FIG. 2(a) PRODUCES THE FLATTENED WAVESHAPE OF THE SOLID CURVE ABOVE.



(a) FIG. 3(a) COMPARATIVE WAVESHAPES OF A PURE TONE AND ITS 5TH HARMONIC.



(b) FIG. 3(b). WHEN THE TWO TONES SHOWN IN FIG. 3(a) ARE ADDED TOGETHER THE RESULTANT WAVE SHAPE IS AS THE SOLID CURVE IN THE ABOVE FIGURE.

3. Elementary Magnetic Theory

There are two main types of magnetic materials—" hard " and " soft ". The difference between them is that the " hard " substances, when magnetized by some external magnetizing force, tend to retain the induced magnetism until subjected to a demagnetizing force, whilst magnetically soft substances though even more readily magnetized (and to a much greater degree by a given magnetizing force) tend to lose their induced magnetism as soon as the magnetizing force is removed. Typical examples of " hard " materials are:

Steel; Ferric Oxide ($F_{e2}O_3$); and Ferrisoferric Oxide ($F_{e3}O_4$).

Typical examples of " soft " materials are:

Soft iron; Stalloy (an alloy of soft iron containing 3·5 per cent. silicon); and Permalloy (an alloy of soft iron containing about 78·5 per cent. nickel).

The " hard ' 'm aterials when magnetized form what are called **permanent magnets.**

The properties of permanent magnets may be summarized as follows:

(*i*) They retain their magnetism indefinitely unless subjected to some demagnetizing force.

(*ii*) A specimen in the form of a bar, if freely suspended by its centre-point, will always orient itself so that one end points to the magnetic north pole of the earth (unless some distracting magnetic force is also present). This end is called the **north-seeking** or **north pole** of the magnet, whilst the other end is called the **south pole.**

(*iii*) If two such magnets are freely suspended it will be found that like poles repel each other whilst unlike poles attract each other.

(*iv*) Around the permanent magnet is a space, called the **field of force,** in which magnetic forces due to the magnet may be detected (Fig. 4). The lines along which the forces act are called **lines of force.** Collectively, lines of force are called **flux.** The engineering symbol used for flux is ϕ (Greek letter *phi*). Each line of force forms a closed loop, completed through the length of the magnet. Flux is measured in **webers** (see Appendix E, page 116).

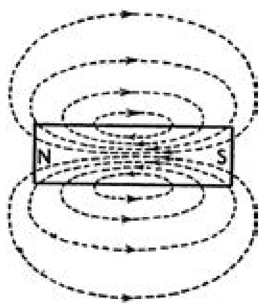


FIG. 4. THE FIELD OF FORCE ROUND A PERMANENT MAGNET OF BAR SHAPE. EACH LINE OF FORCE FORMS A CLOSED LOOP.

(*v*) A magnet has the power to induce magnetism in any ferrous material in its vicinity, the induced pole always being of opposite sign to that of the magnetizing magnet. Thus, the north pole of a magnet induces a south pole in the nearest end of a specimen of magnetic material in proximity. Because unlike poles attract, the material in which the pole is induced is attracted to the magnet. This is why a magnet picks up steel pins though it has no effect on brass ones.

4. The Connection between Electricity and Magnetism

A current-bearing conductor has a field of magnetic force around it—the direction of the lines of force being circular (Fig. 5). It is

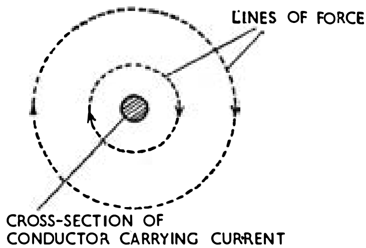


FIG. 5(a). A CURRENT-CARRYING CONDUCTOR HAS AROUND IT A FIELD OF MAGNETIC FORCE.

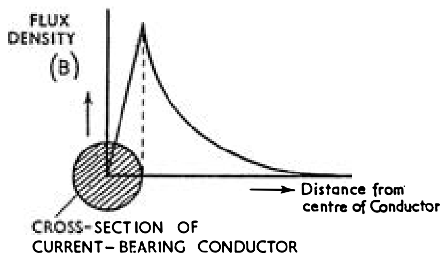


FIG. 5(b). GRAPH SHOWING HOW THE STRENGTH OF FIELD VARIES INSIDE AND IN THE VICINITY OF A CURRENT-CARRYING CONDUCTOR.

often important to know which way the lines of force rotate. Maxwell gave a rule relating the direction of the lines of force and the direction of current flow in a conductor. The rule is that the direction of flow and field bear the same relation to one another as the direction of movement and of rotation of a corkscrew (Fig. 6).

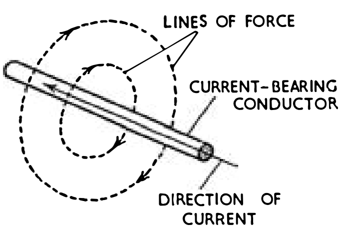


FIG. 6. ILLUSTRATING MAXWELL'S CORKSCREW RULE FOR DETERMINING THE DIRECTION OF THE LINES OF FORCE ROUND A CURRENT-CARRYING CONDUCTOR.

If the wire carrying the current is wound in a loop, the lines of force all pass through the loop in the same direction (Fig. 7).

When the wire is wound in a coil the lines of force around the coil are like the lines of force round a bar magnet (Fig. 8).

If the coil is wound round an iron or steel core the flux per unit area is greatly increased, because the core becomes magnetized. The arrangement is then called an **electro-magnet** and as long as the

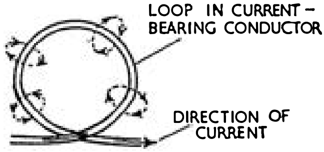


FIG. 7. APPLICATION OF MAXWELL'S CORKSCREW RULE SHOWS THAT IF THE CURRENT IN A WIRE IS CONTINUOUS IN ONE DIRECTION, THE LINES OF FORCE WILL PASS THROUGH A LOOP IN THE WIRE IN THE SAME DIRECTION.

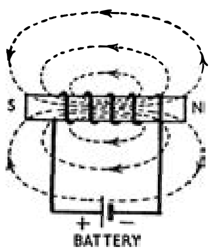


FIG. 8. IF THE CURRENT-CARRYING WIRE IS WOUND IN A COIL ROUND AN IRON CORE, THE MAGNETIC LINES OF FORCE SURROUNDING THE CORE WILL BE LIKE THOSE ROUND A BAR MAGNET. SUCH AN ARRANGEMENT IS CALLED AN ELECTRO-MAGNET.

current is flowing the arrangement behaves just like a strong bar magnet. If the core is of hard magnetic material most of the induced magnetism will remain when the current is switched off. If the core is of soft iron, the induced magnetism will tend to revert to a value little greater than zero.

In a material such as **ferric oxide** (commonly called **jeweller's rouge**) the molecules join up to form tiny particles called **crystal elements.** In that substance each element contains eight molecules of $F_{e2}O_3$. The elements are normally cubic and one side of the tiny cube is 1/10 micron in length (1 micron = 1/1,000,000 metre). Under heat treatment in a magnetic field the material becomes magnetic and the elements are elongated in shape. They bunch together in domains and each domain behaves like a saturated bar magnet. Ordinarily the domains are oriented in random fashion so that the material does not appear magnetized (Fig. 9) but if subjected

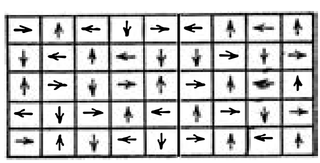


FIG. 9. RANDOM ARRANGEMENT OF THE DOMAINS IN A COMPLETELY DEMAGNETIZED PIECE OF FERROUS MATERIAL. ARROWS SHOW THE DIRECTION OF MAGNETIZATION OF THE INDIVIDUAL DOMAINS. IN THIS CONDITION THE EXTERNAL MAGNETIC EFFECT IS ZERO.

to a magnetizing force the domains swing round into alignment with the magnetizing force and the specimen becomes magnetized (Fig. 10).

An important point is that in some substances—and $F_{e2}O_3$ is one of these—there is not a straight line relationship between induced magnetism and magnetizing force. (Fig. 11 is a graph which illustrates this point).

It will be seen that if a steadily increasing magnetizing force is applied to a specimen originally in a completely demagnetized state, the induced magnetism rises only slowly at first, then more steeply and finally more slowly again. This non-linear reaction causes some

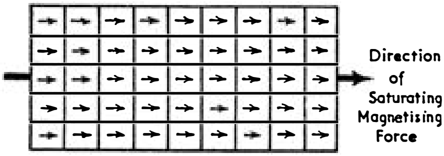


FIG. 10. SHOWING THE REGIMENTED ALIGNMENT OF DOMAINS IN A SATURATED BAR MAGNET.

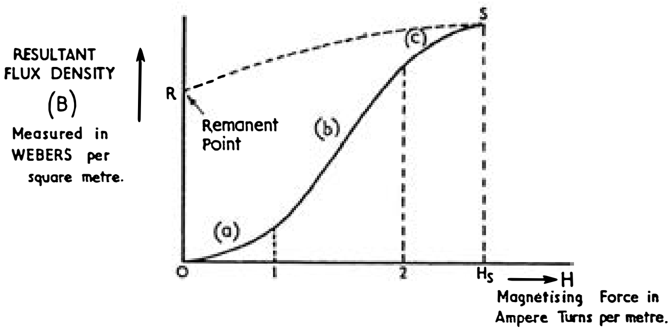


FIG. 11. GRAPH SHOWING THE S-SHAPED INITIAL MAGNETIZATION CURVE OF SOME FERRO-MAGNETIC MATERIALS. STEADILY INCREASING VALUES OF MAGNETIZING FORCE PRODUCE LITTLE INDUCED FLUX DENSITY TO BEGIN WITH (SLOPE (*a*) OF THE CURVE). THEN, WHEN THE MAGNETIZING FORCE IS INCREASED BEYOND A CERTAIN CRITICAL VALUE (1 ON THE *H* SCALE), THE SLOPE OF THE CURVE BECOMES STEEPER (SLOPE (*b*)). WHEN THE MAGNETIZING FORCE APPROACHES THE SATURATION VALUE (H_S), WHEN IT IS (SAY) AT 2 ON THE *H* SCALE, EACH STEADY INCREMENT IN MAGNETIZING FORCE INCREASES THE INDUCED MAGNETISM BY LESSER AND LESSER DEGREES AND THE SLOPE OF THE CURVE BECOMES LESS STEEP (SLOPE (*c*)). *OR* IS THE VALUE OF REMANENT MAGNETIZATION WHEN THE SATURATING MAGNETIZING FORCE (H_S) HAS BEEN APPLIED AND REMOVED. THIS VALUE IS CALLED THE REMANENCE OF THE MATERIAL.

difficulty in magnetic recording and special arrangements have to be made to counter it. When the magnetizing force is sufficient, the specimen becomes saturated with magnetism (saturation is the state when all the domains are oriented in line with the magnetizing force. See Fig. 10).

If a saturating magnetizing force is removed the magnetism in the specimen does not remain at the saturation value but reverts to some lower value depending upon the relative magnetic hardness of the material. This is shown by the dotted curve in Fig. 11.

Figs. 12(a) and (b) show comparative curves for ferric oxide and soft iron. It will be seen that in the case of the latter the remanent flux density after the saturating magnetizing force is removed is practically zero.

In magnetic recording the magnetizing force applied to the tape **varies cyclicly,** *i.e.* it increases in the positive direction, reduces to zero, increases in the negative direction and then decreases to zero once again. It is therefore necessary to examine the changes in the

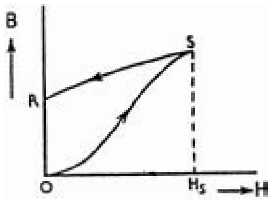


FIG. 12(a). GRAPH OF FLUX DENSITY FOR A GIVEN MAGNETIZING FORCE APPLIED TO FERRIC OXIDE. H_s=SATURATING MAGNETIZING FORCE. *OR* = REMANENCE.

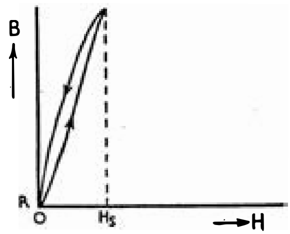


FIG. 12(b). GRAPH OF FLUX DENSITY FOR A GIVEN MAGNETIZING FORCE APPLIED TO SOFT IRON ALLOY. H_s = SATURATING MAGNETIZING FORCE. IN THE CASE OF THIS MATERIAL, *O* and *R* ARE PRACTICALLY COINCIDENT, *i.e.* REMANENCE IS VIRTUALLY ZERO.

induced magnetism when a magnetic material is subjected to a force changing in this way. These are shown in Fig. 13.

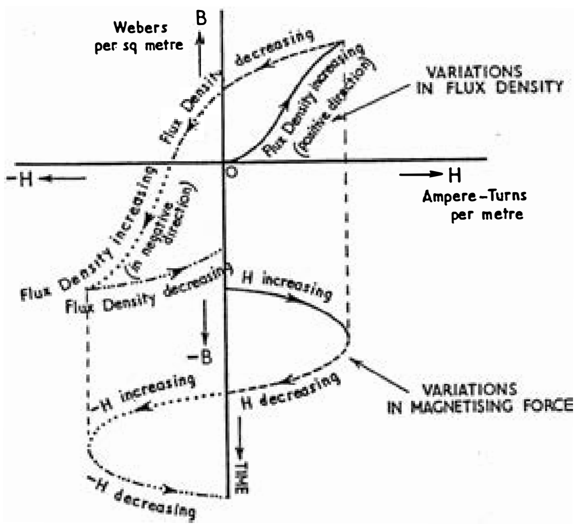


FIG. 13. THIS CURVE SHOWS THE FLUCTUATIONS IN THE FLUX DENSITY WHEN A SINUSOIDALLY VARYING MAGNETIZING FORCE IS APPLIED TO A SPECIMEN PREVIOUSLY IN A STATE OF ZERO MAGNETIZATION. COMPARE THIS WITH FIG. 14 WHICH SHOWS HOW THE FLUX DENSITY VARIES WHEN THE SINUSOIDALLY VARYING MAGNETIZING FORCE HAS BEEN THROUGH AT LEAST 10 REVERSALS.

If the cyclicly varying force is continuous, the curve eventually becomes a closed loop. This is called a **hysteresis loop** (Fig. 14). As the initial state of the medium is a governing factor in deciding the result of applying a given magnetizing force, the loop does not become symmetric until a good many reversals of magnetizing force have taken place.

The meaning of **magnetic hysteresis** is that the resultant magnetization of a specimen always depends not only upon the

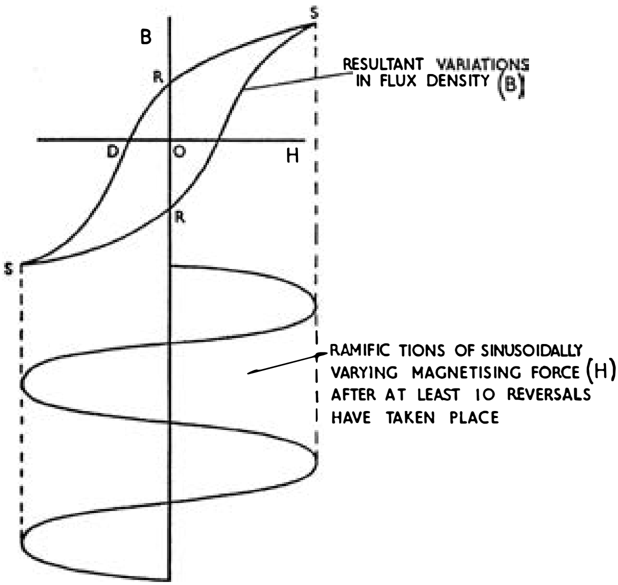


FIG. 14. THE HYSTERESIS LOOP OF A SPECIMEN FORMED IN THE SHAPE OF A RING.
OD IS THE COERCIVITY: THE DEMAGNETIZING FORCE REQUIRED TO REDUCE THE FLUX DENSITY TO ZERO AFTER A SATURATING MAGNETIZING FORCE HAS BEEN APPLIED AND REMOVED.

magnitude and the direction of the applied magnetizing force but also upon the original magnetic state of the specimen.

The process of magnetization gives rise to a dissipation of energy called the **hysteresis loss.** It can be shown that the hysteresis loss is proportional to the area contained within the loop and to the frequency of the alternating magnetic force applied.

Fig. 15 illustrates comparative hysteresis loops for (a) $F_{e2}O_3$ and (b) Permalloy.

It will be seen that the hysteresis loss is far greater in the case of $F_{e2}O_3$ than in the case of Permalloy.

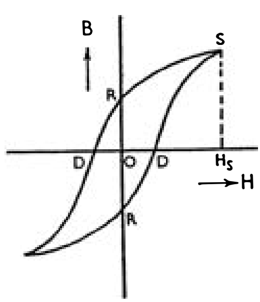


FIG. 15(a). HYSTERESIS LOOP FOR Fe_2O_3. COMPARE THIS WITH FIG. 15(b) — THE LOOP FOR PERMALLOY.

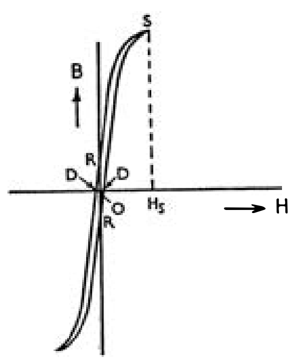


FIG. 15(b). HYSTERESIS LOOP FOR PERMALLOY.

5. Self-Demagnetization

Inspection of the hysteresis loops for long and short bar magnets (Fig. 16) shows that the shorter the bar, of given material and cross-sectional area, the greater the force required to reach saturation and the lower the value of remanence.

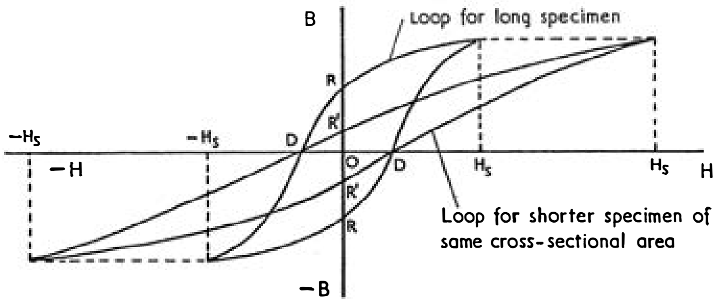


FIG. 16. COMPARATIVE HYSTERESIS LOOPS FOR LONG AND SHORT SPECIMENS OF THE SAME MATERIAL AND OF FIXED CROSS-SECTIONAL AREA.

This effect is called **self-demagnetization** and it is due to the mutual attraction of the salient poles of unlike polarity at the opposite ends of a specimen. The little magnets induced along the length of the tape when recording a signal, are **each half a recorded wavelength long**. (The recorded wavelength is the distance travelled by the tape in the time taken to record one cycle). Thus λ_r (the recorded wavelength) $= \dfrac{\text{tape speed}}{\text{frequency recorded}}$.

The higher the frequency recorded, therefore, the greater the self-demagnetization, up to a point—the self-demagnetization is found never to exceed a certain maximum value for any given specimen. Self-demagnetization is important in magnetic recording for two reasons: (*i*) it is partly responsible for the need for pre-emphasis of high frequencies in the recording system; (*ii*) it helps to remove the unwanted high frequency bias component in the output from the tape, when a.c. bias is used.

6. The Requirements of a Magnetic Recording System

(1) A MEDIUM ON WHICH TO RECORD

This may be a tape, wire, disk or cylinder. The last two are as a rule only used for office dictation types of machine. Wire is not very popular because it tangles too easily when it breaks, cannot be pulled past the heads at so constant a rate as tape and tends to twist under tension. For this reason, only tape recording will be considered here. The tape used has generally a plastic or paper backing coated with a mixture of $F_{e2}O_3$ (70 per cent.) and binder (30 per cent.)—the latter normally **cellulose nitrate.** The standard dimensions are: width = 0·25 inches, overall thickness = 0·002 inches* (backing = 0·0015 inches, coating = 0·0005 inches); length: depending on duration of programme required and spool size.

The coating has to be of very fine molecular structure and it has to be fixed as firmly as possible to the backing.

(2) A MOTOR SYSTEM

(*i*) To pull the tape past the heads at a constant speed;
(*ii*) To spool up the tape as it leaves the constant speed drive; and
(*iii*) To re-spool the tape after each recording or reproduction.
The first requirement is met by a spring-mounted motor, flexibly coupled or directly connected to a capstan against which the tape is pressed by a pressure roller. Either the capstan or the pressure roller is always rubber tyred. Spooling and re-spooling is effected either by a belt drive from the constant speed drive motor to slipping clutches on the spool plates or by using separate motors for spooling and re-spooling. In the latter case the trailing spool motor is generally energized with a power below the maximum so that it is pulled backwards against the direction of drive by the tape—thus keeping the latter taut.

(3) THREE ELECTRO-MAGNETS

These form respectively:

(*i*) The erase head;
(*ii*) The recording head; and
(*iii*) The replay head.

In many home recorders the recording head and the reproducing head are combined. Of course this is a drawback because only one function can then be performed at a time, *i.e.* it is impossible to listen to the reproduction whilst the recording is being made—an essential requirement in professional recorders.

(4) A RECORDING AMPLIFIER

Possibly a microphone pre-amplifier will be included.

(5) A REPRODUCING AMPLIFIER

This can be the same as the recording amplifier (perhaps with an additional stage switched in) if a combined record/replay head is used.

(6) EQUALIZERS (TONE CONTROLS)

For both recording and reproduction.

(7) AN OSCILLATOR

To provide erase current for the erase head and bias current into the recording head.

* Even thinner tapes are now obtainable which enable a greater length of tape to be wound upon a spool, giving a longer playing time per reel.

(8) A RECORDING LEVEL INDICATOR (METER OR "MAGIC-EYE")

(9) AN OUTPUT LEVEL METER

(This is omitted in most domestic type recorders).

7. Ring-Type Heads

The electro-magnets used are usually of the **ring-type.** Two half-rings of laminated permalloy are separated by non-magnetic spacers called **shims.** The coils are wound round the two halves **astatically,** *i.e.* series aiding to programme and in series opposition to stray fields, see Fig. 17.

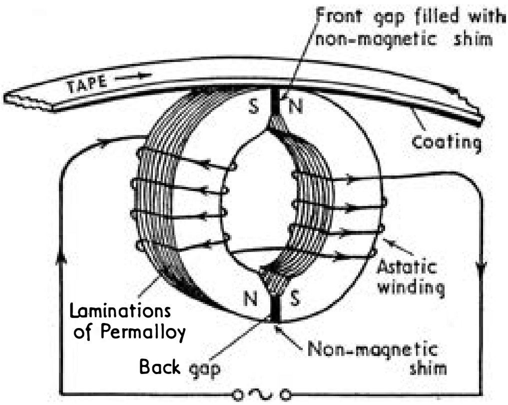


FIG. 17. A RING-TYPE HEAD.

The heads have to be designed to ensure good contact between the tape coating and the front gap. The gaps must be rigidly aligned at right angles to the direction of tape travel and the heads must be well shielded both electromagnetically and electrostatically. (See page 92).

In the recording head, the back shim is made thicker than the front to ensure a more level relationship between magnetizing current and magnetizing flux than would otherwise be the case. If there were no back gap, the head and tape would form a complete magnetic circuit of ferrous material. This is undesirable because the effective permeability (ratio B/H) of iron decreases with increase in the frequency of the magnetizing current. A long back gap introduces into the magnetic path a high reluctance of constant permeability.

8. How unwanted Signals are Erased

A high frequency alternating current is applied to the erase-head coils and this produces an alternating magnetic flux. Because the flux into the tape is not uniform throughout the gap (Fig. 18) the tape passing over it is subjected first to gradually increasing values of alternating magnetizing force and then to gradually decreasing cycles ending at zero. In this way all previous magnetic variations are

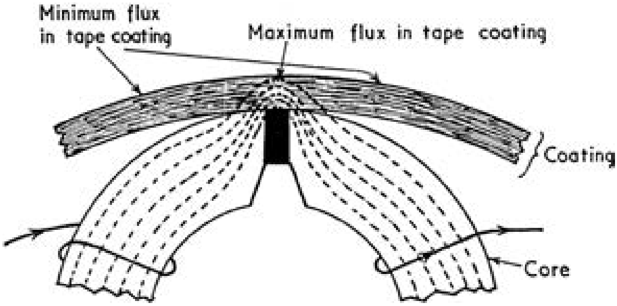


FIG. 18(a). WHEN A CURRENT IS FED INTO THE COIL OF A RING-TYPE HEAD THE LINES OF FORCE INDUCED IN THE CORE PASS STRAIGHT ACROSS THE GAP IN THE ABSENCE OF TAPE, BUT IF THE TAPE COATING IS ACROSS THE GAP THE LINES OF FORCE FIND THE COATING A PATH OF LOWER RELUCTANCE THAN THE NON-MAGNETIC SHIM. THE FIGURE SHOWS THE SPREAD OF THE FLUX IN THE TAPE COATING OVER THE FRONT GAP IN AN ERASE HEAD.

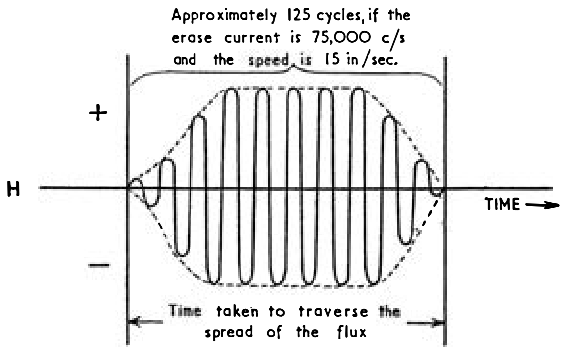


FIG. 18(b) EXAMINATION OF FIG. 18(a) WILL SHOW THAT THE MAXIMUM MAGNETIZING FLUX IN THE TAPE COATING OCCURS AT THE CENTRE OF THE GAP. IF THE CURRENT APPLIED TO THE COIL OF THE HEAD IS ALTERNATING, THE RESULTANT VARIATIONS IN MAGNETIZING FORCE APPLIED TO THE TAPE COATING AS IT TRAVERSES THE GAP WILL BE SMALL AT FIRST, WILL REACH A MAXIMUM WHEN THE SEGMENT REACHES THE CENTRE OF THE GAP AND WILL THEN DIMINISH TO ZERO AS THE TAPE PASSES TO THE FAR SIDE OF THE GAP.

removed and the tape leaves the head in a state of practically zero magnetization. (See Fig. 19 overleaf).

9. Recording Process

DERIVATION OF LOW FREQUENCY TAPE TRANSFER CHARACTERISTIC

A segment of the coating of the tape is subjected to a certain peak magnetizing force as it passes over the gap in the recording head. The magnitude of this depends upon the instantaneous value of the signal flux when the segment reaches a point in the gap. If the corresponding values of remanent induction are plotted against the peak values applied to successive segments of the tape, a curve will be obtained (Fig. 20) which is the low frequency tape transfer characteristic. This characteristic applies only to low frequencies because at high frequencies the tape is subjected to more than half a cycle of signal during the transit of the gap. At these high

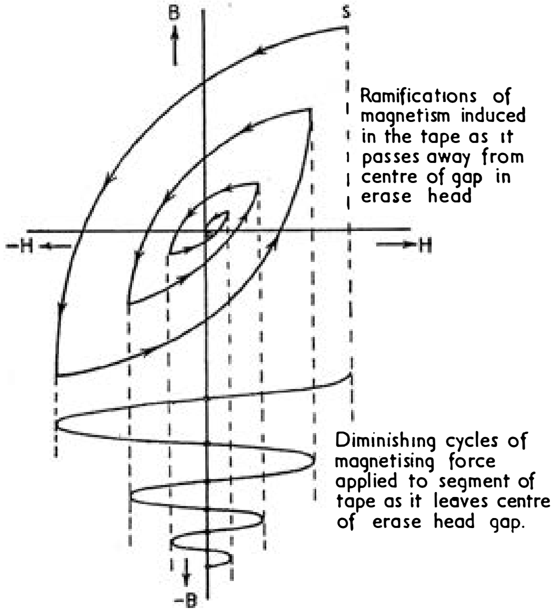


FIG. 19. TO ENSURE ERASURE OF ANY SIGNAL PREVIOUSLY RECORDED, THE CURRENT INTO THE ERASE HEAD IS MADE SUFFICIENT TO SATURATE THE COATING WHEN IT REACHES THE MIDDLE OF THE GAP IN THE ERASE HEAD. THEN, AS THE TAPE LEAVES THE CENTRE OF THE GAP, THE MAGNETIZING FORCE APPLIED PROGRESSIVELY DIMINISHES TO ZERO, LEAVING THE COATING DEMAGNETIZED.

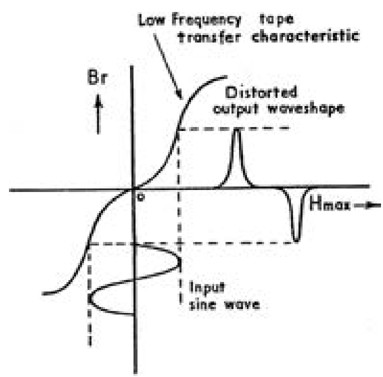


FIG. 20. THE LOW FREQUENCY TAPE TRANSFER CHARACTERISTIC IS DOUBLE-S SHAPED AS SHOWN. THE APPLICATION OF A SINUSOIDALLY VARYING MAGNETIZING FORCE PRODUCES A DISTORTED RECORDING.

frequencies the theoretical transfer characteristic is very complicated and cannot be represented by a single curve. However, it may be experimentally verified that at such frequencies the apparent transfer characteristic is linear. This is due to three factors which tend to attenuate the uneven harmonics of the otherwise distorted high frequency output: reduction in effective permeability with increase in frequency; self-demagnetization; and the frequency discrimination due to the finite length of the gap in the replay head.

Returning to the low frequency characteristic, it will be seen that, because of the insteps in the initial magnetization curves of the tape coating, the transfer characteristic is double-S shaped. This results in severe distortion of all low frequency signals when recorded without bias. For many years after the invention of magnetic recording this difficulty had to be overcome by applying a saturating direct current in the erase head and a direct current bias in the demagnetizing direction in the record head—thus bringing the signal variations on to a straighter part of the hysteresis loop. This system is illustrated in Fig. 21.

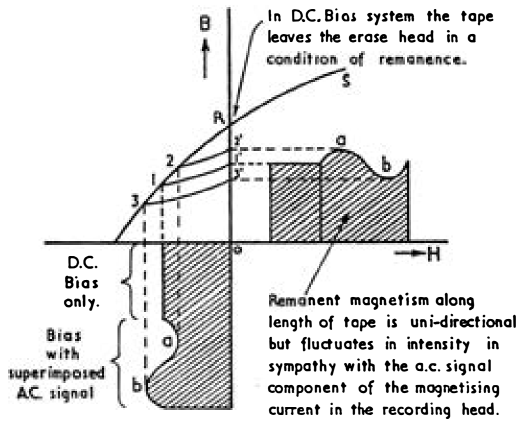


FIG. 21. THE EFFECT OF SATURATING THE TAPE IN THE ERASE HEAD AND APPLYING A D.C. BIAS OF CRITICAL VALUE IN THE RECORDING HEAD IS SHOWN ABOVE. THIS SYSTEM IS NOW RARELY USED FOR AUDIO-FREQUENCY RECORDING.

