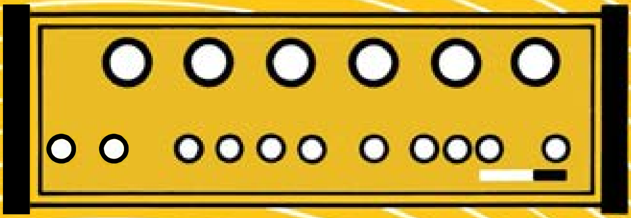
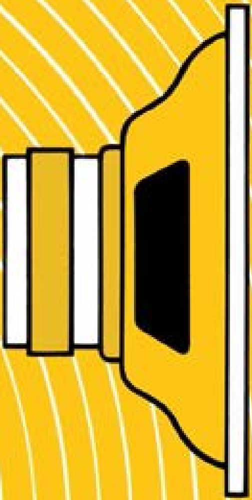


abc of HI-FI



JOHN EARL

abc of HI-FI

This is **not** a dictionary of hi-fi terms, nor is it a mini-encyclopaedia, even though the book is divided into seven main audio subjects and the entries in each chapter are in alphabetical order. This is done to permit quick and easy reference to any particular term or technique used in hi-fi.

The main objective of **abc of HI-FI** is to provide information on particular subjects, to define terms and expressions, to supply facts about equipment, and in general to outline essential aspects of the art of hi-fi listening. In common with all books in this series, the approach is fundamentally practical and has been written by an author heavily involved in the testing, appraisal and enjoyment of audio equipment. He is also engaged in the readers' query services of audio periodicals and is thus in an ideal position to evaluate just what problems puzzle ordinary hi-fi enthusiasts.

Apart from the domestic listener, the book will also be helpful to students and to those engaged professionally in audio. There is a wealth of information hidden in these pages, perhaps not readily apparent from the deceptively simple presentation, which makes it a book that should be ready to hand not only in the home but in reference libraries and the laboratory and service workshop. A quick check of the appropriate sections will unclog the mind on many concepts which are unclear or hazy — whether the reader is an amateur or a full-time professional!

ABC OF HI-FI

DEDICATION

To my son Mike and to all those youngsters
growing up in this new world of hi-fi

ABC OF HI-FI

by John Earl

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Consultant Editor : Norman Stevens

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Other Fountain Press books by John Earl

TUNERS & AMPLIFIERS
PICKUPS & LOUDSPEAKERS
AUDIO TECHNICIAN'S BENCH MANUAL
IMPROVING YOUR HI-FI

PREFACE

MANY ENTHUSIASTS, particularly newcomers, often have the need for a quick reference to a hi-fi term or technique, and this book has been written specifically to satisfy this requirement. It is a practical book, and the aim has been to present the information in a manner which will have maximum appeal to the average hi-fi enthusiast, to the chap whose job it is to answer all those awkward questions and to the student of hi-fi.

The book is designed to complement the other *John Earl* titles, and it is divided into specific aspects of hi-fi, including 'quadraphony', the reference under each topic then being presented in alphabetical order. The subject matter covered is broadly given under seven chapter headings, though it is possible, of course, for a reference to appear in more than one chapter. Distortion is a case in point since this aberration can occur at virtually any point in the channel path from microphone to loudspeaker.

Although inter-references between chapters are given where appropriate, each chapter is essentially complete in itself, so that if a reference or technique pertaining, say, to amplifiers is required, then it will be found in the chapter dealing with amplifiers.

Line drawings and photographs are used to emphasise points of detail, and the general format follows that of the other *John Earl* books.

My thanks are again due to my son-in-law Alan for his help in sorting out useful references and for helping with some of the line drawings. This chap is swiftly developing in the hi-fi sense, and would you believe is now beginning to tell *me* a thing or two!

1975 *John Earl*

CHAPTER ONE

AMPLIFIERS

AMPLIFIER

AN AMPLIFIER IS AN ELECTRONIC DEVICE using valves or transistors which provides an amplification of input source signal. A hi-fi amplifier, considered as an integrated whole, contains various *stages* of amplification, and each stage could rightly be termed an amplifier. A number of such stages are arranged in cascade or in other ways to provide the required overall *amplification factor*. The stages may also be engineered to process the source signal being amplified in terms of frequency response, signal voltage or signal power.

Amplification Factor

The amplification factor can be expressed either as a direct ratio or as a decibel ratio. If, for example, an output of 1V is obtained when an input of 100mV is applied, then the amplification factor would be 10(1/0·1). Expressed in decibels (dB), a 10:1 voltage ratio is 20dB (dB$=$20 $\log_{10}$ V_{in}/V_{out}).

However, this holds accurate only when the input and output impedances are equal. A correction needs to be applied when the output impedance is different from the input impedance, for impedance (symbolised by Z) defines the *power* in a load referred to a specific signal voltage across it.

The overall amplification factor of two or more stages is equal to the product of the direct ratio amplification factors or to the sum of the dB ratios. For example, three stages each of 10:1 amplification factor (i.e., 20dB) will have an overall amplification factor of $10 \times 10 \times 10$, or 10^3 or 1000:1, whichever way you want to view it, or of 20dB$+$20dB$+$20dB, equals 60dB.

Voltage Amplifier

A hi-fi amplifier consists of stages of voltage amplifiers and a power amplifier. In both cases power is being amplified because the signal in the load must consist of both voltage and current, the product of these being power, but the term *voltage amplifier* has come to mean the early stages of the amplifier where the small voltage from the input source, such as pickup cartridge, tape head, radio tuner, etc., is increased to the level required to drive the power amplifier, which in turn drives the loudspeaker which (unless it is an electrostatic type) itself needs power to yield an acoustical output.

The term voltage amplification reflects from the valve days. With transistors, *current amplification* is sometimes more appropriate since transistors in general have lower input impedance than valves and so need current drive rather than voltage drive. But the early term 'voltage amplification' is still often used.

Power Amplifier

The power amplifier which drives the loudspeaker needs to deliver fairly high current which, with a nominal signal voltage, gives rise to relatively high power. Consider an output stage of a fairly powerful hi-fi amplifier which accommodates a peak-to-peak voltage swing to clipping threshold of, say 56, across an 8Ω load.

This means that the peak voltage (half peak-to-peak) is 28 and the r.m.s. (meaning root mean square—or the integrated sum of the squares at discrete levels over the waveform, themselves square-rooted, corresponding to 0·707 of peak) voltage about 20.

Now, the power in the 8Ω load is equal to the r.m.s. voltage squared divided by the load resistance in ohms, which in this case is $20^2/8$, or 50W.

Power is also equal to the product of the r.m.s. voltage across the load and the r.m.s. current through it, which means that when the load is taking 50W the r.m.s. current flowing through it is 2·5A.

Power Gain

Just compare this with an input signal of, say, 10mV across 50kΩ, which corresponds to a mere 2×10^{-9}W. Referred to 50W output, therefore, the power gain is $2{\cdot}5\times10^{10}$, about 104dB.

The signal current through the 50kΩ input load with the 10mV across it works out to 2×10^{-7}A. Thus we have a voltage gain of 2×10^{3} and a current gain of $1{\cdot}25\times10^{7}$, the product of these again yielding the originally computed power gain of $2{\cdot}5\times10^{10}$, which is an interesting point to think about.

Preamplifier

The first voltage (or current) amplifying stages of our hi-fi amplifier are collectively referred to as the *preamplifier*, this merely signifying that the stages precede the power amplifier.

Another term describing the same section is *control unit.* This is really a better term since it describes more acurately the function of the stages; that is, they *control* the signals fed from the sources to the power amplifier.

The term preamplifier is commonly used to describe specific stages in the control unit or *control section.* For example, that stage which is concerned with the amplification and equalisation of the

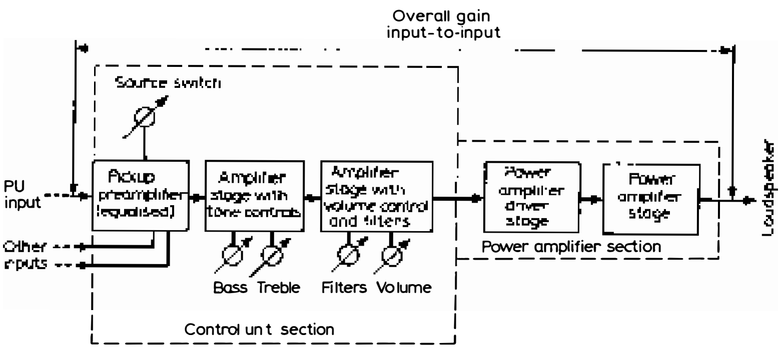


Fig. 1.1 Elementary block diagram of hi-fi amplifier showing the various stages discussed in the text. The overall power gain is the output power (from the power amplifier) divided by the input power (of the source signal). The overall voltage gain is the output voltage divided by the input voltage, and the overall current gain the output current divided by the input current. The product of the voltage gain and the current gain (in direct ratios) also gives the power gain.

gramophone pickup signal, being a part of the control unit, is generally called a preamplifier which is exactly what it is.

The various amplifier stages discussed are illustrated in Fig. 1.1. In addition to these, the control section contains the switching for several input sources and the controls for tone, volume and sometimes filtering.

When the control and power amplifier sections are built into a common housing, energised from the same power supply, the amplifier is sometimes classified as *integrated* (not to be confused with an integrated circuit).

Most hi-fi amplifiers are of the so-called integrated kind, though one or two are still made with completely separate control unit and power amplifier sections, there then being two units to connect together.

AMPLIFIER—FOUR-CHANNEL

As implied, a four-channel amplifier is effectively four complete audio channels in a common housing. A four-channel amplifier is required for four-loudspeaker ('quadraphonic') reproduction. There are

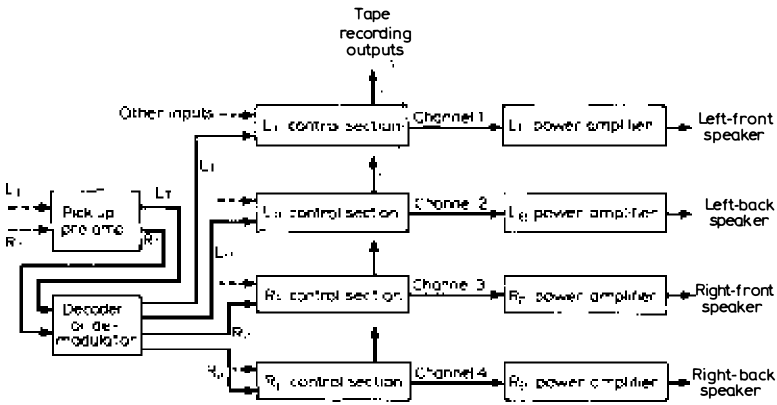


Fig. 1.2 Block diagram of four-channel 'quadraphonic' amplifier. Each channel is composed similar to that shown in Fig. 1.1. The decoder or demodulator may be either external to the four-channel amplifier or built into it. Some models have inbuilt matrix decoding and multiplex demodulating, selected by a front panel switch.

usually common features, such as a common power supply for the four amplifiers and sometimes common volume, tone and filter controls, achieved by ganging. For example, a simultaneous increase in output from the four channels occurs when the common volume control knob is advanced.

Differential regulation of output is provided either by separate signal level controls, one for each channel, with the 'master' volume control regulating the outputs of the four channels together, or by left/right and front/back balance controls.

For each source, including tape recording and replay but usually excluding the pickup, there are four sockets corresponding to left-front (L_F), right-front (R_F), left-back (L_B) and right-back (R_B). The output terminals for connecting the four loudspeakers are correspondingly indicated, though the channels may be marked 1, 2, 3 and 4 which, by convention (JVC), refer respectively to L_F, L_B, R_F and R_B.

Four input sockets are not provided for gramophone pickup because the L_F, R_F, L_B and R_B source signals are encoded into the single groove of a disc either by multiplex (CD-4, JVC. or UD-4, Nippon Columbia) or by matrix (SQ, CBS, or QS, Sansui, for example).

The pickup thus delivers the encoded signals from its two normal circuits and the decoding back into the 'original' four channels occurs either in the amplifier when there is an internal decoder or external to the amplifier when a separate decoder is used. To be pedantic, the term is *decoder* for matrix sources and *demodulator* for multiplex sources.

The block diagram of Fig. 1.2 reveals the points discussed. The signals emanating from the two pickup outputs are called left-total (L_T) and right-total (R_T). These are preamplified and equalised and then applied to the appropriate decoder or demodulator which yields the L_F, R_F, L_B and R_B signals for individual application to the four channels.

AMPLIFIER—HI-FI

This refers to an amplifier whose parameters satisfy a certain minimum requirement. A standard setting out the *minimum* requirements for hi-fi is the German DIN 45-500, but most contemporary hi-fi amplifiers exceed these minimum requirements.

AMPLIFIER—INTEGRATED

This refers to an amplifier in which the control section is built into the same housing as the power amplifier section. Also see under *Amplifier*.

AMPLIFIER—MONO

This is a single-channel amplifier such as illustrated in Fig. 1.1.

AMPLIFIER—POWER

This is the amplifier section which, driven from relatively low power signal from the control section, provides the power required by the loudspeaker for adequate in-room sound intensity.

AMPLIFIER—QUADRAPHONIC

This is a four-channel amplifier. See under *Amplifier—Four-Channel.*

AMPLIFIER—STEREO

For stereo reproduction, an amplifier containing two separately isolated channels is required, which is a scaled-down version of a four-channel amplifier. The vast majority of hi-fi amplifiers are two-channel stereo.

Each channel contains at least the features shown by the block

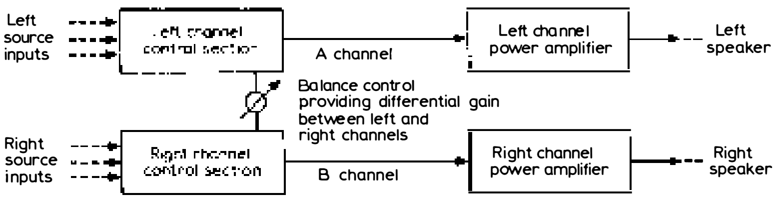


Fig. 1.3 Block diagram of stereo (two-channel) amplifier, showing balance control (when used). This increases or reduces the level of signal at the output of one channel with respect to that of the other. Many controls give a balanced output at the centre setting and decreasing left or right output as the control is turned either side of centre. Sometimes at the extreme setting, one channel is faded to zero while the other channel remains at full output, but with other designs the level of the appropriate channel may only be reduced (not faded to zero) at the extreme setting, depending on the nature of circuitry adopted.

diagram in Fig. 1.1, and differential output level control is achieved either by the use of separate volume controls, one for each channel, or by a balance control, the common volume control then adjusting the outputs of the two channels together. The left channel is sometimes called the A channel and the right channel the B channel. The elementary format of a stereo amplifier is given in Fig. 1.3.

AMPLIFIER—TWO-CHANNEL

See under *Amplifier—Stereo.*

CONNECTIONS

The amplifier needs to be connected to receive power from the mains supply, to receive signal from the various programme sources and to deliver power to the loudspeakers.

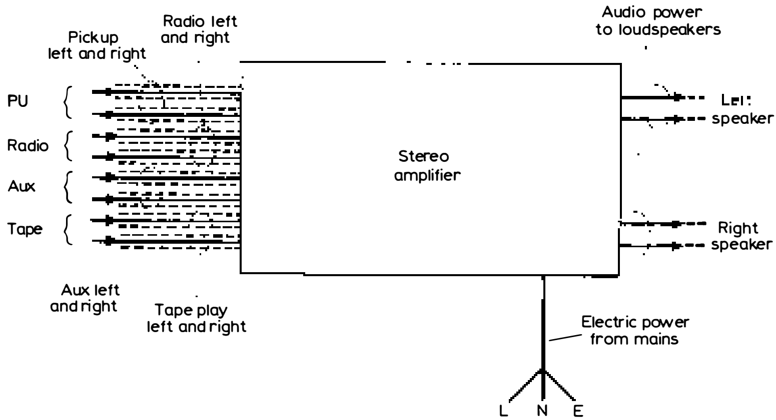


Fig. 1.4 The inputs and outputs of a stereo amplifier. For four-channel multiply by two.

These three primary connections are illustrated in Fig. 1.4 with respect to a stereo amplifier. A four-channel amplifier setup is similar, but caters for four loudspeakers and has four inputs for each source excepting pickup.

Mains Connections

Some amplifiers have a two-conductor mains cable and others a three-conductor cable. In the two-conductor case the connections are made to the live and neutral terminals of the mains and, unless stated in the user's booklet, it does not usually matter which wire goes to which terminal. In the three-conductor case, however, it is extremely important to connect the wires to the correct terminals, for *failure to observe this could result in lethal conditions.*

Where there are three conductors, one is connected directly to the metal parts of the amplifier and this one *must* be connected to the mains plug earth terminal (*on no account must this wire be connected to mains neutral or mains live*). The mains cable wires are colour coded, the standard being green/yellow earth (this is the conductor connected to the metal parts), brown live and blue neutral. If there is any doubt about the connections qualified advice must be sought before operating the equipment.

A three-pin plug mains connection, when there are three conductors in the amplifier's mains cable, automatically earths the amplifier and an external or other earth should not be connected. Also see under *Earthing* and *Hum-Loops.*

Loudspeaker Connections

Each loudspeaker requires a two-conductor cable for connection and the conductors should be coded for identification, which is necessary for correct phasing (see under *Phasing*). As there is no absolute standard for loudspeaker sockets, the connections may be either by screw or spring terminals or by a plug/socket arrangement.

A common plug/socket is that to the DIN standard which has a flat connector for the 'earthy' connection and a pin connector for the 'live' connection (Fig. 1.5). When different plug/socket or terminal arrangements are used instead, the 'earthy' connections are usually

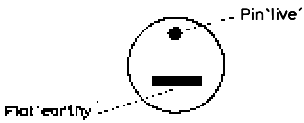


Fig. 1.5 DIN loudspeaker plug/socket connections. Since this plug cannot be reversed, the phasing once established, cannot be inadverently put in error by removing and replacing the plug. See the author's *Improving Your Hi-Fi* for further information.

coloured black or marked ground or negative (—) and the 'live' connections coloured red or marked positive (+).

In some cases—the majority—the negative or 'earthy' connections of both loudspeaker circuits are in communication with the metal parts of the amplifier (chassis), but it is imprudent to connect the 'earthy' sides of the loudspeakers back to the 'earthy' sides of the sources as this can result in a significant increase in distortion and/or hum-loops (see under *Hum-Loops*). In other words, the loudspeaker circuits should be wired in complete isolation.

Owing to the relatively high current flowing through the loudspeaker cables from the power amplifiers to the loudspeakers there is bound to be some power loss in the cables (i.e., I^2R loss), so to relieve this the cable lengths should be as short as possible and the diameters of the conductors as large as possible (within reason!). 5A mains cable is suitable for most runs, but for protracted runs or when the absolute maximum power is required in the loudspeakers 15A cable is more desirable.

An amplifier which produces, say, 9V r.m.s. across 4Ω would be rated around 20W, and if a 4Ω load is connected directly across its output it would dissipate 20W at full drive. Now, if the cable resistance is, say, 2Ω, the 9V would be developed across 6Ω, so that the total power would now be 13·5W. Under this condition the full drive current would be 1·5A, so the load would only receive 9W. The remaining 4·5W would be heating the cable conductors!

In practice, the full drive voltage of the amplifier would probably increase with increasing load resistance so there would be more than 13·5W to be shared between the load and the cable, but the power loss would still be significant. Hence the need for low resistance conductors! The net effect is more complex in practice owing to loudspeaker and circuit impedances.

Source Connections

Each stereo source has two circuits, one each for the left and right channel signals, and each left and right circuit is screened from extraneous electric and hum fields and from each other, the latter to avoid *crosstalk* (see under *Parameters*). The broken lines of Fig. 1.4 source circuits represent the screening, which also acts as the 'earthy' side of the source circuit.

Thus, for say the left radio channel there is an inner conductor 'live' (from the signal point of view) circuit surrounded by a braid (flexible wire mesh) which serves both as the screen and the earth return to the source.

The four stereo sources of Fig. 1.4, therefore, are connected by four pairs of screened conductors to the amplifier source inputs. With a 'quadraphonic' system each source, excepting pickup, which as we have seen (Fig. 1.2) operates on two circuits, would be connected by four screened conductors.

There are two main types of input sockets and matching plugs, called RCA 'phono' (sometimes 'pin') and DIN. Amplifiers with 'phono' and DIN source sockets are shown respectively at (a) and (b) in Fig. 1.6.

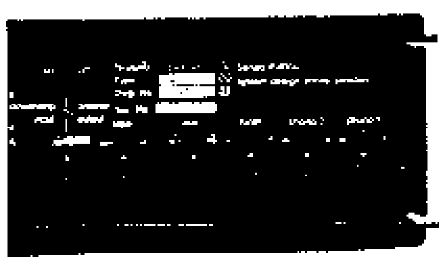
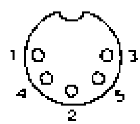


Fig. 1.6 (a) amplifier with phono' source inputs. (b) amplifier with DIN source inputs (also see Fig. 1.20).