Though d.c. bias used in this way effectively minimized the distortion, it resulted in a poor programme-to-noise ratio and an impaired frequency response—50 to 6,000 cycles per second. (The higher audio frequencies could be *recorded* but they had to be filtered in the output because the background noise at frequencies above 6,000c/s was very high). Nearly all modern magnetic recorders now use a.c. bias of supersonic frequency for audio frequency recording. The way this works may be examined by recording without bias and applying a simple additive mixture of two dissimilar low frequencies to the input to the recorder. If the *input* waveform is then examined on a cathode ray oscilloscope it will be seen that the combined waveform consists of the higher frequency varying about a datum line varying at the lower frequency, see Fig. 22.

Similarly examined, the output will be found to be made up of the higher frequency typically distorted, but varying at the lower frequency, see Fig. 23.

If, when recording the two frequencies, the frequency of the higher frequency is gradually raised, inspection of the output on the C.R.O. will show that first the higher frequency becomes less distorted (as it approaches the frequency at which the transfer characteristic begins to become more linear); then, as the frequency of the higher

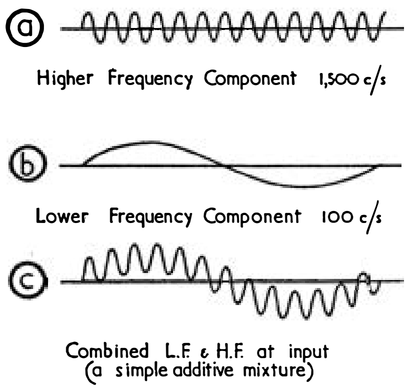


FIG. 22. THE EFFECT OF A.C. BIAS MAY BE EXAMINED BY APPLYING A SIMPLE ADDITIVE MIXTURE OF 100c/s AND 1,500c/s PURE TONE TO THE RECORDING HEAD—WITHOUT ANY SUPERSONIC BIAS

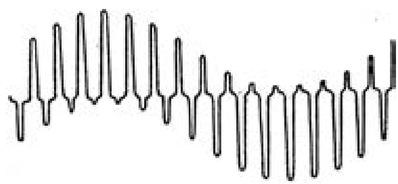


FIG. 23. RECORDED WAVESHAPE WHEN 100c/s AND 1,500c/s ARE RECORDED TOGETHER WITHOUT THE NORMAL A.C. BIAS IN THE HEAD.

tone is still further increased, the combined effects of (*i*) self-demagnetization, (*ii*) reduction in coating permeability with increasing frequency, and (*iii*) the gap effect (see page 27), in the reproducing head, will gradually attenuate the higher frequency component until when the higher frequency becomes supersonic, only the low frequency signal component will be left (see Fig. 24). To achieve this desirable effect, it is necessary that the a.c. bias shall be of a certain optimum amplitude. Fig. 25 is a graph in which three

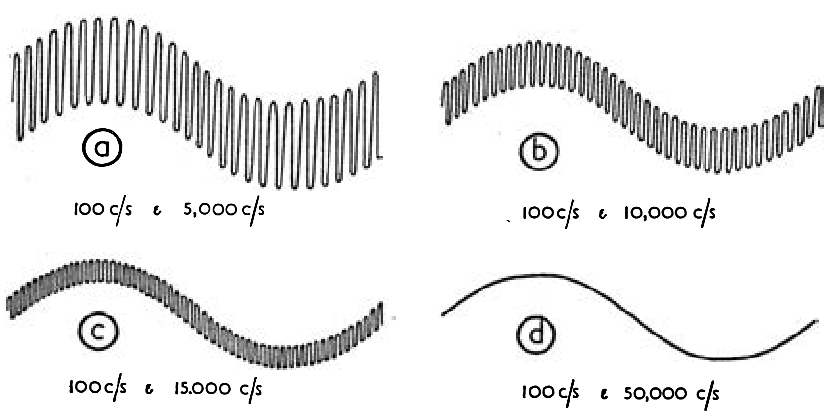


FIG. 24. AS THE HIGHER FREQUENCY IS GRADUALLY INCREASED IN FREQUENCY IT BECOMES PROGRESSIVELY ATTENUATED, AS THE SUCCESSIVE OUTPUT WAVESHAPES SHOW.

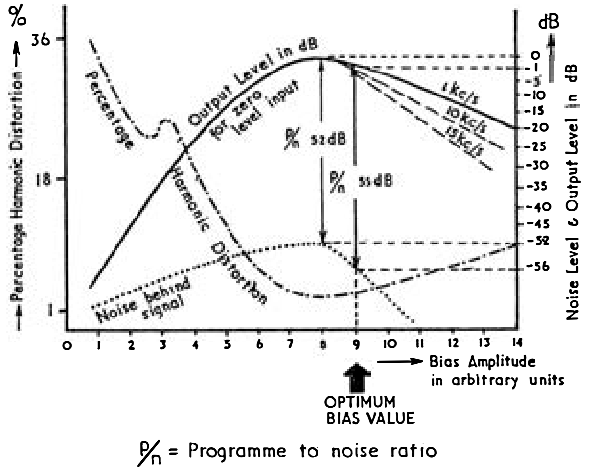


FIG. 25. THIS GRAPH (RELATING (i) PERCENTAGE HARMONIC DISTORTION, (ii) OUTPUT LEVEL FOR ZERO LEVEL INPUT, (iii) BACKGROUND NOISE BEHIND SIGNAL, WITH BIAS AMPLITUDE) SHOWS THAT THE BEST PROGRAMME-TO-NOISE RATIO IS OBTAINED, WITH PRACTICALLY MINIMUM DISTORTION, WHEN THE BIAS AMPLITUDE IS JUST A LITTLE GREATER THAN THAT REQUIRED TO PRODUCE MAXIMUM OUTPUT LEVEL.

things ((*i*) percentage harmonic distortion, (*ii*) signal level out for normal level in, (*iii*) background noise behind the signal) are plotted against bias amplitude, in arbitrary units. It will be seen that the amplitude of bias which gives maximum output for normal input also produces minimum distortion. In practice a slightly higher value of bias amplitude is usual because this produces an improvement in signal-to-noise ratio. If too much bias is used, the wiping effect of the bias begins to attenuate the higher frequencies in the signal.

Some home recorders have a restricted frequency range. When this is the case, a better programme-to-noise ratio can be obtained by using rather heavy bias current.

10. The Reproducing Process

A signal consisting of a pure tone is recorded on magnetic tape in the form of a series of little magnets of alternately opposing polarity, the length of each being half a recorded wavelength (see Fig. 26). These little induced magnets give rise to external lines of force which are distributed as shown in Fig. 27.

When the tape is brought into contact with the gap in a replay head, the lines of force find the core of the head a path of lower reluctance and they close round the core. (See Fig. 28). As the tape is pulled past the gap, the lines of force in the core alternate in direction. One of the important electromagnetic phenomena is that changing lines of force in a core produce an alternating voltage at the terminals of a coil wound round the core. The movement of the tape

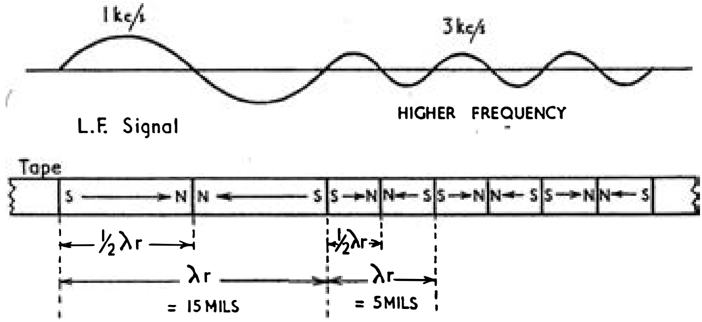


FIG. 26. SHOWING HOW THE SINUSOIDALLY VARYING MAGNETIZING FORCE PRODUCES A SERIES OF MAGNETS ALONG THE LENGTH OF THE TAPE, EACH HALF A RECORDED WAVELENGTH LONG AND OF OPPOSING POLARITY. THE WAVELENGTHS MARKED ARE THOSE OBTAINING WHEN THE TAPE SPEED IS 15 IN./SEC.

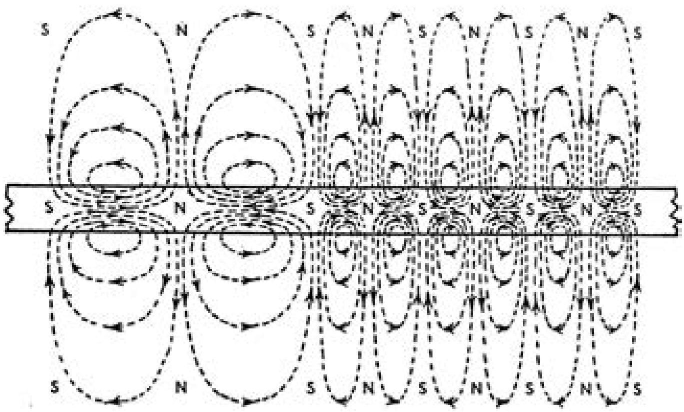


FIG. 27. LINES OF FORCE IN THE TAPE COATING AND EXTERNAL TO THE TAPE. THE ARROWS INDICATE THE DIRECTION OF THE LINES OF FORCE. THE SPREAD OF THE FLUX HAS BEEN PURPOSELY EXAGGERATED FOR CLARITY. IF THE EXPERIMENT IS TRIED OF LIFTING A RECORDED TAPE AWAY FROM THE REPRODUCING HEAD WHILST REPRODUCING, IT WILL BE FOUND THAT THE ACTUAL SPREAD OF THE FLUX IS ONLY A FRACTION OF AN INCH. IF THE TAPE IS SEPARATED FROM THE HEAD BY AS LITTLE AS ¼" THE OUTPUT SIGNAL WILL BE PRACTICALLY ZERO.

therefore produces an output voltage at the terminals of the replay head coil. Because the voltage is proportional to the rate of change of the lines of force, the output voltage doubles if the frequency is doubled (if the recorded intensity is constant despite the change in frequency). The output voltage thus rises 6dB per octave (see Appendix B). This rise has to be compensated by appropriate equalization in the reproducing chain. The 6dB per octave rise is not maintained all over the audio spectrum. At very low frequencies, the longer recorded wavelengths reduce the amount of flux threading the coil (see Fig. 29) and the fall with decrease in frequency increases to 12dB per octave and then to 18dB per octave.

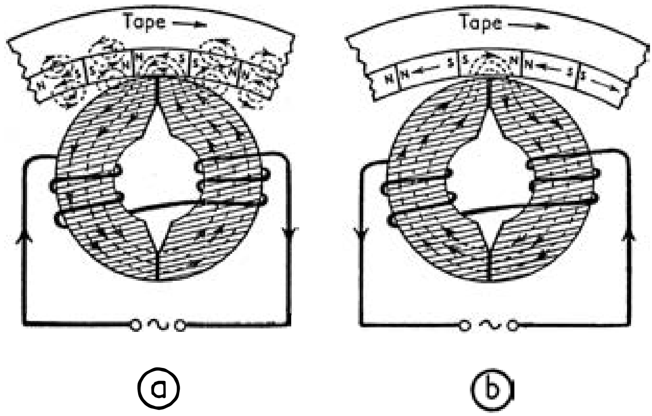


FIG. 28. DISTRIBUTION OF THE FLUX IN THE TAPE AND IN THE CORE OF THE REPRODUCING HEAD WHEN THE RECORDED SIGNAL IS OF MID-AUDIO-FREQUENCY. (a) AT THE PEAK OF THE POSITIVE HALF-CYCLE. (b) AT THE PEAK OF THE NEGATIVE HALF.

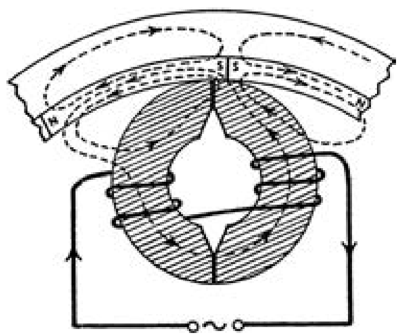


FIG. 29. DISTRIBUTION OF FLUX IN THE CORE OF THE REPLAY HEAD WHEN REPRODUCING VERY LOW AUDIO FREQUENCIES. THIS ACCOUNTS FOR THE FALL IN OUTPUT AT SUCH FREQUENCIES.

At high frequencies the finite length of the gap in the reproducing head has an attenuating effect which reaches a maximum when the recorded wavelength is equal to the effective length of the gap (Fig. 30). This is called the **gap effect** and the recorded frequency at which it occurs is called the **extinction frequency.** Although theoretically affecting all frequencies the attenuation due to the gap effect is not marked at frequencies up to half the extinction frequency. For example, if the extinction frequency is 11,500 cycles per second (which it is when the tape speed is $7\frac{1}{2}$ in./sec. if the effective gap length in the replay head is 0·65 mil) the attenuation due to gap effect is only 3dB at 5,750c/s. A typical unequalized response curve is given in Fig. 31 (overleaf).

In recent years experiments have been made with very short gaps in replay heads—gap lengths of only a few microns (a micron is a millionth part of a metre) have been tried and this has enabled many manufacturers of home recorders to offer extended high frequency response at very low speeds. Unfortunately, in order to achieve the recording of such high frequencies at low tape speeds it is necessary to reduce the amplitude of a.c. bias fed to the recording head and this results in an undesirable increase in non-linear distortion. However, though the additional distortion thus produced would be unacceptable in professional type recorders it is tolerable in domestic ones.

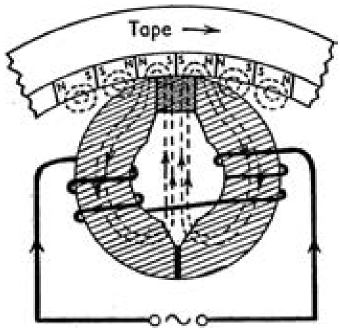


FIG. 30. THE GAP EFFECT. AT THE FREQUENCY AT WHICH THE RECORDED WAVELENGTH IS EQUAL TO THE EFFECTIVE LENGTH OF THE GAP IN THE REPLAY HEAD (A LITTLE LONGER THAN THE PHYSICAL GAP) THE INDUCED FLUX IN THE TWO HALVES OF THE CORE OF THE HEAD IS IN THE SAME DIRECTION IN EACH HALF. THOUGH THE FLUX CHANGES AS THE TAPE IS MOVED, THE INDUCED VOLTAGE IN EACH HALF OF THE COIL IS IN OPPOSITION TO THAT INDUCED IN THE OTHER HALF. A MINIMUM THEREFORE OCCURS IN THE OUTPUT. THE FREQUENCY AT WHICH THIS TAKES PLACE IS CALLED THE EXTINCTION FREQUENCY (f_{ex}).

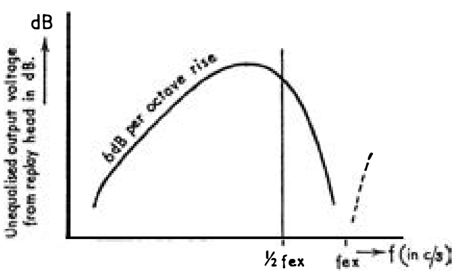


FIG. 31. GRAPH SHOWING HOW THE OUTPUT VOLTAGE FROM THE REPLAY HEAD VARIES WITH FREQUENCY WHEN THE RECORDED SIGNAL IS OF CONSTANT R.M.S. VALUE.

Number of Tracks

Professional recorders use heads which are the full width of the tape and therefore record only one track. Most home recorders have two or four tracks, recorded with heads of the appropriate widths. Whilst four tracks make the most economical use of the tape, the signal-to-noise ratio obtainable is not so good as with the two track variety. If you desire the best quality recording it is therefore better to choose a one or two track recorder—though of course each recording will then cost more.

2
DISTORTIONS

In order to reproduce the original sound as faithfully as possible, both the recording and reproducing system must be as free as possible from unintentional distortions.

Some of the principal kinds are:

1. Attenuation Distortion (often called Frequency Distortion)

In this form, the component frequencies of the sound are reproduced in incorrect comparative amplitude (Fig. 32). In magnetic

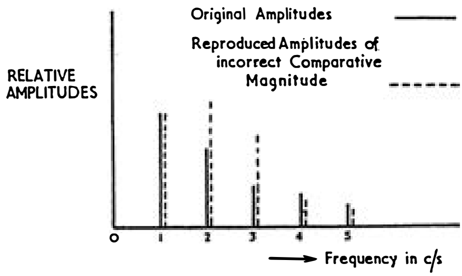


FIG. 32. AN EXAMPLÉ OF ATTENUATION DISTORTION.

recording unintentional attenuation distortion is primarily due to the combined effects of (*i*) the decrease in the effective permeability of the magnetic material with increase in frequency; (*ii*) self-demagnetization; (*iii*) the gap effect in the reproducing system; (*iv*) imperfect contact between tape and head; and (*v*) eddy current and hysteresis losses.

Other causes of frequency discrimination are **reactive elements** (capacitors and inductors) in the recording and replay chains.

Attenuation distortion is countered by using appropriate equalization and by the use of negative feedback circuits in the amplifiers.

2. Non-linear Distortion

This is due to the curvature of some transfer characteristics, typical examples being the S-shaped I_a and V_g curves of valves and the double S-shaped low-frequency tape transfer characteristic mentioned in Chapter 1. There are three main kinds of non-linear distortion:

(1) HARMONIC

This is the introduction of spurious overtones (not necessarily in harmonic sequence). Fig. 33 shows how the parabolic mutual characteristic of a valve may produce a second harmonic of the input signal.

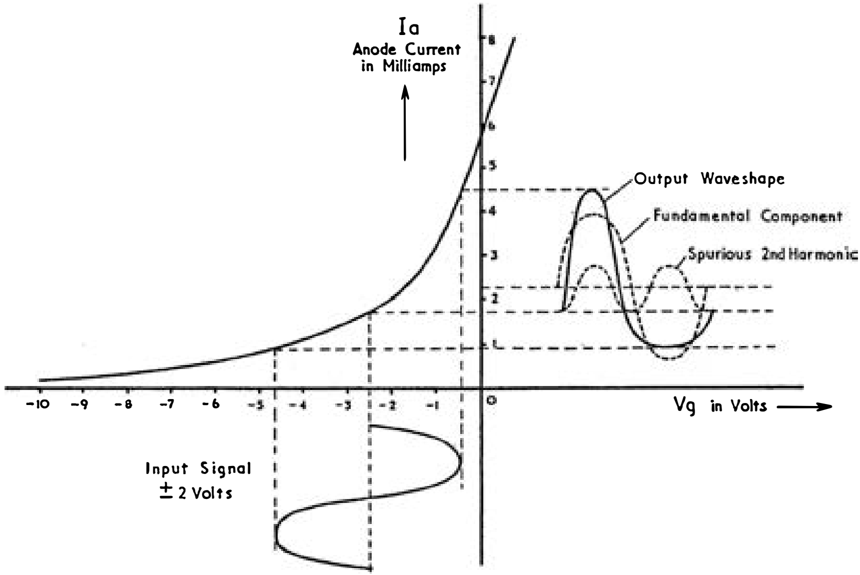


FIG. 33. EXAMPLE OF THE PRODUCTION OF 2ND HARMONIC DISTORTION BY THE PARABOLIC CHARACTERISTIC OF A VALVE. NOTE THE CHANGE IN THE VALUE OF THE D.C. COMPONENT OF THE ANODE CURRENT.

(2) INTERMODULATION

This occurs when two or more pure tones (as in a complex sound) are applied to the input of a non-linear device, producing **sum** and **difference tones** not present in the original sound. For example, if 1,000c/s and 1,500c/s were mixed at the input to a non-linear device, the output would consist of a mixture of the following frequencies: 1,000, 1,500, 2,500 and 500c/s together with harmonics of all these. By using only the linear part of a transfer characteristic, intermodulation distortion is kept to a minimum.

(3) AMPLITUDE

This occurs when, due to a non-linear transfer characteristic, the amplification obtained from an amplifier is dependent upon the size of the input signal. Thus, when the input signal is large enough to reach the knee of the characteristic, further increases in input do not result in proportionate increases in output. The right choice of bias point and careful control of the size of the input signal are factors tending to limit amplitude distortion.

All types of non-linear distortion are reduced by negative feedback and by avoiding overload.

3. Transient

Highly damped (rapidly decaying) waveforms as shown in Fig. 34 (all percussion instruments, drums, cymbals, pianos, etc., as

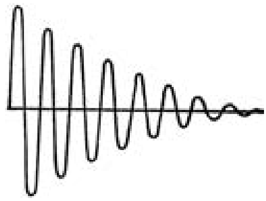


FIG. 34. EXAMPLE OF A TRANSIENT WAVESHAPE.

well as the human voice and extraneous noises are full of these) give rise to distortion because of their shock effect upon electrical and mechanical resonant circuits. They are responsible for rattling on peaks and for a tendency for certain notes to be prolonged in the reproduction beyond their original duration.

Transient distortion is minimized by damping (adding resistance to the circuit) and by the use of negative feedback (feeding a small part of the output voltage back to the input in antiphase to the latter).

4. Noise

Unwanted background noise (hiss, hum, ringing, etc.) should really be regarded as distortion of the original sound—though it is not generally so labelled. A special section is devoted to this (Chapter 10).

5. Flutter

Very short, rapid variations in tape speed or contact causing similar variations in the pitch or volume of the reproduced sound.

6. Wow

Apparent variation in the pitch of sustained notes caused by slow variations in the tape speed.

3

IMPEDANCE AND IMPEDANCE MATCHING

1. Impedance

THE opposition to alternating current flow presented by the elements in an electrical circuit is termed the **impedance of the circuit.** The impedance may be a pure resistance or a pure reactance or a mixture of resistance and reactance. Reactance is the opposition to current flow provided by a coil or a capacitor. Both impedance and reactance are measured in ohms. The reactance, X_L, of a coil is always $2\pi fL$ ohms where L is the inductance of the coil in henrys and f is the frequency in cycles per second and π is a constant $= 3{\cdot}14$. Doubling the frequency does not necessarily double the reactance because, unfortunately, the effective inductance of a coil with an iron core changes with a change in frequency—because the relative permeability of iron (which is one of the deciding factors) changes with change in frequency. A coil always has resistance as well as reactance because there is no such thing as a resistance-less wire. It is possible however to wind a coil in such a way that its inductive reactance is zero. The reactance, X_c, of a capacitor is $\frac{1}{2\pi fC}$ ohms (where C is the capacitance in farads and f is the frequency), and is inversely proportional to frequency. Inductive reactance is always given a positive sign and capacitive reactance a negative sign. When a capacitance is connected in series with a coil the reactances cancel out at the frequency at which $X_c = X_L$. At this frequency, called the **resonant frequency,** the only opposition to the flow of alternating current is provided by any resistance in the circuit. A series resonant circuit is therefore called an **acceptor circuit.** The resonant frequency of an acceptor circuit is $\frac{1}{2\pi\sqrt{LC}}$ cycles per second where L is the inductance in henrys and C is the capacitance in farads. The impedance, Z_s, $= \sqrt{R^2 + (X_L \sim X_c)^2}$ ohms. (The sign $\sim$ means that the term in the brackets is the difference between the magnitudes of X_L and X_c).

A coil connected in parallel (sometimes called in shunt) with a capacitor is called a **rejector circuit.** At its resonant frequency such a circuit offers a very high impedance to alternating current in the external circuit, if the resistance is low. The larger the internal resistance of a rejector circuit, the lower will be its impedance.

Resonance occurs at a frequency equal to

$$\frac{1}{2\pi}\sqrt{\frac{1}{LC} - \left(\frac{R}{L}\right)^2} \text{ cycles per second.}$$

When R (the resistance) is very much smaller in magnitude than X_L, the resonant frequency equals approximately $\frac{1}{2\pi\sqrt{LC}}$ cycles per second, as in the series case.

The parallel impedance $Z_p = \dfrac{\frac{1}{2\pi fC}\left[\sqrt{R^2 + (2\pi fL)^2}\right]}{\sqrt{R^2 + \left(2\pi fL \sim \frac{1}{2\pi fC}\right)^2}}$

At resonance, this simplifies to $Z_p = \frac{L}{CR}$

The behaviour of any piece of electrical apparatus is nearly always modified by the value of its internal impedance and by the impedances connected to it.

Important rules govern the maximum transfer of power from a generator to a load, *e.g.* from a microphone to an amplifier or from an amplifier to a loudspeaker.

2. Matching

The term matching is often very loosely used. A load impedance can truly be said to be matched to a generator impedance only when it is the conjugate of the latter. (The term " conjugate " is explained below). It is usually held that, when the load impedance is matched to the generator impedance, there will be maximum transfer of power from the generator to the load but, as will be shown, this is only true if the generator impedance is fixed and the load impedance is variable. If the load impedance is fixed, there will be maximum transfer of power when the generator impedance is as small as possible.

EFFECT OF MATCHING ON DISTORTION

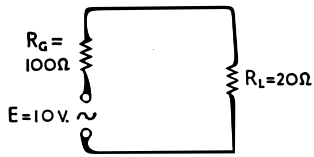
In many cases maximum power transfer is accompanied by an increase in non-linear distortion. A compromise value of load impedance has then to be chosen, *i.e.* the one that will give as much power into the load as it is possible to obtain without exceeding a given percentage of non-linear distortion. This value of load impedance is often erroneously designated the matching impedance. A wrong load impedance can also in certain circumstances produce attenuation distortion.

3. Maximum Power Transfer

CASE 1. **When the generator impedance is fixed and the load impedance is variable,** maximum transfer of power will take place when the resistance component of the load impedance is equal to the resistance component of the generator impedance and the load reactance is the negative of the generator reactance. In these circumstances the load impedance is said to be **the conjugate** of that of the generator. When the load impedance can be changed, but not its reactive element, maximum transfer of power will occur when the overall values of the impedances are made equal.

A simple resistive example is given in Fig. 35.

FIG. 35. AN EXAMPLE OF MISMATCHED LOAD AND GENERATOR IMPEDANCES. INCREASING R_L to 100Ω WOULD INCREASE THE POWER IN THE LOAD FROM 0·139 TO 0·25W.



Here the current produced $I = \dfrac{E}{R_G + R_L} = \frac{1}{12}$ ampere

and the power absorbed by $R_L = I^2 R_L = \dfrac{20}{144} = \dfrac{1}{7{\cdot}2} = 0{\cdot}139$ watt.

If R_L were increased to 100Ω* (equal to R_G), the current in R_L would be reduced to $\frac{1}{20}$ ampere but the power absorbed by the load would be nearly doubled (0·25 watt).

CASE 2. **When the load impedance is fixed but the source impedance is variable,** the maximum transfer of power will take place if the generator's resistive element is zero and its reactive element is the negative of the reactance of the load.

For example, if the impedance of the load is the conjugate of that of the generator, as shown in Fig. 36, and the values are as given,

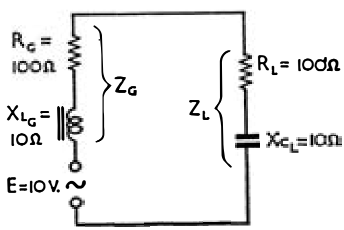


FIG. 36. MAXIMUM POWER IN THIS LOAD IMPEDANCE (Z_L) WILL BE INCREASED IF THE SIZE OF THE GENERATOR RESISTANCE IS REDUCED.

the total impedance $Z_{TOT} = 200\,\Omega$ because the reactances cancel. The current in the circuit will be $\frac{1}{20}$ ampere and the power absorbed, $I^2R_L = \frac{1}{4}$ watt. But if R_G is reduced to zero, the total impedance will be 100Ω, the current will be doubled and the power absorbed in the load will be 1 watt—four times as great as before.

If the generator resistance is variable and the load resistance and reactance are fixed, maximum power will be transferred to the load when the generator resistance is as small as possible, even if the reactances of load and generator do not cancel.

* Ω is the Greek letter *Omega* and is used to signify ohms.

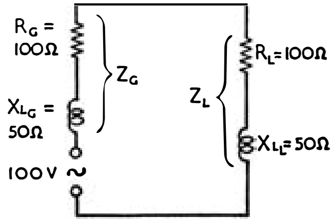


FIG. 37 REDUCING THE GENERATOR RESISTANCE R_G TO 10Ω WOULD INCREASE THE POWER IN THE LOAD FROM 20W TO 45W.

For example, if as shown in Fig. 37, both load and generator consist of resistance in series with inductance of the values shown, the total impedance in the circuit will be $2\sqrt{10^4 + 50^2} = 224\Omega$. The current I circulating will be

$$\frac{E}{Z_{TOT}} = \frac{1}{2{\cdot}24} \text{ ampere}$$

and the power absorbed by the load will be

$$I^2R_L = \frac{10^2}{5} = 20 \text{ watts.}$$

Now, if R_G can be reduced to 10Ω the total impedance will be reduced to $\sqrt{110^2 + 10^4} = 149\Omega$. The current will be increased to $\frac{1}{1{\cdot}49}$ amp. and the power in the load will be $\frac{10^2}{2{\cdot}22} = 45$ watts, *i.e.* it will be more than doubled.

This explains why it is often satisfactory to feed a high impedance input from a low impedance source, though the converse does not hold good. **Even in such a case, a suitable transformer matching the generator impedance to that of the load will increase the power into the load.**

4. Use of Transformers for Impedance Matching

An efficient match can be obtained by the use of a transformer. An audio frequency transformer usually consists of two or more windings on a core of laminated iron alloy.

An alternating current fed into the primary winding produces an alternating magnetic field which cuts the wires of the secondary winding and induces in it an e.m.f. which bears the same relationship to the e.m.f. applied to the primary as the relationship between the respective number of turns in the secondary and primary coils. Thus, 100V applied to a step-up transformer with a turns ratio of 10 : 1 would produce 1,000V at the terminals of the secondary coil. The currents in primary and secondary are also transformed, *i.e.* a transformer which steps up the voltage will step down the current to the same extent.

Assuming no loss of power (many transformers are 99 per cent. efficient) the power in the secondary circuit will be equal to the power in the primary. Now, power is the product of volts and amperes, *i.e.* VI. The power in the primary, V_pI_p, is therefore equal to the power in the secondary, V_sI_s. The primary current $I_p = \frac{V_p}{Z_p}$ where Z_p is the primary impedance, whilst the secondary current $I_s = \frac{V_s}{Z_s}$ where Z_s is the secondary impedance,

$$\therefore \frac{V_p^2}{Z_p} = \frac{V_s^2}{Z_s} \text{ and } \frac{Z_s}{Z_p} = \frac{V_s^2}{V_p^2} = \text{the square of the turns ratio.}$$

Thus, the impedance ratio of a transformer is the square of the turns ratio.

5. Regulation

Another important factor is what is called **the regulation.** If a transformer with 100V applied to the primary steps up the voltage to 200V at the secondary terminals when these are open circuited, the connection of a load resistance which causes current to flow in the secondary circuit will reduce the voltage at the secondary terminals because of the power dissipated in the resistive element of the transformer impedance when the load takes current.

This is analogous to the fall in battery voltage when a battery is connected to a load resistance, because of the power dissipated in the internal resistance of the battery. (See Fig. 38).

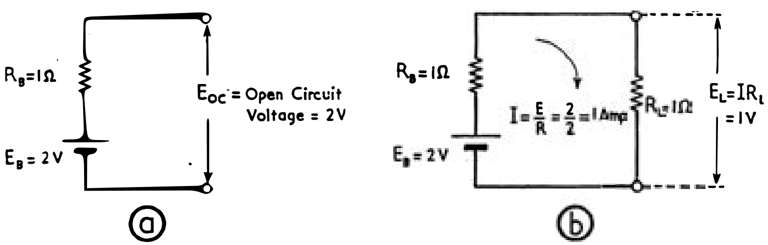


FIG. 38. SHOWING HOW THE INTERNAL RESISTANCE OF A BATTERY REDUCES THE TERMINAL VOLTAGE WHEN A LOAD RESISTANCE IS CONNECTED ACROSS THE OUTPUT.