A 'phono' plug/socket is a two-circuit device designed to accommodate single screened cable such that the inner conductor is connected to a centre connector while the braid is connected to an outer connector. Two 'phono' plugs are thus required for each stereo source and each source input of a stereo amplifier consists of two (left and right) 'phono' sockets.

DIN terminations are different. Here there is one plug and one socket for each source, these carrying both the left and right channel signals as well as a common earth (braid) return. Thus, while eight 'phono' plugs and sockets are needed for the four sources of Fig. 1.4, the DIN method of connection cuts the inter-connections down to four, one for each source.

A standard (fully-fledged!) DIN plug/socket is equipped with five connectors (thin pins on the plug and miniature sockets on the socket to accept them). The connectors are numbered in the way shown in Fig. 1.7, and there is a 'DIN code' correlating the circuits



Connector 2 'earthy' in all cases (connected to braids)

Pickup	Tuner and auxiliary	Tape play	Tape recording output
left 3 right 5	left 3 right 5 (or 1 with connection to 5)	left 3 right 5	(from amplifier) left 1 right 4

Fig. 1.7 DIN plug/socket numbering and some of the chief circuits in terms of connector numbers. In all cases connector 2 is 'earthy' and is connected to the screens of the signal cables. The diagram shows the pin side of the plug and the soldering tag side of the socket. See the author's *Improving Your Hi-Fi* for more details on the DIN standards.

to the different numbers. In all cases, however, connector 2 is 'earthy', this accommodating the screens of the two (or more—see under *Tape Recording Output*) signal circuits. Fig. 1.7 also gives the number code of primary sources.

CONTROLS

Balance

This control adjusts differentially the gain—and hence the outputs—of the left and right channels of a stereo amplifier (see under *Amplifier—Stereo* and Fig. 1.3). Four-channel ('quadraphonic') amplifiers sometimes feature an additional balance control operating similarly but with respect to the front and back channels.

With a stereo amplifier the control is adjusted on a mono source for a centre-point sound image between the left and right loudspeakers.

Loudness

While the volume control regulates the gain and hence output of each channel evenly over the full *bandwidth* (see under *Parameters*), the loudness control, when retarded, reduces the output more over

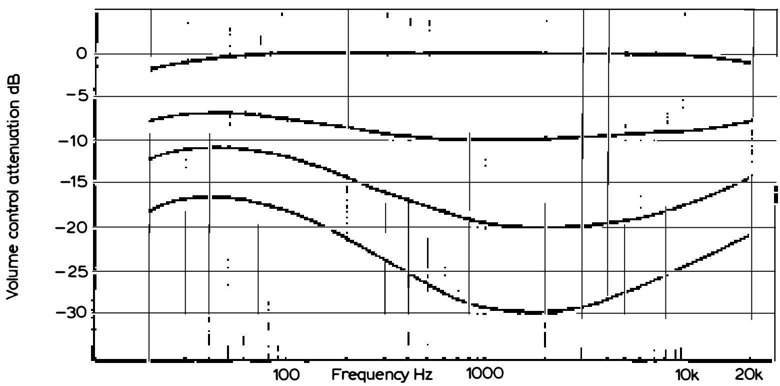


Fig. 1.8 Family of loudness control curves, showing how the middle frequencies are progressively more depressed than the lower and higher frequencies as the control is turned down.

the middle range of frequencies than at the low and high frequencies. This results in bass boost and to a lesser degree treble boost, the boost effects increasing as the control is progressively retarded, as shown by the family of curves in Fig. 1.8.

A so-called 'loudness' switch is often used to change the action of the volume control from frequency 'linear' to frequency compensated to give the loudness effect. On the other hand, the 'loudness' feature may be fixed, it then not being possible to achieve a frequency 'linear' change in volume, which is undesirable.

The loudness function is incorporated to combat the frequency sensitivity of hearing, the sensitivity falling at the lower frequencies and by a smaller amount at the higher frequencies as the sound intensity is decreased. Thus by the increasing bass and treble boost as the control is retarded the sensation of loudness appears to remain constant over the spectrum, an effect which is deprecated by some authorities.

Another 'loudness' function takes the form of a switch which depresses the middle frequencies by a given number of decibels and the lower and higher frequencies by a smaller amount, but is not affected by the setting of the volume control. This is sometimes considered desirable for low-level listening.

Some amplifiers incorporate two controls, the ordinary volume control and a second frequency-compensated loudness control.

The term 'loudness' does not always describe the expected effect. For example, with some designs the average sound level over the spectrum *falls* when the loudness button is operated, in spite of the bass and treble boost. The term implies that the response characteristics of the amplifier are tailored to the curves of equal loudness, such as those after Robinson and Dadson (see page 40 of *Improving Your Hi-Fi*). 'Contour', 'compensated volume control', etc. may sometimes be used instead of loudness to describe the control.

Some circuits provide only for bass boost, which is even less accurate so far as loudness compensation is concerned. It is easily possible to simulate the loudness control effect on amplifiers without a loudness control, merely by applying bass boost and lesser treble boost by the tone controls when listening at low level.

Loudspeaker Switch

Many hi-fi amplifiers are equipped with facilities for operating a headphone set, and in some models the loudspeakers are switched off automatically when the jack plug is pushed into the jack socket. Other models incorporate a loudspeaker switch for this operation.

The switch may also include positions for two or even three pairs of loudspeakers (*pairs* in the case of stereo and quadruples when the amplifier is four-channel) which may be located in different rooms, so that one pair only or two pairs together may be operated.

When two pairs together are used, however, the impedance of each loudspeaker should not generally be much below 8Ω. This is because when two (or more) loudspeakers are connected in parallel the impedance as 'seen' by the amplifier falls, and this could result in output stage overload.

For example, the net impedance when two 8Ω loudspeakers are connected in parallel is 4Ω; it would only be 2Ω if the loudspeakers each had an impedance of 4Ω.

Mode Switch

On a stereo amplifier the mode switch usually has positions for stereo and mono. In the mono position the left and right channel signals are summed to yield a mono signal which passes through both the left and right channels, the loudspeaker output then being termed 'double mono'.

It is desirable to switch to mono when playing a mono record via a stereo system. Double mono output will be achieved even in the stereo mode, but because a stereo pickup produces an output from vertical vibrations of its stylus, rumble, noise and distortion is higher when a mono record is played under this condition. By switching to mono, output due to vertical vibrations is cancelled, the pickup then responding essentially to the lateral cut of the mono record.

When a stereo f.m. tuner is used, the amplifier can be switched to mono to translate a weak, noisy stereo signal to a less noisy mono signal.

Mode switches of more expensive amplifiers have positions for stereo channel reverse and for double mono output when a single channel signal is applied to the left or to the right source input.

The mode switch of a four-channel amplifier may have additional positions for two-channel, four-channel (discrete) and matrix encoding.

Presence Control

This is a control or switch which provides a fixed or variable amount of boost around the middle of the spectrum, as shown in Fig. 1.9.

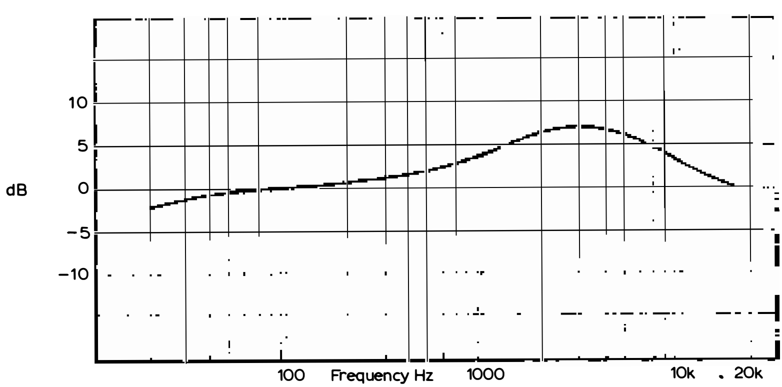


Fig. 1.9 A presence control boosts the middle frequencies. In some designs the boost may occur at a slightly lower frequency than shown by the curve.

Since hearing is most sensitive at the middle frequencies, the effect is that the sound from the loudspeakers is brought forward towards the listener. The response is opposite to that of a loudness control.

Preset Controls

Some amplifiers are equipped with rear or underneath preset controls for input sensitivity adjustment. A single preset may be used (ganged) which operates on the two stereo channels together, or there may be separate presets for the left and right channels. The presets are adjusted to 'normalise' the output so that no significant change in level occurs when switching over the various sources at an established volume control setting.

Similar presets may be present in the tape recording output circuits so that the recording level can be adjusted to suit the input sensitivity of the tape machine used.

Most amplifiers also have internal preset controls for adjusting such things as output stage quiescent current (i.e., Class B biasing) output stage balancing, mid-point voltage, etc., but these are 'service' adjustments which should not normally be interfered with by the user, unless equipped with the necessary test equipment and technical 'know how'.

Source Selector Switch

This is a front panel switch (rotary switch or press buttons or keys) which is used to introduce the required source signals—tuner, pickup, etc. (see under *Source Inputs*).

Tone Controls

Most common tone controls are bass and treble, which operate either side of a centre setting to give progressive boost or cut of the bass and treble frequencies relative to the mid-spectrum frequencies, as shown in Fig. 1.10.

The controls are used to 'equalise' for deficiencies in programme signals, loudspeakers or room acoustics. For example, a loudspeaker which is weak in the upper treble can be corrected to some extent by applying treble boost. Care, however, should be taken with regard to bass boost because some loudspeakers—particularly some of the small ones—are incapable of extended low-frequency reproduction without severe distortion.

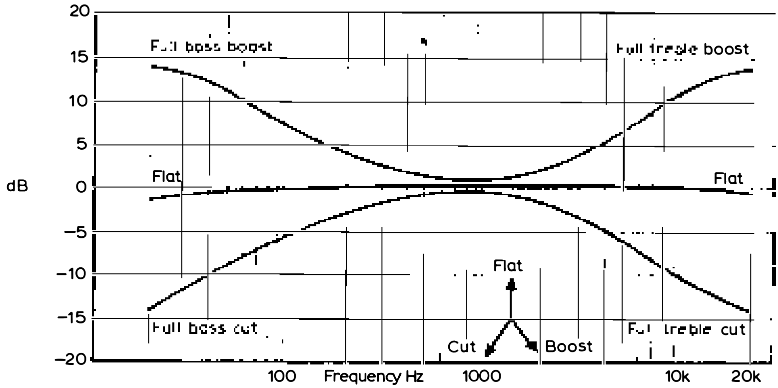


Fig. 1.10 Bass and treble tone control characteristics.

The controls are commonly ganged over the two stereo or four 'quadraphonic' channels, though some stereo amplifiers have separate controls for each channel, either dual-concentric friction-coupled knobs or totally separate knobs. The turnover frequency, where the lift or cut is 3dB from the 'flat' response, is switchable on some models.

A more recent scheme takes the form of a number of controls, each one giving lift or cut over a small part of the spectrum. Slider controls are mostly used, and the set positions of the knobs of the sliders roughly describe the applied equalisation characteristic, and for this reason the term 'graphic equalisers' is sometimes used to indicate the system.

JVC is one firm using the idea, which goes under the name 'sound effect amplifier' (SEA for short). Another is Eagle International, which goes under the name 'sound effect control' (SEC for short). A plot of the JVC SEA is given in Fig 1.11.

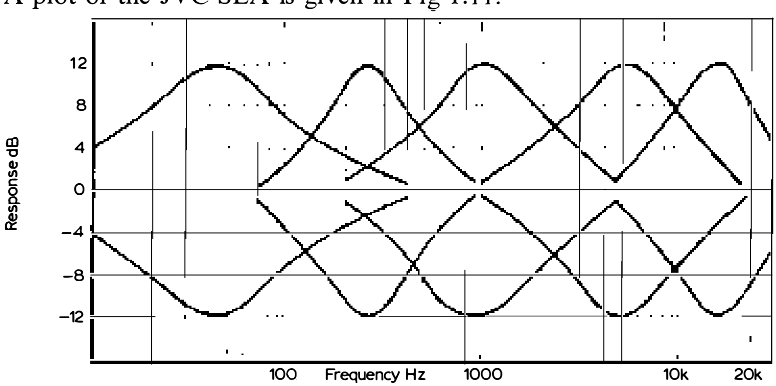


Fig. 1.11 Response characteristics of the JVC SEA 'equaliser'. Five slider controls are used for lift and cut at five discrete frequency bands over the spectrum.

Both JVC and Eagle employ five sliders which make possible continuous and delicate control of practically any part of the musical spectrum. For source, loudspeaker and room equalisation the scheme is far more versatile than the conventional two-control tone system.

Between the 'graphic' arrangement and the two-control system there is the three-control system which, in addition to the bass and treble controls, uses a third control for boost or cut of the middle part of the spectrum. Marantz is but one firm using this idea, the response characteristics of which are given in Fig. 1.12.

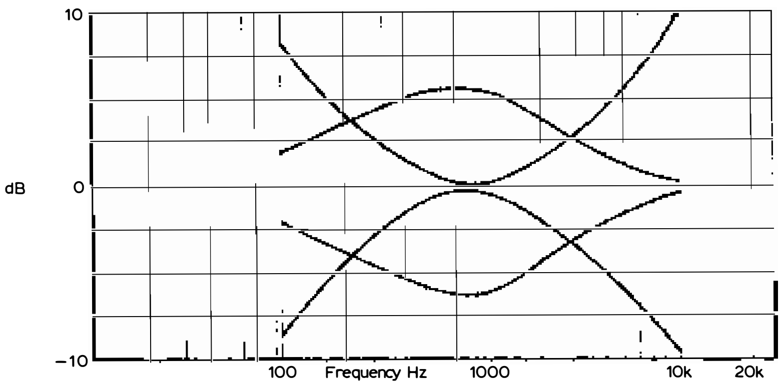


Fig. 1.12 Bass, middle and treble tone control characteristics of the Marantz 4270 'Stereo 2+Quadradial 4 Receiver'.

Volume Control

This is generally placed between the control section and the power amplifier and serves to apply progressive attenuation, thereby decreasing the gain of each channel as it is retarded. It has no effect on the intrinsic power delivery of the amplifier but merely regulates the output level from the rated output downwards. The setting for optimum *dynamic range* (see under *Dynamic Range*) is between 1 o/c and 2 o/c.

When the amplifier features a balance control the volume control works on both channels of a stereo amplifier together. A separate control is sometimes used for each channel, a balance control then not being necessary. Also see under *Loudness.*

CONTROL UNIT

This is the front part of an integrated amplifier or is a separate unit when the amplifier consists of independent control and power amplifier sections—i.e., non-integrated (see under *Amplifier*).

DECIBEL

This is a logarithmic ratio between two common quantities, such as power, voltage, current, sound pressure, etc. allowing linear scale expression (i.e., see the vertical, Y, scale of Fig. 1.10). For power ratios the decibel (dB) value is the logarithm of the ratio multiplied by 10. For example, a power ratio of 1·995:1 (almost 2:1) is equal to 3dB (i.e., half or double power ratio).

For current, voltage or pressure ratios the dB value is the logarithm of the ratio multiplied by 20, so that a voltage or pressure ratio of 1·995:1 (almost 2:1) is equal to 6dB. The difference between the multiplication factors takes account of the fact that with power the voltage or current is squared, and one 'squares' in logs merely by multiplying by 2, which gives the change from 10 to 20.

The factor of 10 is used in the first place (for power) to convert the primary unit Bel to *deci*bel, which of course is a tenth of a Bel. dB tables are readily available to save the toil of calculation; one example is to be found in the author's *Tuners and Amplifiers*, page 159. Also see under *Amplifier*.

DISTORTION

In the amplifier, this refers to any deviation of the output signal with respect to the input signal. Distortion goes under the several headings of harmonic distortion (or, more commonly, total harmonic distortion, THD for short), intermodulation distortion (IMD), crossover distortion, transient distortion (and transient intermodulation distortion, TID for short) and frequency distortion.

THD and IMD (as well as some of the other types) arise from non-linearity of transfer of signal from input to output, which implies lack of correlation between input and output signal *amplitude* changes. With perfect linearity a change in input signal amplitude of, say, ratio x would yield x ratio change at the output. Transfer non-linearity causes the output amplitude to change slightly differently from ratio x.

The nature of the distortion is influenced by the nature of the non-linearity, but in hi-fi amplifiers the distortion produced is very small, which means that such amplifiers are not all that non-linear. Perfect linearity is impossible.

THD is generally measured by applying a very low distortion sinewave signal at the input, developing the signal across the output load (usually a resitive dummy load) and then filtering out the fundamental frequency of the sinewave so that only the distortion components remain, as shown in Fig. 1.13(a). The percentage difference between the voltage of the sinewave and the voltage of the summed distortion components gives the THD of the amplifier at the established test power and frequency.

Hi-fi amplifiers should not produce much more than 0·1% THD at mid-spectrum frequencies at the rated power and below. Distortion generally increases at the higher (and sometimes lower) frequencies, but the rise should not be excessive within the musical spectrum (say, from 20Hz to 20kHz).

IMD occurs when two or more frequencies pass through the amplifier together, as when operating under music conditions. The non-linearity results in intermodulation so that spurious signals are produced at sum and difference frequencies and their multiples.

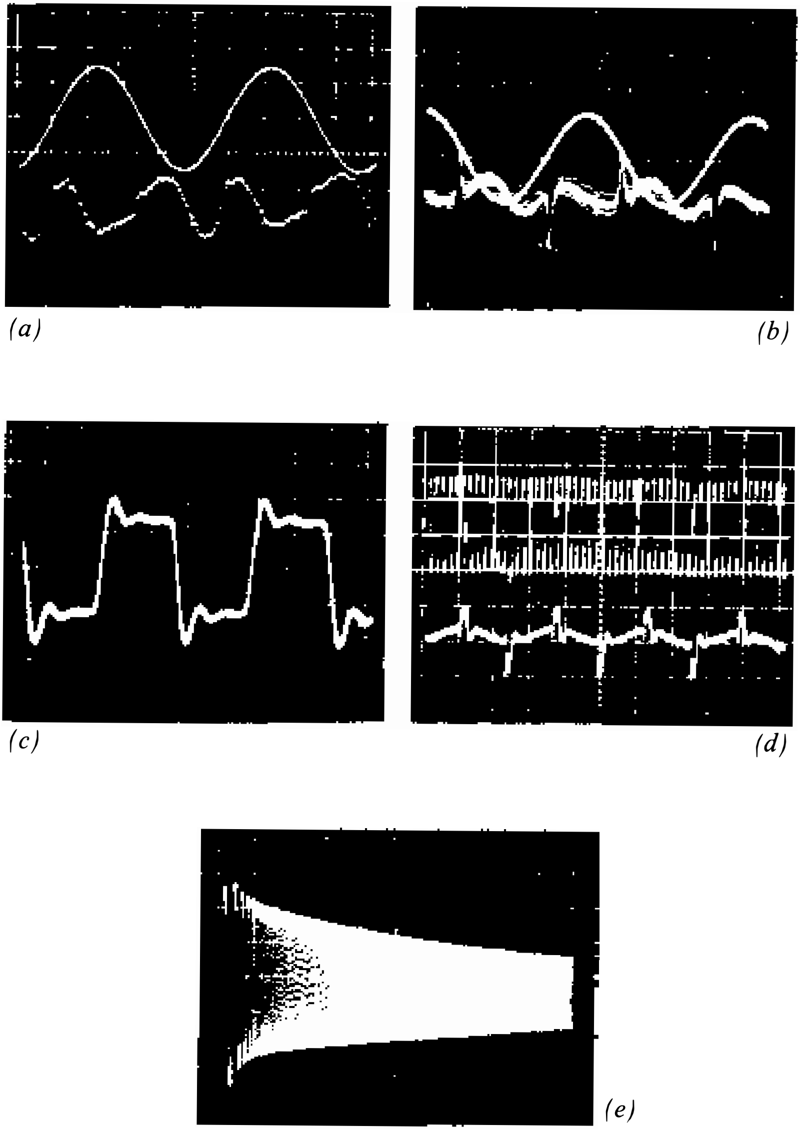
Crossover distortion is a kind of IMD which is caused by lack of correlation where the transfer characteristics of the two push-pull output transistors meet in the middle of the overall push-pull characteristic. This is common to Class B biasing, but by increasing the forward bias of the push-pull stage for a small amount of quiescent current (i.e., towards Class AB) the middle non-linearity is reduced and the crossover distortion is diminished.

Fig. 1.13(b) reveals the presence of crossover distortion during a THD test. Notice both at (a) and (b) that the sinewave signal fails to show the distortion because it is of such a small percentage. Modern Class B amplifiers employ other artifices to minimise crossover distortion.

Transient distortion refers to the change in shape of a very fast transient signal passing through the amplifier. In some cases this can produce tonal changes. To preserve the transients the amplifier must have a power bandwidth extending at least to 20kHz and must be free from 'electrical ringing' effects.

Squarewave signals are often used to appraise transient performance because such signals contain high-order odd-numbered harmonics. Fig. 1.13(c) shows 'ringing' on a squarewave, which is a kind of transient distortion.

Transient intermodulation distortion is more complicated. Basically, it results from the negative feedback of the power amplifier failing to respond as quickly as required to a very fast input signal transient. The transient thus activates the driver stage of the power amplifier before its gain is reduced by the signal coming back anti-phase as negative feedback.



(a) (b) (c) (d) (e)

Fig. 1.13 Types of distortion illustrated. (a) Total harmonic distortion (THD) bottom trace and signal from which it is obtained top trace, (b) crossover distortion effect on THD oscillogram, (c) Squarewave 'ringing' (overshoot) which is a form of transient distortion. (d) Transient intermodulation distortion (TID). (e) Display of frequency distortion (swept oscillogram)—see text.

The result is that the transient overloads the driver stage so that it is 'blocked' for a short time following the transient (that is, until the feedback takes control). Thus, signal information directly following the transient will also be blocked, which results in 100% TID. The oscillogram in Fig. 1.13(d) gives one impression of TID.

Frequency distortion implies that the amplifier gives more or less amplification to some frequencies than others. Frequency distortion is deliberately introduced, for example, when a tone control is adjusted from its centre 'flat' position. An interesting oscillogram of frequency distortion is given in Fig. 1.13(e). This is a sweep-frequency display with the frequency gradually increasing from low to high from left to right. This clearly shows that the low frequencies are amplified more than the higher ones—the amplification decreasing progressively with frequency.

High-order odd-number harmonic distortion, and this includes IMD, is far less palatable than low-order even-number harmonic distortion. Crossover distortion, if in high measure, is also very offensive to the critical ear. Transient distortion can cause disturbing 'hangover' effects and change the tonal quality of music, while TID can be similar in effect to crossover distortion, but the latter usually occurs more at low-level and the former at high-level.

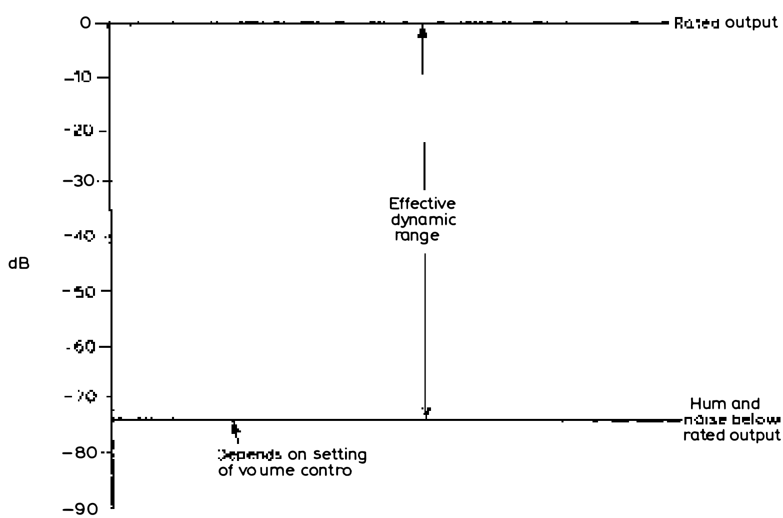


Fig. 1.14 Illustration of amplifier dynamic range.

At high frequencies the power amplifier can produce an effect similar to crossover distortion, called 'notch distortion'. This results from the storage of minority charge carriers, which impairs the high-frequency switching from one transistor of the push-pull pair to the other.

The very recent field effect *power* transistors are expected to eliminate this because such devices work only on majority charge carriers, there being no minority carrier storage.

DYNAMIC RANGE

The potential dynamic range of an amplifier is that range between a 'sandwich' corresponding on one side to the power rating and on the other to the hum and noise output, as shown in Fig. 1.14. The majority of hi-fi amplifiers have a dynamic range so expressed in excess of 60dB, which caters for the dynamic range of the programme sources.

EARTHING

If the amplifier is earthed either to the mains earth or to an outside earth, earthing of the connected programme sources, such as tape recorder, radio tuner, etc. is automatically accomplished by way of the signal cable braids, but for optimum safety each item designed for earthing should be connected direct to an earth point (see p. 112).

A connected system should only be earthed at one point. If more than one earth is connected hum can result (see under *Hum-Loops*). If a mains earth is used an outside earth should not be used, and *vice versa*. Also see under *Connections*.

EQUALISATION

This has common reference to frequency-compensated amplification particularly with regard to magnetic pickups and tape heads, the equalisation compensating for the frequency distortion arising from a specific recording characteristic.

In other words, the frequency distortion introduced by the equalisation is the reciprocal of the distortion resulting from the recording, etc. characteristic, the net result then being a 'flat' frequency characteristic. See under *Equalisation* in Chapters 3 and 6.

In the amplifier the pickup preamplifier contains an equalising network such that the low frequencies are given far more amplification than the high frequencies, as shown in Fig. 1.15.

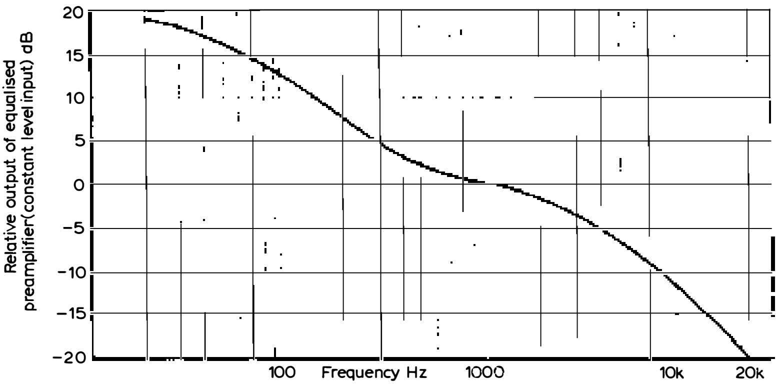


Fig. 1.15 Showing how the magnetic pickup preamplifier equalising boosts the low frequencies and attenuates the high frequencies relative to the middle frequencies. Since the output from a magnetic pickup has a characteristic following the converse of this curve, the net result is a 'flat' output.

FILTERS

Filters come under two headings so far as hi-fi amplifiers are concerned—high-pass and low-pass. The first implies that the high frequencies are passed without attenuation and that the low frequencies are attenuated, and the second that the low frequencies are passed without attenuation and the high frequencies are attenuated.

Some amplifiers feature both high-pass (low cut) and low-pass (high cut) filters. When both are switched on together a 'bandpass' characteristic results. One or two amplifiers have a single switch which activates both filters together, thereby providing only a bandpass characteristic.

The frequency at which the attenuation is —3dB is called the *turnover frequency*. High-pass filtering has a turnover of *circa* 30–80Hz and low-pass filtering a turnover of *circa* 6–8kHz. The slope of the curve after the —3dB point is known as the *roll-off rate*; a simple filter has a rate of 6dB/octave and a multipole filter 18dB/octave or more.

Too rapid roll-off rate can precipitate overshoot and hence transient distortion, while the opposite will attenuate wanted frequencies as well as unwanted ones. Example low-cut and high-cut filter characteristics are given in Fig. 1.16. Some amplifiers have switchable turnover frequencies and adjustable (particularly high-cut) roll-off rate.

Simple, less costly designs have fixed turnovers and relatively

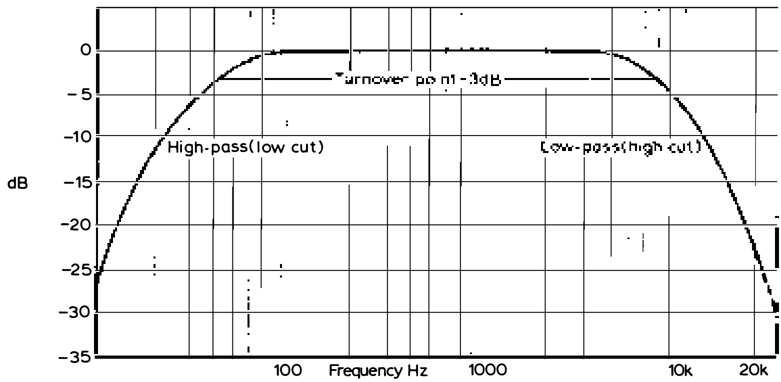


Fig. 1.16 High-pass and low-pass filters. A bandpass characteristic is produced when both filters are active together. The turnover points (—3dB) are shown and correspond to about 40Hz (low-cut) and 7kHz (high-cut). The slope following the —3dB point signifies the rate of roll-off, which is relatively fast in the examples shown.

slow roll-off rates. Roll-off rates of 6dB/octave can be simulated by the tone controls—bass and treble cut, though the turnover frequencies may be too high at the bass and too low at the treble for effective filtering by this means.

Low-cut filtering is required to attenuate unwanted low frequencies, such as turntable rumble, and high-cut filtering for reducing record and radio noise and distortion.

HEAT SINKS

A heat sink is a large mass of metal upon which the power transistors are mounted in good thermal contact. The heat sink conducts heat away from the internal transistor junctions, thereby allowing the transistors to be operated at greater power dissipation without damage.

The photograph in Fig. 1.17 shows two such heat sinks (centre of picture), each accommodating two power transistors. Air needs to flow over the sinks, so ventilation should not be restricted. However, under normal music conditions the sinks remain fairly cool when the output stages are biased to Class B, but with Class A or towards Class A (i.e., Class AB) biasing they operate at significantly higher temperature, even when the amplifier is not driven.

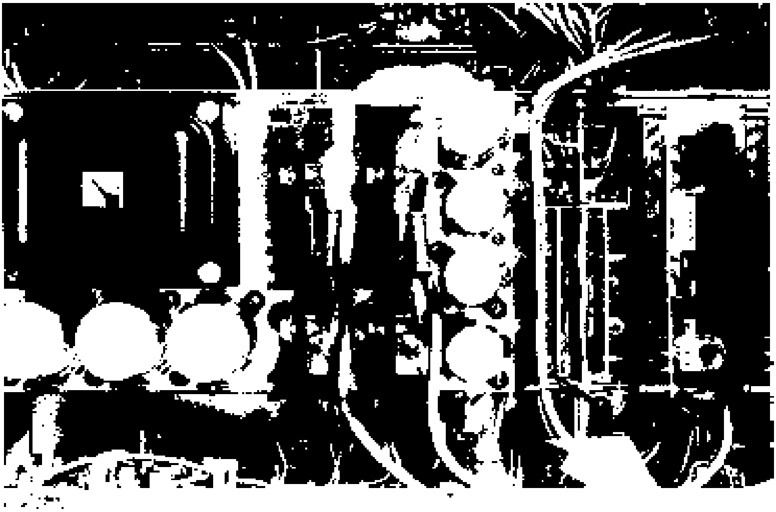


Fig. 1.17 The two heat sinks in the centre of picture (large metal masses with fins) each accommodates two power transistors. Heat sinking is essential when large power is to be drawn.

HUM-LOOP

This condition, sometimes called 'earth-loop', can arise when two earth connections are used, as shown in Fig. 1.18. In the UK, the practice of earthing only the amplifier, relying on the signal lead braids to earth the other items, is incompatible with optimum safety (see under *Earthing*).

If, as shown in the diagram, the source equipment is earthed separately, a loop circuit is formed through the two earths and braids, which precipitates the flow of a minute 50Hz mains current.

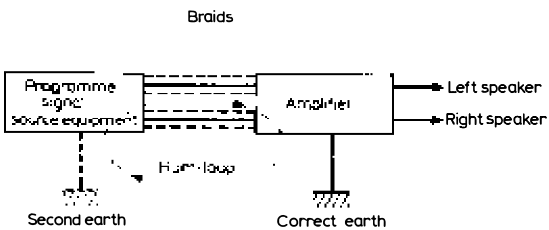


Fig. 1.18 The hum-loop condition illustrated. See text for details.

As this can reflect a small 50Hz voltage across the active source input of the amplifier, hum is superimposed on the programme signal. Thus a background hum accompanies the reproduction.

As the hum voltage is very small, the low-level magnetic pickup source is most susceptible to the effect since this input of the amplifier is the most sensitive. Harmonics of 50Hz (i.e., at 100, 150, 200Hz, etc.) may also enter the amplifier, and these can be responsible for a rough buzz background to the music.

If these sort of effects disappear when the volume control is turned down, then a hum-loop condition is almost certainly responsible, and attention should be directed to the earthing of the equipment. If the hum remains, however, when the volume control is fully retarded, faulty power supply smoothing in the amplifier should be suspected.

INTERFERENCE

Hum is one kind of interference and general background noise another (see under *Noise*). What is more commonly meant by interference, however, are the 'crackles' and 'pops' created by domestic electrical equipment and sometimes breakthrough of radio signals when the amplifier is operating from a magnetic pickup source, particularly when close to a powerful radio or television station.

Transistor amplifiers are especially vulnerable owing to their extended high-frequency (h.f.) response and non-linear transistor junctions which, under certain conditions, can act as radio detectors. Manufacturers are becoming increasingly aware of the problems and are overcoming them by screening and by the inclusion of r.f. filtering in the low-level stages.

Where such filtering is not fitted, however, some relief is often possible by connecting a small ceramic capacitor across the base and emitter terminations of the input transistors, as shown in Fig. 1.19.

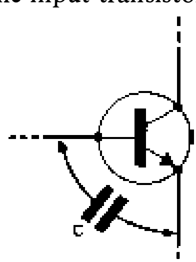


Fig. 1.19 Simple low-level stage r.f. filtering which can help to combat both r.f. and impulsive interference troubles.

In the case of stereo, both channels will require attention. The capacitor can have a value around 1 to 2nF, depending on the frequency or rise-time of the interference.

Inadequate screening of the signal cables or extra long loudspeaker cables can aggravate the trouble, and loudspeaker circuit r.f. filtering may be required in difficult cases. When the interference is mains-borne, additional power input filtering may be required, but this calls for qualified consultation.

MAINS ADJUSTMENT

Many amplifiers are equipped with a mains voltage selector either at the rear (see Fig. 1.20) or inside. The nominal UK mains is 240V (50Hz), and the closest setting to this (or to the known mains voltage) should be used.

Over-powering will result from a lower setting and under-powering from a higher setting, the former ultimately (or immediately!) causing damage to the amplifier and the latter restricting the power delivery.

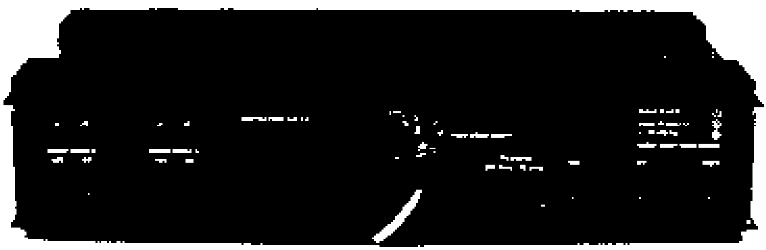


Fig. 1.20 Rear of a Sonab tuner-amplifier, showing mains voltage selector towards the middle of the rear panel. Notice also DIN source sockets (also see Fig. 1.6(b)).

MATCHING

This term, which in hi-fi parlance refers in general to the loudspeaker and source couplings, is rarely used correctly.

Loudspeaker 'Matching'

Almost all amplifiers will accommodate loudspeakers from 8 to 16Ω without trouble; and many will take from 4 to 16Ω. Owing to their low source impedance, hi-fi amplifiers are essentially constant voltage devices.

Consider an amplifier capable of producing 20V r.m.s. across 8Ω. Since power in watts is equal to the r.m.s. voltage *squared*, *divided* by the load in ohms, such an amplifier would deliver 50W at 8Ω. Assuming the same (constant) voltage, the power at 16Ω would be down to 25W, while at 4Ω it would be up to 100W.

Thus, while all the loads (loudspeakers) would be 'matched' those of lower impedance would extract the greatest power. This is the general principle, but in practice the load voltage tends slightly to rise with increasing load and fall with decreasing load, so that the power extracted is further modified.

Current (r.m.s.) flowing through the load is equal to the r.m.s. voltage divided by the load impedance or resistance, so in the case cited the current at 16Ω would be about 1·25A, at 8Ω about 2·5A and at 4Ω about 5A. Since the power transistors of all amplifiers cannot handle the large 4-ohm current, a restriction may be given in terms of minimum load impedance—i.e., 8Ω, as earlier noted.

If a 4-ohm load were connected, a protective device would be likely to operate or a fuse blow; but in the absence of protection the power transistors might well fail at full drive. See under *Protection*.

Source 'Matching'

The most important factor here is that the signal from the source is not wildly different from the amplifier's source input sensitivity. It is not necessary for the source impedance to 'match' the input load impedance of the amplifier.

A lower source impedance may be desirable, in fact, since this could reduce treble roll-off due to coupling lead capacitances. On the other hand, some sources, particularly the magnetic (and possibly ceramic) pickup, are sensitive to amplifier input load impedance, but these are arranged to be compatible by the designers.

For example, most magnetic pickup cartridges require a load from the pickup input of the amplifier of around 47–50kΩ. A lower load may roll-off the treble and a higher one lift the treble unnaturally. See also under *Signal Levels*.

NOISE

Noise in hi-fi usually refers to the background 'hiss' which emanates from the loudspeakers when the amplifier is operated at high gain (volume control fully advanced) with the programme source inactive. Noise is an electrical signal over a wide bandwidth of a random nature which is caused by the random movement of electrons in a passive or active circuit.

The very small noise signals are amplified and appear at the loudspeaker with the programme signal. However, under normal operating conditions the electrical noise power applied to the loudspeaker should not be much greater than 2×10^{-6}W. Noise performance is expressed by signal-to-noise (S/N) ratio. See under *Parameters*. A noise oscillogram is given in Fig. 1.21.

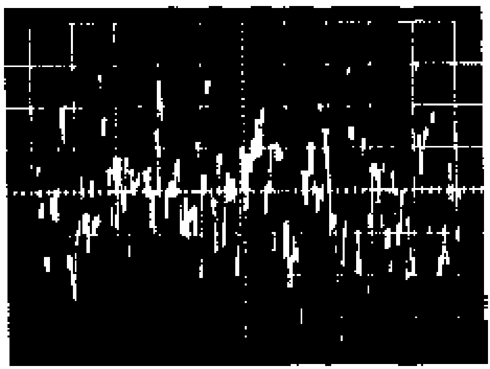


Fig. 1.21 This is what wideband noise signal looks like on the screen of an oscilloscope.

PARAMETERS

Parameters are items of a specification, and the main ones of hi-fi amplifiers are listed below.

Power Output

The full or rated power either over a range of frequencies or at a specific frequency (usually 1kHz) referred to a given load resistance and level of harmonic distortion. With stereo and four-channel amplifiers the per-channel (or total) power should be given with all channels operating simultaneously—a recent requirement of the American Federal Trade Commission (FTC).

The reference distortion is 0·5% or 1% or to waveform clipping threshold, and the power is average sinewave (often incorrectly called 'r.m.s. power'), being the product of the r.m.s. voltage across the load and the r.m.s. current through it (or V^2/R).

Other power expressions, such as 'dynamic', 'peak', 'music', etc. tend to confuse, as these are higher than average sinewave power. Thus when comparing amplifiers for power consider only average sinewave power ('r.m.s. power'), load value (see under *Matching*), frequency or frequency range of power parameter and whether the per-channel power is with one channel only driven or all channels driven together.

The per-channel power with one channel only driven is generally higher than that when all channels are driven together.

Power Bandwidth

Bandwidth over which a given power is delivered, generally referred to 0·5% or 1% harmonic distortion. *Half power bandwidth* is the bandwidth between l.f. and h.f. terminal frequencies where the power has fallen by 3dB.

Damping Factor

The ratio of the load resistance to the source impedance. Thus, if the amplifier's source impedance is 0·1Ω and the load impedance or resistance 8Ω, the damping factor is 80. A large ratio enhances, within limits, the electromagnetic damping of the loudspeaker.

Harmonic Distortion

This is generally expressed in percentage of the output power of measurement when all the harmonics are summed, giving total harmonic distortion (THD). Also see under *Distortion*. A good hi-fi amplifier may have a THD as low as 0·1% over the range 20Hz–20kHz.

Intermodulation Distortion

This is another expression of amplitude non-linearity and the measurement is made by an amplitude analysis of the intermodulation components when two signals of different frequency and specific amplitude ratio are fed simultaneously to the amplifier. Also see under *Distortion*.

The intermodulation distortion (IMD) is generally slightly higher than the THD, though with top-flight designs the two are approximately equal, depending on the nature of the IMD test.