EXAMPLE OF IMPEDANCE MATCHING BY MEANS OF A TRANSFORMER

Fig. 39 shows how a generator with an internal resistance of 1,000Ω could be matched to a 10Ω load by a 10 : 1 step-down transformer. Without the transformer, the current in the 10Ω load would be $\frac{10}{101} = \frac{1}{10{\cdot}1}$ ampere and the power into the 10Ω load would be $I^2R = \frac{10}{102} = \frac{1}{10{\cdot}2} = 0{\cdot}098$ watt.

The transformer would reflect 1,000Ω into the primary and the current in the primary circuit would be $\frac{100}{2000} = \frac{1}{20}$ ampere.

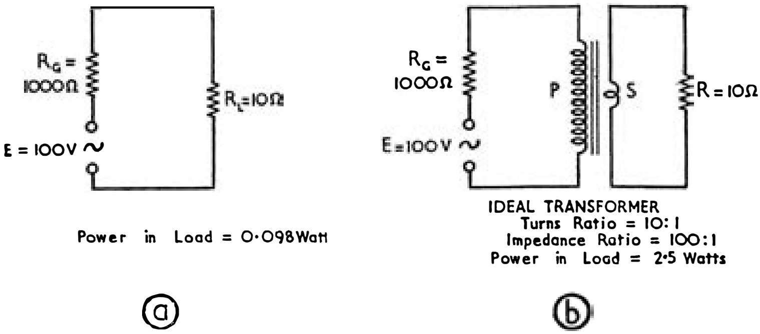


FIG. 39. SHOWING HOW A MATCHING TRANSFORMER CAN INCREASE THE POWER IN THE LOAD.

Assuming no loss in the transformer the current in the secondary would be $\frac{1}{2}$ ampere. The power in the load would therefore be $\frac{10}{4}$ = 2·5 watts—25 times what it would be without the transformer.

6. Impedance Matching by means of an Attenuator

Impedances can also be matched by the insertion of a suitable attenuator pad but such matching always results in a definite insertion loss. Formulae have been worked out that enable resistance values to be chosen to give a minimum insertion loss. For example, supposing it is necessary to match a generator of internal resistance 500Ω to a load resistance of 200Ω, an L-shaped pad consisting of a series resistor value 388Ω followed by a shunt resistor of 257Ω will provide a perfect match, see Fig. 40.

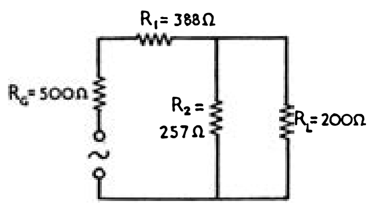


FIG. 40.
SHOWING HOW AN ATTENUATOR MAY BE USED TO MATCH A LOAD RESISTANCE TO THE GENERATOR RESISTANCE.

$$\text{The minimum loss in dB} = 20 \log_{10} \sqrt{\frac{R_G}{R_L}} + \sqrt{\frac{R_G}{R_L} - 1}$$

in this case approximately 9dB. The values of R_1 and R_2 are determined from the consideration that

$$R_G = R_1 + \frac{R_2 R_L}{R_2 + R_L} \qquad (1)$$

$$R_L = \frac{R_2 (R_1 + R_G)}{R_1 + R_2 + R_G} \qquad (2)$$

Hence, if R_L and R_G are known, R_1 and R_2 may be calculated.

If only R_G or R_L require to be matched, a single series or shunt resistor will provide efficient matching and the insertion loss is then less. For example see Fig. 41.

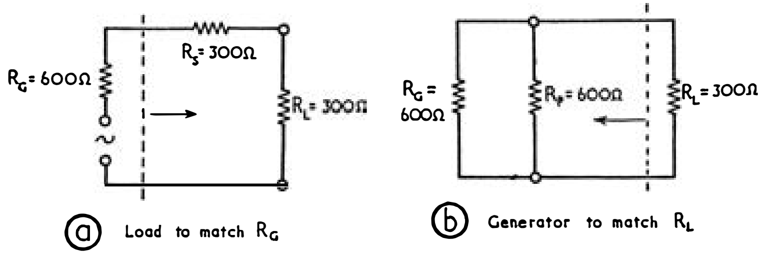


FIG. 41. SHOWING HOW A SINGLE RESISTOR IN SERIES OR PARALLEL WILL PROVIDE EFFECTIVE MATCHING LOOKING FORWARD INTO THE LOAD OR BACK INTO THE GENERATOR RESPECTIVELY.

7. Effect of Load Impedance on Attenuation Distortion

Whenever there are reactive elements in a circuit, the changing of the ratio of generator impedance to that of the load may have a marked effect on the frequency response. Though usually undesirable, this effect is sometimes made use of to obtain equalization (see Chapter 4).

8. Loudspeaker Loads

Especial care must be taken when connecting an amplifier to a loudspeaker. A wrong load here can not only affect the frequency response but may give rise to non-linear distortion as well.

9. Importance of Load Impedance in Prevention of Non-linear Distortion

Non-linear distortion increases in value when the signal into an amplifier reaches a high level and in order to keep this type of distortion to a minimum the input to the amplifier must not exceed a certain maximum value. For maximum undistorted signal output from the final stage, the amplifier must be loaded with the correct impedance. The value of this impedance is determined by the type of output valve (*e.g.* triode or pentode) but only in exceptional cases is its value that which matches the valve impedance and therefore causes maximum power to be delivered to the load. For a triode, the correct load is usually two to three times the r_a of the valve, whilst for a pentode it is considerably less than the r_a. As has been mentioned, the value chosen is a compromise between the value that would give maximum power amplification and that which would give minimum non-linear distortion. It is often very critical especially when the amplifier is working near its overload point. When an amplifier is connected to an attenuator or an equalizer it is essential that the attenuator or equalizer shall be not only of the correct impedance but of constant impedance.

10. Constant Impedance Attenuators

These are of various types but perhaps the simplest and best is that known as the Bridged-T, see Fig. 42. This is a four terminal network in which the variables are shown with arrows across them. The value of this particular arrangement is that it can be made to match both the generator and the load impedance whilst providing fully variable attenuation.

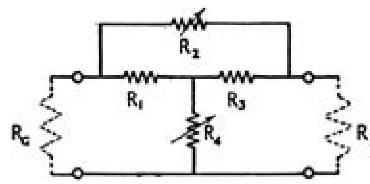


FIG. 42. THE VARIABLE ATTENUATOR KNOWN AS A BRIDGED-T.

$R_3 = R_1 (K-1)$ and $R_4 = R_1 \left(\frac{1}{K-1}\right)$ where the required loss $= 20 \log_{10} K$ or $K = \text{antilog} \left(\frac{\text{required loss in dB}}{20}\right)$ and $R_G = R_L = R_1 = R_2$.

HOW TO FIND OUT THE IMPEDANCE OF A GIVEN PIECE OF APPARATUS

In most cases the manufacturers of microphones and pick-ups state their impedance in an accompanying leaflet, but in case of doubt the retailer or manufacturer will always let you know on enquiry. Similarly, amplifier manufacturers generally state the input and output impedances or will give the necessary information if asked. An easy way of finding out the impedance for yourself, if you have the necessary apparatus, is to connect a source of low value alternating e.m.f. (**not the mains**) to the input or output terminals in series with a suitable milliammeter to measure the resultant current. The voltage applied is then measured with a suitable voltmeter and the impedance at the supply frequency is discovered from the formula

$$Z = \frac{E}{I}.$$

Thus if the voltage applied has an r.m.s. value of 1 volt and the resulting current is 1·67 milliamps the impedance $= \frac{10^3}{1{\cdot}67} = 600\Omega$ at the supply frequency.

MATCHING OUTPUT LEVEL TO REQUIRED INPUT

When you wish to feed the output of a microphone or pick-up into an amplifier, it is necessary to know the approximate output level of the microphone or pick-up and the minimum input level required by the amplifier. If the microphone or pick-up output is lower than the minimum input required a step-up transformer or a pre-amplifier will be necessary.

4

FILTERS AND EQUALIZERS

If an attempt is to be made to improve the frequency response of a recorder some knowledge of electric filters and equalizers is required. Both these are frequency discriminating networks, but whereas the filter tends to discriminate sharply, *i.e.* to pass one band of frequencies and attenuate another, equalizers do not as a rule discriminate so sharply—they introduce attenuation gradually as the frequency is changed. A filter is generally made up of a combination of capacitors and inductors. Equalizers normally include resistances as well as reactances.

Equalizers and filters present one impedance at their input and another (possibly the same value) at their output. Equalizers designed to maintain their impedance at a constant value despite changes in frequency when fed from a given impedance into a similar one are called **constant impedance equalizers.** The design of such equalizers is beyond the scope of this handbook but full details may be found in many textbooks, for example *Elements of Sound Recording* by Frayne and Wolfe (Chapman and Hall, London).

Two-element Equalizers

Simple two-element networks consisting of a resistor in series or parallel with an inductor (coil) or capacitor always offer an impedance which tends to a constant value over a certain range of frequencies and to a value which changes with frequency at another range of frequencies. (See Fig. 43).

Simple equalizers may be constructed from two-element networks as shown in Fig. 44.

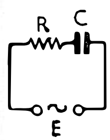
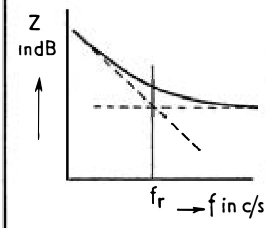
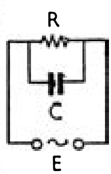
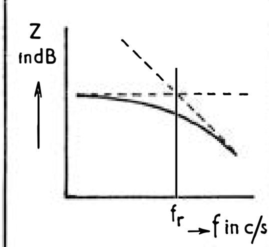
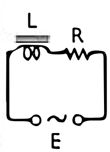
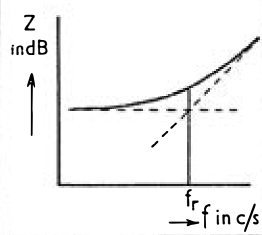
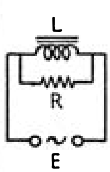
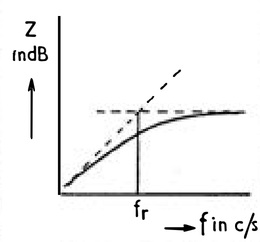
CIRCUIT	IMPEDANCE	IMPEDANCE V_s FREQUENCY	TURNOVER FREQUENCY, f_r	VALUE OF REACTANCE if f_r & R are known
	$\sqrt{R^2+X_C^2}$		$\frac{1}{2\pi CR}$ c/s where C is in Farads and R is in Ohms Z_s at f_r = 1·4 RΩ	$C = \frac{10^6}{2\pi f_r R}$ microfarads
	$\frac{RX_C}{\sqrt{R^2+X_C^2}}$		$\frac{1}{2\pi CR}$ c/s where C is in Farads and R is in Ohms Z_p at f_r = 0·707RΩ	$C = \frac{10^6}{2\pi f_r R}$ microfarads
	$\sqrt{R^2+X_L^2}$		$\frac{R}{2\pi L}$ c/s where L is in Henrys and R is in Ohms Z_s at f_r = 1·4 RΩ	$L = \frac{R}{2\pi f_r}$ Henrys
	$\frac{RX_L}{\sqrt{R^2+X_L^2}}$		$\frac{R}{2\pi L}$ c/s where L is in Henrys and R is in Ohms Z_p at f_r = 0·707RΩ	$L = \frac{R}{2\pi f_r}$ Henrys

FIG. 43. BEHAVIOUR OF SIMPLE TWO-ELEMENT NETWORKS

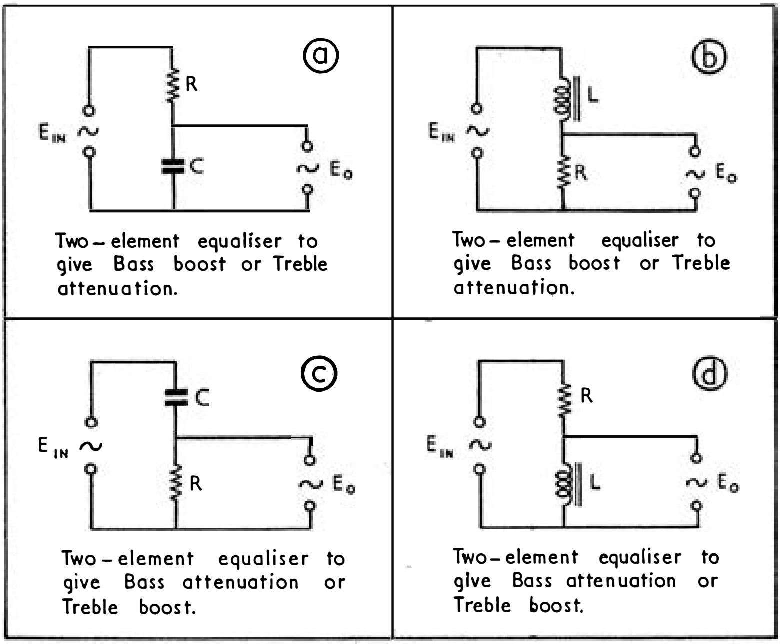


FIG. 44. SIMPLE TWO-ELEMENT EQUALIZERS. MAXIMUM SLOPE 6dB PER OCTAVE.

The source impedance and the load impedance can make a great deal of difference to the effect of equalizers, as the following practical examples show.

(1) EFFECT OF GENERATOR IMPEDANCE ON EQUALIZER RESPONSE

If circuit (a) in Fig. 44 were connected between two pentodes as shown in Fig. 45, the performance would be modified by the load resistance of V_1, R_1 and the resistance R_4 shunted across the capacitor C. This circuit has been investigated by Quartermaine who states that it is equivalent to Fig. 46 and that the type of response produced is as shown in Fig. 47.

The maximum attenuation is determined by the ratio of R_A to R_B. The frequencies f_{Low} and f_{Hi} are determined by equating the reactance of C to R_A and R_B respectively. Values giving the response of Fig. 47 are $R_1 = 220\text{k}\Omega$, $R_2 = 330\text{k}\Omega$, $R_3 = 18\text{k}\Omega$, $R_4 = 500\text{k}\Omega$ and $C = 0{\cdot}006\mu\text{F}$.

Using these values, $R_A = 250\text{k}\Omega$ and $R_B = 17\text{k}\Omega$, giving a ratio of 15 : 1 and maximum attenuation of approximately 23dB. The transition frequencies will be $f_{Low} = 100$c/s approx. and $f_{Hi} = 1{,}500$c/s approx. If the r_a of V_1 and of V_2 are both high, the circuit is equivalent to Fig. 48.

Consider first the potential across C. The time constant* of this circuit will be that of the circuit as a whole.

* See Appendix D.

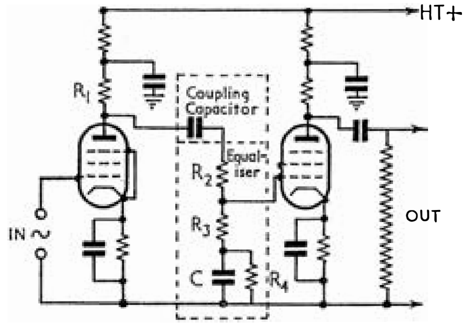


FIG. 45. BASS LIFT EQUALIZER LINKING TWO PENTODES.

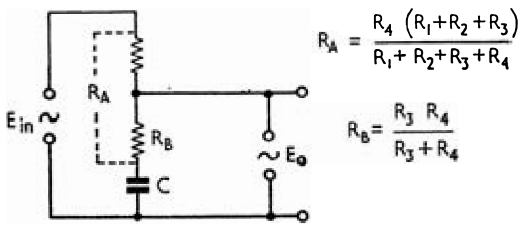


FIG 46. BASS LIFT EQUALIZER AND EQUIVALENT VALUES IN TERMS OF ACTUAL CIRCUIT COMPONENTS.

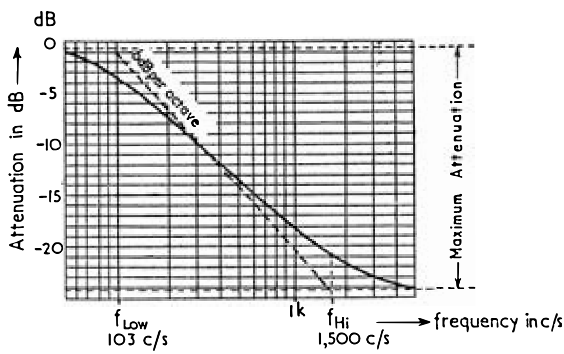


FIG. 47. BASS LIFT EQUALIZER RESPONSE.

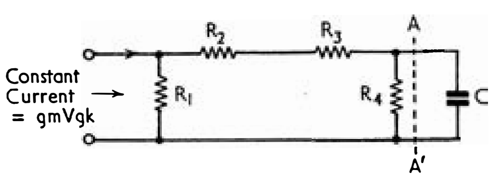


FIG. 48. EQUIVALENT OF BASS EQUALIZER CIRCUIT.

Applying Thévenin's Theorem* to AA′ the circuit of Fig. 48 may be shown to be equivalent to Fig. 49 where the constant current generator is replaced by its infinite internal impedance.

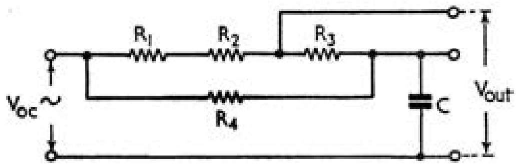


FIG. 49. THÉVENIN EQUIVALENT OF FIG. 48.

$$\text{Then } R_{INT} = R_4 \text{ in parallel with } R_1 + R_2 + R_3$$
$$= \frac{R_4 (R_1 + R_2 + R_3)}{R_1 + R_2 + R_3 + R_4} = R_A$$
$$\text{and } V_{OC} = IR_{INT} \times \frac{R_1}{R_1 + R_2 + R_3}$$
$$= \frac{g_m V_{gk}}{\frac{1}{R_4} + \frac{1}{R_1 + R_2 + R_3}} \times \frac{R_1}{R_1 + R_2 + R_3}$$

The time constant of this circuit, CR_A governs the low frequency transition frequency f_{Low}. (Of course, the output is not taken off across C alone but also across R_3. This adds a little to it, but does not affect the transition frequency). The maximum output will occur at zero frequency where $X_c = \infty$. At this frequency the equivalent is Fig. 50.

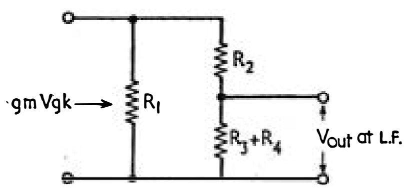


FIG. 50. EQUIVALENT OF FIG. 49 AT VERY LOW FREQUENCIES.

$$V_{out} = \frac{g_m V_{gk}}{\frac{1}{R_1} + \frac{1}{R_2 + R_3 + R_4}} \times \frac{R_3 + R_4}{R_2 + R_3 + R_4}$$
$$= \frac{g_m V_{gk} (R_3 + R_4) R_1}{R_1 + R_2 + R_3 + R_4}$$

Minimum output will occur when, at a very high frequency, X_c tends towards zero and R_4 is therefore shorted. Then

$$V_{out_{HF}} = \frac{g_m V_{gk} R_1 R_3}{R_1 + R_2 + R_3}$$

* See Appendix C

The ratio of maximum to minimum output volts will be

$$\frac{R_3 + R_4}{R_1 + R_2 + R_3 + R_4} \times \frac{R_1 + R_2 + R_3}{R_3}$$

$$= \frac{R_3 + R_4}{R_3 R_4} \times \frac{R_4 (R_1 + R_2 + R_3)}{R_1 + R_2 + R_3 + R_4} = \frac{1}{R_B} \times R_A$$

$\therefore \dfrac{R_A}{R_B}$ gives the ratio of maximum to minimum output.

In the example $R_A = 250\text{k}\Omega$ and $R_B = 17\text{k}\Omega$.
The equivalent circuit may therefore be regarded as Fig. 51.

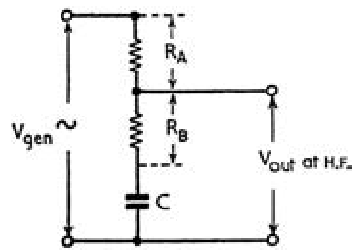


FIG. 51. EQUIVALENT OF FIG. 49 AT HIGH FREQUENCIES.

The voltage generator $= \dfrac{g_m V_{gk} (R_3 + R_4) R_1}{R_1 + R_2 + R_3 + R_4} = V_{gen}$

If $R_A \gg R_B$ and at the frequency $f_{Hi}, X_c = R_B$, the current in the circuit $= \dfrac{V_{gen}}{R_A}$

$\therefore$ Minimum output voltage $= \dfrac{V_{gen} R_B}{|R_A}$

(2) EFFECT OF LOAD RESISTANCE ON EQUALIZER RESPONSE

The way a load resistance can modify the effect of an equalizer is demonstrated by the following example. In Fig. 52 a two-element equalizer is terminated in a very high resistance.

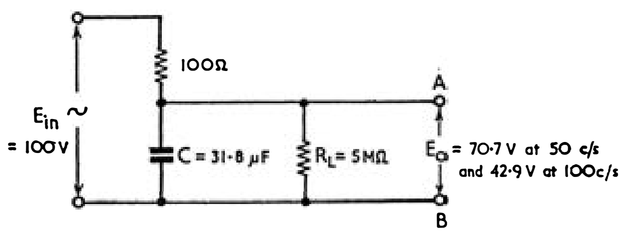


FIG. 52. TWO-ELEMENT EQUALIZER LOADED WITH 5MΩ.

The voltage between *AB* will approach the value which would obtain if the load resistance were infinite. The open circuit voltage is calculated as follows:

$$\text{The value of } X_c = \frac{1}{2\pi fC} \text{ where } C = 31{\cdot}8\mu\text{F}$$

$$\therefore\ X_c \text{ at 50c/s} = \frac{10^6}{2\pi 50 \times 31{\cdot}8} = 100\Omega$$

$$\text{and } X_c \text{ at 100c/s} = \tfrac{1}{2}X_c \text{ at 50c/s} = 50\Omega$$

$$\text{Output voltage at 50c/s} = \frac{E_{IN}\ X_c}{\sqrt{R^2 + X_c^2}} = \frac{10^2 \times 10^2}{\sqrt{2 \times 10^4}}$$

$$= \frac{10^2}{\sqrt{2}} = 70{\cdot}7 \text{ volts}$$

$$\text{Output voltage at 100c/s} = \frac{E_{IN}\ X_c}{\sqrt{R^2 + X_c^2}} = \frac{10^2 \times 5 \times 10}{\sqrt{10^4 + 0{\cdot}25 \times 10^4}}$$

$$= \frac{5 \times 10^3}{10^2\sqrt{1{\cdot}25}} = \frac{5 \times 10}{1{\cdot}12}$$

$$= 44{\cdot}6 \text{ volts.}$$

$$\text{Attenuation in dB per octave} = 20 \log_{10} \frac{70{\cdot}7}{44{\cdot}6} = 20 \log_{10} 1{\cdot}6$$

$$= 20 \times 0{\cdot}2041 = 4\text{dB.}$$

In Fig. 53 the load is reduced to 100Ω and the presence of the load

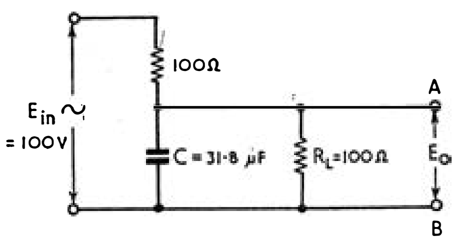


FIG. 53. TWO-ELEMENT EQUALIZER LOADED WITH 100Ω.

resistance can no longer be ignored. Using the well-known parallel to series conversion formulae

$$\text{Effective Series Resistance, } R_S = \frac{X_p^2\ R_p}{X_p^2 + R_p^2}$$

$$\text{and Effective Series Reactance, } X_S = \frac{R_p^2\ X_p}{X_p^2 + R_p^2}$$

where R_p is the shunt resistance and X_p is the shunt reactance.

it can be verified that with the 100Ω resistor across the capacitor

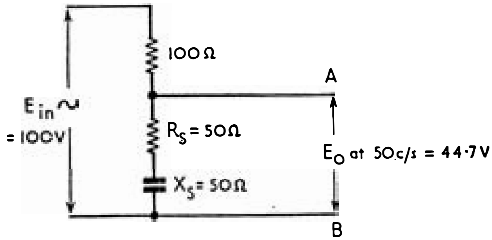


FIG. 54. EQUIVALENT OF FIG. 53 AT 50c/s

the circuit is equivalent at 50c/s to Fig. 54 and the output voltage

$$\text{at 50c/s} = \frac{10^2\sqrt{50^2+50^2}}{\sqrt{150^2+50^2}} = \frac{10^2 \times 70{\cdot}7}{10^2\sqrt{2{\cdot}5}} = \frac{70{\cdot}7}{1{\cdot}58}$$
$$= 44{\cdot}7 \text{ volts.}$$

At 100c/s the circuit is equivalent to Fig. 55 and the output voltage

$$\text{at 100c/s} = \frac{10^2\sqrt{20^2+40^2}}{\sqrt{120^2+40^2}} = \frac{10^2\sqrt{2000}}{\sqrt{120^2+40^2}} = \frac{10\sqrt{20}}{\sqrt{1{\cdot}6}}$$
$$= \frac{44{\cdot}6}{1{\cdot}26} = 35{\cdot}4 \text{ volts}$$

and attenuation in dB per octave is $20 \log_{10} \frac{44{\cdot}7}{35{\cdot}4} = 20 \log_{10} 1{\cdot}264$ $= 20 \times 0{\cdot}1 = 2\text{dB}$.

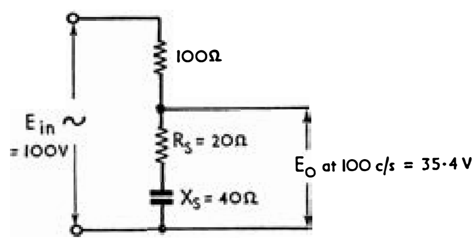


FIG. 55. EQUIVALENT OF FIG. 53 AT 100c/s.

An example of the equalization problems encountered is the " toppy " output of a small loudspeaker. The cause, of course, is the inability of the small cone to radiate low frequencies as efficiently as high frequencies. Manufacturers often compromise by shunting the speech coil with a small capacitor, but any equalizer inserted between the output of the amplifier and the input to the loudspeaker coil will be affected by the low impedance of the coil and the low impedance of the output of the amplifier. The necessary correction is, therefore, better inserted before the final stage of the amplifier. Generally, the recorder will be fitted with a variable treble attenuator and this may require to have its component values adjusted to give more treble attenuation; or, as mentioned elsewhere, it may be possible to increase the amplitude of the a.c. bias.

How to Obtain more than 6dB per Octave Treble Boost

If more than 6dB per octave boost is required it can be obtained by using two or more two-element filters in series, but the amount of boost will rarely be twice that obtained by one such filter and the insertion loss will generally be excessive. A better way to obtain a steeper curve is to use a resonant circuit. Fig. 56 shows a typical

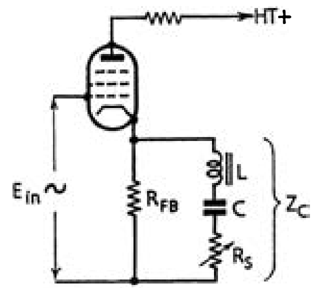


FIG. 56. RESONANT EQUALIZER TO GIVE MORE THAN 6dB PER OCTAVE TREBLE BOOST.

example. Here the feedback resistor in the cathode of a valve is by-passed by a series tuned circuit. The current negative feedback derived from the resistance is such that the gain of the stage with feedback, M', bears the following relationship to the gain without feedback, M.

viz $M' = \dfrac{M}{1 + \beta M}$, where $\beta = \dfrac{Z_c}{Ra'}$ where Z_c is the impedance of the cathode circuit and Ra' is the effective anode load.

Z_c will be least at the resonant frequency of the tuned circuit and at this frequency there will be least current negative feedback and therefore the gain of the stage will be greatest. By controlling the size of the resistance R_S in the shunt arm the degree of treble boost can be controlled. In a typical circuit the treble boost at 15,000c/s relative to 1,000c/s amounted to a maximum of 36dB. The coil used must be very carefully screened against hum pick-up.

The Parallel-T Equalizer

A coil-less circuit that provides a similar amount of treble boost is frequently used. This is known as the **Wien bridge** or **parallel-T.** Though it contains only resistors and capacitors it behaves in some ways like a shunt circuit and at its tuned frequency it is very rejective.

The circuit is as shown in Fig. 57. The values of C and R are very critical. The circuit is most rejective at the frequency at which $|X_c| = R$.

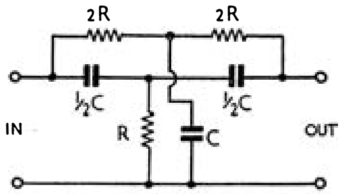


FIG. 57. THE PARALLEL-T EQUALIZER.

Fig. 58 shows how it may be connected in a voltage negative feedback loop to provide feedback which varies with frequency and is least at the resonant frequency. At this frequency, of course, the gain of the stage is greatest.

The amount of frequency discrimination can be varied by shunting the bridge with a variable resistance of high maximum value (2MΩ) or by varying the grid input resistance R_1.

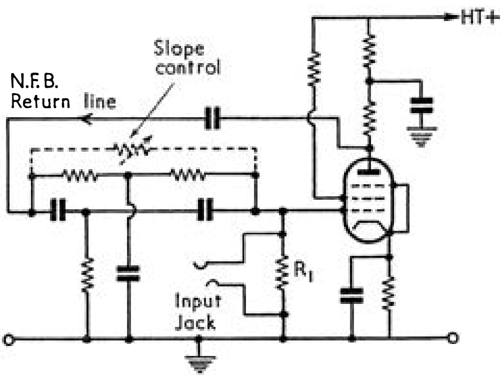


FIG. 58. WIEN BRIDGE EQUALIZER IN VOLTAGE NEGATIVE FEEDBACK CIRCUIT.

Function of the Recording Equalizer

The recording equalizer has to ensure sufficient treble pre-emphasis to offset the effects of changing permeability with increase in frequency, self-demagnetization (which increases with increase in frequency, up to a point), hysteresis and eddy current losses. The response is therefore as shown in Fig. 59.

The slight bass tip up is to offset a certain amount of bass attenuation on reproduction due to the fact that at low frequencies the recorded wavelengths are so long compared with the arc of contact between tape and head that the whole of the external flux does not close round the core of the head.

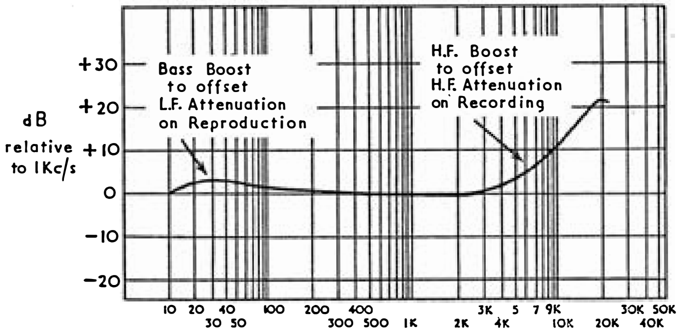


FIG. 59. TYPICAL RECORDING EQUALIZER RESPONSE.