Frequency Response

This is an expression of the deviation of amplification factor with reference to 1kHz, corresponding to a gain of 0dB, when the amplifier is passing a signal of low-level (*circa* 1W). Thus a frequency $\pm$1dB 20Hz–20kHz signifies that the output is constant within +1dB and −1dB, relative to 1kHz 0dB, over the range of 20Hz to 20kHz.

The *frequency range* is essentially meaningless since it states only the range of frequencies over which the amplifier is *responsive* without giving output or gain deviations.

Crosstalk

This refers to the amount of signal leaking from one channel or one circuit to another. The term *stereo separation* is more commonly used to define the crosstalk between stereo or 'quadraphonic' channels. For example, if the signal voltage output of a properly terminated non-speaking channel is 1,000 times below that of the speaking channel, the separation is a 1,000:1, which is 60dB in terms of voltage ratio.

Signal-to-Noise Ratio

This is the ratio in decibels between the rated output of the amplifier and the noise signal output. The measurement may be over the full bandwidth, or the measurement bandwidth may be modified by filters, the latter then giving the *weighted* signal-to-noise (S/N) ratio.

A weighted ratio is larger than an unweighted one of a given amplifier. A 20W channel of 70dB S/N ratio would yield a mere 2μW of noise power.

Hum and Noise

The same as S/N ratio, but the noise readout includes l.f. mains hum/ripple components.

Input Sensitivity

The level of source signal required at 1kHz in terms of V r.m.s. to give the rated output of the amplifier. Also see under *Source Inputs*.

Magnetic Pickup Input Overload Threshold

The amount of 1kHz signal (V r.m.s.) that the magnetic pickup input preamplifier will accommodate up to waveform clipping or a specified level of harmonic distortion.

The *overload margin* is the ratio of this input to the rated sensitivity in decibels. For example, if the overload threshold is 100mV and the input sensitivity 2mV, the overload margin is 50:1 or 34dB.

Tone Controls

See under *Controls*.

Filters

See under *Filters*.

PHASING

For correct stereo or four-channel imagery, correct phasing of the loudspeakers is necessary. Correct phasing implies that when all channels are simultaneously fed with a common signal, the cones of all the connected loudspeakers move in and out in unison. If one moves in while the other moves out the phasing between those two channels is in error.

It is impossible to check by cone movement, of course! One check is to place the two stereo loudspeakers close together, side-by-side, switch the amplifier to mono and play a record containing heavy bass—such as deep organ music. Misphasing is indicated by lack or thinness of very low bass. Changing round the connections of one of the loudspeakers will either bring the bass back or, if it was present in the first place, significantly diminish it.

The terminations of amplifiers and loudspeakers are identified (plus and minus, black and red, etc.) so that phasing can be correctly established in the first place. Also see under *Connections*.

It should also be remembered that the phasing can be destroyed by a reversed source connection (at the pickup, for example).

PROTECTION

Hi-fi amplifiers are protected by fuses, and when replacement becomes necessary the correct values *must* be used. Electronic protection may also be included. This sort of protection is activated when heavy signal current flows through the power transistors, such as would occur when driving hard into a very low value load (i.e., short-circuit).

Amplifiers with direct coupling from the power transistors to the loudspeakers may also include additional protection which automatically disconnects the loudspeakers in the event of a fault producing a rise in offset voltage. Without such protection d.c. would flow through the loudspeakers, and when high enough a fuse would blow.

QUADRAPHONY

See Chapter 4.

SIGNAL LEVELS

These generally refer to the 'average' levels of the source signals, and they should not differ wildly from the input sensitivities of the amplifier. For example, the average output from a magnetic pickup is 3–5mV, matched by the pickup input sensitivity of the amplifier. A radio tuner may have an average output around 150mV, and most amplifiers have a radio (tuner) input which will accommodate this.

In some cases attenuation is required to avoid the amplifier operating at full blast with only a small volume control setting. Amplifiers with preset level controls (see under *Controls—Preset*) can be adjusted accurately to the source signal level.

SOURCE INPUTS

Most amplifiers have inputs suitable for accepting source signals from a magnetic pickup, sometimes ceramic pickup, radio tuner, tape recorder and auxiliary source, such as second radio tuner, perhaps ceramic pickup, etc.

The sensitivity of each input is engineered to correspond fairly closely to the signal level of the source (see under *Signal Levels*), and each source input represents a suitable load for the source. Although there are no absolute standards in these respects, there is reasonable correlation between the items of equipment from various manufacturers.

The various sources, with the exception of tape monitor, are selected by the source switch on the front panel, which may be rotary or press buttons or keys. The tape monitor facility is commonly under the control of a press button or key, and when this is operated the control section is disconnected from the power amplifier section (all channels simultaneously).

The diagram in Fig. 1.22 shows the basic scheme on one channel.

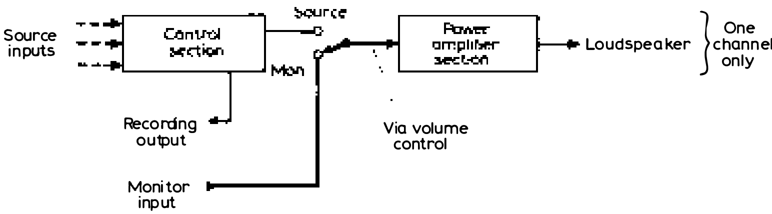


Fig. 1.22 Showing how the tape monitor switch operates. The disconnection of the control and power amplifier sections provided by this switch is sometimes exploited for the connection of a matrix 'quadraphonic' decoder/amplifier (see Chapter 4).

In the monitor mode, therefore, signal from the replay head of a tape recorder can be fed direct to the power amplifier, usually via the volume and tone controls, while at the same time the selected source signal from the control section can be applied to the recording head, assuming, of course, that the tape machine is equipped with separate recording and replay heads.

This, then, makes it possible to compare the source signal with that just recorded on the tape by operating the button between source and monitor positions.

TAPE MONITORING

See under *Source Inputs* (also Fig. 1.22).

TAPE RECORDING OUTPUT

From 'phono' type tape recording output sockets the signal level is around 200mV, but to cater for the input sensitivity of the different recorders, the level may be adjustable by a preset control. The Armstrong Model 621 amplifier, shown in Fig. 1.23, is one example with such a level preset; the tone controls also operate on the recording signal with this amplifier.

Fig. 1.23 Armstrong 621 amplifier which features a tape recording level preset.

When DIN socketry is used, and there may be a DIN socket in addition to 'phono' type sockets for a tape machine, the DIN socket accepting tape relay signal as well as delivering recording signal (see Fig. 1.7), the level of the recording output signal is lower.

The level, in fact, depends on the input load of the recording machine, the higher the load value, the greater the signal level. This is because so-called 'constant current' coupling is used, which merely constitutes a high value resistor in series with the recording signal output, as shown in Fig. 1.24.

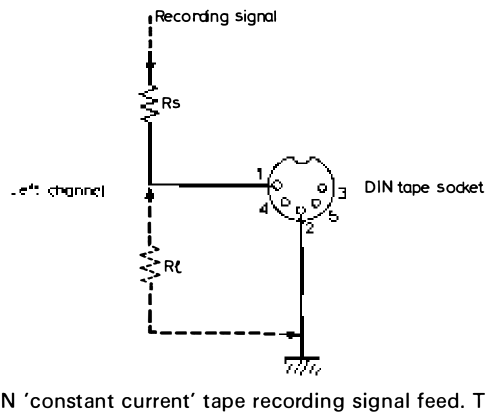


Fig. 1.24 DIN 'constant current' tape recording signal feed. The signal is fed to the appropriate pin on the amplifier DIN socket through a high value resistor R_S, and when the tape recorder is connected the input load corresponds to R_L. Thus the recording signal is divided down by the potential-divider effect of R_S and R_L.

The DIN 'standard' in this respect is 0·1–2mV of recording output for every 1kΩ of load resistance, represented by R_L in Fig. 1.24. An output of 40mV at about 50kΩ is fairly common. See the author's *Audio Technicians Bench Manual*.

CHAPTER TWO

LOUDSPEAKERS

ACOUSTICAL RESISTANCE UNIT

THIS IS A DEVICE, sometimes called ARU for short, pioneered by Goodmans, which fits over the normal port or over an extra vent on a reflex loudspeaker enclosure. One effect is to reduce the sound emission from the port; but the most important advantage is that it provides even damping of the two bass peaks which are characteristic of the reflex type of enclosure.

It also makes possible the design of a reflex enclosure of smaller air volume than is otherwise feasible, and a regulating factor is imposed upon the resonant or Q characteristics of the loudspeaker system. See also under *Reflex Enclosure.*

AUXILIARY BASS RADIATOR

This is another device associated with the reflex enclosure principle, called ABR for short. It takes the place of the usual port and is of the form of a cone/diaphragm of carefully controlled mass and compliance such that it is activated by the air pressure built up at bass frequencies in the enclosure (Fig. 2.1).

It comes into play at a low frequency, determined by its design and that of the enclosure, and at the frequency where the bass output would normally be falling off the ABR itself commences to vibrate in phase with the cone of the bass driver (woofer) so that the bass delivery is augmented.

Since it permits the displacement of a greater volume of air, the enclosure dimensions can be made smaller than those of an ordinary reflex enclosure without undue loss of bass response. The principle is adopted by Goodmans, Celestion, Grundig and Lowther. See also under *Enclosure.*

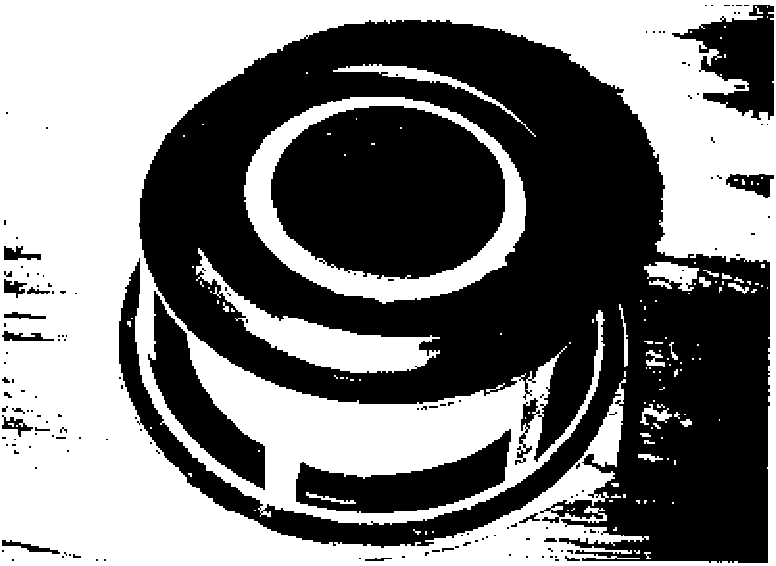


Fig. 2.1 Auxiliary Bass Radiator (ABR) as used in the Celestion Ditton 15 loudspeaker system.

ACOUSTICAL SUSPENSION

This is a type of loudspeaker system in which the bass driver (woofer) is loaded acoustically by the air trapped in the sealed enclosure. In other words, the main compliance for the cone is provided by the air in the enclosure rather than by restoring force devices of the driver. The acoustical stiffness added to the cone-restoring stiffness raises the intrinsic bass resonance frequency of the driver.

Thus, for a good bass response the driver must possess a natural free-air resonance of 30Hz or less since the method of loading will increase this significantly, particularly in a small volume enclosure. At higher frequencies, the enclosure exhibits resonance modes, rather like standing waves in a room, and these are damped by lagging the enclosure internally with acoustically absorbent material (Fig. 2.2).

In some designs, additional damping is provided by lagging on the sides of the enclosure and by suspending bonded acetate fibre (BAF) or similar acoustically absorbent material down the centre.

The craft of good design lies in the correct marrying of enclosure to driver or *vice versa.* Over the years some very good examples of this type of loudspeaker system have emerged—some remarkably

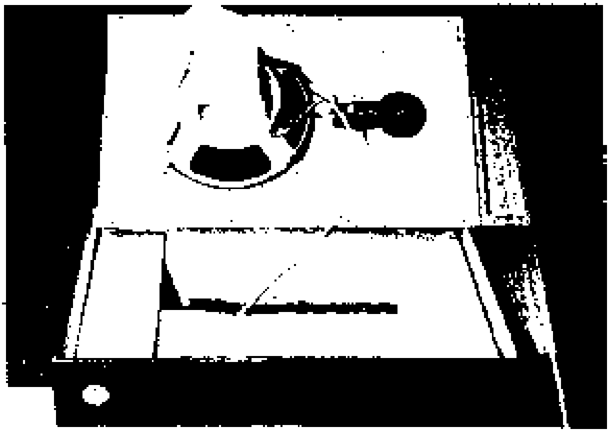


Fig. 2.2 Example of acoustical damping in small 'infinite baffle' loudspeaker system.

small for the bass they are able to provide. Most of the contemporary systems of smaller size adopt the acoustical suspension principle and amongst them are some very good and very bad examples.

The term 'infinite baffle' (IB) is commonly used to describe this type of system because owing to the sealed loading there is no air

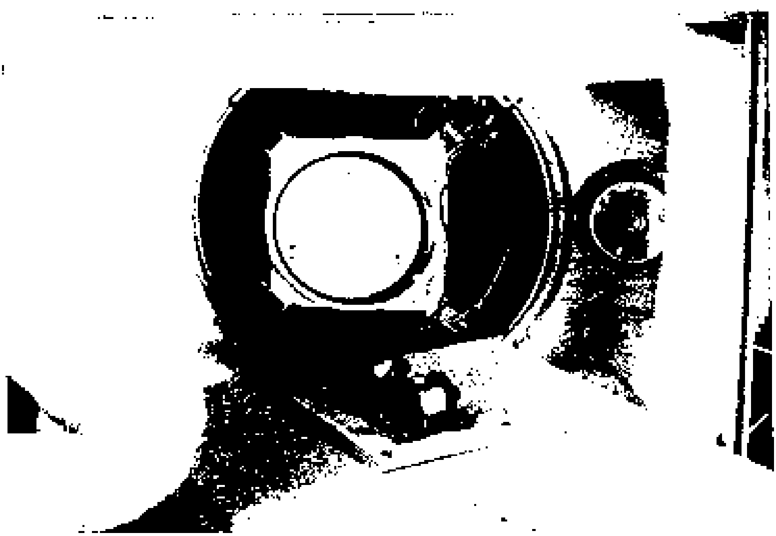


Fig. 2.3 Example of inside of 'infinite baffle' loudspeaker system, showing frequency-divider.

path between the front and rear of the driver cone, an effect which, of course, would also be obtained with a flat baffle of infinite size.

One example of the inside of the acoustical suspension system is given in Fig. 2.2. Another is given in Fig. 2.3, also showing the frequency-divider.

BAFFLE

The simplest baffle consists of a flat piece of wood with a hole in it behind which the loudspeaker unit is mounted, as shown in Fig. 2.4.

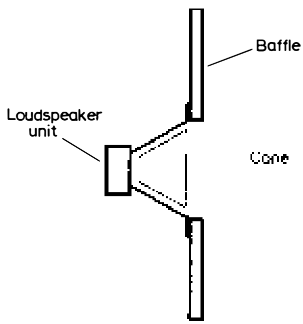


Fig. 2.4 Flat baffle which prevents low-frequency sound from the rear of the unit cancelling out sound from the front of the unit.

Without some kind of baffle, sound waves from the rear of the cone would interfere with those simultaneously produced from the front of the cone.

The effect is that cancellation of sounds of lower frequencies occurs owing to the rear sounds being out-of-phase with the front sounds. The baffle increases the sound path length from the rear to the front of the cone and therefore reduces such cancellation effects.

The cut-off wavelength of a baffle is about 21m, corresponding to a frequency of about 16Hz, but baffles of smaller size will, in conjunction with the driver, roll-off the bass earlier, at a rate of 12dB/octave below the critical frequency.

Clearly, the use of such large baffles, particularly for stereo and four-loudspeaker reproduction, is impractical in the contemporary domestic scene, hence the need for a different, smaller type of enclosure which facilitates a reasonable bass response. However, some enthusiasts have used a dividing wall between two rooms, for example, as a large baffle to good effect, but a more scientifically derived enclosure is generally required for the best acoustical loading of today's drive units.

COLOURATION

The aim of the hi-fi enthusiast is to reproduce sound with the least change from the original. So many factors affect the reproduced sound, including shortcomings in the recording/replay processes, various distortions—albeit, small ones—in the amplifier and loudspeakers and the changes in the tonal quality imparted by the acoustics of the listening room and the loudspeakers themselves.

The term colouration often refers to an alteration in the character of sound for which the loudspeakers are responsible. All loudspeakers yield their own blend of colouration, though some designs are far less 'coloured' than others, which is why loudspeakers of different type, make, etc. tend to 'sound' differently in a given environment and when energised from a common signal source.

The preference of a critical listener is for a loudspeaker system with the least obtrusive of colouration, which generally resolves to a design in which the resonances have been carefully tamed (by good design) and whose frequency response characteristic is free from violent undulations. Certain parts of the spectrum may be particularly coloured, giving coin to such terms as bass colouration, middle colouration, etc.

The listening room, too, can be responsible for colouration of the reproduction owing to resonances, and as this can work with or against the colouration of the loudspeakers it is desirable, wherever possible, to try out a loudspeaker (pair or quadruple) in the intended listening room before making a final decision to purchase.

Pink noise is often used to detect undue colouration due to response peaks, dips and resonances, but this test requires a 'control' loudspeaker to compare any differences in noise output against.

CONE BREAKUP

Usually, at low frequency, the cone of a loudspeaker unit tends to move as a whole—up and down, rather like a piston or air pump. At higher frequencies, the inner area of the cone becomes more inclined to vibrate in its own mode relative to the outer area, the cone then ceasing to move as a whole.

As the frequency is raised, a multitude of different, unconnected vibratory modes may occur over the entire cone. This, called cone breakup, manifests as mild or severe undulations in the frequency response characteristic and impairs the transient performance.

Cones are designed to avoid this unfortunate effect as much as possible, and by employing two or more units, each for a part of the frequency spectrum, cone breakup troubles are further minimised.

Cone rigidity is a factor required to reduce cone breakup, but to cater for high frequencies the mass must of necessity be small. A cone designed with these aims in view is that in the Leak Sandwich loudspeaker system, based on a 'sandwich' of thin lightweight metal and plastics material.

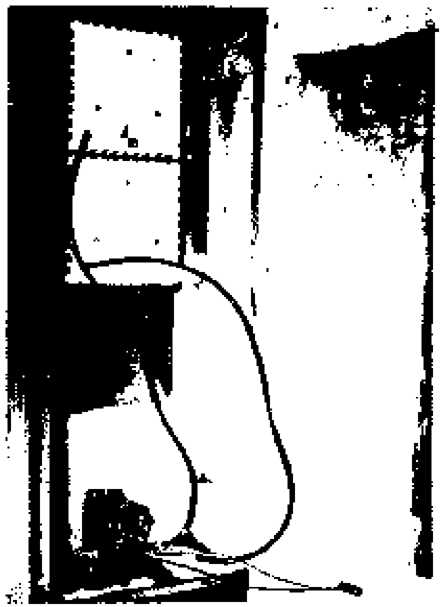


Fig. 2.5 Two Jordan-Watts 'Module' loudspeaker units mounted in vented enclosure. These units are full-range and use lightweight metal cones of special design such that as the frequency is raised a reducing inner area of cone is utilised, with automatic decoupling from the outer area.

Some cones are designed deliberately so that as the frequency is increased the inner area becomes progressively more decoupled from the outer area. Thus, at low frequency the whole cone area is utilised, while at high frequency a smaller inner area only is utilized, the outer area then being effectively 'decoupled'. The Jordan-Watts 'Module' loudspeaker units (Fig. 2.5) adopt this principle, the cones being specially engineered of lightweight metal.

CROSSOVER UNIT

This is a form of frequency-divider (see under *Frequency Divider*) which splits the loudspeaker signal into two frequency parts, one for

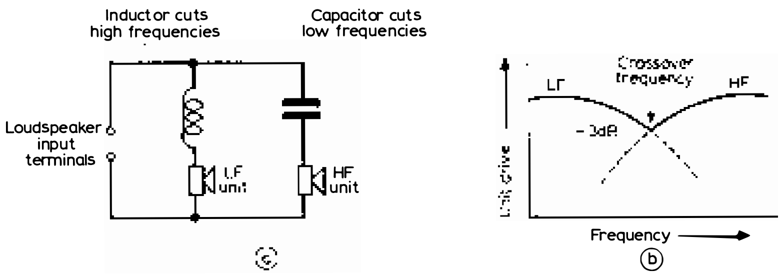


Fig. 2.6 Simple crossover circuit (a) and its nature of response (b). With 15Ω units a 5kHz crossover frequency can be achieved by making the inductor 0·5mH and the capacitor 2μF.

operating the low-frequency unit and the other for operating the high-frequency unit, as shown in Fig. 2.6.

CUT-OFF FREQUENCY

This is the frequency at the effective limit of the spectrum, higher or lower, and is commonly the frequency where the output has fallen by 3dB (half power) relative to the 'nominal' output. In the case of a loudspeaker, the lower cut-off frequency may be defined as in Fig. 2.7. Another term is turnover frequency.

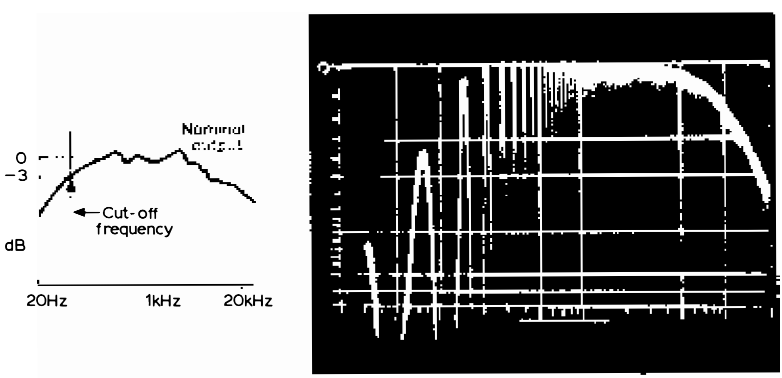


Fig. 2.7 (a) Definition of cut-off frequency. (b) Sweep-frequency response of f.m. tuner (sweep 20Hz–20kHz) which shows the —3dB point at approximately 15kHz.

DAMPING

To avoid resonance peaks, resistive damping is applied to electronic circuits and acoustical and/or mechanical damping to *transducers*, such as loudspeakers, gramophone pickups and microphones. To reduce the effects of resonances in loudspeaker enclosures acoustical damping is commonly used.

There are various acoustically absorbent materials, including bonded acetate fibre (BAF), which can be applied to the inner surfaces of the enclosure and rolled or suspended inside to inhibit standing waves which might otherwise be responsible for peaks in the response at the middle and high frequencies (see Figs. 2.2 and 2.3, for example).

DAMPING FACTOR

The loudspeaker is also damped electromagnetically. If we take a moving-coil loudspeaker unit and deflect the cone downwards by applying pressure with a finger, we will find that significantly more pressure is required to secure a given deflection when a short-circuit is applied across its terminals than when the terminals are open-circuit.

The reason for this is that when the cone is deflected, the moving-coil cuts the magnetic lines of force and an e.m.f. (electromotive force) is generated in the winding, rather like a dynamo, and a short across the terminals (and hence the moving-coil) results in a flow of current through the coil. This in turn produces a magnetic field round the winding which counteracts with the field from the unit's magnet such that more pressure is required to obtain a deflection.

This is the basic principle of electromagnetic damping, but clearly it is impossible for us to operate with a short-circuit across our loudspeakers, for to do so would bypass all the signal power and blow the amplifier's fuses or output transistors!

Instead, we arrange for our amplifiers to possess a low source impedance: that is, the impedance 'seen' by the loudspeaker when it is connected to the amplifier. The source impedance is a function of the design of the amplifier and the negative feedback used to reduce the distortion and smooth the frequency response.

Some amplifiers have a source impedance as small as $0{\cdot}1\,\Omega$ which, from the loudspeaker's point of view, would be virtually a short-circuit. The damping factor is the ratio of the rated load impedance of the amplifier to the source impedance. Thus, if the rated load is $8\,\Omega$, and this is the load value connected to the amplifier and involved in the source impedance in easutment, the damping factor would be 8/0·1, or 80.

However, in practice things are not as clear-cut as this because the loudspeaker does not provide a resistive load over its entire frequency range and its impedance tends to alter with frequency, being nominally resistive around 400Hz. Moreover, the loudspeaker unit is separated from the source impedance of the amplifier by the resistance of its connecting cable and possibly by the impedance of the crossover unit or frequency-divider. Hence, the source damping applied to the unit will be significantly less than the source impedance of the amplifier.

Nevertheless, the damping factor of an *amplifier* is a useful parameter to know, particularly when measured at low-frequency as

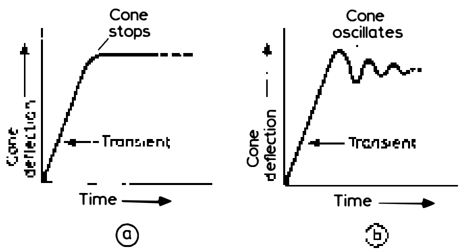


Fig. 2.8 The cone of a loudspeaker with optimum damping will follow a transient as at (a), while inadequate damping tends to encourage overshoot and 'ringing' (b).

per BSI (British Standards Institution) because it is generally the low-frequency unit which requires most damping of this kind, since it is one expression of the low-frequency coupling and feedback performance attained by the designer.

Looking now at the loudspeaker unit under dynamic conditions, when this receives a transient signal there is a sudden and swift deflection of the cone. As soon as the transient ends the cone should immediately come to a standstill, as shown at (a) in Fig. 2.8.

Without sufficient damping, however, the cone will tend to overshoot and oscillate to and fro with diminishing vigour, as shown at Fig. 2.8 (b). The overshoot and damped oscillation is a form of transient distortion which impairs the tonal quality of the reproduction, sometimes being responsible for 'boomy' bass or so-called 'one-note bass', and causes 'over-hang' effects.

There is reason to believe that a negative amplifier source impedance (provided it is not so negative as to result in amplifier instability or oscillation) can sometimes be used to advantage to counteract the positive impedance of the loudspeaker cables and cross-over unit or frequency-divider to improve the electromagnetic damping of some loudspeaker systems, but this is an area of hi-fi which warrants more detailed attention.

It should be understood, of course, that the low source impedance of an amplifier itself does not consume power. From the damping point of view the e.m.f. produced by overshoot or 'ringing' tendency (Fig. 2.8(b)) operates the negative feedback circuit of the amplifier in a manner which inhibits oscillatory movement of the cone—a kind of 'servo' control.

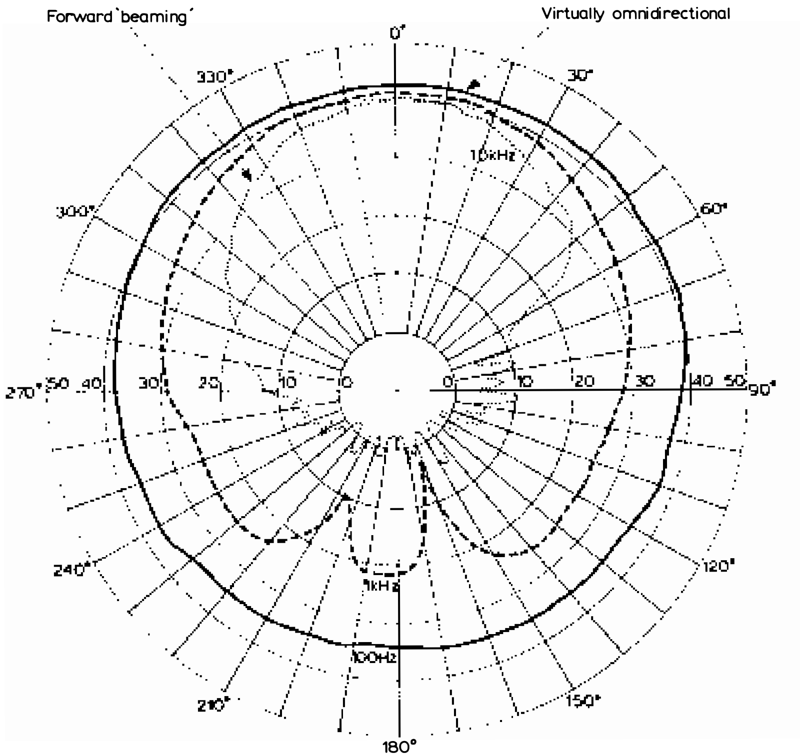


Fig. 2.9 Example 360° polar diagram, showing the directivity at three different frequencies.

DIRECTIVITY

At low-frequency, the sound radiated from a loudspeaker system is essentially omni-directional, meaning that its output is equal in all directions. As the frequency rises so the angle of radiation tends to diminish, until at high treble the sound is virtually 'beamed' forward over a small angle.

Good high-frequency dispersion of sound is the aim of most designers, for excessive beaming with increasing frequency can affect stereo reproduction by causing unnatural wandering of the sound image.

A polar characteristic shows the output on a two-dimension 360° or 180° scale, and such plots at different frequencies show how the dispersion decreases as the frequency is raised (Fig. 2.9). Few loudspeaker manufacturer's, though, issue plots of this kind.

DISTORTION

A loudspeaker system can suffer from various kinds of distortion, including transient (see under *Colouration*, *Hangover* and *Damping Factor*) frequency and non-linear. Frequency distortion manifests as undulations in the pressure response characteristic and as bass and treble roll-off while, as with amplifiers, non-linearity of the transfer characteristic results in harmonic (see under *Frequency Doubling*) and intermodulation distortion.

Although the fundamental electromagnetic deflection of the cone is linear, non-linearity can result towards the extremes of deflection owing to 'tightening' of the suspension (Hook's Law not followed) and to the speech coil moving into a zone of weaker magnetic field. Hence non-linear distortion is more likely when a loudspeaker is driven hard.

Distortion effects can also occur owing to inadequate damping of enclosure resonances, particularly internal resonances which pass through the cone of the bass driver. Excessive distortion yields a general 'mushiness' combined with lack of clarity.

Intermodulation distortion can also cause a high-frequency note to be modulated by a low-frequency one. *Doppler distortion* refers to a change of pitch when the cone radiating high-frequency has a large low-frequency deflection.

DUAL-CONCENTRIC

This is a type of loudspeaker unit in which the low-frequency and high-frequency cones or diaphragms are mounted concentrically in a common housing but driven, via a suitable crossover unit, separately.

DUAL CONE

This is a loudspeaker unit which has a large cone for low/middle frequencies and a small, rigid cone for the high frequencies, both driven from a single moving-coil assembly. The term *double cone* is sometimes used.

At low frequencies the large cone is operative, the small one having little influence on the sound output. At high frequencies, however, the lower mass, more rigid cone takes over so that the frequency response is extended. The technique allows one common unit to handle the full range of music frequencies.

EFFICIENCY

This refers to the ratio of acoustical power (P_a) delivered by a loudspeaker to the electrical power (P_e) applied to it. For example, per cent efficiency equals $P_a \times 100/P_e$. Thus, if the electrical power needs to be, say, 8W to yield 25mW acoustical power (which is fairly typical of contemporary loudspeaker systems), then the efficiency would be 0·3125%.

The acoustical power is not particularly easy to measure, but based on a hemispherical free-space condition (where the underside of the loudspeaker is prevented from radiating (such as if the loudspeaker were buried in the ground up to the surface of the front panel) the low-frequency acoustical power is close to 25mW when the sound pressure is 96dB at 1m from source (see also *Sensitivity*).

ELECTROSTATIC LOUDSPEAKER

This is a kind of loudspeaker system or unit whose *modus operandi* is based on the electrostatic principle, rather than (as with the moving-coil unit) on the electromagnetic principle. A very well known example is the Quad Electrostatic (full range doublet) by the Acoustical Manufacturing Company Limited.

For further information on this principle refer to the author's book entitled *Pickups and Loudspeakers* by the publisher of this book.

ENCLOSURE

An enclosure is the box in which the driver units are mounted. It provides the 'baffling' for the low-frequency unit and its design is critical. For the best results its nature and dimensions have to be related to the low-frequency driver (woofer). See also under *Acoustical Suspension*, *Infinite Baffle*, *Reflex*, etc.

FREE-AIR RESONANCE

The mass of a loudspeaker cone or diaphragm resonates with its compliance at a frequency corresponding to $1/2\pi\sqrt{MC}$, where M is the mass and C the compliance (in SI units expressed respectively in terms of 10^{-3}kg and 10^{-3}m/N, where m/N is metre/Newton). Thus the low-frequency resonance of the bass driver would be a trifle above 11 Hz based on a mass of 10×10^{-3}kg and a compliance of 20×10^{-3}m/N.

The impedance versus frequency characteristic of such a unit, therefore, may appear as in Fig. 2.10, where a significant peak in

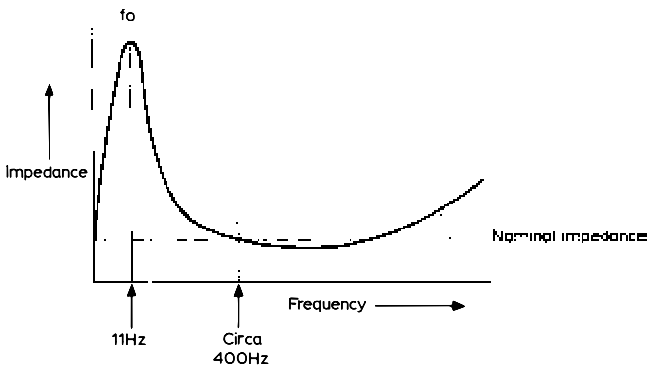


Fig. 2.10 Impedance versus frequency characteristic of loudspeaker unit without baffling. The resonance frequency (f_o) rises when the unit is mounted in, for example, an 'infinite baffle' enclosure owing to the air in the enclosuring adding to the cone-restoring stiffness of the unit (see text).

impedance occurs at the resonance frequency. This is the free-air resonance of the low-frequency unit.

Now, when the unit is loaded acoustically into a suitable enclosure the air therein adds acoustical stiffness to the cone-restoring stiffness thereby *reducing* the compliance (because compliance is the reciprocal of stiffness) and hence increasing the resonance frequency.

Most forms of 'baffling', with the exception of the acoustical labyrinth, increase the bass resonance frequency, the acoustical suspension or 'infinite baffle' being particularly notable in this respect. See under *Acoustical Suspension.*

FREE-FIELD

This refers to the acoustical environment where sound waves are permitted unrestricted propagation from a sounding source. Free-field conditions can occur in the open air when the sounding source is clear of impeditive and reflecting surfaces.

Certain loudspeaker measurements are often required under free-field conditions, but, owing to the difficulties of making such measurements in the open air, free-field conditions are often simulated in a specially prepared room or chamber designed to possess acoustically absorbent boundaries. Such an environment is called *anechoic*, meaning devoid of reverberation.

FREQUENCY-DIVIDER

This is a passive or active (the latter with transistors or valves) filter which divides the audio signal applied to a loudspeaker system into smaller bands of frequencies for operating two or more loudspeaker units. When the loudspeaker system features two units, one for bass and the other for treble, the term *crossover unit* is sometimes adopted (see under *Crossover Unit*).

A three-way frequency-divider block diagram is given in Fig. 2.11.

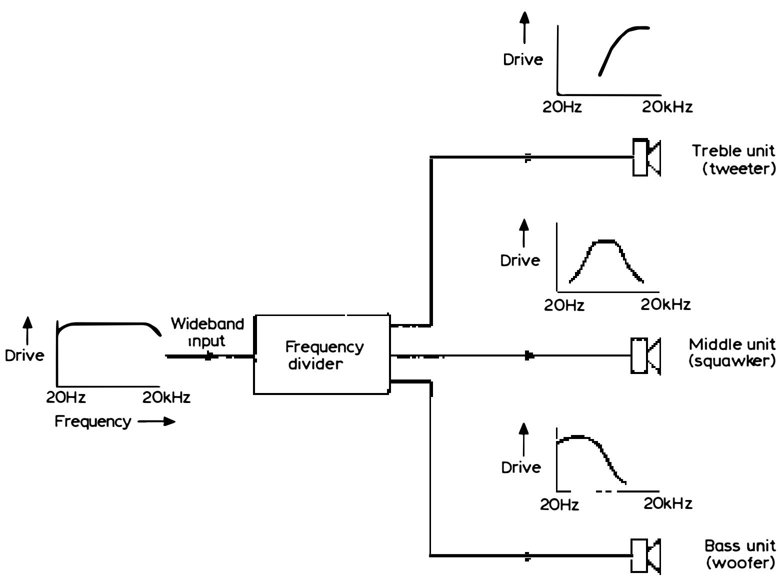


Fig. 2.11 Block diagram of frequency-divider, showing filter characteristics.

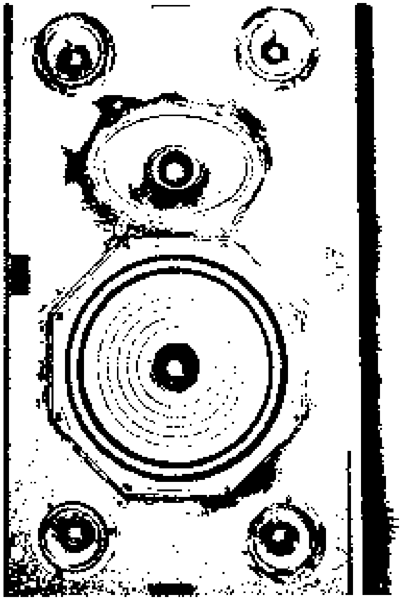
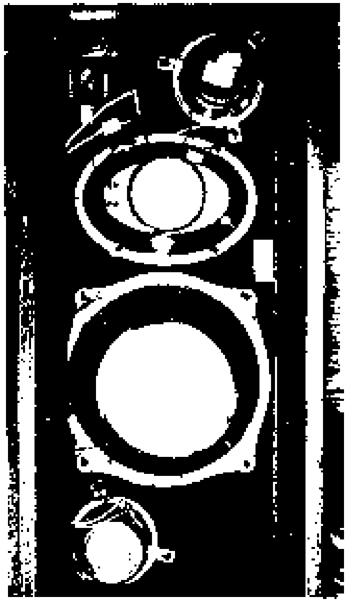



Fig. 2.12 (left) Loudspeaker system having six drive units. Fig. 2.13 (right) Inside of four-unit loudspeaker system, showing frequency-divider in one corner.

This is the sort of thing found in many loudspeaker systems using three drivers, the frequency dividing being done 'passively'; that is, with inductors and capacitors. There are three filter sections—high-pass for the treble unit, bandpass for the middle unit and low-pass for the bass unit.

Design is critical since the filters have to suit the impedances of the drive units and the turnover frequencies have to be carefully selected to avoid undue drive power being passed to a unit at a frequency outside its normal handling range. Moreover, the integration of the three outputs needs to result in a pressure response characteristic free from peaks or dips or violent undulations at the crossover frequencies.

The filters may be single-pole, giving 6dB/octave rate of roll-off, or two-pole, giving 12dB/octave rate of roll-off. In some designs single- and two-pole filters may be used.

With some loudspeaker systems it is possible to adjust the energy applied to the treble unit, and also sometimes the rate of treble roll-off, to secure the best balance in the listening room.

An active frequency-divider may have similar filter characteristics, but valves or (more usually) transistors are used in the design of the filters. The dividing in this case is generally done at low power, the outputs from the filter sections then feeding power amplifiers, one for each range of frequencies, which in turn feed the separate loudspeaker units.

When more than three drivers are incorporated in the system design a more elaborate frequency-divider may be used or two or more of the units may be connected in parallel, depending on the design. Fig. 2.12 shows one design using six drivers. The inside of another system employing four drivers, showing the passive frequency-divider in one corner, is depicted in Fig. 2.13.

FREQUENCY DOUBLING

When an input close to the bass cut-off frequency is applied to some loudspeakers, an output is produced which has twice the frequency of the input signal. This is frequency doubling, and results from non-linearity in the electrical input/acoustical output transfer characteristic.

At low-frequency and at fairly high power the cone deflection can be quite large. Towards maximum deflection, the proportional deflection with signal current drive may not hold, and the resulting non-linear deviation tends to produce second harmonic distortion, which is responsible for the frequency doubling.

The effect is aggravated when extra heavy drive is applied in an endeavour to get a small, intrinsically bass-light loudspeaker system to produce low bass, such as by turning on excessive bass boost at the amplifier. Some loudspeaker systems are more susceptible to the effect than others, particularly when the low-bass acoustical loading is incorrectly engineered or inadequate.

FREQUENCY RESPONSE

This is a sound pressure output versus frequency plot of a loudspeaker system, as shown in Fig. 2.14. There are various conditions of test, a primary one being that the loudspeaker should be measured under free-field conditions (i.e., in an anechoic chamber) or in such a manner that the acoustical effects of the test environment are not reflected into the response characteristic. The main measurement is made on axis at a given distance from the loudspeaker and at a stated power.