Function of the Reproducing Equalizer

The reproducing equalizer is required to offset the rise of 6dB per octave (due to the fact that the voltage output from the reproducing head is proportional to the rate of change of flux in its core) and to offset the gap effect at high frequencies and eddy current and hysteresis losses. A typical response curve is shown in Fig. 60.

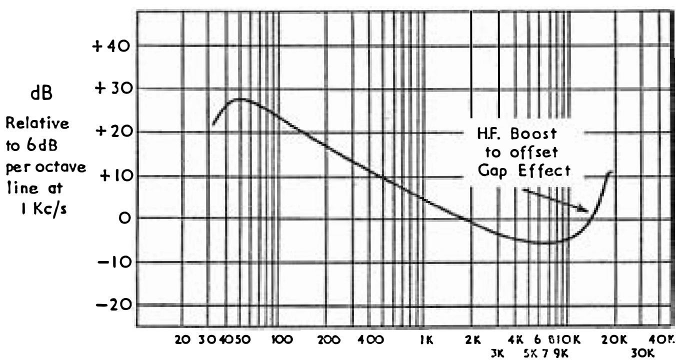


FIG. 60. TYPICAL REPRODUCING EQUALIZER RESPONSE.

The Design and Use of Filters

Filters can be used to suppress any undesired frequency. The simplest forms are the acceptor and rejector circuits: C in series with L or C in shunt with L. Examples of the use of resonant circuits as filters are given on pages 88 and 89.

5

MIXERS

It is often desired to mix at the input to the recorder the outputs of two or more microphones or the output of one or more microphones and the output of a gramophone or radio. An additional requirement is to control the output level of each source independently. Ideally this must be achieved without mutual interference between the sources and without changing the source impedance presented to the input of the recorder.

The simplest way to deal with these requirements is to buy or build a mixer unit with (say) four inputs and two outputs: one low level, the other high.

The low level section of the mixer is then used for microphones and the high level section for radio and gramophone inputs. Fig. 61 shows the front of such a mixer. Fig. 62 shows the circuit of a simple two-channel mixing unit for a pair of low impedance microphones.

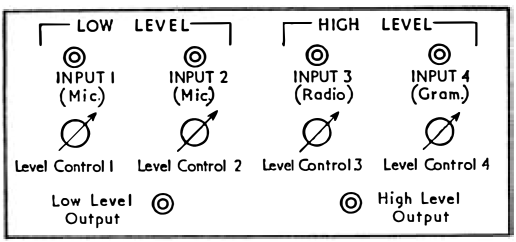


FIG. 61. FRONT PANEL OF MIXER DESIGNED TO MIX THE OUTPUT FROM TWO MICROPHONES WITH THE OUTPUT FROM A RADIO RECEIVER AND A GRAMOPHONE.

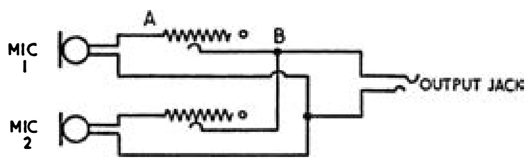


FIG. 62. SHOWING CONNECTIONS TO POTENTIAL DIVIDER.

Here the volume controls consist of simple series faders made from Carbon Potential Dividers connected as shown in Fig. 62(a). The value of the resistance in the Potential Divider is governed by the impedance of the microphones to be used. For moving coil and ribbon

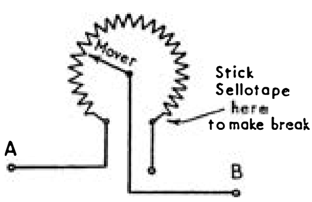


FIG. 62(a) CONNECTIONS TO A POTENTIAL DIVIDER USED AS A SERIES FADER WITH AN *OFF* POSITION. IF SELLOTAPE IS USED IN THE POSITION INDICATED IT WILL BREAK THE CONNECTION BETWEEN THE WIPING BLADE AND THE CARBON RESISTOR. IN PRACTICE THE SELLOTAPE DOES NOT LAST LONG BECAUSE IT TENDS TO BE CUT THROUGH BY THE WIPING BLADE WHICH USUALLY HAS A POINTED CONTACT. A BETTER WAY TO ACHIEVE THE BREAK IS TO SCRAPE OFF THE CARBON FROM THE PLASTIC FORMER FOR A DISTANCE OF ABOUT $\frac{1}{8}$" FROM THE RIGHT-HAND END (LOOKING AT THE SIDE OPPOSITE THE KNOB). THE RECOMMENDED DIVIDER HAS A TOTAL RESISTANCE OF 10,000Ω AND IS OF THE LOG VARIETY, *i.e.* THE INCREMENTS IN SERIES RESISTANCE ARE PROGRESSIVELY INCREASED IN SIZE AS THE KNOB IS TURNED ANTI-CLOCKWISE TOWARDS THE *OFF* POSITION.

microphones a suitable value is 1,000Ω. This arrangement is unsuitable for crystal microphones and though it will serve well enough, it is not ideal even for low impedance microphones, especially if the lines from the mixer to the recorder are more than a few yards long.

An elaboration is shown in Fig. 63. It is called a **balanced series fade unit.** In this, a variable resistance is placed in each

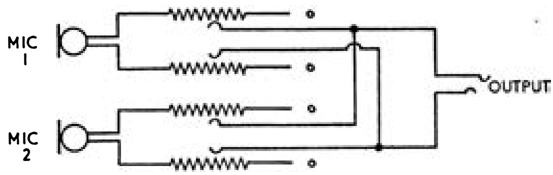


FIG. 63. BALANCE SERIES FADE UNIT.

wire from the microphone, then, when the knob of the volume control is turned, both the sliders move together.

In professional type balanced series fade units the faders are not carbon potentiometers but are wire-wound faders with stud contacts. By varying the resistance inserted between the studs the loss in dB, introduced by changing from stud to stud, is kept constant. A high resistance of the order of 20 megohms is introduced between the last stud of the variable resistance and the "off" stud in one leg of each microphone circuit. This is called the **static leak** and its presence minimizes the risk of a click due to static discharge each time the microphone is faded up. (See Fig. 64). Neither of the arrangements above described is of the constant impedance variety.

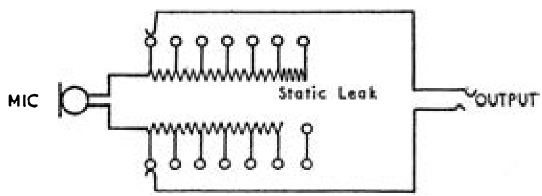


FIG. 64. PROFESSIONAL TYPE BALANCED SERIES FADE UNIT, SHOWING HIGH RESISTANCE CONNECTED BETWEEN LAST TWO STUDS ON ONE SIDE TO FORM A STATIC LEAK TO PREVENT CLICKS.

Fig. 65 shows a typical two microphone mixer using constant impedance potentiometers. The moving arms of R_1, R_2 and R_3 are mechanically coupled and move in such a way that when the series resistors R_1 and R_2 are increased the shunt resistor is decreased. In this way the total impedance of the circuit (combined with the input or output impedance) is kept constant despite the setting of the fader. The value of the series resistor R is important and it can be calculated from the formula

$$R = R_L \frac{n-1}{n+1}$$

where n equals the number of microphones and R_L equals the

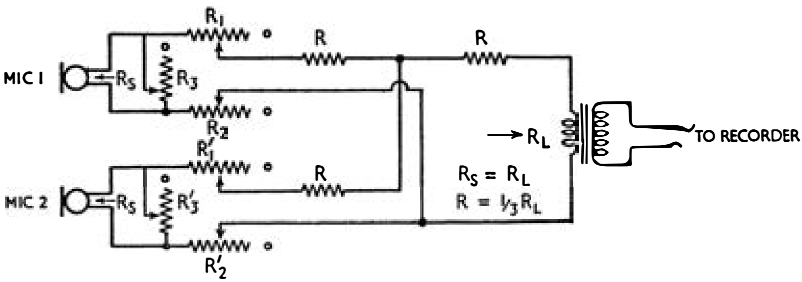


FIG. 65. A TWO-MICROPHONE MIXER WITH A CONSTANT IMPEDANCE.

constant impedance from which the mixer inputs are fed and the constant impedance to which the output of the mixer is connected.

The transformer in the output serves a double purpose: it is used for matching and as a repeating coil to ensure that interference picked up on the leads to the recorder is kept to a minimum. If R_L is known and R_3 is chosen such that $R_3 \gg R_L$, the value of R_1 and R_2 in the fader can be calculated from the formula

$$R_1 = R_2 = \frac{R_L{}^2}{2\,(R_3 - R_L)}$$

For example, if $R_L = 300\Omega$ and $R_3 = 1{,}000\Omega$, $R_1 = 64{\cdot}3\Omega$.

If it is required to introduce any equalization in the output of the individual microphones, the equalizer insertion loss will almost certainly make it necessary to use an additional amplifying stage. The pre-amplifiers can then be coupled together to form a mixer.

Several makes of recorder provide h.t. and l.t. output from a spare socket which can be used to feed the pre-amplifier. A typical circuit is shown in Fig. 66.

In building a separate power supply for such a pre-amplifier, especial care must be taken to use efficient smoothing because the hum content must be kept very low indeed.

The mixer controls for the radio and gramophone outputs may

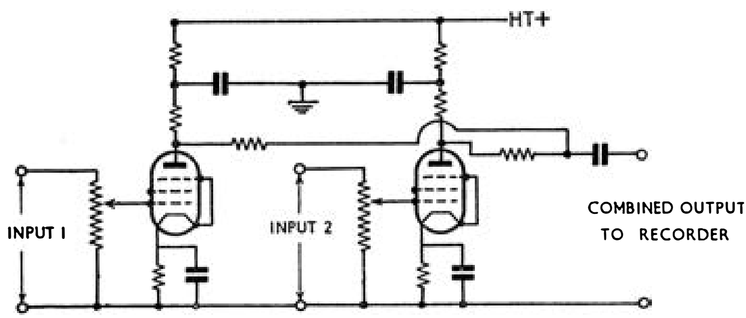


FIG. 66. TWO-MICROPHONE MIXER INCORPORATING PRE-AMPLIFIERS WITH SUCH AN ARRANGEMENT IT IS POSSIBLE TO MIX MICROPHONES WITH DIFFERING IMPEDANCES. (SCREEN CONNECTIONS OMITTED FOR SIMPLICITY).

be similar to those described for microphones or may take the form shown in Fig. 67. Such an arrangement is also very suitable for high

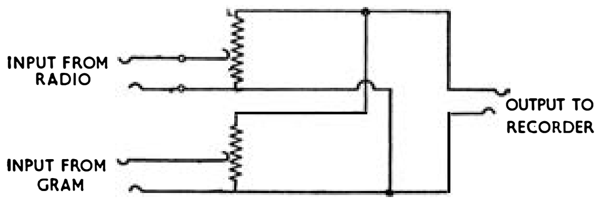


FIG. 67. A SIMPLE MIXER FOR RADIO AND GRAMOPHONE OUTPUTS.

impedance crystal microphones, though, even so, high impedance microphones should not be used if the leads need to be long. The value of the potential divider used depends very much on the output impedance of the microphones or other apparatus. If the direct output of the pick-up is, for instance, fed to the potential divider, great care must be taken in the choice of the potential divider resistance, because a pick-up may be regarded as a voltage generator in series with its own impedance. The internal impedance of the pick-up therefore forms part of the potential divider (see Fig. 68) and may

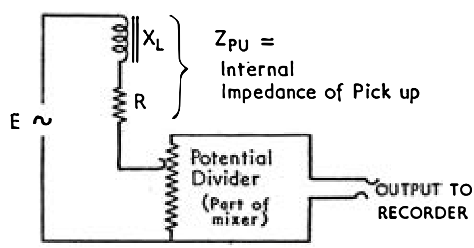


FIG. 68. SHOWING HOW INTERNAL IMPEDANCE, Z_{PU}, OF A PICK-UP MAY MODIFY THE OUTPUT FROM THE MIXER, IF THE MIXER IMPEDANCE IS LOW COMPARED WITH Z_{PU}.

alter the frequency response if the load resistance is small compared with the internal impedance of the pick-up. Of course, the output of the pick-up must be equalized to offset the bass attenuation and treble pre-emphasis used in recording the gramophone record. A deliberate mismatch is often used to provide part of this equalization.

Fig. 69 gives the circuit of a two-position microphone mixer with constant impedance attenuators (a) and (b), and Fig. 70 that of a transformer mixer. This achieves the desired results with less attenuation.

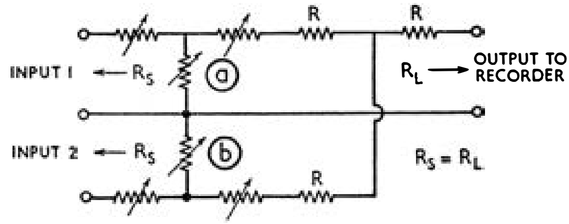


FIG. 69. CIRCUIT OF A TWO POSITION MICROPHONE MIXER WITH CONSTANT IMPEDANCE ATTENUATORS (a) AND (b).

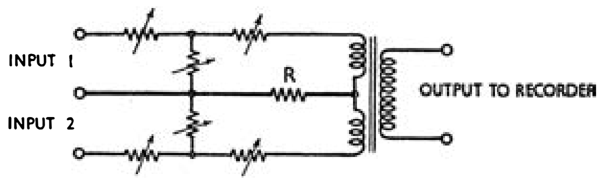


FIG. 70. A TRANSFORMER MIXER. THIS ACHIEVES THE RESULTS DESIRED WITH LESS ATTENUATION. $R = R_0 \frac{(n-1)}{2}$ WHERE R_0 = MIXER CIRCUIT RESISTANCE AND n = THE NUMBER OF POSITIONS IN THIS CASE TWO.

6

ACOUSTICS

THE acoustical properties of both the place in which a recording is made and the place in which the reproduction is effected are major factors in deciding the quality of the reproduced sound.

Sound Absorption

Any surface against which a sound wave strikes absorbs some of the energy in the air wave and reflects a proportion of it. The degree of absorption and the degree of reflection depend upon a number of things: *e.g.* the material of which the surface is made, the smoothness or roughness of that surface, the shape of the surface, its resilience and the frequency of the impinging sound.

Reverberation

When a sound is made in a closed space, the surfaces of the walls, floor, ceiling and furniture (include people in the latter category) cause multiple reflections of the sound. In addition, therefore, to the direct sound wave, the microphone receives a large proportion of reflected sound. As the path travelled by the reflected sound is longer than the path of the direct sound, the reverberation has the effect of prolonging the sound after the sound source has ceased to vibrate. The reverberation time is the time that an abruptly terminated sound takes to decay to one-millionth of its original intensity, *i.e.* to decay through 60dB.

The amount of reverberation required depends upon the type of performance being recorded and the size of the studio. Many modern studios are arranged to have a " live " end and a " dead " end. The live end is used for music and the dead end for speech. Much less reverberation is permissible for the latter, because the more reverberation there is, the less clear will be the diction. Of course, the closer the microphone is set to the sound source the greater will be the ratio of direct to indirect sound but the microphone must not be set too close because, with pressure operated microphones, the high frequencies, and with pressure gradient types, the bass frequencies, will be accentuated. This is explained in Chapter 7. It is possible to get some idea of the comparative reverberation time of a room by recording the sound of a single handclap and then measuring the length of tape upon which the sound is recorded. Try this in a tiled bathroom (where the reverberation time will be long) and in a carpeted and fully furnished sitting-room, taking care that the microphone and sound source are in the same relative positions. With a tape speed of 7½ inches per second the sound will cover about 2½ inches when recorded in the bathroom and only about 1 inch when recorded in the sitting-room. For more definite results use a standard source of sound, such as a cap-pistol.

Here is a table of suitable reverberation times—this is only a rough guide and the final criterion must always be the effect on your own ears. Generally, the larger the room the longer will be the most effective reverberation time.

Talks	0·4—0·6 seconds
Piano	0·8—1·25 ,,
Small Instrumental groups—trios and quintets	0·8—1·25 ,,
Orchestras	1—2 ,,

Control of Reverberation

By careful treatment of the surfaces it is possible to control the reverberation time.

Porous materials like cotton-wool and felt are good absorbers of high frequencies but they have little effect on low frequencies. To absorb low frequencies, panels, membrane or cavity absorbers are required.

A panel absorber can be a thin plywood panel firmly tacked across a window. If backed up by wadding, such a panel will absorb well frequencies up to 150c/s.

Membrane absorbers are lid-less boxes covered at the top with tightly stretched roofing felt. The frequencies absorbed depend upon the depth of the box. It is useful to have handy a number of boxes 4″, 6″, 8″ and 12″ in depth treated in this way. Their effect on the acoustics can then be tried singly or collectively. Several boxes of one size are required to control a particular frequency.

Cavity absorbers are bottle-shaped absorbers which depend for their action on the resonance between the mass of the air in the neck of the bottle with the compliance of the air in the bottle itself. Various shapes and sizes can therefore be designed to absorb any audio frequency up to about 1kc/s. Care must be taken to damp the resonance of the bottle by (say) lightly packing the neck with cotton-wool or the absorber may reinforce rather than absorb the sound. Although some fun may be had by experimenting with cavity absorbers, their practical use must be left to the experts who have all the necessary formulae for calculating the resonance.

Standing Waves

Sound produced in a closed space is built up by reflections from any hard parallel surfaces such as floor and ceiling or opposite parallel walls. The reinforcement of the sound due to this effect may be as much as 25dB, at the frequency of which the wavelength is twice the distance between the surfaces. Harmonics of this frequency are also reinforced, though to a lesser degree. For example, if the distance between two walls is 10 feet, the frequencies reinforced will be 56, 112, 168, 224 and 280 cycles per second.

The effect of room resonances may be minimized by resonant absorbers which are tuned to the particular frequencies it is desired to attenuate. The condition can also be minimized by carpeting the floor, sputtering the ceiling and avoiding parallel wall surfaces.

Microphones should be placed as far as possible from reflecting surfaces because of the selective reinforcement of the sound there; but the position midway between parallel reflecting surfaces should also be avoided, because at this point the sound pressure variations are at a minimum and the particle velocity of the air is at a maximum, so that both pressure operated and pressure gradient operated microphones will have abnormal response (one abnormally low, the other high) to the resonant frequencies.

Microphone Correction Units

If a room is very boomy some alleviation may be obtained by using a microphone correction unit. This is an equalizer inserted between the microphone and the recorder to provide a bass cut. A typical circuit is shown in Fig. 71.

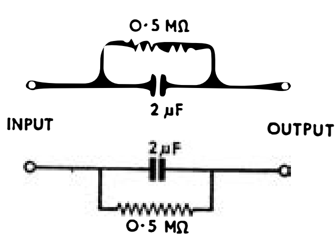


FIG. 71. A TYPICAL MICROPHONE CORRECTION UNIT USED TO MINIMIZE THE EFFECT OF BOOMINESS. SUCH A UNIT DOES NOT REDUCE THE BOOMINESS OF THE ROOM ITSELF WHICH MAY DISTRACT A SPEAKER.

Flutter Echoes

These are likely to occur in all rooms with parallel wall surfaces. The remedy is to use screens to mask off one or more of the walls or to line the walls with a sound-scattering surface such as corrugated paper.

Turning your Sitting-Room into a Studio

The usual problem for the home recordist is to increase the reverberation time in a sitting-room when recording music or singing. It may be necessary to roll up the carpet and to cover the curtains with newspaper. Removing an upholstered suite and turning out the audience will also work wonders.

No hard and fast rules can be laid down—even experts proceed by trial and error. The thing to do is to make repeated trial recordings under varying conditions until a satisfying result is obtained.

Acoustics for Reproduction

Never play back your trial recordings in the acoustically treated recording room. Far less reverberation is required in the reproducing room. Remember, reverberation in the latter is added to the reverberation in the recording room, which is recorded on the tape. After each test recording, therefore, it pays to take the recorder into the next room before making the test reproduction. In this way you will be able to judge the effect better. It may take quite a long time and much patience before you get results which please, but once settled the same arrangement may be used every time a similar type of recording has to be made.

Acoustics of Halls

The acoustic treatment of a local hall may well present a much more difficult problem for the amateur. Generally, it will be far too " live ", *i.e.* there will be too much reverberation. The remedy is to hang absorbent material round the walls and this of course can be a costly business. A point to remember is that the audience will have a good deal of effect on sound absorption. If you have the choice, always record with a full house. Quite a lot can sometimes be done with available material. A village hall that was very echoey and boomy was made reasonably usable by (*i*) placing on end round the hall a set of 20 forms borrowed from the school next door, each form being draped with a coat or blanket. The sound was thus scattered as well as absorbed. (*ii*) Hanging a screen usually used for cinematograph shows high up half-way down the hall. This tended to reduce an undesirable flutter echo. As a further precaution a cardioid microphone was used to reduce the proportion of indirect sound received.

Use of Echo Chambers

Occasionally, the trouble is that the hall is not reverberant enough and this presents a still more difficult problem. One way to overcome it is to use an echo chamber. This can be a corridor or cellar or any place unlikely to be disturbed which has a fairly long reverberation time. The output of the microphone in the "dead" hall is fed to an amplifier and loudspeaker in the echo chamber. The output of the loudspeaker, suitably coloured by the reverberation of the chamber, is picked up by a microphone and combined with the output of the microphone in the hall. By careful manipulation of the mixer faders it is possible to get any desired colouration in this way. (See Fig. 72).

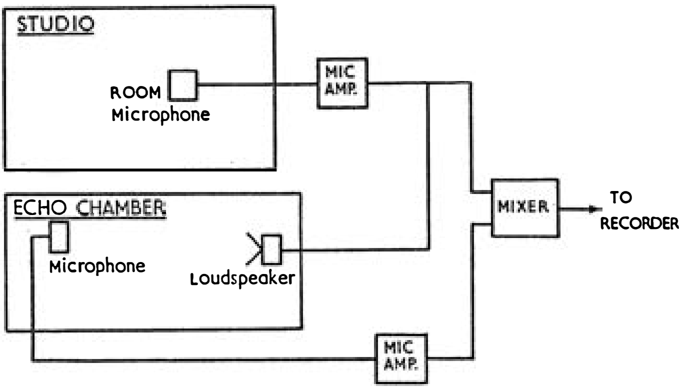


FIG. 72. ILLUSTRATING THE USE OF AN ECHO CHAMBER TO COLOUR THE SOUND FROM A ROOM WITH TOO LITTLE NATURAL REVERBERATION.

Artificial Reverberation Machines

Because suitable echo chambers are sometimes hard to find, many different kinds of artificial reverberation machines have been devised. In one of these the sound is magnetically recorded on the outer circumference of a rotating wheel and reproduced from a succession of reproducing heads spaced around the periphery. By varying the spacing and controlling the output level from each head and mixing the combined output with the direct output of the microphone, good colouration can be obtained. In another machine the amplified output of the microphone is fed to a moving coil vibrator attached to a large sheet of metal. A pick-up needle connected to a reproducing head is used to detect the multiple vibrations set up in the sheet. The output from the pick-up is mixed in with the amplified output from the microphone. This also gives good colouration.

7

MICROPHONES

THE first requisite for good recordings is to get a good microphone and to know its limitations. There is little point in purchasing a microphone having a frequency response greatly in excess of that of your recorder, but on the other hand it is important not to restrict the overall fidelity by using an inferior one. The manufacturers may provide you with a polar diagram or directivity pattern. This is a graph showing the relative voltages output from the microphone for sounds of constant loudness but differing frequency arriving from various angles around the microphone and (sometimes) from differing planes.

A perfect omnidirectional microphone would give an equal output from sounds of a fixed loudness and distance from it, no matter from what angle they arrived or what their pitch and such a microphone would generate no noise.

Microphones do not in practice ever achieve this " ideal " response and, of course, for many purposes an omnidirectional microphone is not required.

If a variable frequency tone source and a suitable loudspeaker are available it is possible on a still day to make a test in the open air of the response of your own microphone, but the results you obtain will only be very approximate because the frequency response of the recorder, loudspeaker and microphone amplifier will also be determining factors. Scientifically polar diagrams are prepared (with the use of specially calibrated loudspeakers and amplifiers) by testing the response of the microphone in a " dead " room—one in which floor, walls and ceiling have been specially treated to prevent any reflection from these surfaces.

What a Microphone is

A microphone is a transducer or energy converter for converting the acoustical energy in a sound wave to electrical energy.

There are three main kinds of microphones: (1) Pressure operated; (2) Pressure gradient operated; and (3) Double-transducer or " cardioid "—*i.e.* a mixture of types (1) and (2) in the same case.

1. Pressure Operated Microphones

Any microphone of which the moving part is open to the air on one side only is pressure operated, *i.e.* the magnitude of the force applied by the air wave depends upon the instantaneous pressure in the sound wave at the diaphragm. Where there is a cavity in front of the diaphragm, the response tends to show a peak at some frequency due to a resonant effect determined by the size and shape of the cavity. In addition, at high frequencies (where the wavelength of the sound is so short that the microphone acts as a reflector), the actual wave pressure tends to be increased to up to double its normal value. This explains why it often pays to speak across rather than into the diaphragm of this type of microphone.

The polar diagram of a pressure operated microphone is practically spherical at low frequencies, but at high frequencies due to the sound shadow cast by the case, the response is better at the front than at the back. Fig. 73 shows a polar diagram of a typical pressure

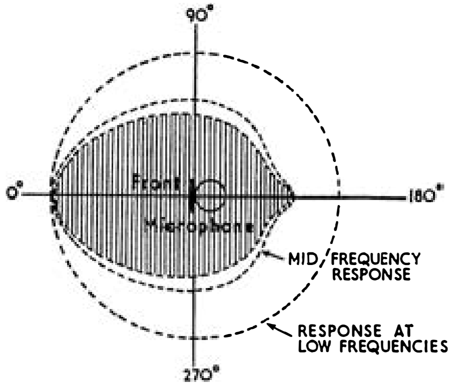


FIG. 73. POLAR DIAGRAM OF A TYPICAL PRESSURE OPERATED MICROPHONE. (SHADED AREA=HIGH FREQUENCY RESPONSE).

operated microphone, drawn in the plane at microphone level. The sensitivity of the microphone is shown by the diameter of the diagram at any given angle with respect to the microphone. The graph shows that at low frequencies the response is uniform all round the microphone but that at high frequencies the response falls off at the back. Very small pressure operated microphones, though of course less sensitive, have polar diagrams which show a uniform response up to quite high frequencies.

There are many different kinds of pressure operated microphones. Of these the most common are:

(1) CRYSTAL MICROPHONES

These make use of the fact that, when a bimorph of Rochelle-salt or some similar ferro-electrical crystal is bent or twisted, an e.m.f. is generated which is proportional to the displacement of the crystal. The air pressure may either be applied direct to the crystal or to a diaphragm to which the crystal is attached. Many crystal microphones are comparatively inexpensive and they have a reasonable frequency response and comparatively large output. Their main disadvantage is their high internal impedance which makes it necessary to place them as near to the pre-amplifier as possible, *i.e.* long leads will not do. The leads must be efficiently screened and the screen earthed. Very often the screen itself is used as one lead. In that case it must be connected to the earthy side of the input jack.

(2) CONDENSER MICROPHONES (SOME TYPES)

In these, the diaphragm forms one plate of a capacitor which is polarized by a d.c. potential applied through a resistance to the two

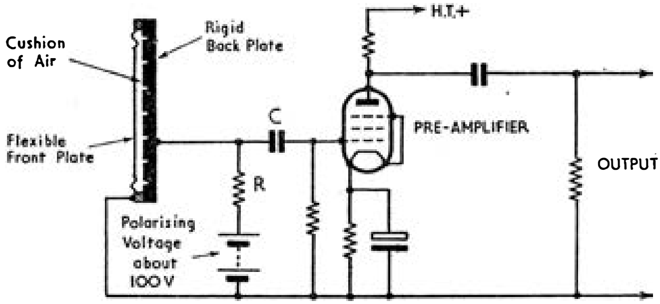


FIG. 74. THE CONDENSER MICROPHONE.

plates. (See Fig. 74). When an air wave vibrates the diaphragm, the capacity of the capacitor is varied, causing a charge and discharge current to flow in the resistor. The e.m.f. developed across the resistance is applied to the input to the pre-amplifier which must be very close to the microphone because of the high internal impedance of this type. For this reason the pre-amplifier is often housed in the microphone holder and this means that power leads must also be connected.

(3) MOVING COIL MICROPHONES

In this type, the diaphragm is attached to a coil which vibrates in a magnetic field. Good ones are comparatively costly. The output impedance is low and a matching transformer is required for most input circuits. They are fairly robust and not affected by changes in temperature or humidity but in most cases they have a resonant peak at from 5-8kc/s. Nevertheless they often deliver better quality sound than the crystal types, *i.e.* they are less readily overloaded. Because of their low impedance they can be used with quite long leads provided these are well screened. They are excellent for outdoor use when fitted with a windshield.

Working too close to a pressure operated microphone tends to over-emphasize the treble response. Sibilants such as *s* and *f* will therefore in this circumstance be given undue prominence in the reproduction.

2. Pressure Gradient Microphones

When both sides of the diaphragm are open to the air, as in the ribbon microphone, the sound wave arrives at both the front and the back of the microphone (not necessarily in the same phase) and the resultant force is determined by the difference in magnitude of the pressures at the front and back of the diaphragm. Thus, if a positive peak of pressure arrives at the back of the diaphragm at the same time as a positive peak of equal value arrives at the front, the net pressure change at the diaphragm will be zero. If the pressures are out of phase the difference in the pressure at the front from the pressure at the back will produce a certain finite change in pressure on the diaphragm. Because the path travelled by the sound arriving at the front is shorter than the path travelled by the sound arriving at the back, there is generally *some* difference in the amplitude of the pressure at the front and back. This difference is determined by two things: (*i*) the difference in path length; and (*ii*) the frequency of the sound.

The phase difference due to travelling the extra distance is very small at low frequencies and is often quite large at high frequencies (see Fig. 75). It follows that, up to a given frequency, the force

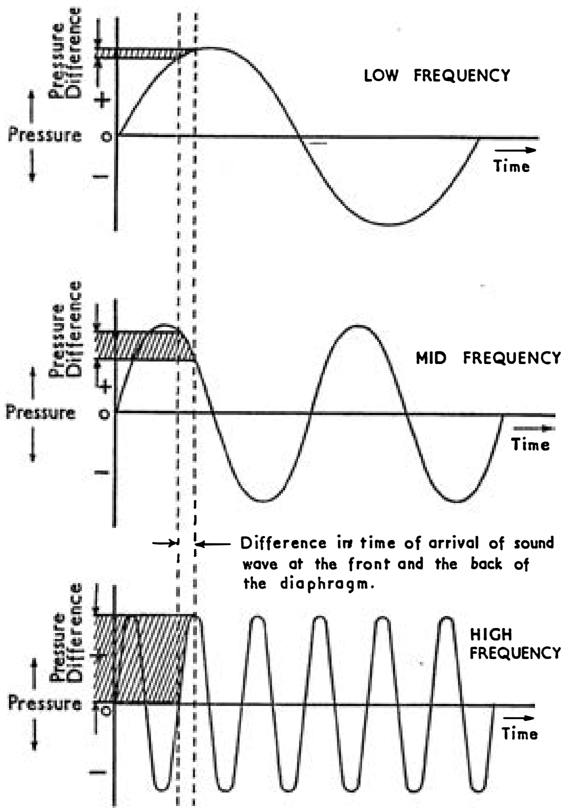


FIG. 75. PRESSURE GRAPHS OF THREE SOUND WAVES OF DIFFERING FREQUENCY SHOWING HOW INCREASE IN FREQUENCY INCREASES THE PRESSURE DIFFERENCE BETWEEN THE FRONT AND THE BACK OF THE DIAPHRAGM.

applied to the diaphragm increases with increase in frequency. By careful design it is possible nevertheless to arrange that the output level does not so increase. Pressure gradient microphones are particularly sensitive to close operation and if the speaker is too close to the microphone all the bass frequencies of the voice will be unduly noticeable—the effect will be to give the voice a dark brown quality. The minimum separation between speaker and microphone (unless this special effect is desired) is about two feet. An exception to this rule is the lip ribbon microphone described later (page 72).

The polar diagram of a pressure gradient microphone (in the plane of the microphone) is a figure of eight. (See Fig. 76).

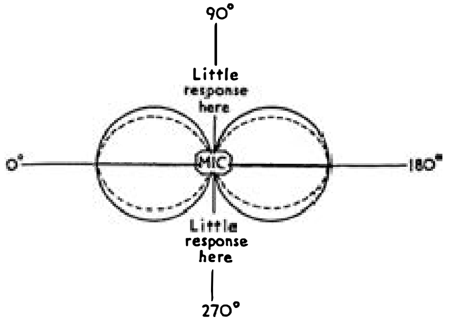


FIG. 76. RESPONSE OF PRESSURE-GRADIENT OPERATED MICROPHONE (HIGH FREQUENCY RESPONSE DOTTED).

These microphones are alert to sounds arriving on lines at right angles to their two faces but have little response at the sides. (Just how little depends upon how much reflection takes place. In a small room so much sound is reflected from the walls that the ribbon microphone appears to have quite good sensitivity to sounds made at the side—though of course it rarely appears *as* sensitive at the side as at the front).

The most common type of pressure gradient microphone is the ribbon microphone. In this the diaphragm is a small, thin, light-weight metal strip, stretched between the extended poles of a permanent magnet. Electrical contact is made between the two ends of the ribbon and the terminals of the microphone. The output is usually rather low compared with other types but the frequency response is excellent so long as the sound arrives at right angles to the ribbon. They are of very low impedance and require a suitable matching transformer. Ribbon microphones are not as a rule suitable for outdoor work because they are too susceptible to wind noises. Sometimes only one face of the ribbon is exposed to the sound and the other face is enclosed by an acoustic labyrinth of maybe variable acoustic resistance. This type of ribbon microphone is pressure operated.

Some modern crystal microphones are pressure gradient operated —and these of course exhibit a figure of eight response like the double-sided ribbon.

3. Cardioid Microphones

If a pressure operated unit is combined with a pressure gradient one, the polar diagram is heart-shaped as shown in Fig. 77. This sort of response can be obtained by housing a double-sided ribbon microphone in the same box as a moving coil microphone. Or, some-times the ribbon of a ribbon microphone is damped at the centre as

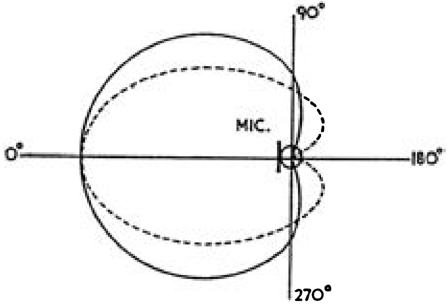


FIG. 77. HEART-SHAPED RESPONSE OF A CARDIOID MICROPHONE (HIGH FREQUENCY RESPONSE DOTTED).

well as at the ends and one half of the ribbon is backed by an acoustic labyrinth, whilst the other half is open to the air on both sides. Another way is to use a special type of capacitor microphone in which a series of holes in the rigid plate of the capacitor allows the sound pressure to affect both the front and the back of the diaphragm. In some types of microphone a simple switching device enables the microphone to behave at will as a cardioid, a pressure operated or a pressure gradient operated type, but such general-purpose microphones do not as a rule have quite such a good response in any condition as more straightforward types which have only one polar diagram.

How to get to know the Limitations of a Microphone without Professional Test Gear

1. Set the microphone up near a piano and record all the notes in succession from the lowest to the highest—then reproduce your recording and gauge by ear if the notes reproduce with about the same comparative loudness as when you sounded them on the piano. The best position for the microphone is just behind the piano, a little to the right of its centre.

2. If a gramophone is available, reproduce a test record of a frequency run, and record the output from the gramophone as picked up by the microphone. This will only give a very approximate idea of the microphone's actual response because the position of the latter with respect to the loudspeaker and the frequency response of both the loudspeaker and of the recording machine will be determining factors. Not to be misled by room resonances, place the microphone close to the bell of the loudspeaker. Even here, if it is a pressure gradient operated microphone, it will be affected by reflected sound impinging on the side remote from the loudspeaker. Don't, however, place a baffle behind such a microphone when testing it, because that will also change the frequency response.

3. To get an idea of the polar diagram, mark out a circle on the ground and place the microphone on a stand at the centre. Then, whilst recording the output of the microphone, take up positions round the circle (always facing the microphone) at various degrees, calling the front of the microphone 0° or 360°, the left-hand side 90°

and the right-hand side 270°. Say the same sentence or make a musical note on some instrument at 0°, 30°, 60°, 90°, 120°, 150°, 180°, 210°, 240°, 270°, 300°, 330°, endeavouring to make equally loud sounds at each spot. Announce each time the position occupied. If you do this with a pressure gradient microphone you will, on replay, be scarcely able to hear any output at 90° and 270°—the dead part of the microphone field—provided that the experiment is made well away from any building.

A pressure operated microphone will give you practically a uniform response all round the microphone, whilst the cardioid type will give you very little output at 180°, little at 90° and 270° but plenty at 0°, 30°, 60°, 300° and 330°.

If you repeat the experiment indoors, you will find that reflection will lessen the fall in output in the dead areas of the pressure gradient and cardioid types of microphone. This is a very useful experiment to make and it will tell you more about the response of your microphone than reading any number of textbooks.

With the aid of a tall step-ladder it is possible in a similar way to test the vertical response of the microphone. The comparative response at the dead side of a ribbon microphone tested in this way is a good indication of the reverberation qualities of the room in which the test is made.

The Voice Test

If you are sure that the response of your recorder is good, a useful test of a microphone is to use it to record the sound of a well-known voice and to listen to the reproduction and compare it with the original. Do not use your own voice for this test because you will find it difficult to recognise—unless of course you have heard your recorded voice many times.

Treat Microphones with Care

All microphones are very delicate pieces of mechanism and are easily upset by rough treatment. It pays, therefore, to handle them with about the same care as you would a valuable piece of china. Many microphones are seriously affected by dirt and damp, so when not in use wrap them up and store them carefully in a cool dry place.

8

BALANCE AND CONTROL

A WHOLE book could easily be written on the subject of microphone placing. Only general rules can be propounded here, but the beginner will find the basic principles which must be observed and which will enable him to learn more quickly by experience.

When considering where to place the microphone or microphones four things must be borne in mind:

(*i*) The type of microphone and the particular limitations which are imposed by its polar diagram and its impedance.
(*ii*) The acoustic properties of the place in which the microphone is to be used.
(*iii*) The type of programme that is to be recorded.
(*iv*) The overall effect that the reproduction will be required to give (*i.e.* whether it is to sound as if made in a large hall, in the intimate surroundings of a small room, or in the wide open space outdoors).

Typical Problems

(1) THE RECORDING OF A SINGLE SPEAKER

If the recording is to be made indoors, the type of microphone is not of great importance provided that it is a good one.

The microphone should preferably be hung over a table or mounted on a stand supported by a piece of thick felt or sponge rubber on a table. There should be space in front of the microphone for the speaker's notes, and, except for the microphone and perhaps a script rest, the table top should be clear. The position of the microphone in the room can modify to a great extent the results obtained. Generally, the best results will be obtained if the microphone is not too close and placed diagonally with respect to the walls. The microphone should be on a level with the lips of the speaker who should be comfortably seated about two feet, rarely less, from the microphone. Professional microphone tables have special cellular tops that cause little sound reflection. You will find by experiment that the sort of surface covering the top makes quite a difference to the quality of the recorded sound. Remember, the less reflection from the table top the better. If the room is small you may have trouble from the boominess due to room resonances. If you do, try placing the microphone table in various positions and if the condition cannot be overcome in this way, try another room. Having decided by trial and error where best to place the microphone and the best distance from it to seat the speaker, you must give the latter some training if he is not an experienced recorder. There are a number of errors which must be avoided. The beginner, though generally very microphone conscious, will nevertheless apparently forget that he is talking into a microphone and move his head away from the direct line to the microphone. If he is reading a script, he will either talk down to the script instead of keeping his head up or he will try to talk through the script. A suitable script rest is a great help in overcoming this fault but the rest must be a low one in order not to obscure the microphone.

Another usual fault is to get too close to the microphone—this over-emphasizes either the low or the high frequency components of the voice according to whether it is a pressure gradient or pressure operated microphone.

A common error is to use too little voice. Ordinary quiet conversational tones do not as a rule record successfully—but of course a loud declamatory style must equally be discouraged.

Another common error is to speak too fast when reading. Some of these faults will undoubtedly be corrected by the speaker for himself if you let him listen to some trial recordings. Sibilant speakers present a difficult problem. If a ribbon microphone is being used, an improvement can usually be effected by tilting the diaphragm so that it takes up an angle of about 45° with respect to the speaker. This produces some attenuation of the high frequencies and hence reduces the apparant sibilance—the hiss of the latter being mainly high frequency. With other types of microphone the position which produces the best results must be found by experiment. A rather drastic expedient is to introduce a good deal of treble attenuation in either the recording or reproducing chains. This may be effected by equalization or by overbiasing the recording head. Beginners are rather prone to tap the table or the floor whilst talking and to fiddle with the microphone leads. A few trial shots will soon convince them that it is better to sit still.

In home recording there are usually a few spectators and these must be warned not to mar the recording by making extraneous sounds by laughing, coughing, tapping their feet, walking across the room or closing a door.

(2) RECORDING TWO OR MORE SPEAKERS

A microphone with a figure of eight polar diagram is ideal for two speakers who can face opposite sides. Good results can also be obtained by using a moving coil or cardioid microphone pointing straight up or down between the two speakers. The latter plan will usually cope with several speakers in a discussion, so long as the table is not too large to allow proper distancing to be arranged. Each individual speaker must be given a trial recording to ensure that the voices are balanced in level. The criterion here must be the ear—even a professional type peak programme meter gives little indication of the apparent loudness of a speaker and only a test recording will tell whether the individual levels match. If a speaker has much too much voice, let his head be half turned from the microphone.

(3) RECORDING DRAMATIC PRODUCTIONS

The amateur recordist often volunteers or is asked to record an amateur dramatic production. Before attempting this it is as well to consider the copyright and the legal positions. (See Chapter 14). It is practically impossible to obtain a worthwhile recording from the actual stage whilst the public performance is being made. The cast must be specially placed with respect to the microphone as described above and carefully balanced. Contrasted acoustics must be made use of for each change of scene. An indoor scene must be recorded indoors and an outdoor scene out of doors. The actors must of course be allowed reasonable freedom of movement but they must be warned not to step into the dead side of a microphone or it will sound as though they have moved a considerable distance. The directional property of a ribbon microphone can of course be used for dramatic effect: as the actor says farewell he can move to the dead side and will sound as though walking away.

Great fun can be had by introducing suitable effects noises. It is not, for instance, necessary to find a waterfall to imitate the sound of one. Pouring water from one bucket to another will often sound very similar. Thunder can be imitated by rattling a sheet of tin and a sharp tap of a walking stick on to a table top will simulate the sound of a shot; or some real sound effect can be pre-recorded and mixed in with the sounds of the production if another reproducer is available.

(4) USE OF SCREENS

A screen can be used very effectively to alter the sound pattern in a room—it can, for instance, mask off an unwanted reflection from the walls and be used to reduce the liveliness of an otherwise boomy room. Screens must, however, be used with care because they rarely have any marked effect on long wavelength (low frequency) sounds and such frequency discrimination may be very noticeable.

(5) SOLO ARTISTE

The aim here must be careful balance between the artiste and the accompaniment. A straight singer should be at least three feet from the microphone and must be warned not to move backwards and forwards and not to swing the head from side to side—though a very slight turning of the head on *fortissimo* passages will probably assist the control. Crooners, of course, can be allowed to use a much closer technique but they must then be warned to use very little voice and to avoid breathing noises and teeth clicking—a few trial recordings will soon convince them of that necessity. A windshield on the microphone is also often advantageous. If only one microphone is used, several relative positions of microphone, singer and accompaniment will have to be tried. No hard and fast rule can be propounded because room acoustics play so large a part in the finished result. If two microphones are used, the best balance for each microphone must be found by trial and then the best setting of the mixer controls must similarly be determined. Microphones interact and only by trial can the best relative positions be found. If two directional microphones are used, a setting at right angles to each other will generally produce the best results. Reversing the connections to one of the microphones may have to be tried.

(6) CHOIRS

The choice of microphone is most important here. A cardioid microphone is probably the best because of its wide angle of pick-up, but good results can be obtained from two or more microphones correctly placed and phased. The danger of multi-microphone technique is that there are bound to be areas in which the sound made there is picked up by two or more microphones and this can lead to undue emphasis on that part of the choir. Of course, this can be used intentionally to give prominence to a soloist. Remember that balance can be achieved not only by moving the microphones but also be rearranging the choir. A distant technique is usually best but of course the farther away the microphone is placed the more indistinct will be the words and the final position is a compromise between clarity and reverberation.

(7) CHOIR WITH ORCHESTRA

Another even more difficult assignment is the choir with orchestra. This generally requires at least two microphones—one for the choir, the other for the orchestra. Ribbon microphones at right angles to each other are ideal for this purpose but it may take quite a lot of trial recordings before the best positions are found.

(8) CLOSE-HARMONY GROUPS

Success here depends upon careful placing of the voices. Do not forget that heads may be arranged one above the other as well as side by side. Always remember the polar diagram of the microphone that you are using and that to obtain best results the sounds you wish to pick up must come from somewhere within the live angle of the microphone (as in photography—the subject must be within the angle of view).

(9) PLACING OF INSTRUMENTALISTS

In recording the sound of a group of musicians the placing of the instrumentalists is of great importance. Brass instruments need to be well separated from stringed and wood-wind instruments. Care must be taken that the sound of a particular instrument is not masked from the microphone by some other instrument or player—the violoncello is particularly likely to give trouble in this respect. A good general rule is to avoid placing the microphone in line with the bell of any brass or wood-wind instrument, except muted trumpets.

(10) ODD RESONANCES

Vases, music-stands and some lamp-shades may be set in vibration by musical notes or by the sound of percussion instruments. They then make their presence felt by jangling sounds and by the apparently undue prolongation of some sounds. Be on the lookout for this and remove or blanket the offending articles.

(11) SONGS AT THE PIANO

These are amongst the most troublesome assignments of the amateur recordist, the difficulty being that, if the microphone is in the right position for the voice, too much piano sound will be picked up. A cardioid microphone suspended above the head and a little to the front of the performer or a ribbon microphone in the same position but tilted at an agle of 45° with respect to the keyboard will usually give good results and the performer can help by endeavouring to play the accompaniment more softly than usual.

(12) QUIET AT THE BEGINNING AND END

Remember to impress upon the performers that complete quiet for at least 30 seconds is required both before the start and at the end of the recording to enable a clean commencement and fade-out to be made. It is a great nuisance if unwanted sounds have afterwards to be edited out.

(13) RECORDING LEVEL

It is essential to record at a reasonably high level to obtain a good programme-to-noise ratio but it is always better to under-record (*i.e.* to record at so low a level that the programme-to-noise ratio is impaired) than to record at such a level that loud passages are distorted. Only repeated test recordings with careful note of the setting of the volume control for each test can enable the right one to be decided. If the programme has plenty of light and shade it will often not be feasible to use a fixed setting. Soft sounds will have to be recorded at a higher level and loud sounds at a lower level. This means that the volume control will have to be adjusted during the recording. A knowledge of the piece or, if you can read music, a musical score is then a great help. The artistes can also help if the limitations of the recorder are explained, *i.e.* they can limit the softness of *pianissimo* passages and the loudness of *fortissimo* passages. Do not worry about the loss of light and shade. The ear is very complacent about that. For broadcasting purposes the dynamic range of an orchestra, which is often as wide as 70dB or more, is controlled to 22dB with very little apparent diminution of the apparent contrast between loud and soft passages. Such drastic limitation is not necessary with magnetic recording—tape will accommodate a range of 36dB at least—but the risk of overload distortion and of printing between layers makes it essential to limit the volume of loud passages, whilst the inherent noise in the system makes it necessary to record the soft passages at a certain minimum level. Beware, however, of sudden abrupt alterations of the setting of the volume control: it must be gently eased back in anticipation of peak volume sounds and as gently brought up before a soft passage is expected. Of course, facility in this sort of control is only acquired by dint of much practice—but it is well worth while.

(14) POSITION OF RECORDER

It is a good plan to keep the recorder as far as possible away from the microphone in order to avoid the pick up of machine noises. For similar reasons keep the microphone well away from clocks or, better still, remove all clocks from the recording room.

(15) RECORDING OUT OF DOORS

Outdoor recordings can be peculiarly effective and satisfying. The bird songs, for instance, recorded by Ludwig Koch and Eric Knight Simms have delighted millions of listeners. The sound of an express train roaring through a station is always most impressive. The swish of the surf and the thunder of breaking waves will be a pleasant reminder of your holiday. But outdoor recordings are a real test of skill and generally of patience. The greatest difficulty is to avoid wind noises which often cause alarming plops, rattles and bangs. Never attempt to use an ordinary ribbon microphone out of doors—except on very still days. The **lip ribbon microphone** used by commentators is of course a different matter. That is a special design in which the microphone is held very close to the lips and is shielded from the wind by the lip shield. The resultant rise in bass response is corrected by means of an acoustic impedance. These microphones discriminate against the crowd noises and if " atmosphere " sounds are required they are picked up by a separate microphone and mixed with the output from the lip ribbon.

A windshield is almost essential for outdoor work. It consists of a spherical cap of hard woven material fitting tightly over the diaphragm, or a similar cap of wire mesh covered with thin material. If you do not possess a windshield, a scarf or handkerchief may be wrapped round the microphone but these are not very effective substitutes.

In order to collect sound from a wide area and concentrate it at the microphone, a parabolic reflector may be used. The microphone should be placed inside the reflector and pointing towards its centre as shown in Fig. 78.

The sound is collected by the paraboloid and reflected, in concentrated form, on to the diaphragm. This is the arrangement often used to collect bird sounds.

(16) RECORDING THE OUTPUT OF A RADIO RECEIVER
(See also legal aspect, Chapter 14)

The simplest way—though this is not to be recommended—is to place the microphone in front of the loudspeaker of the receiver and to record the output from the microphone.

A slightly better way is to connect the high level input to the recorder to the socket on the receiver marked extension loudspeaker (or if there is no such socket, to the terminals of the internal loudspeaker).

A good listening level is required from the receiver in order to minimize the effect of noise and hum components in the receiver

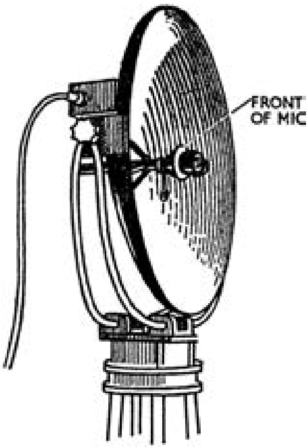


FIG. 78. A PARABOLIC REFLECTOR. THESE REFLECTORS MUST BE QUITE LARGE (3 FT. OR MORE IN DIAMETER) IF THEY ARE TO BE OF MUCH USE.

output. Such a high level will mean that the volume control on the recorder will have to be set fairly low to prevent overloading the latter. A point to remember is that, if the receiver is one in which connection to external loudspeaker socket automatically disengages the connection to the receiver's internal loudspeaker, a resistor equivalent to the internal loudspeaker's impedance (usually 3 ohms) will have to be shunted across the external loudspeaker sockets to maintain the correct load on the output valve of the receiver.

(17) CONNECTING DIRECT TO DETECTOR STAGE

A better, but slightly more complicated, method is to connect the low level input of the recorder to the output of the detector stage of the receiver. This will avoid all hum and distortion in the receiver's output stage. The normal arrangement in a superhet is as shown in Fig. 79. The blocking capacitor *C* (value say $0{\cdot}1\mu F$)

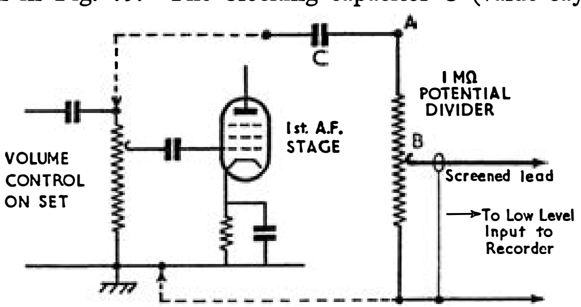


FIG. 79. CONNECTION DIRECT TO DETECTOR STAGE OF RADIO RECEIVER. (THIS WILL NOT DO IF HEAVY NEGATIVE FEEDBACK IS RETURNED TO THIS STAGE).

may not be necessary if d.c. is not present in the volume control. If the leads from the radio to the recorder are long, the shunt capacitance of the leads may produce treble attenuation. This can be offset by shunting the top half of the 1MΩ potentiometer (*AB* in Fig. 79) by a small capacitor. If negative feedback is taken back to the volume control, this arrangement may not work satisfactorily, but the output from receivers of this type is so good that connection to external loudspeaker sockets will do.

A number of manufacturers now market small pre-set radio tuners (sometimes switchable to two or more stations) which may be plugged direct into the high impedance input of a magnetic recorder. No other connection is necessary other than a good aerial and earth to enable radio programmes to be recorded.

9

LOUDSPEAKERS

THE loudspeaker is a double transducer: it first converts the electrical energy of the amplified output of the reproducing electro-magnet into the mechanical energy of a vibrating member, generally a coil, and then converts this mechanical energy into the acoustical energy of the sound wave.

In the dynamic variety of loudspeaker, the vibrator is a coil positioned in the gap of a cylindrical magnet by a spring holder called a **spider** to which a cone is attached (see Fig. 80). The

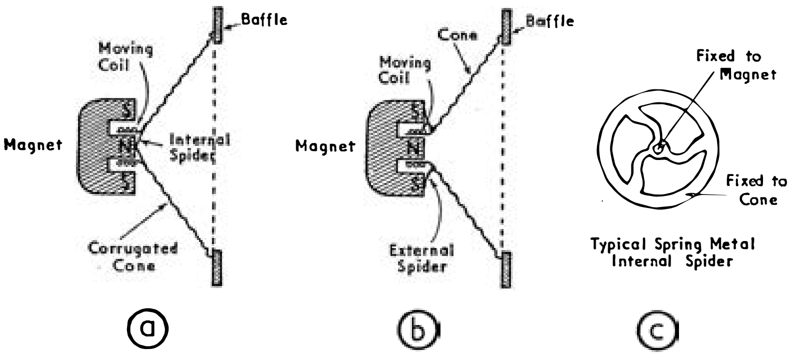


FIG. 80. MOVING COIL LOUDSPEAKERS WITH (a) INTERNAL SPIDER; AND (b) EXTERNAL SPIDER.

function of the cone is to apply to the air in the most efficient way the force developed upon the coil by the interaction between the magnetic field due to the current and that of the permanent magnet.

The waves generated at the back of the cone are in antiphase to those generated at the front, and unless the path length between the front and back of the cone is appreciable they tend to have a cancelling effect at low frequencies. The cone must therefore be mounted in a cabinet or upon a baffle board. To radiate low frequencies efficiently the cone should be large and so should the baffle or cabinet. This accounts for the "toppy" reproduction obtained from most small diameter loudspeakers. As the average home recorder can only house a small loudspeaker, the remedy is to use an external one. Care must be taken to match the external loudspeaker to the output of the recorder.

As mentioned in Chapter 3 on impedance matching, an incorrect load can lead to both harmonic and attenuation distortion. Fortunately, a suitable transformer will always make matters right. Remember that the impedance ratio of a transformer is the square of the turns ratio.

To maintain the response at high frequencies, the cone is often corrugated but corrugations only help—they cannot ensure perfect radiation at high frequencies. Many loudspeakers, therefore, have two units, one for low frequencies, the other for high. The latter is called a **tweeter.** In such systems a cross-over filter is used to ensure that low frequencies are fed only to the low frequency unit and high frequencies only to the tweeter.

There are at least four well-known types of cross-over network. A typical arrangement is shown in Fig. 81. This is called a **constant**

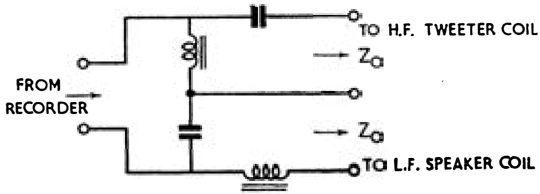


FIG. 81. A CONSTANT IMPEDANCE CROSS-OVER NETWORK FOR L.F. COIL AND TWEETER.

$$C \text{ (in farads)} = \frac{\sqrt{2}}{2\pi f_T Z_0} = 15\mu F, \text{ when } Z_0 = 15\Omega \text{ and } f_T = 1\text{kc/s.}$$

$$L \text{ (in henrys)} = \frac{\sqrt{2}\, Z_0}{4\pi f_T} = 1{\cdot}68mH, \text{ when } Z_0 = 15\Omega \text{ and } f_T = 1\text{kc/s.}$$

impedance network because, if the load presented by the loudspeakers is Z_0 in each case, the input to the cross-over network will also be Z_0 and constant despite change in frequency. Care must be taken that the cross-over network does not upset the negative feedback arrangements of the recorder's output stage.

Many experimenters who are dissatisfied with the high frequency response of their loudspeakers invest in separate tweeters, but these are not so successful as those built into l.f. loudspeakers. It is not a good thing to improve the response of the loudspeaker to such an extent that its range is wider than the frequency response of the recorder. The wider frequency response of the loudspeaker cannot improve the response of the recorder and will make audible many high frequency noises which are best left unheard.

Loudspeakers can only radiate up to a certain power without distortion and it is imperative not to overload them.

The positioning of the loudspeaker can have a great deal of bearing upon its apparent response. Usually, the best effect is obtained if the loudspeaker is mounted fairly high up in one corner of the room, but no general rule can be stated because everything depends upon the shape and dimensions of the room and the relative positions of loudspeaker and listener.

A loudspeaker cannot radiate correctly if the coil is not free to move. If the spider becomes bent, the coil former will foul the magnet and distortion will result. Generally speaking, recentering a coil is a job for the expert and the amateur will be well advised not to attempt it, because the clearance between coil former and magnet is often very small indeed.

Choosing a Loudspeaker

The volume of sound required to be delivered by a loudspeaker depends very much upon the size of the room in which it is to be used. A large cabinet loudspeaker, which sounds wonderful when tested in a shop, may be quite unsuitable for a small living-room. Whenever possible, therefore, it is best to try out a loudspeaker in your own home before buying. Most dealers will gladly arrange such a trial.

Phase-Inverter Type Cabinets

By careful design of the cabinet it is possible to collect the low frequency sound from the back of the cone and deliver it to the room in phase with the low frequency sound from the front of the cone. This, of course, greatly improves the low frequency response. Some acoustic treatment of the inside of the cabinet is necessary to ensure that high frequencies radiated at the back of the cone do not get transferred to the vent. A typical vented cabinet is shown in Fig. 82.

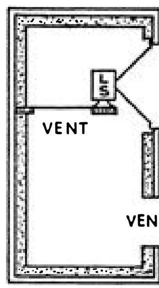


FIG. 82. CROSS-SECTION OF A VENTED LOUDSPEAKER CABINET.

Hoekstra* has worked out an approximate expression relating the various design parameters, *viz* $A = \frac{\pi V^2}{4} \left(\frac{\omega}{C}\right)^4$

where A equals area of vent in square feet, V equals volume of cabinet in cubic feet, ω equals $2\pi f$ where f is the resonant frequency of the loudspeaker and C equals velocity of sound in air in feet per second.

Thus, if the cabinet were made 36″ high, 24″ wide and 18″ deep and the loudspeaker resonance occurred at 50c/s, the vent area would have to be approximately 57 sq. inches. The cabinet is best made of plywood lined with $1\frac{1}{2}$″ thick slag wool covered with needle-loom carpeting.

Directional Properties of Loudspeakers

Loudspeakers usually radiate low frequencies over a wide area but the high frequency radiation is limited to a narrow area along the axis of the speaker. (See Fig. 83). This does not mean that high frequencies are inaudible at the side of the room—reflection ensures a good coverage—but merely that, for best results, you should place yourself on the axis of the loudspeaker and face the cone.

* C. E. Hoekstra, "Vented Speaker Enclosure". Reprinted by permission from March 1940 issue of *Electronics*, a McGraw-Hill Publication, copyright 1940.

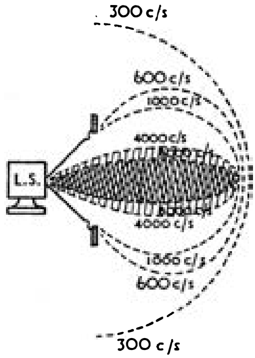


FIG. 83. RADIATION PATTERN OF A TYPICAL 12-INCH CONE SPEAKER.

Electrostatic Type Loudspeakers

A number of recently developed tweeters work on electrostatic principles. Fig. 84 shows a balanced push-pull electrostatic speaker.*

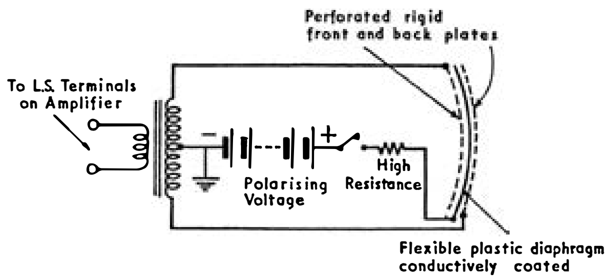


FIG. 84. SCHEMATIC DRAWING OF BALANCED PUSH-PULL ELECTROSTATIC LOUDSPEAKER (H. J. LEAK & CO. LTD).

It is claimed that this loudspeaker has a measured distortion characteristic over its working range of only 0·1 per cent.

The use of constant-charge electrostatic principles has enabled electrostatic loudspeakers to be designed to cover any desired range. The first wide range unit to be designed commercially† covers the frequency range 45-18,000c/s with a uniformity of ±3dB. It is claimed that it introduces negligible distortion within this range, at a sound level of 95 phons in an enclosure of 5,000 cubic feet, with average reverberation.

* H. J. Leak, "Why Electrostatic Loudspeakers," *The Gramophone*, May, 1955.

† Acoustical Manufacturing Company.

10

TESTING AND FAULT TRACING

PROFESSIONAL recording engineers make use of a great deal of complicated and relatively expensive test gear to check the condition and performance of tape recorders (*e.g.* peak reading valve voltmeters; multi-range current/voltage/resistance meters; bridge meggers which are instruments for measuring resistance with great accuracy; decade attenuators; valve-testers; distortion-factor meters; cathode ray oscilloscopes; harmonic analyzers; wow-meters and the like).

Equipment Required

The average amateur cannot afford such an array of equipment, nor will he find it necessary. Little, however, can be done without *any* equipment at all—other than the ear—and some form of meter for measuring alternating currents and voltages together with some source of pure tone are practically essentials.