Obviously, the acoustical characteristics of the listening room will

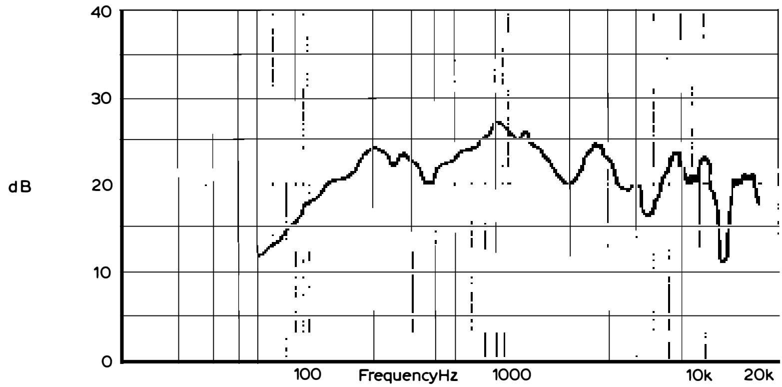


Fig. 2.14 Pressure-versus-frequency plot of Jordan-Watts 'Juno' loudspeaker system which is based on a reflex enclosure. This was taken at 1 m from the loudspeaker at a power of 2 watts with Brüel and Kjoer automatic pen recording instrumentation.

influence the inroom sound pressure, so a basic frequency plot made under free-field conditions will have little correlation with the practical listening conditions.

Attempts are being made to secure improved correlation by measuring under reverberant conditions to simulate the 'average' listening room, but so far the best appraisal of a loudspeaker is to listen to it under the proposed domestic conditions.

Nevertheless, the pressure response curve is useful for evaluating specific design features, particularly associated with the 'crossover' points and the integrated sound pressure of a multi-unit system.

HANGOVER

This is a form of transient distortion resulting from inadequate damping, particularly at low frequency, as explained under *Damping Factor*.

HORN

A horn sometimes replaces the conventional loudspeaker enclosure. The driver unit is coupled to the throat and the mouth delivers sound over a frequency range for which the horn is designed. A horn can be regarded as an acoustical transformer which converts high pressure, low velocity, sound at the throat to high velocity, low pressure, sound at the mouth, the latter constituting a more efficient coupling to the surrounding air of the listening area.

A horn loudspeaker has an efficiency of some 30 to 50% within its operating frequency range, which is astonishingly high compared with the 1% or less efficiencies of 'box' enclosed loudspeaker systems. Such high efficiency, therefore, makes it possible to achieve hi-fi-scaled sound intensities from a moderately-powered amplifier, such as of Class A biasing, which may be preferred by some critical listeners.

The transformer effect results from the nature of the horn, which has a cross-sectional area expanding in accordance with a mathematical law, such as parabolic, conical, hyperbolic, exponential, etc. The exponential contour is that commonly adopted for hi-fi horn loudspeakers, and the ideal horn in this respect would be of straight circular tube of logarithmically expanding cross-sectional area from a small throat to a large mouth.

To load the driver at low bass frequencies, the horn requires a mouth of some 2 to 3 square metres and a length of some 7 to 9 metres, which puts a pair in this form outside the scope of the contemporary hi-fi listening room!

Several compromises, therefore, are adopted, including square cross-section construction and folding into a cabinet of more conventional shape, though of significantly larger dimensions than contemporary loudspeaker systems based on the 'infinite baffle' or reflex principle.

By placing a specific tailoring of a folded horn in the corner of a room, the latter part of the flare is effectively extended by the walls, ceiling and floor, thereby lowering the cut-off frequency, so that a design of more acceptable dimensions and yet with an acceptably low bass response can be achieved.

Such comprises, however, detract from some of the desirable features of horn loading unless very carefully handled, and in general a horn of compromise design cannot be expected to perform as well as one of optimum design.

To avoid too many compromises, some enthusiasts build low-frequency horns actually into the listening room, in some cases utilising room corners, and in other instances building the horns below floor level and exhausting the bass into the room through vents in the floor!

When a driver unit is loaded by a horn, the movement of the cone is much less and the acoustical damping much greater than provided by some other methods of loading for a given acoustical power output. Thus horn loading tends to reduce non-linear distortion and enhance the transient performance.

It is a revelation to listen to the 'true' bass delivered by a well designed horn in comparison to the 'woolly' or 'honky' bass delivered by some of the less well engineered smaller makes of loudspeakers!

Since the cut-off frequency is a function of horn dimensions, middle- and high-frequency horns represent far less of a space problem, and horn loading of high-frequency units is commonly practiced. Some middle-frequency units may also have similar loading, and to achieve good sound dispersion multi-cellular construction may be adopted. Alternatively, diffusing lenses may be used with high-frequency horns to spread the sound over a wider angle.

IMPEDANCE

This is the opposition presented by a circuit to a flow of alternating current, such as signal current, and it results from the complex integration of inductive reactance ($X_L+2\pi fL$), capacitive reactance ($X_c=1/2\pi fC$) and pure resistance (R), such that impedance (Z) equals $\sqrt{X^2+R^2}$), where f is the frequency in Hz, L the inductance in Henries, C the capacitance in Farads, R the resistance in ohms and X the integration of X_L and X_c.

Since the components of a loudspeaker system include inductance (of the speech coils and crossover or divider inductances), capacitance (of the crossover or divider capacitances and the winding capacitances of the inductive elements) and resistance (of the wire of the inductive elements, etc.), it follows that the terminals will reflect a given value of impedance to the amplifier.

Like resistance and reactance, impedance is measured in ohms, and, because it is frequency dependent, the value of Z will not be the same at all frequencies applied to the loudspeaker. The impedance is a loudspeaker parameter, and the value quoted can thus only be nominal. It is generally the impedance at 400Hz or 1kHz (see Fig. 2.10).

As shown in Chapter 1, some amplifiers are critical to loudspeaker impedance, more power usually being delivered into lower values than high ones, but a too low value can precipitate protection device operation, or fuse or transistor failure. By applying a simple ohm-meter across the terminals of a loudspeaker system or unit, the value read is the d.c. resistance *not* the impedance.

INFINITE BAFFLE

This is the term commonly used to describe a loudspeaker loading where there is no free path between the front and the rear of the driver cone. It can also refer to a flat baffle of 'infinite' size. See also under *Acoustical Suspension*.

ISOTHERMAL

The presence of damping material inside a loudspeaker enclosure yields an *isothermal* (i.e., constant temperature) effect owing to heat storage resulting from air pressure changes. Without damping an *adiabatic* (i.e., without heat transference) condition obtains.

As the effective compliance due to the air is increased by a factor of about 1·4 from the adiabatic condition to the isothermal condition, it follows that the damping material will increase the effective volume of the enclosure and hence the bass resonance by about 17%. See, for example, under *Acoustical Suspension*.

LABYRINTH

This is a type of loudspeaker enclosure where the rear of the driver cone drives along a long folded tube whose mouth exhausts at the bottom front of the enclosure or underneath to deliver the bass frequencies, the middle and treble frequencies emanating from the front of the cone, since the driver is usually mounted over an aperture on the front panel, as shown in Fig. 2.15.

The inside surfaces are lagged with an acoustically absorbent

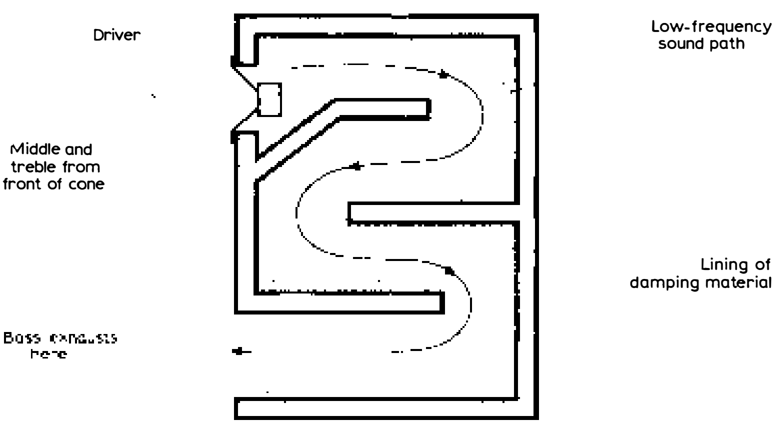


Fig. 2.15 Principle of acoustical labyrinth enclosure. See text for more details.

material to avoid upper-frequency standing waves and associated resonances, and the overall length of the 'tube' is arranged to suit the loaded resonance of the driver. For a linear frequency response, the frequency corresponding to the wavelength of four times the length of the 'tube' should fall close to the loaded resonance of the driver.

For example, if the 'tube' is, say, 2·5m in length, four times this will be 10m, and the frequency corresponding to 10m is 34·4Hz (i.e., velocity of sound taken as 344 metres/sec. divided by 10), so the loaded resonance of the driver should be around 34Hz.

This method of loading tends to reduce the natural (unloaded) resonance of the driver; if the loading is, say, for 40Hz, then the natural, unloaded resonance of the driver would be around 50Hz.

Fig. 2.16 shows the driver end of an acoustical labyrinth type of enclosure (LC94 by Stentorian–Whiteley Electrical (Radio) Company Limited) in which the acoustical damping material is clearly visible.

Fig. 2.16 The drive unit end of the Stentorian LC94 labyrinth enclosure, showing the acoustical damping material lining the 'tube'.

LOUDSPEAKER UNIT

This refers to the transducer or driver which is loaded acoustically by means of a horn, baffle or other type of enclosure. Units are made for bass, middle, treble and super-treble frequency ranges. Others, like dual-cone units and the Jordan-Watts 'modules', are designed to handle the entire frequency range. See also under *Moving Coil Unit.*

MATCHING

With loudspeakers, this generally means that the impedance of the loudspeaker is of a value suitable for the loading requirement of the amplifier. Most amplifiers will accept an impedance from 8 to 16Ω. Some will accept down to 4Ω without distress, but other designs are unhappy with a loading as low as 4Ω. The power of the amplifier commonly increases with decreasing load impedance (see Chapter 1).

MOVING-COIL UNIT

A moving-coil unit is an electro-magnetic transducer, sometimes called a *dynamic* loudspeaker unit. It is the type most commonly used in hi-fi loudspeaker systems, though the ribbon and electrostatic driver are sometimes used for treble reproduction.

In general, the larger the unit, the lower the frequencies it is most suitable for. This is why bass units are physically larger than units for mid-range and treble reproduction. This has to do with the mass of the cone or diaphragm and the compliance of the suspension. A large mass and large compliance means a low bass resonance (see under *Free-Air Resonance*), and such a resonance may be used to help offset other factors reducing efficiency at that order of low-frequency.

Higher-frequency units, particularly tweeters, call for very low mass diaphragms and less compliant suspensions, since the aim here is to reduce inertia so as to achieve faithful reproduction of fast occurring transient-type signals.

A moving-coil loudspeaker unit is shown in Fig. 2.17 and the main features in Fig. 2.18. The operation results from the motor effect produced by interaction of the magnetic field yielded by signal current passing through the moving-coil (or speech coil as it is usually called) and the fixed magnetic field produced by a high flux permanent magnet.

The speech coil moves freely between pole pieces across which a concentrated fixed magnetic field is developed. This means that the speech coil, and hence the cone which is attached to this coil, moves like a piston in sympathy with the signal current passing through the coil from the amplifier, the cone then producing compression and refraction waves in the surrounding air.

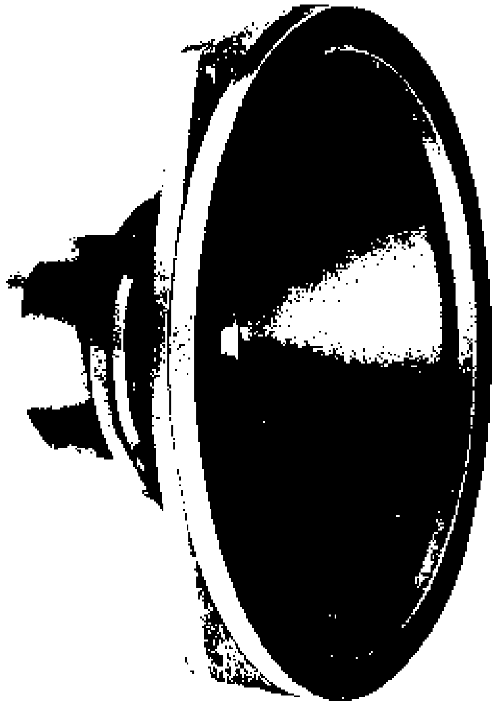


Fig. 2.17
Moving-coil
loudspeaker unit.

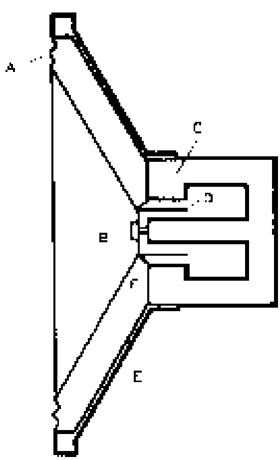


Fig. 2.18 Features of moving-coil loudspeaker unit, where A is the cone surround and compliant suspension, B the speech coil centring device and also sometimes part of the suspension, C high flux permanent magnet which produces a strong flux across the gaps of the pole pieces in which the speech coil D moves, and F a covering to exclude the entry of dust, etc. from the pole pieces and moving-coil. E is known as the loudspeaker chassis, which should be resonance free.

OMNIDIRECTIONAL

Most loudspeaker systems tend to become mildly directional with increasing frequency (see under *Directivity* and Fig. 2.9). However, excessive beaming is undesirable since it make treble sounds virtually

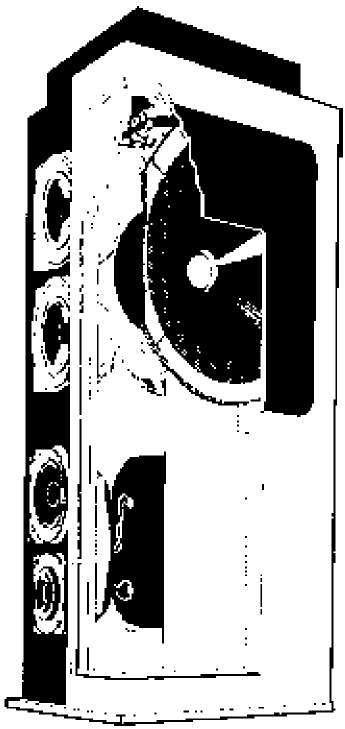


Fig. 2.19 Multi-unit loudspeaker system by Goodmans, called Dimension-8. This employs one 'slave radiator' (ABR), four 5in bass units, two 4in mid-range units and two 1in dome tweeters. Crossover frequencies are 80Hz, 800Hz and 4kHz, and the frequency response is 30Hz–22kHz.

inaudible to listeners located, say, 30° off axis. Moreover, loudspeaker systems which become significantly directional with increasing frequency exhibit a sharply defined, small sound source and contribute to listener fatigue.

Good designs endeavour to retain a reasonably broad and constant radiation pattern by various schemes, an important one being by the use of two or more driver units (sometimes two or more for treble) each covering a relatively small frequency range (Fig. 2.19).

At the other extreme, some designers favour omnidirectional characteristics, meaning that the polar response is essentially spherical and that the system has equal output in all directions.

All loudspeaker systems are essentialy omnidirectional at low frequency, but true omnidirectional models are designed by the use of multiple units, sometimes integrating the reflective effects of walls and the ceiling, so that the omnidirectivity holds at all frequencies.

There are various views on the merits and demerits of this sort of reproduction, particularly so far as stereo is concerned, but there is not room here to delve into this. One firm producing omnidirectional loudspeaker systems is Sonab of Sweden.

PHASING

When two or more loudspeakers are contributing to a common sound field, such as in stereo and quadraphony, the cones or diaphragms of all the units should move in the same direction on a common signal component. When this happens the loudspeakers are said to be in-phase.

Should the diaphragm or cone of one loudspeaker move out while that of another loudspeaker moves in (on a common signal), then the loudspeakers are said to be out-of-phase. Misphasing tends to cancel low bass frequencies and confuse the stereo image.

To help establish the correct phasing, the terminals of the loudspeakers and those of the amplifier for connecting the loudspeakers are either colour-coded or marked *plus* and *minus*. If in doubt, the two stereo loudspeakers should be placed close together and operated from a mono source or from a stereo source with the amplifier in the mono mode.

The programme material should have a heavy bass content, and the anti-phase condition will be evidenced by lack of bass in the reproduction. If the bass increases when *one* pair of loudspeaker connections are changed round, then that is the correct way of connecting for the in-phase conditions.

PIPE LOADING

With this type of loading, the enclosure is in some respects like a 'pipe' and the driver is mounted so that the rear of its cone is loaded at its resonance frequency by the column of air in the pipe. For this reason, the enclosure is sometimes called '*column*'.

There are various styles, the driver either being mounted at one end, or the pipe may be tapered and the driver mounted a calculated distance from the narrow end. There are also versions where the pipe is folded or of a special shape. Heavy lagging is generally applied to the inside surfaces, as with an acoustical labyrinth. In fact, in some respects the pipe or column enclosure is similar to the labyrinth.

The basic scheme is for bass loading of the driver at or near its resonance frequency, the bass sound then exhausting from the far end of the pipe, with the middle and treble frequencies emanating from the front of the cone or from separate drivers. One design aspect is for the reduction of the quarter-wave fundamental resonance and the unwanted odd harmonics of the pipe.

POLAR RESPONSE

This is a graph of loudspeaker output versus angular position which shows how the output alters with position. A typical polar response is given in Fig. 2.9. The polar response at various frequencies may be included on the same graph, as in Fig. 2.9.

POWER HANDLING CAPACITY

This refers to the maximum amount of electrical power that can be fed safely to a loudspeaker. Various expressions of this parameter are in current use, and in some cases the power quoted by the loudspeaker manufacturer may not correlate with the continuous sinewave power output of the amplifier channel.

For example, a rating of 40W music power for a loudspeaker means that the maximum continuous wave average power (sometimes erroneously referred to as 'R.M.S.' power—see Chapter 1) handling would be about 62% of 40W, or about 25W.

On the other hand, some manufacturers rate the power capacity on a continuous wave basis, or indicate that a loudspeaker is suitable for, say, a 40W amplifier channel provided the amplifier is carrying only music signal (*not* continuous sinewave signal).

Overdriving a loudspeaker will result in significant increase in distortion, especially at the bass end, overheating of the speech coils of the drivers and possibly early failure of the tweeter. Excessive overdrive could turn the cone of the bass driver inside out!

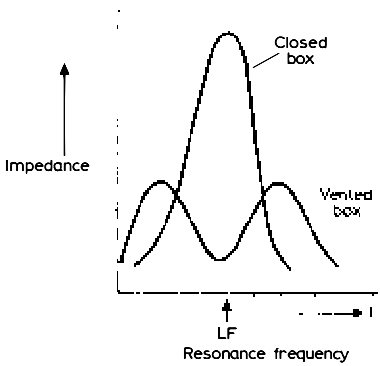
REFLEX ENCLOSURE

When an aperture is cut in a closed box the performance is changed considerably, and the resonance behaviour so introduced, when correctly handled, can be exploited for enhancing the bass loading and reproduction. This kind of enclosure is called reflex or *vented*, and the aperture the *vent* or *port*.

The dimensions of the enclosure and vent need to be carefully related to the fundamental resonance of the driver for the best results, and the capacity of the enclosure is generally greater than that of the 'infinite baffle' enclosure.

At a frequency depending on the dimensions, the air mass in the vent oscillates when excited by the bass driver and the acoustical loading at the rear of the cone maximises. Around this frequency, the emission from the vent is in phase with that from the front of the cone so that the bass output is augmented.

Either side of the resonance frequency, the cone loading obviously falls so that the impedance characteristic takes the form of two peaks, one either side of resonance, but each of smaller amplitude than the single peak which may occur with 'infinite baffle' loading, as shown in Fig. 2.20.



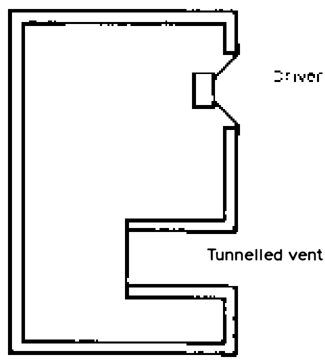


Fig. 2.20 (left) Impedance characteristic of reflex enclosure compared with closed box (see text). Fig. 2.21 (right) By 'tunnelling' the vent as shown a low resonance frequency can be obtained with a smaller enclosure volume.

The lower the fundamental resonance of the driver, the greater the volume needed for the enclosure which, in general, means that the lower the bass output required, the larger the enclosure. However, it is possible to achieve a fairly low-frequency resonance in a smaller enclosure by 'tunnelling' the vent, as shown in Fig. 2.21. A tube extending from the vent inside the enclosure may also be used to similar effect, as shown in Fig. 2.22.

To help reduce the amplitude of the two peaks (Fig. 2.20) and provide better cone control over a wider low-frequency range an

Fig. 2.22 'Tunnelling' is sometimes achieved by a tube as this photograph shows.