Fortunately, test tone (440c/s and 1,000c/s) is provided by the BBC on the Third Programme wavelength ten minutes before the start of the programme, and test tapes with carefully recorded frequency runs are available at very moderate cost. (E.M.I. Frequency Test Tape TBT 1, for example, consists of 60 seconds of 8,000c/s pure tone recorded at 7½ inches per second for azimuth adjustment, followed by 30 seconds at 1,000c/s and 15 seconds each of 40, 60, 110, 200, 500, 1,000, 2,000, 4,000, 6,000 8,000 and 10,000c/s).

A HOME-MADE A.C. VOLTMETER

Accurate a.c. voltmeters can be quite expensive but it is easy to make one that will do.

Fig. 85 shows a simple form of a.c. voltmeter that may be constructed quite cheaply. The rectifier can be either a germanium crystal diode (type OA71) or a metal rectifier (WX6).

The calibration of such a meter is unimportant as it is comparative readings which are required, but it must be remembered that a change in level of 6dB will halve the voltage reading.

An even more useful meter can be made by incorporating a valve. Fig. 86 shows a simple valve voltmeter.

A CATHODE RAY OSCILLOSCOPE (C.R.O.)

This instrument is also extremely useful, enabling the shape of a waveform to be observed on the screen of a cathode ray tube. An electronic " gun " sends a stream of electrons on to the screen, which fluoresces where they impinge. A focusing arrangement narrows the beam to a tiny spot. A sawtoothed waveform (Fig. 87) is applied to deflecting coils or plates to pull the beam slowly across

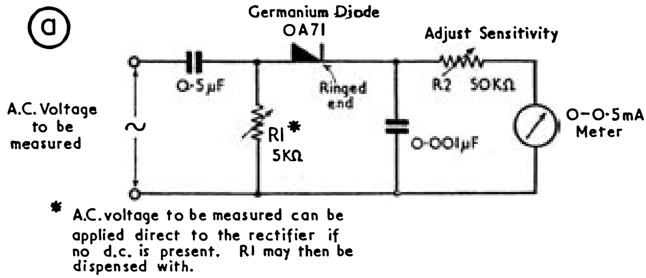


FIG. 85(a). A SIMPLE A.C. VOLTMETER.

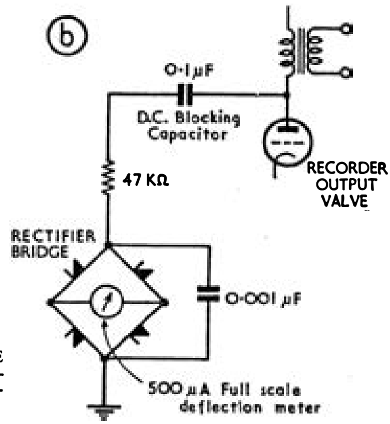


FIG. 85(b). CONNECTIONS FOR BRIDGE RECTIFIER METER. THE 0·001 CAPACITOR SERVES TO BY-PASS ANY BIAS COMPONENTS IN THE OUTPUT.

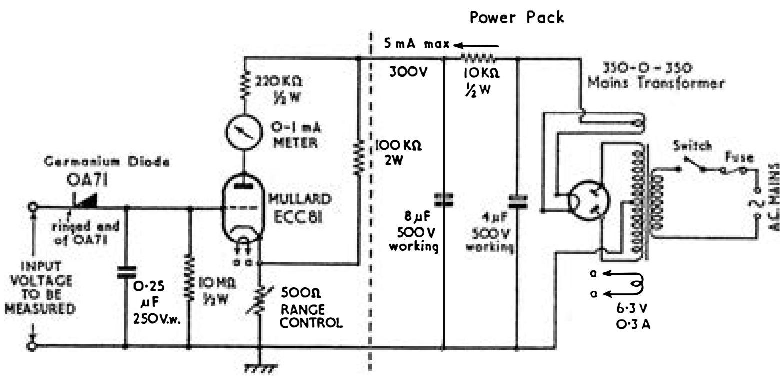


FIG. 86. SIMPLE VALVE VOLTMETER* AND ITS ASSOCIATED POWER PACK.

ECC81 is a double triode. Use either one side only or connect to each other the respective anodes, grids and cathodes. If the voltage to be measured contains a d.c. component, an input capacitor as shown in Fig. 85 (a) will be required.

FIG. 87.
SAWTOOTHED WAVEFORM OF C.R.O. TIMEBASE

Recor Circuit suggested by H. L. York in part 2 of series *Experimental High Fidelity Tap* **dings*, "Hi-Fi News," Vol. 2, No. 3, August, 1957.

the tube from left to right and then to make it fly back to the beginning again. In many oscilloscopes the beam is suppressed during the flyback period. The signal of which the waveshape is required to be examined is fed to another pair of deflecting coils, or plates, at right angles to the ones to which the sawtoothed waveform is applied. (Fig. 88). The result is that the flying spot is displaced

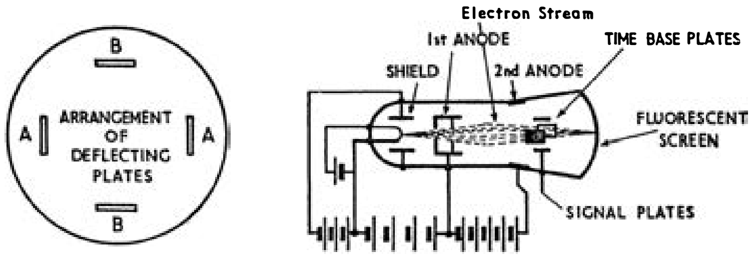


FIG. 88. THE CATHODE RAY OSCILLOSCOPE.

AA =HORIZONTAL PLATES TO WHICH THE SAWTOOTH WAVEFORM IS APPLIED.
BB =VERTICAL PLATES TO WHICH SIGNAL IS APPLIED.

from the horizontal datum line by an amount proportional to the instantaneous value of the alternating signal.

In this way a good copy of the signal waveshape is caused to appear on the screen. By varying the frequency of the **timebase** (as the sawtoothed waveform is called) the displayed wavelength can be changed to suit the particular frequency of the signal.

Little more in the way of testing equipment is absolutely necessary.

The maintenance of a recorder falls into two main categories: MECHANICAL and ELECTRICAL but of course the two are often considerably intermixed. Mechanical maintenance is dealt with in Chapter 11 and only electrical maintenance will be detailed in this chapter.

Testing the Amplifier

The amplifier or amplifiers should be regularly tested to ensure that they are free from distortion and have the correct frequency response. The professional method of testing an amplifier is:

(*i*) The valves are tested on a valve-tester for correct mutual conductance and grid-cathode resistance.
(*ii*) The l.t. volts and h.t. feeds of the various valves are measured to ensure that they are within tolerance.
(*iii*) The percentage harmonic distortion in the output, when the input is fed with pure tone at a given level, is measured with a distortion-factor meter.
(*iv*) A calibrated peak-programme meter is used to measure the output levels for a fixed level of input, at a series of frequencies covering the desired frequency range.

If the reader does not have a valve-tester and wishes to get his valves tested, he should take them to a radio dealer who may test them free of charge the first time—though no doubt a small charge will be made if they are taken in too regularly.

The feeds of the various stages are generally specified by the manufacturer but unless special facilities are provided, they are rather difficult for the amateur to measure. Most instruction books give the voltages which should appear at various parts of the amplifier and if a suitable meter is available and the right spots can be located (either from the circuit or from the chassis diagram) it is a good plan to make a regular check. Professional tape recorders provide a switchable feed meter to enable head currents to be measured but in the average home recorder no such provision is made. If it is required to measure the current into a recording or erasing head the following arrangement may be used. Connect a 100Ω resistor in the earthy side of the coil and measure the voltage across it with an a.c. millivoltmeter. (Fig. 89).

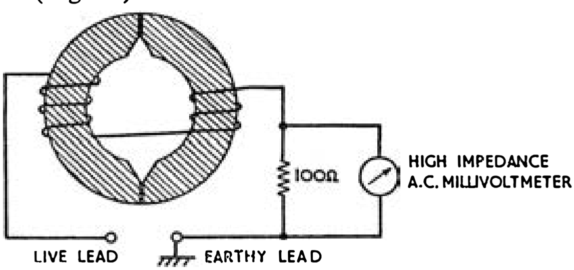


FIG. 89. A SIMPLE METHOD OF MEASURING HEAD CURRENT.

As $I = \frac{E}{R}$, the head current in amperes will be equal to the measured voltage (in *volts*) divided by 100. Thus, if the meter reads 17mV the head current will be equal to $\frac{17 \times 10^{-3}}{100} = 0{\cdot}17\text{mA}$.

Testing the Frequency Response of a Tape Recorder

If a standard test tape is available the frequency response of the reproducing system is readily adjusted as follows:

(*i*) Clean the heads, capstan and pressure roller.
(*ii*) Lace up the test tape.
(*iii*) Connect the test meter to the output terminals.
(*iv*) Adjust the alignment of the reproducing head (instructions for this are given on page 93).
(*v*) Reproduce the test run and note the readings on the output meter, adjusting the reproducing equalizer BASS and TREBLE controls to give the same reading on the meter at all frequencies.

If the test meter is a peak programme meter, it will be possible to compare the output levels in dB obtained for each recorded frequency. The tolerance is usually ± 2dB from 50c/s to 5,000c/s and ± 3dB from 5,000c/s up to the limit of the frequency response of the recorder.

When the reproducing system has been tested in this way, it may be used to test the recording response, if a variable frequency tone source is available: this is an oscillator the output of which can be varied both in frequency and amplitude.

What is required is to inject tone into the input at a constant level but varying frequency and to record each frequency for about 15 seconds. The test recording is then reproduced and the comparative output levels compared with the comparative output levels obtained from the standard tape. Care must be taken to make the test recording at a low level, or distortion components will make accurate reading of the comparative output level impossible.

With certain recorders, and particularly with certain American pre-emphasis networks, unless the frequency sweeps are made at very low level, complete saturation of the tape occurs at the high frequencies, not only giving low comparative output, but many spurious inter-modulation products.

If the recorder has a meter for measuring input level, take care that this meter registers the same reading for each input frequency. If the recorder has a magic eye level indicator it will probably not give sufficient indication of the input level (it is used to indicate maximum levels and, for the frequency response test, low levels of input must be used). It will therefore be necessary to connect a meter across the input to ensure that the level is constant.

The home-made meter previously described will do if no other is available.

In the case of many of the slower speed recorders ($3\frac{3}{4}$ in./sec.), the frequency response is only linear up to about 5,000c/s: the maker's instruction manual should be your guide here.

Fault Tracing

Quick fault tracing depends very largely on knowing the circuit of the recorder. Many manufacturers will only supply circuit diagrams to registered dealers, while others will supply them to users on request. Of course, it is always possible to trace the circuit but this might well stump the average amateur.

Here is a list of various common faults and their probable causes:

FAULT	PROBABLE CAUSES	ACTION
Pronounced attenuation of high frequencies.	Tape laced up the wrong way round (coated side away from head).	Check tape lace up.
	Head dirty, *i.e.* covered with magnetic dust from the tape coating or other foreign matter such as adhesive from patches.	Clean heads and check their position and azimuth adjustment.

FAULT	PROBABLE CAUSES	ACTION
Pronounced attenuation of high frequencies —*continued.*	Wrong azimuth adjustment.	
	Wrong equalizer setting.	Check equalizer settings.
	Part of the mumetal head screen in contact with the tape.	Check that the screen is not in contact with the tape.
	Tape strained and twisted so that it does not make sufficiently good contact with the gap in the replay head.	
	Too much bias current into the record head.	Adjust bias amplitude.
	Misplaced or worn heads preventing close contact between the tape and the head.	
Pronounced resonant peak at some high frequency.	Dry joint (badly soldered connection) giving virtually capacitative coupling to the input to one or other of the valves; or dirty contact on volume control or valve cap.	Check that the peak is not intentionally provided by the manufacturer in order to increase the intelligibility of speech.
	(Audible high frequency emphasis is often due to the use of too small a loudspeaker but sometimes a gross mismatch between amplifier output and loudspeaker input will give the same sort of result even if a good speaker is used).	
Attenuation of very low frequencies.	This is generally due to insufficient low frequency boost in the recording chain. Very often, in cheap recorders, it is intentional, because full l.f. boosting tends to emphasize any hum components. When l.f. attenuation is due to an excessive bass cut in the reproducing equalizer, reducing the cut will bring up hum components and emphasize rumble due to motor vibration or imperfections in the constant speed drive system.	
Non-linear distortion.	Overmodulation; incorrect value of a.c. bias into record head; a soft rectifier or a faulty valve; loudspeaker coil off centre. Incorrect bias on a valve or valves. Drop in h.t. volts. Rejector circuit in output of rec. amp. off tune (this would allow the output of the bias oscillator to saturate the core of the output transformer of the recording amplifier).	Check recording level and bias value. Check valves. Check tuning of bias trap. Measure h.t. voltage.

FAULT	PROBABLE CAUSES	ACTION
Recorded tape gives no output.	Selector knob left in record position. Input plug left in. Unconnected output to loudspeaker plug left in. Break in connection to loudspeaker. Piece of splicing tape sticking over gap in head. Gain control at zero. Amplifier fault. H.T. fuse blown. No recording on tape, due to fault in recording chain.	Check replay chain first by replaying a known good recording, taking care to put selector into replay position before pressing start button.
Tape records but is insufficiently erased.	Mains voltage low. Break in output of erase oscillator. Erase current control wrongly set.	Check mains volts.
	Failure of the oscillator valve. This valve is generally pretty hard worked to provide sufficient power and after a long life it may lose its emission.	Check conductance of oscillator valve.
	Erase head dirty (a layer of coating dust only a quarter of a thousandth of an inch thick will provide a shorting path for erase flux).	Clean erase head.
Tape is erased but machine will not record.	Input plug not properly connected or break in wire or disconnection at socket. Short on input cable. Valve failure in amplifier. Disconnection between amplifier and head. Head winding open circuited.	Check input circuit. Feed tone to input and check meter reading. Check head winding for continuity, using tone for test. Never attempt to measure d.c. resistance of head as this would magnetize the core.
Recording is low level or high frequencies are attenuated.	Recorded or replayed with backing side of tape against head. Meter sensitivity wrong. Bias current of too low value due perhaps to wrong setting of bias adjustment control or to failure of oscillator valve. In the latter case, of course, the erase current will be too low as well.	Check lace-up. Adjust meter sensitivity. Check bias level setting. Check conductance of oscillator valve.
Low pitched hum during playback.	Instrument being operated over power cables or near equipment with high external fields. Microphone in hum field. Microphone transformer or leads insufficiently screened or earth on screen faulty.	Check if hum is there even if tape is stationary. To see if hum is on tape try reproducing at different speed—pitch of recorded hum will change.

FAULT	PROBABLE CAUSES	ACTION
Low pitched hum during playback *—continued.*	Orientation of mains transformer incorrect. Faulty earth connection (see section on earthing on page 91).	In many recorders the angle of the mu-metal wing on the head pressure arm finally decides the hum content. When readjusting take care that the pressure pad contact area is not reduced. Try changing the polarity of the mains connection.

Principal Causes of Excessive Background Noise

The background noise on a tape recording should be very low indeed—in fact barely audible. Amongst the main causes of excessive noise are:

(1) HEADS COVERED WITH POWDERED TAPE COATING OR DIRT

All manufacturers stress that the heads must be kept clean but many users do not realize quite how important this is. It cannot be too strongly emphasized that the heads must be kept meticulously clean. A dirty erase head will not erase because the erase flux is shorted through the oxide powder. A dirty recording head will not record high frequencies at correct amplitude because the high frequency magnetizing flux is similarly shorted by the oxide powder and the increased separation between head and tape also tends to reduce the high frequency response. A dirty reproducing head will not reproduce correctly because the external flux of the tape coating is partially shorted by the powdered oxide. Never allow adhesive from jointing patches to come in contact with the head.

The best cleaner to use is a piece of mutton cloth, preferably lightly damped with methylated spirits. Avoid the use of carbon tetrachloride—it leaves a greyish white deposit which will cause nearly as much noise as the powdered iron oxide. Oil, water and petrol should also be avoided.

(2) STRETCHED, CURLED TAPE

With repeated use the tape always stretches slightly and it does not then make proper contact with the heads even on decks fitted with pressure pads. A tape in this condition should be scrapped if it is required to make good recordings.

(3) UNEVEN TAPE COATING

As the coating is gradually worn off, some parts of the coating become of uneven thickness. Again, the only remedy is to scrap the tape or at any rate the parts which are unevenly coated.

(4) MAGNETIZED HEADS

Though the heads are made of Permalloy or similar ferrous materials which have normally little or no remanence, they do tend to become magnetized after a time. In some makes of machine it is arranged that, if the selector knob is moved slowly from the recording to the rewind position, the currents in both the record head and erase head die away slowly—thus subjecting the heads to decreasing cycles of magnetization. Several makes of head demagnetizer which can be plugged into the mains are available, or one can be made by winding many turns of insulated 24 S.W.G. copper wire on to a U-shaped laminated core (Fig. 90). The number of turns depends

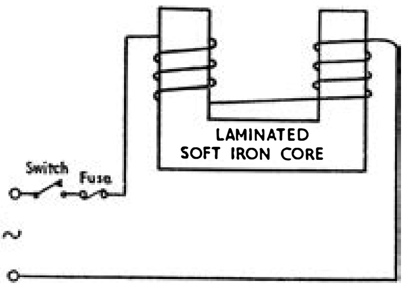


FIG. 90. A USEFUL HEAD DEMAGNETIZER. WINDING CONSISTS OF MANY TURNS OF 24 S.W.G. COPPER WIRE.

on the size of the core but there should be sufficient to ensure that the coil takes no more than 1·5mA when directly connected to the mains.

Do not bring the demagnetizer too close to the heads. The correct procedure is to switch on the demagnetizer whilst it is still several feet away from the heads. The demagnetizer should then be advanced to $\frac{1}{4}$ inch from the gaps and withdrawn very slowly to a distance of 3 feet before switching off. (Be careful to put your wrist-watch out of harm's way before using the demagnetizer—it is so powerful that it may damage even an " anti-magnetic " watch and it is disastrous to bring an ordinary hair spring within 3 feet of it).

(5) HUM

Though the heads are carefully screened both magnetically and electrostatically and the coils are astatically wound, hum troubles are by no means abnormal. Look first for improper earthing of the screens and try reversing the polarity of the mains leads. Unwanted hum may be due to disturbing the orientation of hum-bucking coils—beware of changing the position of any apparently haphazard loops in wire connectors. It may well be that the designer has found by trial and error that a certain loop in a certain place will minimize hum. Hum may be due to a faulty smoothing capacitor in the mains unit or to low heater-to-cathode insulation in the first valve stage. Do not forget hum may be picked up by the microphone if it is placed too near a mains lead. (See also section on earthing, page 91).

(6) VALVE NOISES

Such as thermal noise from first stage, or microphony: a valve with a loose electrode will tend to ring if subjected to vibration. Very often a valve which sounds microphonic in the first stage will work satisfactorily in a later one.

(7) RESISTANCE NOISES

Many of the ceramic types of resistor tend to get noisy with age. The only remedy is replacement.

(8) INCORRECT BIAS

If the bias current is a little too low the background noise will rise noticeably.

(9) RECORDING MADE AT TOO LOW LEVEL

A good programme-to-noise ratio can only be achieved if the level recorded is sufficiently high—but beware of recording at too high a level because printing between layers will occur (see page 99) and this will give rise to spurious pre-echo and post-echo signals.

(10) UNEARTHED SCREEN OF MICROPHONE LEADS

(11) RUMBLE

This may be due to a fault in the vibration insulation between motor and capstan or a flat on the rubber pressure roller. Dirt on capstan and pressure roller can also produce it. Mechanical vibration of the microphone during recording will also cause rumble as will direct microphone pick-up of motor noise during recording.

(12) SECOND HARMONIC IN OSCILLATOR OUTPUT

If the bias or erase currents have any second harmonic component, the current waveshape becomes flattened on one side. The magnetizing flux in the bias or erase head will then have an effect similar to a mixture of bias current and d.c. signal. Owing to minute irregularities in the contact between tape and head and to small changes in the thickness of the coating and to slightly uneven coercivity of the coating along its length, this virtually unidirectional component of the flux will produce minute variations in the unidirectional component of the induced magnetism. This produces an audible hiss in the output. In the more expensive recorders a great deal of trouble is taken to keep the oscillator output free from second harmonic distortion. A typical arrangement is shown in Fig. 91. If the rejector circuit marked 1 or the acceptor circuit marked 2 in Fig. 91 comes off tune, or if the oscillator output changes in frequency (due perhaps to change in temperature affecting the capacitance of the capacitors) whilst the rejector and acceptor circuits remain tuned to the original frequency, these circuits provide less protection and noise may result.

(13) REJECTOR CIRCUIT IN OUTPUT OF RECORDING AMPLIFIER COMING OFF TUNE

This rejector circuit (Fig. 92) is to prevent the bias current

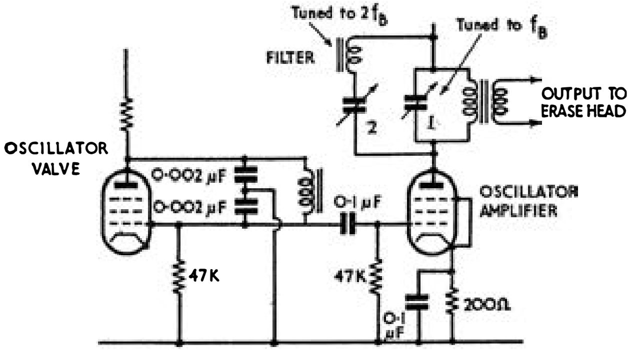


FIG. 91. 2ND HARMONIC FILTER IN OUTPUT CIRCUIT OF ERASE OSCILLATOR AMPLIFIER.

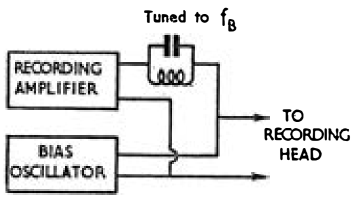


FIG. 92. THE REJECTOR CIRCUIT IN THE OUTPUT OF THE RECORDING AMPLIFIER PROTECTS THE LATTER FROM BIAS OSCILLATOR OUTPUT.

from saturating the core of the output transformer of the recording amplifier. (The ratio of bias current to signal current is about 3 : 1). If the rejector comes off tune the indication will be increased distortion and noise in the output.

(14) WORN OR MISPLACED HEADS

After a time the core of the head wears flat and the tape no longer makes good contact with the gap. The core can be reground into the right shape but this is not an operation to be undertaken by the amateur. If the head is a cheap one, it will generally be cheaper to fit a new one than to have it reground. If the head securing screws work loose it is possible—on some machines—that a head may change its position relative to the tape so that the latter no longer properly contacts the gap. This condition is shown—exaggerated—in Fig. 93. The remedy is to replay the test tape and slowly alter the position of the head until maximum output is obtained. The head should then be locked in that position.

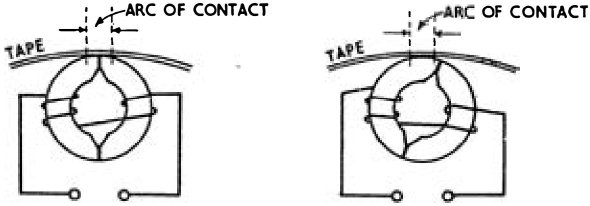


FIG. 93. SHOWS HOW DISPLACEMENT OF RECORDING HEAD MAY REMOVE THE GAP FROM THE ARC OF CONTACT BETWEEN TAPE AND HEAD.

Checking the Distortion Content

Even without suitable test gear any appreciable non-linear distortion is detectable by the ear, especially when reproducing the sound of a piano. A more accurate assessment can be made by recording a pure tone and displaying the reproduced waveform on the screen of a cathode ray oscilloscope. If no distortion is present, the reproduced waveform will be sinusoidal. Any departure from the sine-wave shape indicates the presence of spurious harmonics. The actual shape obtained will depend upon whether the distortion is mainly second or third harmonic. Typical output waveshapes when second harmonic is present are shown in Fig. 94.

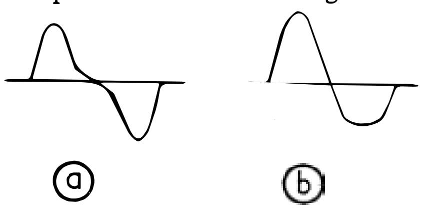


FIG. 94. TYPICAL WAVESHAPES WHEN A SPURIOUS 2ND HARMONIC IS ADDED TO THE FUNDAMENTAL. THE SHAPE DEPENDS UPON THE PHASE RELATIONSHIP BETWEEN 1ST AND 2ND HARMONICS.

When third harmonics are present, the waveform tends to be flattened on both sides. If the distortion is due to too little bias, the reproduced low frequency waveform will look like Fig. 95(b).

Absolute measurements of harmonic content can only be made with a distortion factor meter or harmonic analyzer, but if a separate replay head is available some idea of the total distortion can be obtained by connecting the output of a test oscillator to the vertical or *Y* plates of a cathode ray oscilloscope and to the recorder input, recording the tone and connecting the output of the replay amplifier to the horizontal or *X* plates instead of the timebase signal. With low distortion content, the figure on the cathode ray oscilloscope screen will be a circle or an ellipse. Any distortion in the output will make the figure of irregular shape.

If no separate replay head is available a similar method may be used to detect distortion in the amplifier only. Connect the input to the amplifier to the tone source and to the *X* plates of a CRO and connect the output of the amplifier to the *Y* plates (the timebase oscillator must, of course, be switched off or disconnected).

Using a CRO to Detect Wow, Flutter and Hum

A similar method may be used to detect wow and flutter. Record tone and replay, connecting the output of the oscillator to the *X* plates and the output of the recorder to the *Y* plates. The rate of change of the axis of the ellipse and the degree of change indicate the amount of wow. Another way is to record tone and reproduce it, connecting the output of the recorder to the *Y* plates of the CRO. It should be possible by sync. control to keep the waveform on the screen steady. Wow and flutter will make the trace oscillate, but a good deal of experience is required to interpret the degree of wow from the observed oscillation. A more informative test for the amateur is to record music which includes long sustained notes. If the reproduction of such notes sounds as though they are changing in pitch, too much wow is present. A more severe test is to record music at the higher tape speed and reproduce it at the lower. This will have the effect of emphasizing any wow that may be present in the drive system.

When 1kc/s tone is recorded and the reproduced waveform is displayed on the CRO, the presence of hum in the output will be shown up as a ripple superimposed on the 1kc/s waveshape. (See Fig. 95(c)).

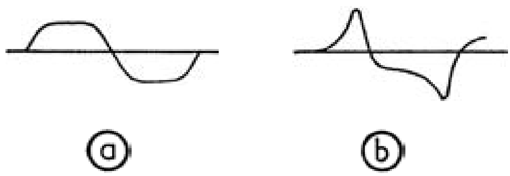


FIG. 95(a) AND (b). TYPICAL WAVESHAPES WHEN UNEVEN HARMONICS ARE SUPERIMPOSED UPON THE FUNDAMENTAL.

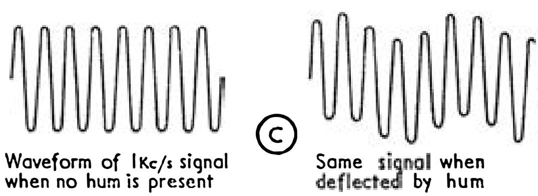


FIG. 95(c) WAVETRACES ON C.R.O. WHEN REPRODUCING 1 kc/s TONE, WITH AND WITHOUT HUM COMPONENT.

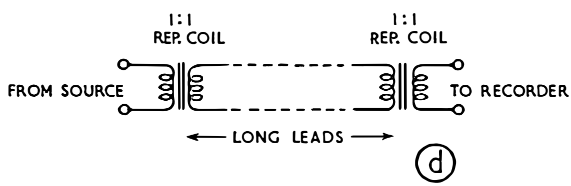


FIG. 95(d). ILLUSTRATING THE USE OF REPEATING COILS TO OVERCOME NOISE INDUCTION ON LONG LEADS. (SEE PAGE 92).

Earthing

Even when every piece of apparatus in use appears to be adequately earthed, mains hum may still be audible. This may be due to an undesirable multiplicity of earthing points. It is almost impossible to prevent small differences in potential between different earthing points and, of course, even a difference of a few millivolts will cause mains frequency currents to flow in the earthy side of connectors and this may produce audible hum if the hum voltage is applied to the input to the amplifier.

The remedy is to use as far as possible only one common earth point, connecting the earthy side of each piece of apparatus to it by heavy gauge wire. Earthing problems are always mitigated by the use of suitable coupling transformers.

Repeating Coils

These are 1 : 1 transformers inserted at opposite ends of long leads. They minimize induction because stray fields cut both leads in the same direction and the induced currents cancel in the primary of the output transformer.

Theory of Magnetic Screening

If two coils are in proximity, an alternating current in one of them will produce alternating lines of force which will cut the conductors of the other coil and induce a voltage in it. But if it is required to inject alternating current into either coil without inducing any voltage in the other, the coils can be magnetically screened from each other. This is achieved by enclosing one or other of the coils in a box of permeable magnetic material such as mumetal. To be effective the box must be fairly thick. This is expensive because mumetal is about 75 per cent. nickel—a comparatively costly metal. The cost may be reduced by using a screen consisting of two thinner mumetal boxes, one outside the other—an arrangement which is as effective as if mumetal filled the space between the boxes.

The function of the screen is to provide a path of low reluctance for the magnetic flux. Thus, the lines of force due to the current in the coil close through the mumetal box. (See Fig. 96).

If the mumetal box is earthed it will form an electrostatic screen too.

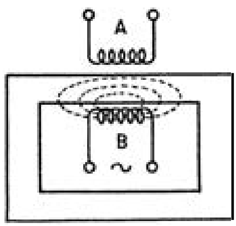


FIG. 96. BOX OF MUMETAL FORMING COMPLETE MAGNETIC SCREEN AROUND COIL, *B*. LINES OF FORCE DUE TO CURRENT IN *B* DO NOT INDUCE ANY VOLTAGE IN COIL *A*.

Theory of Electrostatic Screening

If a generator, with one side earthed, is connected to two plates *A* and *B* of a capacitor, the alternating voltage applied to the unearthed plate *A* will not cause any change in the potential of plate *B* because plate *B* is directly connected to earth. (Fig. 97(a)).

If, however, plate *B* is connected to earth through an impedance Z_1 as shown in Fig 97(b), the application of the alternating potential to plate *A* will cause an alternating current *I* to flow through Z_1 and will build up a voltage IZ_1 on plate *B*.

The build-up of this voltage can be prevented by placing a metal screen *S* between plates *A* and *B* and connecting *S* to earth. Sometimes the screen has to be connected to earth through an impedance Z_2 and it is then ineffective because the voltage *E* produces a current I_1, through Z_2 producing a potential difference

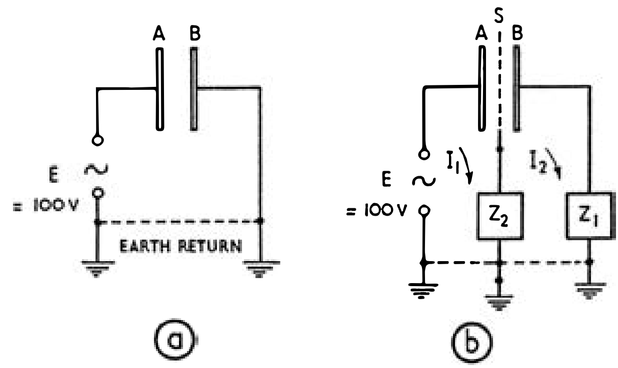


FIG. 96. ILLUSTRATING THE THEORY OF ELECTROSTATIC SCREENING.