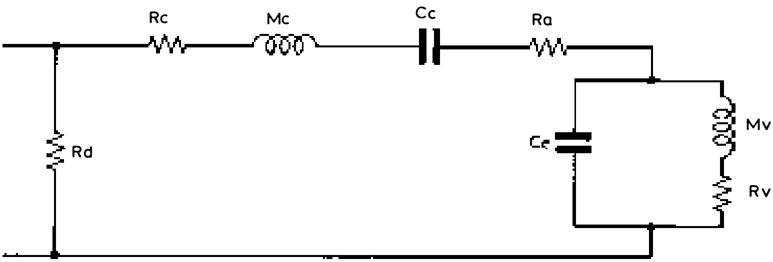


Fig. 2.23 Electrical analogue of reflex enclosure, where the parallel tuned circuit represents the enclosure whose impedance is maximum at resonance. C_e is the reactance (i.e., the 'stiffness' of the enclosed air), M_v is the mass of air in the vent and R_v is the resistance of the air in the vent. The remaining components refer to the driver, where R_d is the electro-magnetic damping, R_c the resistance of the suspension, M_c the mass of the cone, C_c the compliance (stiffness factor) of the cone and R_a is the resistance of the air into which the cone radiates.

acoustical resistance unit is sometimes used to cover the whole or part of the vent (see under *Acoustical Resistance Unit*).

The electrical analogue of the reflex enclosure is given in Fig. 2.23.

RIBBON UNIT

This is a high-frequency driver which uses as its diaphragm a thin, low mass corrugated metallic ribbon suspended between the gaps of a powerful magnet.

The ribbon forms both the sound emitting diaphragm and the conductor through which the high-frequency signal current is passed. Thus, when signal current passes through the ribbon a magnetic field is produced which interacts with the field of the fixed magnet such that the ribbon vibrates at the signal frequency.

The efficiency is often increased by horn loading, such as with the Decca DK30 and 'London' ribbon units. The low mass of the ribbon ensures very good transient performance. To improve the dispersion, Decca also make and market an acoustical lens for the units.

SENSITIVITY

This is similar to efficiency (see under *Efficiency*) but refers to the electrical power input required for a given sound *pressure* at a given distance from the loudspeaker under defined free-field conditions (see under *Free-Field*).

The German DIN (Deutscher Industrie Normenausschuss) reference pressure is 96dB* (corresponding to 12·62 microbars—given nominally as 12μb—which is equivalent to 12·62 dyne/cm^2, 1·262 N/m^2—SI units, where N is Newtons and 1·262 Pa, where Pa is the unit pascal) at 1m under *hemispherical* free-field conditions (see under *Efficiency*).

Thus, a loudspekaer system with a DIN sensitivity of, say, 9W for 96dB, means that 9W electrical power input is required to produce a sound pressure of 96dB (based on noise signal) at 1 metre from the loudspeaker when the loudspeaker is arranged to radiate into hemispherical free-space.

SQUAWKER

The name sometimes given to the middle-frequency driver unit.

SYSTEM

While a driver is generally called a 'unit', a complete loudspeaker is a 'system'.

*0dB sound pressure corresponds to 20μN/m^2 (SI units), to 0·0002μb, 0·0002 dyne/cm^2 and 20μPa It also corresponds to 10^{-12}W/m^2.

TWEETER

The name given to the high-frequency driver unit.

WOOFER

The name given to the low-frequency driver unit.

CHAPTER THREE

PROGRAMME SOURCES AND SIGNALS

ACOUSTICAL FEEDBACK

THIS REFERS TO A positive feedback loop sustained from the programme source, through the amplifier and loudspeakers and then back to the source via an acoustical path in the listening room.

A source often causing this sort of trouble is the record playing unit, since when the stylus of the pickup cartridge lies in contact with the gramophone record the unit somewhat sensitive to sound waves falling upon it—rather like an inefficient microphone.

The phenomenon can be demonstrated by placing the stylus of the pickup on a stationary record, turning up the amplifier's volume control and then tapping the unit. The taps will be heard through the loudspeakers.

Now, if the microphonic sensitivity is high, a low-frequency howl will tend to develop when the volume control is well advanced. This is acoustical feedback resulting from sound from the loudspeakers vibrating the source. The frequency of howl is influenced by any low-frequency resonances existing in the feedback loop as a whole, including loudspeaker, room and unit resonances.

If the frequencies of several resonances have coincidence, the stability margin will diminish, and a lesser setting of the volume control will precipitate the howl. The trouble is aggravated by poor acoustical damping of the record playing unit or by placing the unit too close to the loudspeakers.

Another susceptible source is, of course, the microphone. If this is placed in line-of-fire of the loudspeakers, then it will be virtually impossible to eliminate the howl at a reasonable setting of the volume control.

The effect can also sometimes occur with a radio tuner, the sound from the loudspeakers vibrating the plates of the tuning capacitor of the local oscillator, thereby resulting in frequency-modulation and hence feedback.

BANDWIDTH

This is the width of the frequency band between the lower and upper limits of the programme source. However, unlike the bandwidth of an amplifier whose lower and upper terminal frequencies are commonly referred to −3dB or half power, the bandwidth of the source is generally defined by a frequency response plot.

For example, the bandwidth of a good f.m. tuner would be from 20Hz to 15kHz within ±1dB. A stereo pickup, on the other hand, would have a bandwidth in the order of 20Hz–20kHz, again defined by dB deviations from the 1kHz nominal.

The high-frequency bandwidth indicates *rise time* potential, which is the time it takes a 'perfect' step signal to rise from 10% to 90% of its maximum value, as shown in Fig. 3.1. The relationship is

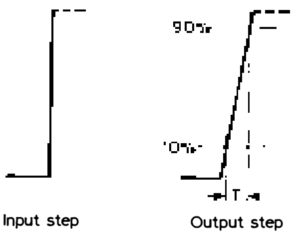


Fig. 3.1 Definition of rise time. This is related to the upper −3dB response (see text). In general, the rise time potential of source and amplifier electronics is in excess of the fastest programme signal.

$T=0{\cdot}35/f_{-3dB}$, where T is the rise time in seconds and f_{-3dB} the high-frequency terminal frequency where the response is −3dB from 1kHz.

It is noteworthy that the rise time capabilities of most amplifiers and source electronics are in excess of the rise times of the programme signals themselves!

CROSSTALK

This refers to the breakthrough of signal in one circuit to another circuit and is commonly expressed in decibels. Thus, if a tuner connected to the radio source input of an amplifier produces an output, say, a hundred times less when the amplifier is switched to gramophone pickup, then the crosstalk between the two source input circuits would be 40dB because 100:1 voltage ratio corresponds to 40dB (note that in terms of power a 100:1 ratio is 20dB—see under *Decibel* in Chapter 1).

Crosstalk can also occur between the two stereo or four 'quadraphonic' channels of a source, but in the former case this is commonly referred to as *stereo separation*, which is really a similar measurement indicating the amount of signal from a talking channel breaking into a non-talking channel. Gramophone pickups should have a stereo separation of at least 20dB at middle frequencies, corresponding to 10:1 voltage ratio or 100:1 power ratio.

It should be understood, of course, that other items of the system will also be responsible for degrees of crosstalk, which must be taken into account when establishing the overall crosstalk. For example, if the pickup has a stereo separation of 20dB and the pre- and power-amplifier sections combined 40dB, then the overall crosstalk will be less than 20dB.

In this case, it will be less than 1dB below 20dB, since when the crosstalk difference is great the overall crosstalk is not very much less than the item of most crosstalk.

DE-EMPHASIS

This is a form of 'equalisation' applied to the f.m. radio tuner source deliberately to introduce high-frequency roll-off to compensate for corresponding pre-emphasis of the modulation signal at the transmitter, as shown in Fig. 3.2.

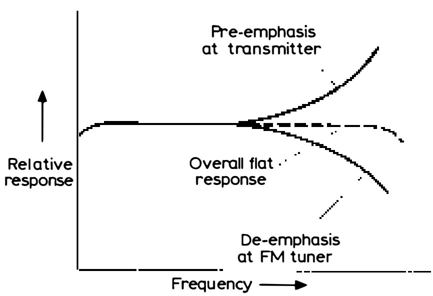


Fig. 3.2 Pre-emphasis at the transmitter and de-emphasis at the tuner result in a flat overall frequency response characteristic as shown. The de-emphasis not only equalises for the treble boost at the transmitter, but it also reduces high-frequency noise, which is the intrinsic purpose of the scheme. With Dolby encoding the pre-emphasis is reduced to 25μS from the European 50μS or American 75μS such that the peak treble modulating power is reduced, thereby avoiding peak limiting. When Dolby decoding is used at the tuner the de-emphasis is also reduced to 25μS and a significant improvement in S/N ratio obtains. Dolby encoding together with the 25μS pre-emphasis provide compatibility with 50 or 75μS tuner de-emphasis (see text and also Chapter 5).

The pre-/de-emphasis is based on a time-constant, being 50μS in Europe, including the UK, and 75μS in America, the 3dB point corresponding respectively to frequencies of 3·184 and 2·123kHz. In both cases the ultimate rate of roll-off is 6dB/octave.

Some US f.m. stations are currently incorporating Dolby noise reduction and a decrease in pre-emphasis to 25μS (3dB point at 6·369kHz), which means that without peak limiting there is less likelihood of the transmitters being overmodulated at high energy treble frequencies.

The idea behind pre-/de-emphasis is noise reduction and hence signal/noise ratio enhancement, and with Dolby there is a further improvement even with the smaller time-constant. It is noteworthy that a tuner of 75μS de-emphasis working from a Dolby encoded signal will have approximately the same overall frequency response as when working from a non-encoded signal of 75μS pre-emphasis. However, significant signal/noise ratio advantage accrues when a Dolby decoder is coupled to the output of the tuner and the de-emphasis is reduced to 25μS and when the aerial signal is Dolby encoded. Also see Chapter 5.

DISTORTION

Source signals suffer similar distortion as the signals passing through the main amplifier (see under *Distortion* in Chapters 1 and 2). For hi-fi reproduction, therefore, it is essential to make sure that the source yields as little distortion as possible.

It is in general true that the distortion on the source signals is several magnitudes greater than the distortion to which the signals are subject in passing through the main amplifier, but this is no good reason why the amplifier should not be designed for the lowest possible distortion yield. Every item of the hi-fi reproducing chain must be designed for the lowest possible distortion and hence maximum amplitude linearity.

The f.m. tuner is one of the lowest distortion sources, excluding, of course, the distortion on the modulation signals at the transmitter; but on a 'live' transmission conveyed through the latest pulse code modulation studio-centre-to-transmitter links the overall distortion can be very small.

Some programme sources have their own distortion characteristics, which include mistracking distortion of a record playing unit (see Chapter 6) and multipath distortion of an f.m. tuner (see Chapter 5).

DOUBLE-MONO

When a mono signal is applied simultaneously to the left and right inputs of a stereo amplifier the two loudspeaker reproduce the same signal, which is known as double-mono.

Double-mono will also occur when the mode switch of the amplifier is set to mono and the source is stereo because then the left and right signals are added which converts the sum to a mono signal, which again is applied to the two channels. Under balanced conditions, a double-mono signal will appear to emanate from a point midway between the left and right loudspeakers.

DYNAMIC RANGE

The dynamic range of a large 'live' orchestra may be as great as 70dB, meaning that the peak power rises 10^7 times above the threshold power (or the sound pressure 3,162 times above the threshold pressure).

Source signal dynamic range is less than this, being about 60dB at hi-fi quality, which corresponds to 10^6:1 power ratio and 10^3:1 sound pressure ratio. The limit is established by peak overload at *fff* and noise masking at *ppp*, as shown in Fig. 3.3. Also see under *Dynamic Range* in Chapter 1.

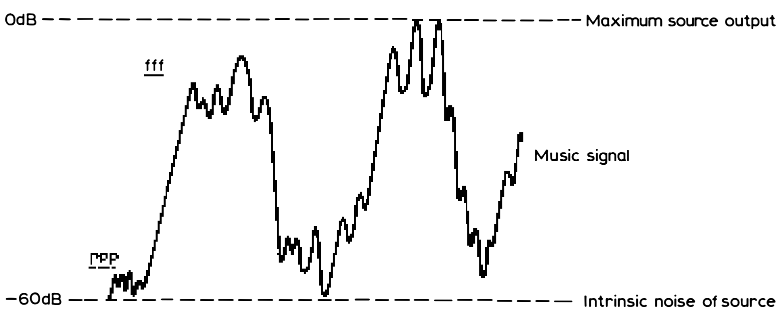


Fig. 3.3 Expression of source dynamic range. Also see under *Dynamic Range* in Chapter 1.

EARTHING

Programme sources are commonly earthed from the amplifier, which itself is earthed either to the earth circuit of the mains socket or to an external earth point (as explained in Chapter 1), via the braids of the source-to-amplifier connecting cables, but for optimum safety all sources should be earthed direct, though this may precipitate a hum loop condition.

EARTH-LOOP CONDITION

When the amplifier is earthed as explained above and an additional earth is connected to a source an earth-loop condition can develop whereby small hum signals are introduced into the source circuits (see under *Hum-Loop* in Chapter 1).

EQUALISATION

Source equalisation is sometimes necessary to compensate for the characteristics of the signals introduced during recording or transmission. One example of source equalisation is f.m. tuner de-emphasis (see under *De-Emphasis* and Fig. 3.2). Also see under *Equalisation* in Chapters 1 and 6.

During disc recording, the signal is given bass cut to avoid intergroove breakdown on strong low-frequency signals and treble boost (pre-emphasis). Thus, for replay, reciprocal equalisation is required; that is, bass boost and treble cut, the latter for improving the signal/noise ratio.

Similar equalisation is required for tape replay (see Chapters 1 and 6).

HUM

This refers to a background low-frequency tone accompanying the required signal. The fundamental frequency corresponds to 50Hz (60Hz in America, etc.) mains frequency, but non-linearity in the nature of the hum signal coupling can result also in harmonics at 100, 150, 200Hz etc. when the fundamental is 50Hz.

One cause of hum is inefficient screening of the leads carrying the low-level signals from a source to the amplifier. Another cause is lack of earthing of the main amplifier or more than one earth circuit (see under *Earth Loop Condition*).

If hum is present with the volume control of the amplifier fully retarded, then a fault in the smoothing circuits of the amplifier would most likely be responsible (i.e., open-circuit or reduced value electrolytic capacitor).

Hum can also be injected into the amplifier due to hum field coupling from the turntable unit motor to the magnetic pickup cartridge.

INTERFERENCE

This refers to the entry of spurious signal information to the source or its amplifier coupling circuits. Interference to radio reception is usually due to a faulty or incorrectly operated radio tuner (see Chapter 5).

Radio breakthrough or impulsive (crackles, pops, etc.) interference when the amplifier is switched to other sources may indicate inadequate screening of the connecting cables (see under *Hum*), particularly when accompanied by background hum, or a fault in the source circuit itself.

In areas of high radio or television signal fields high-frequency filtering may be required in the amplifier to eliminate breakthrough (see under *Interference* in Chapter 1).

LEVEL ADJUSTMENT

Some programme sources, notably radio tuners, are equipped with signal level controls or presets whose purpose is to adjust the level of the signal delivered by the source to correspond to the sensitivity of the amplifier's source input. As explained in Chapter 1, the sensitivity refers to the r.m.s. voltage at 1kHz required at the appropriate input for full power drive of the amplifier when the volume control is at maximum, balance control centre and other controls 'flat'.

For maximum dynamic range of the amplifier, the volume control usually needs to be set to about —20dB, corresponding approximately to the 1 o'clock position. Thus, with the volume control so adjusted, the source level control should be adjusted to provide maximum (before clipping) peak output of the amplifier. After such 'normalisation', the output is then adjusted in the usual way by the amplifier's volume control.

Some amplifiers also feature input level presets, sometimes one (or a pair for the left and right channels) for each source. These make is possible to adjust the input levels of all sources so that the output at a given setting of the volume control is the same on all sources. The level control at the rear of a radio tuner is shown in Fig. 3.4. This tuner also has a pair of outputs which deliver full signal irrespective of the level control setting, which refers only to the other pair of outputs.

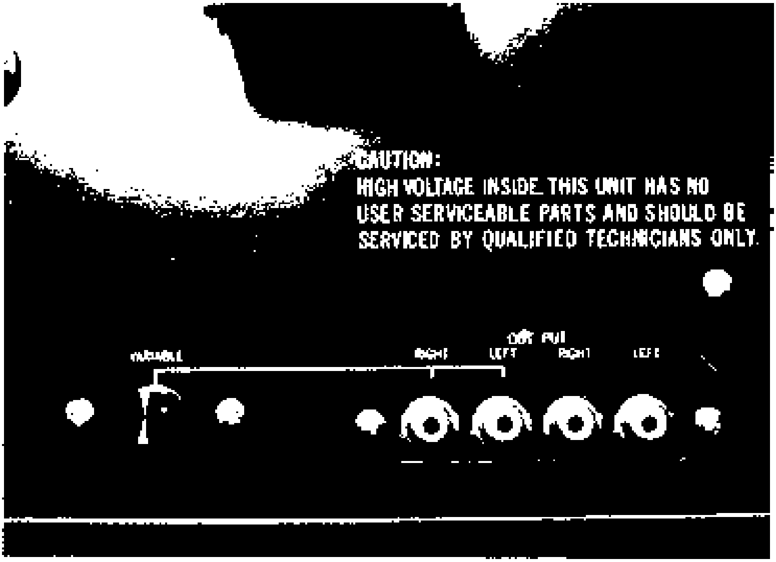


Fig. 3.4 Example of level control at rear of a radio tuner. This operates on one pair of outlets. The other pair shown delivers full-level signal.

Level adjustment may also be found on the amplifier for the tape recording signal. With the recording level control on the recorder set to about 1 o'clock, this preset is adjusted for the correct recording level as indicated by the VU meter, for example.

MATCHING

In general, this refers to matching the level of the signal delivered by the source to the sensitivity of the amplifier at the appropriate input (see under *Matching* in Chapter 1, under *Level Adjustment* above and under *Output Impedance*).

MATRIXING

This is a four-loudspeaker technique whereby source signals corresponding to the front left and right channels (L_F and R_F respectively) and the back left and right channels (L_B and R_B respectively) are reduced to two output circuits corresponding to the left total (L_T)and right total (R_T). This is called a *coding* or *encoding* matrix. For *decoding*, a reciprocal matrix is used which receives the L_T and R_T

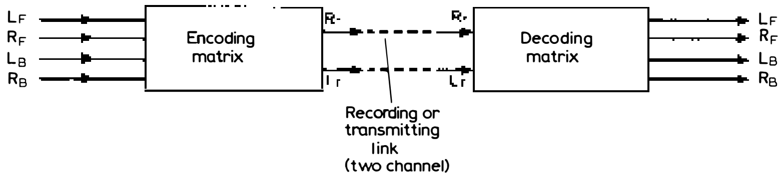


Fig. 3.5 Block diagram showing source encoding matrix and output decoding matrix. At the expense of crosstalk, the four signals can be carried on a two-channel (stereo) medium.

signals at separate inputs and delivers separately signals corresponding to L_F, R_F, L_B and R_B information.

A block diagram of an encoding and decoding matrix is given in Fig. 3.5, which shows that a four-channel source can be conveyed through a two-channel (stereo) medium. The four output signals differ from the original source signals owing to crosstalk between the outputs, which is a characteristic of basic matrixing.

MICROPHONE

This well known programme signal source translates sound pressure waves falling upon its diaphragm into equivalent electrical signals. The principle of operation can be electromagnetic (as the moving-coil or ribbon loudspeaker unit), electrostatic (or capacitive) or piezo-electric (where a p.d. is produced by the sound waves resulting in stressing of a crystal element or a ceramic element prior polarised to possess piezo-electric properties).

Moving-coil and ribbon microphones are low impedance so they usually need to be connected to the input of the amplifier through a step-up transformer, as shown in Fig. 3.6. The turns ratio required of

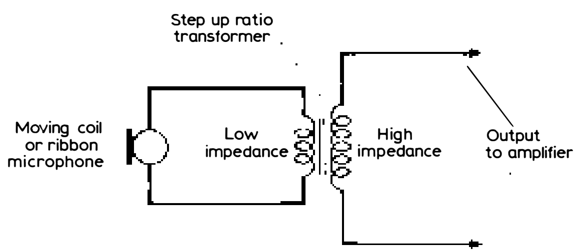


Fig. 3.6 The low output impedance of a moving-coil (dynamic) or ribbon microphone is translated to a higher impedance for connection to the amplifier by means of a step-up transformer as shown. A transistor stage can also be arranged as an impedance converter, and such is sometimes used instead of a transformer.

the transformer is equal to $\sqrt{Z_{out}/Z_{in}}$, where Z_{in} is the input impedance (of the microphone) and Z_{out} the output impedance to the amplifier. For example, if the required output impedance is 50kΩ and the microphone impedance 10Ω, the turns ratio would be 70.7:1 (step-up).

Ceramic (piezo electric) and capacitor microphones are high impedance, and for operating into a long line a step-down transformer would be required to avoid frequency distortion and interference pick up on the line.

The lines, of course, should be screened to avoid hum pick up, and when a low impedance line is of extended length hum pick up can be reduced by operating into a balanced circuit as shown in Fig. 3.7. Since the circuit is balanced, hum signals are cancelled in the centre-tapped transformer primary winding.