I_1Z_2 across Z_2 and as Z_2 is in the circuit Earth, Z_1, B, S, Z_2, Earth, the voltage across Z_2 is applied to the latter circuit and produces a current I_2 in it and therefore a potential I_2Z_1 appears on plate B. The remedy is to connect the screen to some fixed potential. Changes in the potential of plate A then cause no change in the potential of the screen and hence no change in the potential of plate B.

Azimuth Adjustment

Unless the alignment of the gap in the reproducing head is the same as that of the recording head, treble attenuation will result. The misaligned reproducing head gap behaves as though its effective length had been increased. (See Fig. 98). In machines with com-

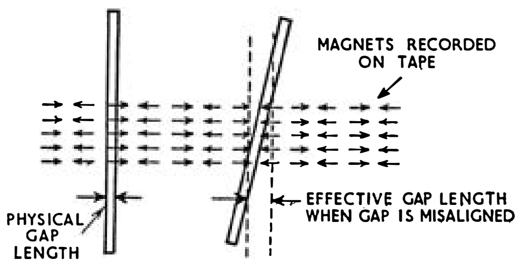


FIG. 98. SHOWING THE EFFECT OF MISALIGNMENT OF REPRODUCING HEAD GAP.

bined record/reproducing heads, the alignment is of course automatically correct for all tapes recorded *on the same machine*, but tapes recorded on other machines will not be satisfactorily reproduced unless the alignment of the recording head on the machines upon which they were recorded is the same as that of the reproducing head on the machine on which they are reproduced. The standard gap alignment is at right angles to the direction of tape travel. The method of adjustment varies considerably with the make of head. In some professional recorders, the heads are balanced on knife-edges and held in position by springs working against an adjustable stop (Fig. 99). In other makes the heads are balanced on a piece of wire

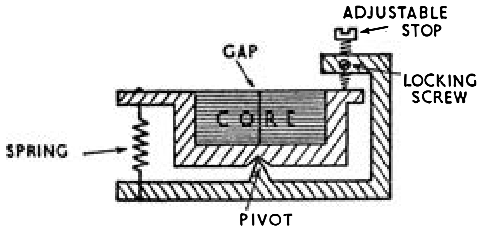


FIG. 99. A PRACTICAL SYSTEM OF AZIMUTH ADJUSTMENT.

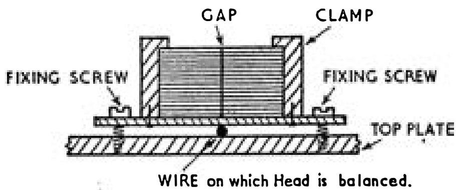


FIG. 100. SIMPLE SYSTEM OF AZIMUTH ADJUSTMENT.

and held in position by four screws (Fig. 100). With this sort of mounting do not attempt to adjust all four screws. It is generally sufficient to tighten either the left-hand or right-hand screw in front of the head.

Do not be tempted to remove the screens to examine the head. It is often quite difficult to replace the screening if once removed unless a special jig is available.

A special test tape, with a fairly high frequency recorded upon it, is reproduced and the alignment of the head is adjusted for maximum output. Recording heads (on machines with separate recording heads) are aligned by first aligning the reproducing head and then recording high frequency tone and adjusting the recording head gap alignment for maximum output from the already aligned reproducing head.

Adjusting Bias

(1) ON MACHINES WITH SEPARATE REPRODUCING HEADS

Record tone at normal input level and reduce bias to zero. Then gradually increase the bias until the output level is maximum. Go on increasing the bias until the output level drops slightly (about 1dB). This is the optimum setting.

(2) ON MACHINES WITH COMBINED RECORD/REPRODUCING HEADS

Record tone at normal input level and reduce bias to zero. Stop the machine and mark the tape " 1 ". Record again and increase the bias by a small amount, marking the setting of the adjust bias control " 1 ". Stop the tape and mark it " 2 ". Record again with slightly increased bias, noting the new setting of the control. Do this, step by step, until bias is at its maximum. Then replay the tape and note the mark on the tape when the output signal is greatest. The next higher bias setting is the correct one, *unless a measure of top cut is desired.* If another similar recorder is available the tape may be threaded up to run past the heads on both machines and if the drive on both machines is switched on simultaneously (at the same speed) the signal recorded on the first machine will be reproduced on the second—always supposing that one machine is set to RECORD and the other to REPLAY. Adjusting the bias on the first machine is then a simpler matter, as it can be effected as explained in (1).

11

MECHANICAL MAINTENANCE

THE tape drive system has three important tasks to perform: (*i*) it must pull the tape past the heads at a constant rate; (*ii*) it must wind the tape up evenly upon the take-up spool; and (*iii*) it must rewind the tape smoothly after each recording or replay.

Motor Maintenance

In looking after the motors it is essential to follow the manufacturer's instructions, because some motors require regular oiling and others have grease-packed bearings which require no oiling for very long periods. Even the types that require oiling should be oiled very sparingly—it is generally much more harmful to over-oil than to under-oil any electric motor. Commutator-type motors depend upon the commutator being kept meticulously clean, otherwise sparking will occur at the brushes and the commutator will become pitted. If the motor is of the type which employs a capacitor for phase splitting, the capacitor must, if faulty, be replaced by another of exactly the same capacity and rating. A wrong capacity here might cause the magnitude of the capacitative reactance at the supply frequency to equal that of the inductive reactance of the winding and this would short circuit the mains. The resultant heavy current might burn out the winding or at best blow a fuse.

Constant Speed Drive Systems

The tape is pressed between a constant speed drive capstan and a pressure roller. One of these is mirror smooth, the other is rubber covered. When the capstan is smooth its driving surface is made wider than the tape—the capstan then drives the rubber pressure roller which in its turn drives the tape. It cannot be too strongly emphasized that the pressure roller must not be left engaged with the capstan when the recorder is not in use. If it is, a depression will be formed in the rubber tyre which will cause flutter when the next recording or reproduction is attempted. Some makes of tape deck have a protective arrangement whereby the pressure roller is automatically disengaged from the capstan whenever the machine is switched off—a refinement not common to all makes. Care must be taken to keep the capstan and pressure roller free from oil and dirt.

Braking Systems

The need for rapid rewinding and winding-on makes it necessary to employ a braking system to bring both spools to a stop simultaneously. In some machines reduced braking is applied to the trailing spool when winding forward or backward. The brakes are of many types, the most ingenious being an electro-magnetic clutch. Many decks employ band brakes which are mechanically linked to the stop switch or are operated by a relay when the stop switch is pressed. The smooth action of the recorder depends upon accurate braking and the maker's instruction book should be consulted when they need adjustment. A friction type brake cannot work successfully if the surfaces become oily and great care must be taken to prevent this happening. Two common braking faults are: a sticking plunger on brake relay (the remedy is to grease the plunger with the type of grease recommended by the maker); and armature of relay spaced too far from the relay coil, due to brake arm coming out of adjustment.

Many quite expensive tape decks do not provide for sufficient braking on the trailing spool during rewinding. Slight manually applied braking, with finger to outside of trailing spool, must then be used to ensure that the tape is tightly wound on the take-up spool. A tape can be completely ruined by loose winding. On the other hand, excessive braking may cause the spool to collapse or may over-stretch the tape.

If the tape deck is one that incorporates a belt drive from the constant speed motor to effect spooling, it is essential to check which way the belt is laced up in order that, when it is required to fit a replacement, the correct lace-up may be used.

Common motor faults are as follows:

INDICATION	PROBABLE CAUSES	ACTION
Constant speed motor will not run or runs slowly and gets excessively hot.	Blown fuse. Faulty switch or relay or fault in relay circuit. Lead to motor fractured or disconnected. Dry joint in motor leads circuit. Shorted turn in field winding. Winding burnt out. Bent spool causing friction between spool and top plate.	(1) Check fuses. (2) Measure voltage at motor input. (3) Check continuity of windings, measure their resistance and compare it with the value quoted by maker.

INDICATION	PROBABLE CAUSES	ACTION
	Cooling fan fouling the stator. Reverse drive on trailing spool too great. Braking being applied to take-up spool during forward running. In the case of battery operated d.c. motors, the motor brushes may be worn or the commutator dirty. Another common cause of trouble is poor tension on the brushes due to weak or broken springs. Failure of phase splitting capacitor.	(4) Disconnect phase splitting capacitor and measure its resistance. If resistance appears high, apply charging voltage and check to see if it holds charge by disconnecting the supply and checking that a discharge current flows when the terminals are connected to a meter. A flash lamp battery may be used to supply the charging voltage. (5) Test to see if motor and spools are free to rotate. (6) Check switches for continuity. (7) Clean and adjust switch contacts.
Motor runs erratically.	Motor bearings dry. Oil on pressure roller, capstan or tape. Bent spool fouling deck. Excessive braking on trailing spool. Fluctuations in drive to take-up spool. Dirty switch contacts causing momentary breaks in motor circuit. Phase-splitting capacitor beginning to fail. Misplaced stator lamination fouling the rotor.	Check that tape, capstan and pressure roller are free from oil and that the tape is not stretched or curled. Check that spool is not bent and fouling the deck. Check that braking is not being inadvertently applied to the take-up spool during forward runs or that excessive braking is not being applied to the trailing spool.
Tape slips in drive; wow or flutter.	Dirty or oily capstan or accumulation of tape deposit on capstan roller. Bad splice in tape sticking in guides or on heads. Sticky tape due to joint adhesive spreading over several turns.	Clean drive capstan and pinch roller. Replace buckled spool. Check pinch roller tyre for concentricity.
	Buckled reel scraping the deck. Oversize tape. Bent motor spindle. Uneven rubber tyre on pinch roller. Worn bearings or motor fault.	Check that there is no excessive play in motor or capstan bearings and that the stator is not fouling the rotor.
Constant speed motor runs too fast.	One or more of the poles on the stator shorted.	Check continuity of windings.

How to Check Tape Speed

Make a clear mark on the back of the tape with a yellow china-graph pencil, or stick to the backing a small white patch only a fraction of an inch long. Set up the spool with this marked place several turns in on the left-hand spool. Start the machine and using a stop-watch, or a watch with a second hand note the exact time at which the marked piece of tape passes a certain place on the recorder. After a fixed time—say 10 seconds—stop the machine and mark the piece of tape then at the same fixed place. Carefully measure the length of tape between the two marks.

The correct length, if the speed is accurate, is given in the following table:

TAPE SPEED	LENGTH OF TAPE IF TIME IS 10 SECS.	30 SECS.
3·75 inches per sec.	37·5 inches	112·5 inches
7·5 " "	75 "	225 "
15 " "	150 "	450 "

The percentage difference in speed is given by

$$\frac{(\text{Difference between correct and measured length}) \times 100}{\text{Correct length}}$$

For example, if the correct length is 75 inches and the measured length is 78 inches, the tape is travelling $\frac{300}{75} = 4$ per cent. fast, and if the measured length is 72 inches the tape is running $\frac{300}{75} = 4$ per cent. slow.

12

PRINTING BETWEEN LAYERS

UNLESS certain precautions are taken there is a risk that when a recording is spooled up, the signal in one layer of the tape may print an attenuated copy of itself into adjacent layers. In really bad cases the print signals may penetrate up to five or more layers on either side of the original, giving a disconcerting pre-echo and post-echo effect. This trouble was quite unknown in the early magnetic recorders, because signals printed in this way are always quite low level (about 55dB with respect to the original) and, though they must have been there, their presence was masked by the rather high background noise, which was a feature of magnetic recording in those days.

When improvements were made in the recorders and the background noise was reduced, the print signals began to give trouble. Fortunately, the tape manufacturers have now so improved the tape coating (mainly by increasing the coercivity) that printing between layers is not the serious problem it used to be. The cause of the printing is illustrated in Fig. 100(a). The external lines of force,

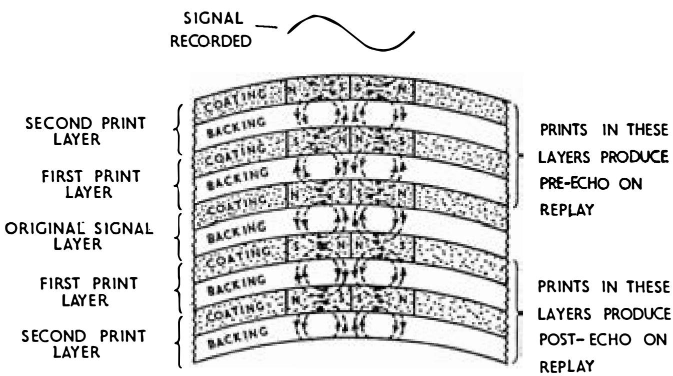


FIG. 100(a) THE FLUX DISTRIBUTION IN MAGNETIC LAYERS ADJACENT TO THE ORIGINAL SIGNAL LAYER.

due to the little magnets recorded in the tape coating, close through the coating in the adjacent layers and thus magnetize them. Once a print signal has been induced in this way, it gives rise to lines of force which close through the next layer and so on. In this way, the signal is printed from one layer to another in ever diminishing intensity.

It has been found that the level of the print signal depends upon a number of factors of which the principal are:

(*i*) The recorded intensity of the original signal (therefore take care never to over-record).
(*ii*) The coercivity of the tape coating.
(*iii*) The duration of storage.
(*iv*) The temperature during storage.
(*v*) The amount of stray magnetic field to which the tape is subjected during storage.

Precautions that must be taken to minimize printing between layers

To keep printing to a minimum, take care not to record at too high a level. Store the tape in a moderately cool place (too low a temperature may make the backing brittle) and do not put it near any electrical apparatus which may subject it to stray magnetic fields.

If the tape has only been stored for a short time, the print signal may be diminished by spooling backwards and forwards fairly rapidly. If it has been stored for any length of time (exceeding about 14 days) the only way to get rid of any print signal that may be present is to cut it out or to wipe the tape completely. When sending a tape through the post it should always be packed in a metal container in order to prevent the accidental application of stray fields which, if weak, will increase printing and, if strong, may actually erase the original recording.

13

TAPE JOINTING AND EDITING

Two pieces of magnetic tape are best joined in the following way. First trim the ends diagonally as shown in Fig. 101(a) and bring the diagonals together as shown in Fig. 101(b). A jig can be used with advantage to ensure that the diagonal cuts are at exactly the same angle. Another way is to overlap the two ends of the tape keeping the sides parallel and to cut diagonally across the overlap. (See Fig. 102).

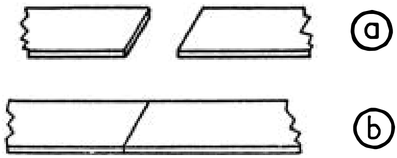


FIG. 101. A DIAGONAL JOIN MAKES THE JOINT MORE SILENT WHEN PASSING OVER THE REPLAY HEAD.

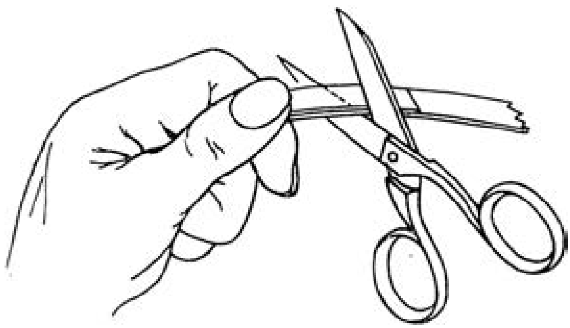
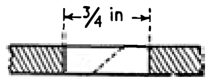


FIG. 102. IF YOU HAVE NO JIG, OVERLAP THE TWO ENDS AND CUT DIAGONALLY ACROSS THE OVERLAP AS SHOWN.

The uncoated side of the butted ends must be carefully cleaned with a dry cloth to ensure that they are free from greasy finger marks or crayon and then a patch of adhesive jointing tape* about $\frac{3}{4}$ inch long must be fixed to the uncoated side directly over the joint. (Fig. 103). Great care must be taken to remove any air bubbles

FIG. 103. ATTACH THE JOINTING PATCH TO THE SHINY (UNCOATED) SIDE OF THE BACKING.



beneath the patch by rubbing with the thumb nail; and to ensure that the ends of the tape are absolutely touching. It is better to have a fractional overlap than to allow part of the patch to show through on the coated side. Any part of the patch which shows at top and bottom of the tape on the coated side must be trimmed away. The danger is that some of the adhesive will spread to the next layer

* Always use tape with "hard" adhesive specially made for splicing. Ordinary cellulose tape will not do because the adhesive is soft and spreads under pressure to adjacent turns.

causing binding on replay. Many professionals advocate the application of a fine talcum powder after applying the patch, but this can cause noise on reproduction if the amount of powder used is excessive. Do not use carbon tetrachloride (C.T.C.) for cleaning up the joint as this is apt to attack the tape coating.

If a steel cutting tool (razor blade or scissors) is used, it must be demagnetized if a quiet joint is aimed at. Plastic and phosphor-bronze scissors are now obtainable, the latter being more effective than the former though of course more expensive. Tapes may also be joined by using a cement but as this method is not suitable for the amateur it will not be described here.

Editing

Provided that only one track is required, the tape may be edited by cutting and joining or by careful erasure.

The first requisite is to find the right place and mark it. This involves moving the tape by hand backwards and forwards across the reproducing head. As, in many home recorders, the reproducing amplifier is only switched into circuit when the selector is put in the reproducing position or the button is pressed for replay, special steps have to be taken to ensure that the tape is free to be moved and that it is engaged with the reproducing head whilst the editing is in progress. It is not possible to detail the required procedure for every type of recorder and only two representative cases will be quoted.

(1) THE FERROGRAPH

Put selector knob to REPLAY. This switches the reproducing head to the input of the amplifier and the loudspeaker to the output. Provided that the start lever is not pulled the pressure roller will not be pressed against the capstan and the tape can be wound backwards and forwards by hand.

(2) PHILIPS RECORDERGRAM TYPE AG8107

The selector knob must be turned to replay and the pressure roller disengaged from the capstan. One way of doing that is to drill a small hole in the top cover in such a place that a small screwdriver can be inserted to make contact with the arm holding the pressure roller. A better way is to design a small eccentric cam which when rotated will hold off the roller or allow it to engage at will. A switch will require to be inserted in the lead to the motor so that this may be switched off without de-energizing the amplifier. This model has now been superseded by Type EL3538.

To locate a particular space, say between two words, first replay the first word and as soon as it is heard press the stop button. Then wind the tape backwards and forwards by hand until the gap between the two words is clearly located. The speed of tape movement must be sufficiently fast to make the recorded sound audible but not so fast that it is unintelligible. Next move the tape till the first part of the unwanted word is heard (position 1, Fig. 104).

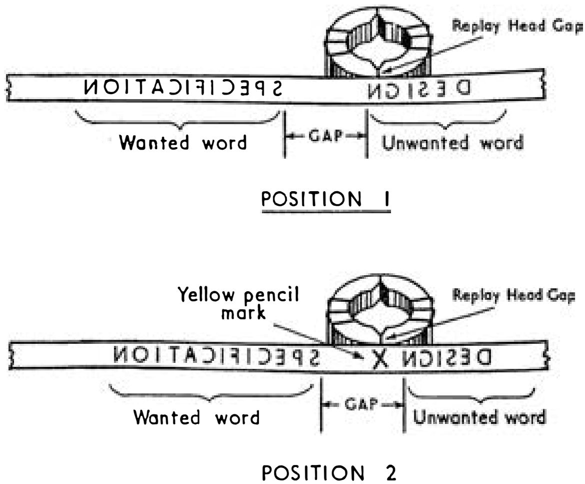


FIG. 104. TAPE EDITING CAN BE MOST EFFECTIVE BUT TAKES MUCH PRACTICE. THE SKETCHES SHOW HOW THE CORRECT CUTTING PLACE MUST BE LOCATED. POINT *X*—THE CUTTING POINT—LIES IN THE GAP NEXT TO THE UNWANTED MATERIAL.

Now move the tape in the reverse direction until the gap is reached and mark the back of the tape with a yellow chinagraph pencil at a point in the gap as near as possible to the unwanted word.

With some machines the head assembly is completely boxed and it will not be possible to mark the spot directly. A cross must be made on the tape at the point where it enters or leaves the assembly. Careful measurement and testing will tell you how far the mark you have made is away from the required spot (see Fig. 105), which can thus be accurately marked on all future occasions.

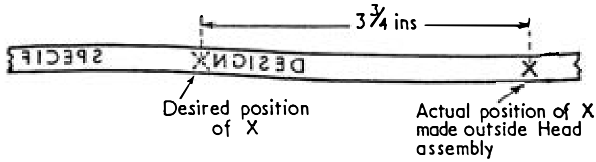


FIG. 105. HOW TO LOCATE POINT *X* WHEN THE HEADS ARE TOTALLY ENCLOSED IN THE HEAD UNIT.

Once this point is marked it will be one cutting point. Another cutting point, at the other end of the unwanted matter, is similarly located. If on joining and testing, the space thus left appears too long, it is a simple matter to cut off some tape on either side of the join and to rejoin. A similar procedure may be used to remove unwanted clicks and bangs.

Sometimes very close editing is required, as when an unnecessary aspirate has to be removed (*e.g.* removing the " h " from " hextraordinary "). This can best be done by a process of attrition. First locate the word and then remove about $\frac{1}{4}''$ of tape at the beginning of it. Rejoin and test. If the unwanted aspirate still sounds, remove another $\frac{1}{4}''$ and so on until the aspirate is removed completely.

The Wiping Fade

It is sometimes required to reduce the length of a music recording. This necessitates a gradual fade-out of the music. In some recorders provision is made for varying the erase current. When that is so, the wiping fade may be achieved by reducing the erase current to zero, recording with recorder input to zero and gradually re-introducing the erase current. The bias current into the record head provides about 6dB of erasure and the bias current must, therefore, be reduced to zero whilst the wiping fade is being made; or alternatively the tape must be held clear of the recording head though firmly against the erase head during the wiping process. Do not forget to reset the bias current to the normal value if it has been reduced to zero whilst making a wiping fade.

A simpler way is to hold the tape away from both the record and erase heads to begin with and then to allow the tape gradually to come into contact with the heads. This method will present some difficulty in the case of machines provided with pressure pads but such machines very often have the facility mentioned above for fading out the erase current.

Interpolating Announcements into a Recording already made

It is sometimes required to interject announcements into a tape already recorded. The most satisfactory method is to record the announcements on a separate tape so that each may be joined into the recording at the appropriate place. Nevertheless, sometimes it is more effective to record the announcement on top of the previous recording. This can readily be done on those recorders which make provision for removing the erase current from the erase head. The bias into the record head will reduce the previous recording level by 6dB so that the announcement will be recorded at 6dB higher level than the background. If the background sounds require to be attenuated to a greater extent than 6dB, they may be passed over the erase head when the latter is energized with a smaller amount of current than that required for complete erasure. To do this accurately takes a good deal of practice, so it is as well to try it out on a scrap recording first of all.

On some recorders it is not possible to diminish the erase current whilst the recording is being made. Announcements can, however, still be superimposed as described above if the tape is held away from the erase head by, for example, a piece of cardboard.

Cleaning up a Speech

By careful editing it is possible to reduce the duration of a speech without removing a single word. First, all "er's" and "ah's" are cut out and then any space between words exceeding one inch in length is reduced to that length. The resultant speech is crisp, devoid of hesitation and generally a great improvement on the original.

Marking a Given Place

To mark a given place in the recording insert a small piece of numbered paper between turns, or stick a numbered patch on the back of the tape.

To mark the beginning of a recording insert a white leader tape. To mark the end insert a red trailer tape.

Index Counters

Many home recorders now have built-in index counters. A counter makes it possible to note the location of any particular section of a tape and thus makes it easier to find, but beware of trusting the index counter too implicitly when editing because slip makes the *exact* location of a given place to within an inch or so a matter of chance.

Trick Editing

Great fun can be had by trick editing. By a process of attrition, for instance, the aspirates can be cut from some words and, if you are really clever, you can then add them to others. Imagine the chagrin of a rather pompous individual who records, " My name is Edward Henry . . .", if, when the tape is replayed, he hears himself say (all too distinctly): " My name is Hedward 'Enery . . ." Once you have tried your hand at trick editing, you will find it fascinating. Although the effects obtained are always amusing and sometimes astounding, your greatest pleasure will come from the pride of achievement, because editing of this kind is always extremely difficult to do.

Reversing Speech and Music

It is always good fun to hear the sound of speech and music played backwards and if you can do it on your machine it is interesting and instructive to listen to the reversed sound of various instruments.

In machines with whole-track heads the two reels may be transposed in order to reverse the track, but on machines with narrow-track heads (most home recorders come in this category) reversing the tape in this way is of no use as the other track is then automatically brought against the heads. It is, however, possible to get a backward replay on many such machines by lacing up the tape so that it runs round both the constant speed drive capstan and the pressure roller in the opposite way to normal. This is illustrated in Fig. 105(a) overleaf.

Whether this arrangement is successful or not depends upon the amount of counter drive applied to the left-hand spool during normal forward running, but it is worth trying.

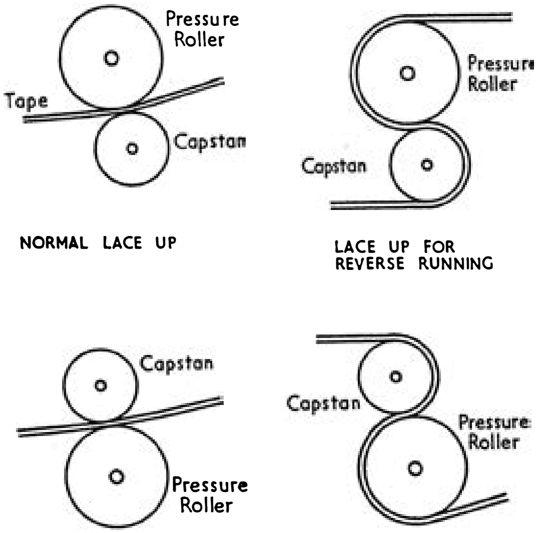


FIG. 105(a) METHOD OF TAPE LACE-UP, IF IT IS DESIRED TO REPLAY BACKWARDS.

Mixing Live and Recorded Material

If another tape recorder is available, it is amusing to try re-recording the output of one of the recorders whilst mixing in other sounds picked up by a microphone. A singer, for instance, may record a song and then sing a duet with his reproduced voice. The re-recording when reproduced will sound like a duet though both voices on the record are those of the same singer. If the singer again chimes in and is again recorded, the duet will become a trio and so on. Or, a violinist may record the first fiddle part and then when the recording is reproduced he may add the second fiddle part, and so on.

Each first recording must be preceded by a handclap to act as a starting cue for succeeding recordings. Best results are obtained when the output of the reproducer is fed direct to the input of the recorder whilst the performer listens to his previous recording on headphones.

14

COPYRIGHT AND LEGAL ASPECTS OF HOME RECORDING

IN view of the fact that users of home recorders may unwittingly break the law when making recordings and reproductions and thereby lay themselves open to penalties and/or costs, they would be well advised to familiarize themselves with the legal enactments which deal with copyright and the protection of artistes. Copies of the relevant acts, the COPYRIGHT ACT 1956 and the DRAMATIC AND MUSICAL PERFORMERS PROTECTION ACT 1925, may be obtained from H.M. Stationery Office and should be studied by all home recordists.

The Mechanical-Copyright Protection Society Ltd. which acts on behalf of individual copyright owners in the collection and distribution of mechanical fees and royalties and takes steps to safeguard the interests of its members in matters affecting their mechanical copyrights, have very kindly given the following brief summary of the application of the COPYRIGHT ACT 1956 in regard to home recording of copyright music.

1. *It is an infringement of copyright to make a recording of copyright musical work or any other copyright material for any purpose whatsoever without the consent of the owner or his agent.*

2. *It is only permissible to record the programmes of BBC and ITA for private use but this does not extend to any copyright material included and such recordings would infringe the copyright unless prior permission had been obtained from the owner or his agent.*

3. *If permission is given to a home recordist to make a record of copyright music, it is usual to restrict this to private and domestic use only and permission for public performance or public play-back is withheld except in very special circumstances.*

4. *It is not permissible to record an amateur dramatic performance (or in fact any other public performance involving copyright material) and a clause forbidding this is usually inserted in the licence given by the copyright owner to the amateur dramatic society whereby they are authorized to perform the work in public.*

5. *It is an offence under the provisions of the* DRAMATIC AND MUSICAL PERFORMERS PROTECTION ACT 1925, *as amended by the* COPYRIGHT ACT 1956, *to record the performance of an artiste without his written consent although it can be an answer to a charge under this Act that the recording was made solely for private use. Here again any copyright material involved in the performance would have to be taken into consideration before a recording could be made.*

In view of the foregoing it is clear that the private individual who makes an unauthorized recording of a musical work *even for his own private purpose* is committing an infringement of copyright.

An interesting booklet *Copyright in Sound Recordings* by Peter Ford, LL.B. (Hons), F.R.S.A., of Gray's Inn, Barrister-at-Law, is published by The British Sound Recording Association, 3 Coombe Gardens, New Malden, Surrey, price 1*s.* 3*d.* post free.

The address of the Mechanical-Copyright Protection Society Ltd. is 29 Maddox Street, London, W.1.

Some of the many other bodies interested in protecting the rights of authors, artistes and composers, etc. are:

The Performing Right Society, Ltd.,
33 Margaret Street, London, W.1.
British Actors' Equity Association,
8 Harley Street, London, W.1.
The Musicians' Union,
29 Catherine Place, London, S.W.1.
The Institute of Journalists,
2 Tudor Street, London, E.C.4.
The Imperial Arts League,
Royal Institute Galleries,
195 Piccadilly, London, W.1.
The British Broadcasting Corporation,
Broadcasting House, London, W.1.
The Independent Television Authority,
14 Princes Gate, London, S.W.7.
The League of Dramatists,
84 Drayton Gardens, London, S.W.10.
The Society of Authors, Playwrights and Composers Inc.,
84 Drayton Gardens, London, S.W.10.

APPENDIX A

Elementary Trigonometry

In order to understand the meaning of the term **sine wave,** it is necessary to know something about elementary trigonometry—the science of angles.

The difference in direction between two straight lines is called

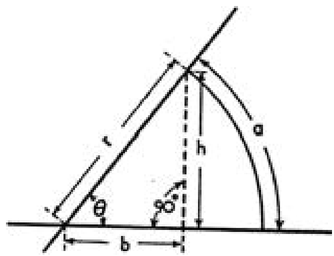


FIG. 106. IN THIS CONSTRUCTION a IS THE ARC OF A CIRCLE OF RADIUS r.

an angle, θ (Greek letter *theta*). In the diagram (Fig. 106) certain ratios of the lengths shown are constants for the given angle. These ratios have the following names:

$\dfrac{a}{r}$ is called the size of the angle in radians $= \theta$

$\dfrac{h}{r}$ is called the sine of the angle θ (generally written $\sin\,\theta$)

$\dfrac{b}{r}$ is called the cosine of the angle θ (generally written $\cos\,\theta$)

$\dfrac{h}{b}$ is called the tangent of the angle (generally written $\tan\,\theta$)

In one complete revolution of r, the length of the arc $= 2\pi r$, *i.e.* there are $\dfrac{2\pi r}{r} = 2\pi$ radians in one complete rotation. (As there are also 360°, 2π radians $= 360°$).

If we plot a graph of $\sin\,\theta$ against θ, we obtain the shape shown in Fig. 107(a). This is called a **sine wave.** Any graph of this shape is called **sinusoidal.** Compare Fig. 107(a) with that of (b)—a graph of the pressure variations in a sound wave. They are of the same shape because the pressure at any point in the transmission path of a free progressive tone varies sinusoidally with time. The time taken for the pressure to pass through one complete set of variations is called the period, T, or the time for one cycle. The number of cycles per second is called the frequency, $f = \dfrac{1}{T}$. The instantaneous

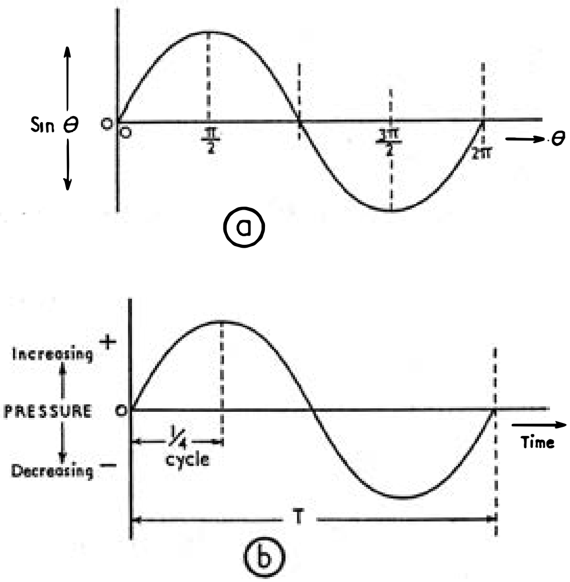


FIG. 107. GRAPH OF A SINE WAVE (a) COMPARED WITH THE GRAPH OF THE PRESSURE VARIATIONS IN THE PATH OF A PURE TONE (b).

pressure, p, at a time t seconds after the start of a cycle, is always equal to the maximum pressure multiplied by sin θ, *viz*

$$P_{inst} = P_{max} \sin \theta$$

where $\dfrac{\theta}{2\pi} = \dfrac{t}{T}$, *i.e.* $\theta = 2\pi \dfrac{t}{T} = 2\pi ft$ (sometimes written ωt): see Fig. 108.

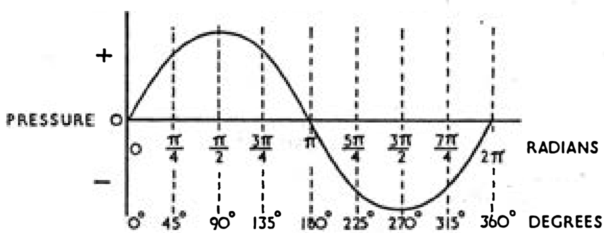


FIG. 108. A GRAPH OF PRESSURE AGAINST RADIANS OR DEGREES.

A good sine wave can be drawn with ordinates spaced 30° apart on the horizontal axis. Thus, from the tables, sin 0° = 0, sin 30° = 0·5, sin 60° = 0·866, sin 90° = 1, sin 120° = 0·866, sin 150° = 0·5, etc. (See Fig. 109).

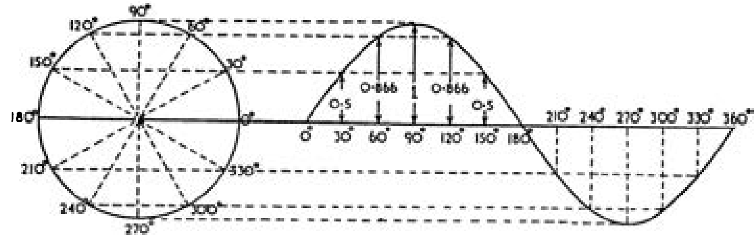


FIG. 109. CONSTRUCTION OF A SINE WAVE.

APPENDIX B

The Decibel (dB)

This is a unit used to express the ratio of two amounts of power, pressure, velocity, current or voltage.

If P_1 and P_2 designate two amounts of power, then the ratio of P_2 to P_1, in dB, is given by

$$N(\text{dB}) = 10 \log_{10} \frac{P_2}{P_1}$$

and if currents, voltages, pressures or velocities are compared the ratio becomes

$$N(\text{dB}) = 20 \log_{10} \frac{X_2}{X_1}$$

where X_1 and X_2 are the two amounts compared. As an example, if doubling the frequency (*i.e.* increasing it by one octave) causes a doubling of the output voltage, the ratio becomes $20 \log_{10} 2 = 20 \times 0{\cdot}301 = 6\text{dB}$ approx.

The Phon

At 1,000c/s there is a separation of 140dB between the threshold of hearing and the threshold of pain (see Fig. 110). It will be seen that if the power in the air wave is increased the loudness increases —but not in step because the ear has a logarithmic response to sound. But the decibel cannot be used as a scale of loudness because the human ear is not equally responsive to all frequencies. A related unit, the **phon** is used to express loudness in terms of the equivalent loudness of a standard reference tone: the loudness of a sound, in phons, is numerically equal to the sound intensity in decibels of an equally loud 1,000 cycles per second pure tone. Sound intensity is the power per unit area of the sound conducting medium, measured in watts per square metre.

SOUND POWER RATIOS	LOUDNESS AT 1,000c/s IN dB	SOUND PRESSURE RATIOS	
10^{14}	140	10^{7}	THRESHOLD OF PAIN
10^{13}	130	$10^{6\cdot5}$	
10^{12}	120	10^{6}	
10^{11}	110	$10^{5\cdot5}$	
10^{10}	100	10^{5}	
10^{9}	90	$10^{4\cdot5}$	
10^{8}	80	10^{4}	
10^{7}	70	$10^{3\cdot5}$	
10^{6}	60	10^{3}	
10^{5}	50	$10^{2\cdot5}$	
10^{4}	40	10^{2}	
10^{3}	30	$10^{1\cdot5}$	
10^{2}	20	10	
10	10	$10^{0\cdot5}$	
1	0	1	THRESHOLD OF HEARING

FIG. 110. AT 1,000c/s THERE IS A SEPARATION OF 140dB BETWEEN THE THRESHOLD OF HEARING AND THE THRESHOLD OF PAIN. IT WILL BE SEEN THAT IF THE POWER IN THE AIR WAVE IS INCREASED, THE LOUDNESS INCREASES—BUT NOT IN STEP BECAUSE THE EAR HAS A LOGARITHMIC RESPONSE TO SOUND.

APPENDIX C

Thévenin's Theorem

Any linear network, which contains one or more sources of e.m.f. and has two output terminals, is equivalent to a generator having an e.m.f. equal to the open circuit voltage at the output terminals and an internal impedance equal to the impedance between the output terminals. For example, suppose the network consists of a generator of e.m.f. = 100V, of internal impedance, $R_g = 100\Omega$, connected to terminals A and B which are also bridged by a 1,000Ω resistor (R_1) (Fig. 111). The connection of a load resistance

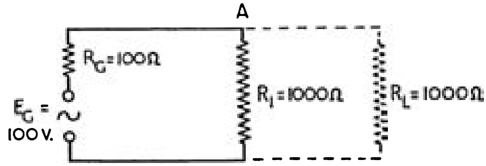


FIG. 111. NETWORK TO ILLUSTRATE THÉVENIN'S THEOREM.

$R_L = 1{,}000\Omega$ to the terminals completes a circuit equivalent to Fig. 112 in which the open circuit voltage across AB is applied to R_G

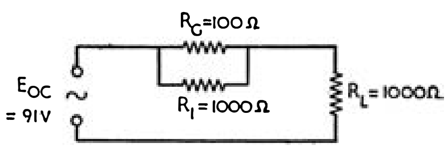


FIG. 112. THÉVENIN EQUIVALENT OF CIRCUIT FIG. 111.

in parallel with R_1 and in series with R_L. This circuit simplifies to Fig. 113 because the total resistance of two resistors in parallel is

FIG. 113. SIMPLIFICATION OF THÉVENIN EQUIVALENT OF FIG. 111.

equal to the product of the two resistances divided by their sum

$$i.e. \quad \frac{R_G.R_1}{R_G + R_1} = \frac{10^2.10^3}{10^2 + 10^3} = \frac{10^3}{11} = 91 = R_G'$$

The current in the load is therefore equal to

$$\frac{E_{OC}}{R_G' + R_L} = \frac{91}{1091} = 0{\cdot}0834 \text{ ampere.}$$

APPENDIX D

Time Constants

(1) CAPACITATIVE CIRCUITS

If a capacitor is charged through a resistor, the potential difference built up between the plates rises at a rate determined by the respective magnitudes of the capacitance and the resistance. The rate of discharge is similarly controlled. Theoretically, an infinite time is required to bring the capacitor to full charge (*i.e.* to make the p.d. across the plates equal to the applied voltage) or to discharge the capacitor completely. When the charging current flows, the electron drift builds up a back e.m.f. on the plates which tends to oppose the applied voltage. The longer the current flows the greater the back e.m.f. and the smaller the net e.m.f. acting in the circuit and hence the smaller the current and the smaller the successive rises in p.d. across the plates. Conversely, on discharge the electron drift reduces the p.d. between the plates and automatically cuts down the discharge current, making the successive drops in p.d. smaller and smaller.

Curves of rate of change of p.d. with time are shown in Fig. 114.

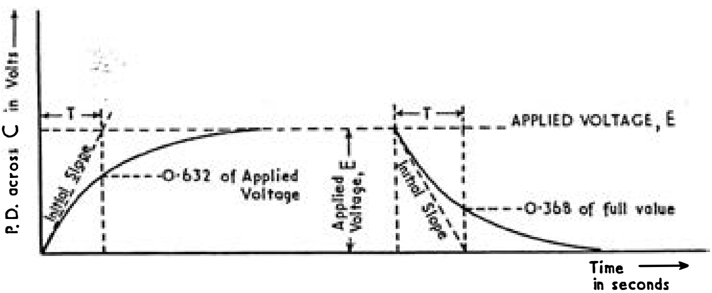


FIG. 114. GROWTH AND DECAY OF P.D. ACROSS A CAPACITOR IN SERIES WITH A RESISTANCE

Because the curves are asymtotic, an arbitrary basis must be established for comparison of the rapidity with which the circuit conditions become adjusted to sudden changes. The basis ordinarily used for simple circuits consisting of capacitance and resistance is the time that it takes the p.d. to rise to 0·632 of its final value or to fall to 0·368 of its initial magnitude. This is called the time constant, T, of the circuit and, in seconds, it is numerically equal to the product of C and R, *i.e.* $T = CR$ seconds where C is the capacitance in farads and R is the resistance in ohms.

(2) INDUCTIVE CIRCUITS

When a voltage is applied to a circuit containing inductance and resistance, the current builds up rapidly at first and then more and more slowly because as the current rises, the lines of magnetic force due to the current cut the adjacent turns of the coil and produce a back e.m.f. which opposes the applied e.m.f. This back e.m.f. is proportional to the rate of change of the current and the rate of change is great when the voltage is first applied. The back e.m.f. therefore nearly equals the applied e.m.f. and the net e.m.f. acting in the circuit is small so that the current is small. As the current gradually grows towards the $\frac{E}{R}$ value, the rate of change of current diminishes, the back e.m.f. falls and the net driving voltage increases.

Similarly, when the applied voltage is removed the current tends to fall rapidly. The lines of force due to the diminishing current cut the adjacent turns in the opposite direction and produce a back e.m.f. acting in the same direction as the originally applied e.m.f. The current, therefore, tends to fall more slowly than would otherwise be the case. The time constant T of a circuit containing inductance and resistance is the length of time in which the current in the coil rises to 0·632 of the $\frac{E}{R}$ value or, alternatively, it is the time in which it falls to 0·368 of that value after the voltage is removed. (See Fig. 115).

$T = \frac{L}{R}$ seconds, where L is the inductance in henrys and R is the resistance in ohms.

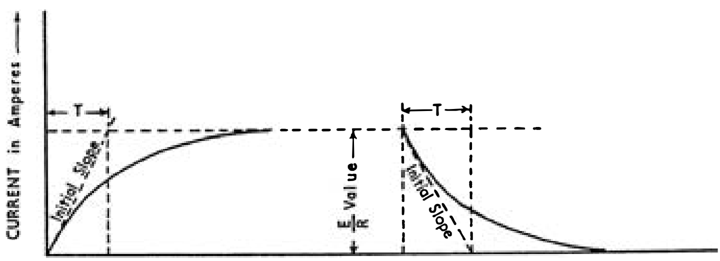


FIG. 115. GROWTH AND DECAY OF CURRENT THROUGH AN INDUCTOR IN SERIES WITH A RESISTANCE.

APPENDIX E

TERMS USED IN MAGNETIC RECORDING

TERM AND DEFINITION	C.G.S. ELECTRO-MAGNETIC UNIT	RATIONALIZED M.K.S. UNIT
COERCIVE FORCE (H_c) The demagnetizing force required to reduce the magnetic flux density in a substance from the remanent value to zero. It depends upon the initial magnetization.	**OERSTED** (1 oersted = 1 dyne per unit pole).	**AMPERE-TURN PER METRE** (1 ampere-turn per metre = $\frac{4\pi}{10^3}$ oersteds).
COERCIVITY* The value of coercive force when the initial magnetization has the saturation value for the substance.	**OERSTED**	**AMPERE-TURN PER METRE**
INTENSITY OF MAGNETIZATION (J) The strength of magnetization measured by the flux per square centimetre or the quantity of webers per square metre. This must not be confused with MAGNETIC FLUX DENSITY (B), with which it is related by the formulae: (*i*) (in C.G.S. units) $B = H + 4\pi J$ (*ii*) (in M.K.S. units) $B = \mu_0 H + J$	$\frac{\textbf{GAUSS}}{4\pi}$ (1 gauss = 1 line of force per square centimetre).	**WEBER PER SQUARE METRE** (1 weber per square metre = 10^4 lines per square centimetre = 10^4 gauss).
MAGNETIC FIELD* The space in the neighbourhood of a current-bearing conductor or of a permanent magnet throughout which the forces due to the current or magnet can be detected.		
MAGNETIC FLUX (ϕ) Collectively, the lines indicating the direction along which the forces in a magnetic field act. Individually these lines are called magnetic lines of force.	**MAXWELL** (1 maxwell = 1 line of force. It is that flux the removal of which, from a circuit of unit resistance causes one electromagnetic unit of electricity to flow).	**WEBER** Flux changing at 1 weber/sec. induces an e.m.f. of 1 volt in a single conductor linked with the field. 1 weber = 10^8 maxwells.

* Definitions of Terms marked * are abstracted by permission from *British Standard* 205, *Part* 1. Official copies of the publication may be purchased, price 6/–, from BRITISH STANDARDS INSTITUTION, 2 Park Street, London, W.1.

TERM AND DEFINITION	C.G.S. ELECTRO-MAGNETIC UNIT	RATIONALIZED M.K.S. UNIT
MAGNETIC FLUX DENSITY (B) The magnetic flux per unit area. $B = \dfrac{\phi}{A}$ where A is cross sectional area. B_0 = Flux Density in air.	**GAUSS** (1 gauss = 1 maxwell per square centimetre).	**WEBER PER SQUARE METRE**
MAGNETIZING FORCE (or Field Strength) (H) In the C.G.S. electromagnetic system magnetizing force is the force in dynes which would act on a unit north pole placed in a given field. In the M.K.S. system, field strength is simply measured in ampere-turns per metre. Magnetizing force and flux density are related by the E.M. formula $B = \mu H$ where μ is the absolute permeability of the medium. The corresponding M.K.S. formula is $B = \mu_0 \mu_{rel} H$. (For definitions of μ_0 and μ_{rel}, see PERMEABILITY).	**OERSTED** (1 oersted = 1 dyne per unit pole.) In a solenoid $H = \dfrac{4\pi IN}{10L}$ where I is the current in amps, H is the number of turns and L is the length of the solenoid in centimetres.	**AMPERE-TURNS PER METRE** In a solenoid $H = \dfrac{IN}{L}$ where I is the current in amperes, N is the number of turns and L is the length of the solenoid in metres. (1 ampere-turn per metre = $\dfrac{4\pi}{10^3}$ oersteds).
MAGNETOMOTIVE FORCE ($M.M.F.$) The product of the magnetizing force and the length of the magnetic circuit. It is also equal to the product of the flux and the reluctance, *i.e.* $MMF = HL = \phi S$ where S is the reluctance. In the E.M. system it is the work done in moving a unit north pole around the circuit. In the M.K.S. system it is the product of the current in amperes and the number of turns.	**GILBERT** (1 gilbert = $\dfrac{10}{4\pi}$ ampere-turns).	**AMPERE-TURN** (1 ampere-turn = $\dfrac{4\pi}{10}$ gilberts).
MAGNETIC HYSTERESIS* The phenomenon by which the intensity of magnetization of some materials depends not only on the present magnetizing force but also on the previous magnetic state. It causes a dissipation of energy (the hysteresis loss) when the material is subjected to cyclic magnetization (as in magnetic sound recording).		

* See footnote opposite

TERM AND DEFINITION	C.G.S. ELECTRO-MAGNETIC UNIT	RATIONALIZED M.K.S. UNIT
MAGNETIC HYSTERESIS LOOP* (B/H loop) The closed figure formed by plotting the values of magnetic flux density (B) against the magnetizing force (H) when the latter is taken through a complete cycle. The hysteresis loss is proportional to the area of this loop.		
PERMEABILITY* (Absolute) (μ) The permeability of a material is the ratio of the magnetic flux density to the magnetizing force producing it: $\frac{B}{H}$. The symbol for the absolute permeability of free space is μ_o. In the E.M. system $\mu_o = 1$. In the M.K.S. system $\mu_o = \frac{4\pi}{10^7}$ henrys per metre and $\mu_o = \frac{B_o}{H}$.		**HENRY PER METRE** [μ (M.K.S.) = $\frac{4\pi}{10^7}$ μ (C.G.S.)]
PERMEABILITY (Incremental) (μ_i) The ratio of a small change in magnetic flux density to the small change in magnetizing force producing it: $\frac{\triangle B}{\triangle H}$.		**HENRY PER METRE**
PERMEABILITY (Relative) (μ_{rel}) The ratio of the magnetic flux density produced in the medium to that produced in a vacuum by the same magnetizing force. $\mu_{rel} = \frac{\mu}{\mu_o}$		(μ_{rel} is the same value in both M.K.S. and C.G.S. systems).

* See footnote on page 116.

TERM AND DEFINITION	C.G.S. ELECTRO-MAGNETIC UNIT	RATIONALIZED M.K.S. UNIT
RELUCTANCE (S) The ratio of the magnetomotive force acting in a magnetic circuit to the resulting magnetic flux. i.e. $S = \dfrac{MMF}{\phi}$. The reluctance of a magnetic circuit $= \dfrac{l}{\mu A}$ where l is the length, A the cross-sectional area and μ the absolute permeability of the medium.	**GILBERT PER MAXWELL**	**AMPERE-TURN PER WEBER** (1 ampere-turn per weber = $\dfrac{4\pi}{10^9}$ gilberts per maxwell).
REMANENCE (B_r (Sat)) The remaining magnetic flux density in a medium after a saturating magnetizing force has been removed.	**GAUSS**	**WEBER/METRE^2** 1 weber/metre^2 = 10^4 gauss.
REMANENT FLUX DENSITY (B_r) The magnetic flux density remaining in a substance when, after any value of initial magnetization, the magnetizing force is reduced to zero.	**GAUSS**	**WEBER/METRE^2**
REMANENT POINT (R) The point on the hysteresis loop to which the magnetic flux density returns after a saturating magnetizing force has been removed.		

GLOSSARY

ACETATE FILM—The super-smooth transparent plastic film which forms the base for approximately 90 per cent. of the magnetic recording tape made in the world to-day.

AUDIBLE TONES—Sounds with wave frequencies which the average human can hear and which range from 30 to 15,000 cycles per second.

AZIMUTH ADJUSTMENT—Method of adjusting the gap on the record/playback head to a standard alignment in order to facilitate the interchangeability of recordings between machines.

BIAS—A high frequency alternating current fed into the recording circuit to eliminate distortion.

BINAURAL RECORDER—A tape recorder which employs two separate recording channels or systems, each with its own microphone, amplifier, recording and playback heads and headphones. Recordings using both systems are made simultaneously on a single magnetic tape on two parallel tracks, which, upon playback, reproduce the original sound with depth and realism unequalled by any other recording method. Use of headphones for listening is necessary for true binaural effect.

BULK ERASER—An a.c. mains device used to erase an entire reel of magnetic tape instantaneously without running it through a recorder. It uses a strong magnetic field which neutralizes the magnetic patterns on the tape.

CAPSTAN—The spindle or shaft, often the motor shaft itself, which rotates against the tape, pulling it along at a constant speed on recording and playback.

C.C.I.R.—Comité Consultatif International des Radiocommunications (International Radio Consultative Committee). This body specifies standard replay chains for various tape speeds. Recordings for programme interchange should give a response within certain recommended tolerances when reproduced on a standard replay chain.

CYCLES PER SECOND—The unit for measuring the frequency, or " pitch " of any sound. Abbreviated cps. or c/s.

DECIBEL—Abbreviated " dB ", it is a relative measure of sound intensity or " volume." It expresses the ratio of one sound intensity to another. One dB is the smallest change in sound volume that the human ear can detect.

DECK—See MOTOR BOARD.

DISTORTION—Any difference between the original sound and that reproduced by a recording machine. Distortion takes on many forms, and although it can never be completely eliminated, it can be reduced to a minimum in a good recording and reproducing system. Tape offers the maximum potential in distortion-free recording.

DUAL TRACK RECORDER—Usually a tape recorder with a recording head that covers half of the tape width, making it possible to record one track, then turn the reels over and record a second track in the opposite direction. Sometimes called a " half-track recorder." On some modern machines a double set of heads are incorporated, avoiding the need for reels to be turned over.

DUPE—Sometimes called a " dub," or " dubbing." A copy of a tape recording made by recording on one machine what another machine is playing. Tape recordings are easy to duplicate simply by re-recording, and there is a minimum loss in quality from the original to the copy.

DYNAMIC RANGE—The ratio between the softest and loudest sounds a tape recorder or other device can reproduce, without undesirable distortion. Usually measured in dBs.

EDITING—Selecting certain sections of a tape recording, or of a number of different tape recordings, then splicing them together in the desired sequence. Magnetic tape is unsurpassed for editing purposes, since it can be easily cut and spliced.

ERASURE—Neutralizing the magnetic pattern on tape by placing it in a strong magnetic field, thereby removing the recorded sound from the tape. An ERASE head on the tape recorder does this automatically to any sound previously recorded on the tape just before the tape reaches the RECORD head. A permanent magnet can also be used to erase magnetic tape.

EQUALIZATION—Either boosting or decreasing the intensity of the low, middle or high frequencies in the recording or playback amplifier or both. This compensation is made automatically by the recorder and serves to correct any deficiences in the recording system and to increase the signal-to-noise ratio. The standard equalization adopted by the U.K. is according to the recommendation of C.C.I.R. (International Radio Consultative Committee).

FLAT RESPONSE—The ability of a sound system to reproduce all tones, low and high, in their proper proportion—amplified to the same extent—within the specified frequency range.

FLUTTER—Very short, rapid variations in tape speed causing similar variations in sound volume and pitch, not present in the original sound. A form of distortion.

FREQUENCY RANGE—The range between the highest and lowest pitched sounds which a tape recorder or other sound system can reproduce at a usable output, or volume, level.

FREQUENCY RESPONSE—The output level of a recorder or sound system over a given range of frequencies. A more specific term than FREQUENCY RANGE. Usually in the form of a curve plotted on a chart.

GAIN—The ratio between the input level and the output level of a piece of sound equipment. Gain is increased by means of an amplifier.

GAP—The tiny distance between the poles of the recording head, measured in mils (0·001 in.). The head gap of home recorders may range from 1 mil down to ¼ mil The smaller the gap, the higher the frequency response of the tape recorder can be.

HEAD—The ring-shaped electro-magnet across which the tape is drawn, and which magnetizes the iron oxide-coated tape in a series of patterns. Most tape recorders employ a combination record-playback head and also an erase head. Some professional machines also employ a monitor head for listening to the recorded sound a split second after it has been put on the tape.

INDEX COUNTER—A counter which makes it possible to note the location of any particular section of a tape, thereby making it easier to find. Many late model tape recorders feature built-in index counters.

INSTRUMENTATION TAPE—Magnetic tape which has been specially manufactured under extra-carefully controlled conditions. For use where extreme uniformity and accuracy is demanded, such as in tellemetery, data recording, computers, measuring equipment, automatic machine control, etc.

INPUT—An electrical voltage fed into an amplifier.

IPS—See TAPE SPEED.

LEADER AND TIMING TAPE—Special, tough, non-magnetic tape which can be spliced to either end of a tape to prevent damage or breaking off of the magnetic tape ends and possible loss of part of the recorded material. Used as a timing tape, it can be spliced between musical selections on a tape providing a pause of a given number of seconds, depending on the tape speed.

LEVEL INDICATOR—A device on the tape recorder to indicate the level at which the recording is being made, and which serves as a warning against under-recording or over-recording. It may be a neon bulb, a MAGIC EYE, or a VU METER (*q.v.*).

MAGNETIC TAPE—A high-quality plastic tape which has been precision-coated by the manufacturer with a layer of magnetizable iron oxide particles. The result is a recording medium that is subject to virtually no wear, can be erased and re-used, and offers the highest fidelity of reproduction possible to-day.

MOTOR BOARD—Also called TAPE TRANSPORT MECHANISM. The platform, or assembly, of a tape recorder on which the motor (or motors), the reels, the heads and the controls are mounted. It includes those parts of the recorder other than the amplifier, pre-amplifier, loudspeaker and case.

MAGIC EYE—An electronic indicator which, by a change of shadow angle, visually indicates the peak record level.

OUTPUT—An electrical voltage coming from an amplifier and normally fed into a loudspeaker.

OXIDE—Microscopically small particles of ferric oxide dispersed in a liquid binder and coated on a tape backing. These oxides are magnetically " hard " *i.e.*, once magnetized they remain magnetized permanently, unless they are demagnetized by exposure to a strong magnetic field.

POLYESTER BASE—Plastic film used for magnetic tape. It is very tough and flexible and has outstanding mechanical properties over a wide range of temperatures and, because of its extremely low water absorption, it retains these properties at high humidities and is highly resistant to fungi and bacteria. Although thinner than standard tape it is as strong and enables 50 per cent. more tape to be wound on a spool.

POWER AMPLIFIER—An amplifier designed to operate a loudspeaker.

PRE-AMPLIFIER—An amplifier that raises extremely weak signal levels such as those from a microphone, magnetic playback head, or a gramophone pick-up to a level usable by the power amplifier. Some tape recorders combine the pre-amplifier and the power amplifier. Others, especially the tape recorders designed for use in high fidelity music systems, may feature a separate pre-amplifier. In such cases the pre-amplifier includes an equalization circuit. In addition, the bias oscillator (necessary to record on tape) is often mounted in a unit with the pre-amplifier.

PRESSURE PADS—Felt pads mounted on arms which hold the magnetic tape in close contact with the heads on some machines.

PRESSURE ROLLER—Also called " capstan idler." A rubber roller which holds the magnetic tape tight against the capstan by means of spring pressure, to ensure constant tape speed and prevent slippage.

PRINT THROUGH—Transfer of the magnetic field from layer to layer of tape on the reel. Virtually non-existent in high quality magnetic tape to-day.

P.V.C. (Pol-vynil-chloride) BASE—Plastic film base which is tensilized or stretched especially for making magnetic tape. It will not easily break and is suitable for long-term storage.

RAW TAPE—A term sometimes used to describe tape that has not been recorded. Also called " virgin " tape.

RECORDED TAPE—A recording on tape that is commercially available. Also called a " pre-recorded " tape or, in the case of music, " music on tape." However, any tape that has been recorded, whether commercially recorded or not, is called a recorded tape.

RECORDING NOISE—Noise induced by the amplifier and other components of the recorder. High quality magnetic tape itself is inherently noise-free.

SELF-POWERED RECORDER—Tape recorder containing its own power supply, either a combination of wet and dry cells to power the unit, or dry cells in conjunction with a spring-driven motor.

SIGNAL-TO-NOISE RATIO—The ratio between the loudest, undistorted tone recorded and reproduced by a recorder and the noise induced by the recording system itself. Normally measured in decibels.

SINGLE TRACK RECORDER—A tape recorder which records only one track on the tape. Usually a full-track recording head is used which covers the full width of the ¼in. tape although some machines use a narrower, half-track recording head which records a single track down the middle of the tape. Output of a full-track recording is theoretically double that of a half-track recording, although actually the output is only slightly greater because of improved half-track head design.

SPLICING TAPE—A special, pressure-sensitive, non-magnetic tape used for splicing magnetic tape. Its " hard "adhesive will not ooze and consequently will not gum up the recording head, or cause adjacent layers of tape on the reel to stick together. (Cellulose tape should never be used).

STEREOPHONIC RECORDERS—Similar to binaural recorders, but in general the two microphones are positioned differently and loudspeakers positioned apart from each other are used in place of headphones. Normally " stacked " or in-line heads are used, but stereo tapes are also available in America made with " staggered " heads where one track on the tape is used slightly in advance of the other. Stereo tapes made on one system are not suitable for playing on the other.

TAPE GUIDES—Grooved pins of non-magnetic material mounted at either side of the recording head assembly to position the magnetic tape on the head as it is being recorded or played.

TAPE LOOP—A length of magnetic tape with the ends joined together to form an endless loop. Used either on standard recorder, special " message repeater " type units, or in conjunction with a cassette or cartridge device, makes it possible to play back a recorded message repetitively without rewinding the tape.

TAPE SPEED—Speed at which the tape moves past the recording head. Standard tape speeds for home use are $3\frac{3}{4}$in. per second (abbreviated in./s or ips) and $7\frac{1}{2}$in./s. Faster speeds are 15in./s and 30in./s. Slower speeds sometimes used are $1\frac{7}{8}$in./s and 15/16in./s. Faster speed makes possible improved high-frequency response, while slower speed means greater tape economy. If a tape is recorded at $3\frac{3}{4}$in./s, then played back at $7\frac{1}{2}$in./s, all sounds will be raised one octave in pitch. Cutting the speed in half lowers a tone one octave.

TELEPHONE PICK-UP—Type of induction coil device in close proximity to a telephone receiver, or upon which entire telephone may rest, used to pick up both voices during a telephone conversation for recording on tape.

THREADING SLOT—Slot in recording head assembly cover-plate into which tape is slipped in threading up the reels for use of the recorder.

TONE CONTROL—Control knob on tape recorder amplifier used to vary bass and treble response to achieve most desirable balance of tone.

VOLUME—An acoustic, rather than electrical, measurement, which refers to the pressure of the sound waves in terms of dynes per square centimetre. The louder the sound, the greater the pressure. Most technicians prefer to talk in terms of decibels.

VTR—Video tape recording.* Recording and reproducing television picture tube signals on standard, but highest quality, magnetic tape. It is extremely difficult to design a tape recorder capable of handling a wide frequency range up to 4 million cycles per second. One method uses several magnetic tracks, recorded side by side on a $\frac{1}{2}$in. tape at a considerably higher speed than used in home recording, each track recording a certain range of frequencies. Improved quality and lower operating cost are expected to enable it to replace ciné film for television use.

VU METER—A " volume unit " meter which indicates the relative levels of the various sounds being recorded by measuring the electrical voltages.

WOW—Slow variations in tape speed causing similar variations in sound volume and pitch not present in the original sound. A form of distortion.

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