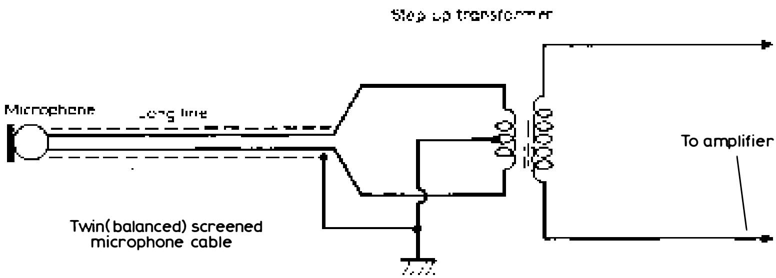


Fig. 3.7 When the microphone line is extended, improved signal/noise ratio and reduced susceptibility to hum pickup can be achieved by using a balanced circuit of the kind shown.

MONO

This is short for *monophonic* or *monophony*, not to be confused with *monaural*, which means 'one-eared'. Mono thus refers to a single channel amplifier, source, system or medium.

NOISE

Electrical noise results from the random movement of electrons in a conductor, valve, transistor or, indeed, in any circuit. Noise produced by a programme source is greatly magnified by the amplifier, and unless of sufficiently low level will manifest as 'hiss' from the loudspeakers.

If the source-to-amplifier coupling impedances are wildly in error the noise output may increase such that the signal/noise ratio will be impaired. Also see under *Noise* in Chapter 1 and under *Output Impedance* below.

OUTPUT IMPEDANCE

This is the impedance from which the source signals are delivered, and in many cases it corresponds to the source impedance in series with the source e.m.f., as shown in Fig. 3.8. From this diagram it can be seen that the source impedance may appear in series with the amplifier input impedance (shown in broken-line), which means that

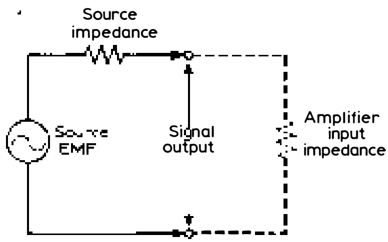


Fig. 3.8 Basic configuration of source output circuit (see text).

the output signal then depends on the ratio of source to input impedance.

A common input impedance value is 47kΩ, and the rated source output signal voltage is often based on the source being shunted with a load of this value. Clearly, then, the output will be higher with a higher value load and lower with a lower value load.

Pickup cartridges are particularly load sensitive (see under *Source Matching* in Chapter 1), but other higher-level sources, such as tape deck, radio tuner, etc., are more accommodating of load impedance. Some radio tuners, in fact, have a relatively low source impedance (perhaps, less than 100Ω), which reduces the problems when connecting to an amplifier impedance of higher value.

When the source impedance is high, there is a greater possibility of early treble roll-off due to the capacitance of the screened connecting cable operating in conjunction with the source impedance to form a simple, single-pole low-pass filter. With a given cable capacitance, the frequency of turnover rises with decreasing source impedance. Thus with tuners, etc. of high output impedance long, high capacitance coupling cables should not be used for fear of early treble roll-off.

Moreover, the noise performance of the input stages of the amplifier can be impaired when the source impedance is particularly high, which is one reason why it is undesirable to attempt to attenuate the source signal to the amplifier by the inclusion of series resistance, and this applies especially to the pickup input.

When a magnetic pickup is connected, the amplifier 'sees' a relatively low value source due to the essentially inductive makeup of the cartridge. This keeps the noise of the preamplifier stage at a low level. If series resistance is included then the noise will almost certainly rise (see under *Noise*).

PROGRAMME SOURCE

This refers to any item of equipment which produces a programme signal to be fed to the amplifier. Typical programme sources are radio tuner, record playing unit, tape deck, microphone, etc.

PHASING

Source phasing is equally as important as loudspeaker phasing for proper stereo and four-loudspeaker reproduction (see under *Phasing* in Chapters 1 and 2). The phasing must be maintained from source to loudspeakers, and if one source is incorrectly phased, relative to the other sources, while the loudspeakers are correctly phased, poor stereo or four-loudspeaker production will result from the incorrectly-phased source.

If the loudspeakers are re-phased to suit the incorrectly-phased source, then the other source will be poorly reproduced. It is best to establish first that the loudspeakers are correctly phased relative, say, to a mono signal from a radio tuner (see under *Phasing* in Chapter 2) and then use that phasing as a reference to check the phasing of the other sources, such as record playing unit and tape deck.

'QUADRAPHONIC' SOURCE

This is a source which is capable of ultimate discrete or matrix four-loudspeaker reproduction. For example, a four-channel tape recorder in replay mode provides four separate channels of information (see Chapter 4) which can be fed correspondingly to the separate inputs of a four-channel amplifier for four-loudspeaker reproduction.

The tape programme *medium*—currently referred to as programme *software*—would, of course, need to carry four lots of information, corresponding to the signals of left front, right front, left back and right back, each lot utilising its own tape track.

Another 'quadraphonic' source could be two-*channel* software, such as tape or disc, with the four channels matrixed into two channels whose outputs then correspond to total left (L_T) and total right (R_T) signals (see under *Matrixing*). In this case, the tape machine or record playing unit would incorporate a two-channel (stereo) head or cartridge which would deliver the L_T and R_T signals.

These two signals would then be fed to a decoding matrix, from the output of which would be obtained the signals corresponding to left front, right front, left back and right back. These resulting four signals would be directed separately to the inputs of a four-channel amplifier for four-loudspeaker reproduction.

A different approach is adopted with so-called discrete (CD-4, for example) discs. Again, a two-channel (stereo) pickup is used, but the four-channel encoding on to the disc utilises a 30kHz carrier which is frequency/phase modulated with the left and right rear channel information (see Chapter 4).

With this technique, the pickup needs to respond up to some 45kHz to define the carrier-modulated components, which, for the best results, calls for a special type of stylus (i.e., Shibata stylus), and the signals corresponding to the original four channels are obtained by feeding the L_T and R_T signals to a special demodulator.

RADIO TUNER

This is a rather special type of radio receiver minus the audio output stages (see Chapter 5). It may be f.m. (frequency modulation) only or f.m./a.m. (a.m. corresponding to amplitude modulation). Only f.m. is capable of providing hi-fi quality reception, and only this system provides for stereo.

Audio signal is delivered at relatively low power (*circa*, 250mV across 47kΩ for 30% modulation level), and this is communicated to the radio or tuner inputs of the amplifier. The radio tuner is thus a programme source.

For stereo or four-channel matrix reproduction a *stereo decoder* is coupled to the output of the f.m. detector, and this yields the left and right signals of a stereo programme or the total left and total right signals of a matrixed programme (however, at the time of writing there are no known matrix transmissions in the UK).

RECORD PLAYER

This is another programme source which consists of a turntable and pickup system (usually arm and detachable cartridge). Most record players (or record playing units) are suitable for the replay of matrix discs, but for the best replay of discrete four-channel discs, such as CD-4 (see Chapter 4), a cartridge responsive up to at least 45kHz with a good high-frequency separation is essential.

This sort of cartridge is usually equipped with a special stylus (Shibata or similar) for maximum definition of the high-frequency modulation and for assisting in the maintenance of a high, high-frequency resonance.

SEPARATION

See under *Crosstalk.*

SIGNAL LEVELS

With regard to the programme sources, the signal levels refer to the r.m.s. voltages at 1kHz delivered across the rated load values, taking account of the recorded or modulation levels. For example, the output of a magnetic cartridge is commonly 5mV per channel at 5cm/S recorded velocity. A radio tuner commonly yields about 250mV per channel for 30% modulation level. (See also under *Source Matching* in Chapter 1, and *Level Adjustment* in this chapter).

SIGNAL-TO-NOISE RATIO

The signal-to-noise, signal/noise or S/N ratio is the noise signal output of the source referred to the signal output at a given recorded

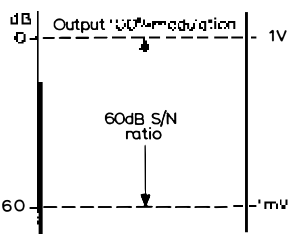


Fig. 3.9 Definition of S/N ratio. When referred to maximum output, this is rather like dynamic range.

or modulation level. For example, if the noise-signal voltage of a radio tuner is 1,000 times less than the signal output at, say, 100% modulation, then the S/N ratio would be 1,000:1 or, expressed in decibels as it usually is, 60dB (see Fig. 3.9).

100% or 30% modulation level may be used, but the level should be stated. A level of 30% would produce a signal output almost 10dB below the 100% level output, so referred to that lower level the S/N ratio would be around 50dB.

The S/N ratio of a record playing unit is generally referred to 10cm/S velocity or recorded level.

An S/N ratio much below 50dB referred to maximum signal output (i.e., 100% modulation) is inadequate for hi-fi reproduction since the background 'hiss' would tend to interfere with *ppp* reproduction.

STEREO SOURCE

Almost all hi-fi programme sources are stereo, including tape, disc and radio. The term implies that the source produces two outputs, one corresponding to left-hand signals and the other to right-hand signals.

SOFTWARE

This refers to the programme *medium*, such as disc and tape records.

TAPE DECK

This is another programme medium whose software is magnetic reel-to-reel or cassette tape. Such sources are available in single-channel (mono), two-channel (stereo) and four-channel ('quadraphonic') versions, but at the time of writing the cassette deck has not been very extensively developed as a four-channel source. Like the radio tuner, the signal output is at a relatively low power, and thus needs to be connected to a hi-fi amplifier for reproduction.

Most tape decks can usually be switched to allow signals delivered by the parent hi-fi amplifier (or separate microphone) to be recorded on the tape.

TELEVISION TUNER

This is rather like the radio tuner but is designed either to tune over the television bands or to respond to spurious sound signal radiated by the intercarrier circuits of 625-line television receivers so as to deliver the sound accompaniment of a television programme, which can then be channelled into the hi-fi system.

This source signal, of course, is in mono, so the amplifier would be switched to double-mono mode for its reproduction.

600-OHM LINE

Some programme sources deliver signals from 600Ω line outlets. These outlets work into 600Ω loads so as to produce a *power* of 1mW at full modulation, which corresponds to 0·775Vr.m.s.

Tape machines are often equipped with such outlets (perhaps, in addition to others), and there is at least one radio tuner with a 600Ω outlet, the Armstrong Model 624 (and Model 623, a.m. as well as f.m.). A power of 1mW into 600Ω is often referred to 0dB.

CHAPTER FOUR

QUADRAPHONY

AMBIOPHONY

A HOST OF NEW TERMS are being coined to describe certain aspects of four-loudspeaker reproduction, and this is one of them. It commonly implies the addition of one or two loudspeakers placed at the sides or back of the listening area, and wired to the regular stereo system so as to respond to the differential of the two stereo channels.

Differential or 'difference' information arises between the two 'live' output terminals of the two power amplifiers when the outputs

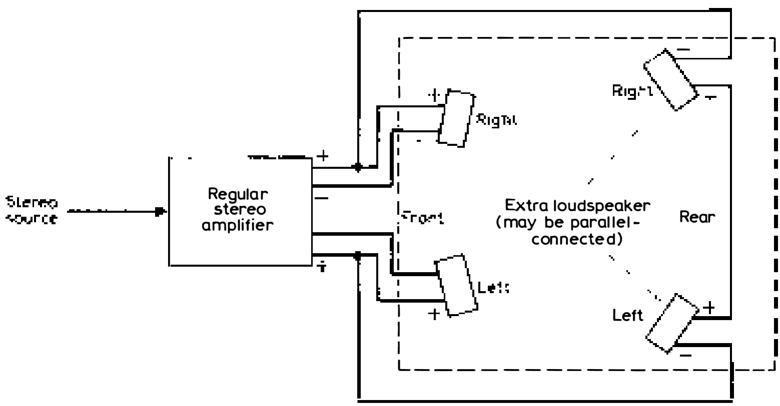


Fig. 4.1 Simple technique (after Hafler) for extracting 'differential' information from a two-channel stereo source. The success of this technique depends to some degree on the nature of the original two-channel recording or transmission. The two back loudspeakers can be connected in series or parallel, though their inter-phasing will effect the net result. Some users of this technique prefer to face the rear loudspeakers (or one rear loudspeaker) to the wall to help diffuse the sound information. Blending between the two front channels may also be adopted as a means of improving the spatial characteristics of the sound field (see under *Blending* and *Blend Coefficient*).

of the two stereo channels deviate from the centre-source mono condition.

Since the differential can carry reflected signals, enhanced listening pleasure can result when the information is fed to back or side loudspeakers, since then a form of surround-sound effect is achieved.

This mode of four-loudspeaker reproduction derives the power for the side or back loudspeakers from the normal two stereo channels, so it only requires a two-channel stereo amplifier. Stereo amplifiers and tuner-amplifiers are sometimes equipped with extra loudspeaker sockets wired in this way, which is basically as shown in Fig. 4.1.

In some cases, however, a low-pass filter may be included in the circuits to the rear loudspeakers so that these loudspeakers respond essentially to the lower frequency sounds. There may also be a level control for the back loudspeakers to secure the most desirable balance between the direct and reflected sounds. The amplifier or receiver may incorporate a switch to communicate the 'differential' drive to a pair of loudspeakers.

AMBIO-STEREO

Sometimes called 'ambi-sterophony', this is rather the same as ambiophony, just described. That is, it is a technique for deriving back or side reverberation from two-channel programme sources or software.

AMBISONIC

A term suggested (see footnote on page 000) to describe a multi-loudspeaker sound reproducing system from which the listener experiences directionality and reverberance acceptably approximating those features of the original sound.

BLENDING

To improve subjectively four-loudspeaker reproduction via a matrix technique, the signals in two circuits may be deliberately mixed. This is known as blending.

BLENDING COEFFICIENT

This is the factor which determines the value of the mixing or crosstalk (i.e., blending). Different matrices have adopted different blend coefficients, but in March 1972 the Japanese EIA-J standardised the *regular matrix* along with a series of blend coefficients for standard monitor matrix decoders.

Blending is used to counter subjectively the effects of the intrinsic crosstalk which exists between channels of the basic matrix (see also under *Matrixing* in Chapter 3).

CARRIER TECHNIQUE

This is a 'quadraphonic' technique whereby the additional information required for four-loudspeaker reproduction over stereo is modulated on to a carrier wave. Techniques currently adopting this principle include the JVC CD-4 discrete four-channel gramophone record (see under *CD-4* Technique) and the Nippon Columbia UD-4 four-channel gramophone record, the latter being a development of the QMX disc record (see under *UD-4 Technique*).

The four separate signals for replay are derived from a special kind of demodulator (see under *Demodulator*).

Various carrier techniques have also been put forward for the radio transmission of four channels. This can be regarded as an extension of the two-channel stereo f.m. multiplex system with re-engineering to allow the transmission of extra signals for the back two channels. Such a system of *multiplexing* is achieved on a single

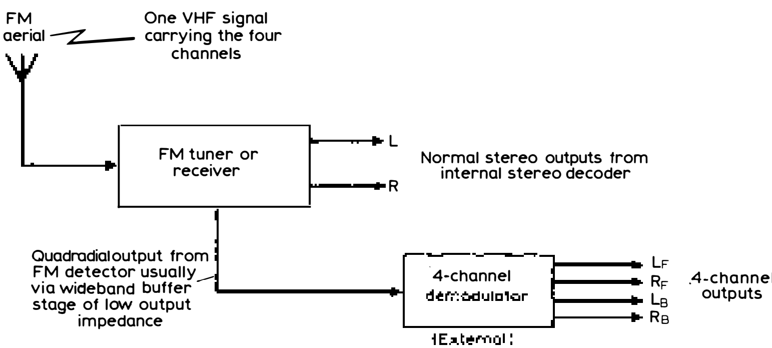


Fig. 4.2 A 'multiplex' method of four-channel broadcasting, based on a single v.h.f. carrier, will require the connection of a four-channel demodulator to the tuner or receiver at the f.m. detector output (prior to the de-emphasis) as shown. Some hi-fi tuners and tuner-amplifiers are already equipped with such an output, sometimes called 'quadradial'.

v.h.f. carrier wave, which is frequency modulated, with the extra information being carried on a modulated subchannel or subchannels.

To reclaim the four-channel signals a special four-channel demodulator or decoder would need to be connected to the output of the tuner's f.m. detector, and some hi-fi receivers and f.m. tuners are already equipped with such an output labelled 'quadradial', the connection from this being shown in Fig. 4.2.

CBS TECHNIQUE

This is a matrix technique developed by CBS Laboratories Inc. of the US, whose software is being promoted jointly by CBS records and Sony of Japan (see under *SQ Technique*).

CD-4 TECHNIQUE

This is a four-channel discrete disc technique developed by JVC of Japan (having American RIAA approval), whereby the left (inner) wall of the groove carries the sum of the left front and the left back

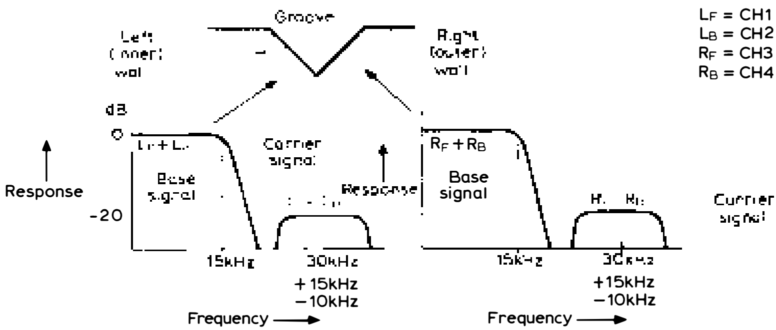


Fig. 4.3 Features of the JVC CD-4 discrete disc technique. See text for explanation.

information and the right (outer) wall the sum of the right front and the right back information. Each wall also carries a difference signal, being L_F–L_B on the left wall and R_F–R_B on the right wall.

The difference signals carry information up to 15kHz and frequency/phase modulate a 30kHz carrier, also recorded in the groove, over +15kHz and −10kHz, as shown in Fig. 4.3. The instantaneous carrier frequency, therefore, can be over the range 20 to 45kHz, for which reason the cartridge used for playing the record must be responsive at least up to 45kHz.

The cartridge is two-channel, as for stereo, but its design is focused on good high-frequency response and good high-frequency separation, and to these ends, and also to achieve optimum high-frequency tracing, a special stylus is generally incorporated for the best CD-4 results. One such stylus for this application is the *Shibata* (see under *Shibata Stylus*).

Since the groove walls carry L_F+L_B and R_F+R_B, CD-4 records can be played in normal stereo (see under *Compatibility*). To obtain the four lots of information in isolation for four-channel replay, the two-channel cartridge outputs are fed to a CD-4 demodulator (see under *Demodulator*) which yields L_F, L_B, R_F and R_B outputs for application to a four-channel amplifier and four loudspeakers.

The CD-4 demodulator incorporates a matrix which processes the signals as follows:

$$L_F=\tfrac{1}{2}\ [(L_F+L_B)+(L_F-L_B)]$$
$$L_B=\tfrac{1}{2}\ [(L_F+L_B)-(L_F-L_B)]$$
$$R_F=\tfrac{1}{2}\ [(R_F+R_B)+(R_F-R_B)]$$
$$R_B=\tfrac{1}{2}\ [(R_F+R_B)-(R_F-R_B)]$$

COMPATIBILITY

This refers to how well four-channel or matrix software plays in stereo or mono mode. For example, since the two walls of a CD-4 disc carry respectively L_F+L_B and R_F+R_B information, this kind of 'quadraphonic' record plays in stereo just the same as any regular stereo record, and likewise in mono.

Matrix discs have different degrees of stereo and mono compatibility. They will all play in stereo or mono mode, but owing to the nature of the matrix phasing, etc. in mono some of the back centre sounds may be attenuated, and in the stereo mode there may be some loss of separation or a widening of the front sound stage.

However, it must be realised that when any disc is recorded for four-loudspeaker reproduction the conditions for this nature of reproduction are generally optimised (microphone and musician locations, etc.), which may not be the requirement for the best stereo or mono reproduction. The same reasoning can apply when a stereo disc is played in mono mode.

Any 'quadraphonic' method of f.m. broadcasting will also need to be compatible for stereo or mono reception. Some American and Japanese stations are currently transmitting matrix discs which can thus be received in stereo or mono with appropriate receivers or in four-loudspeaker mode when a suitable matrix decoder is connected to the audio output circuits of a stereo tuner and fed to four loudspeakers by way of four separate amplifier channels.

DECODER

This term is generally applied to the matrix type of 'quadraphonic' technique at the reproducing end. The complementary matrix at the source end is called the *encoder* (or sometimes merely 'coder').

DEMODULATOR

This is the device which retrieves the four signals from a carrier type of four-channel technique (see under *CD-4 Technique*).

DERIVED REAR CHANNEL

Sometimes called *derived 'four-channel'*, this generally refers to the simple type of loudspeaker matrixing (see under *Matrixing* and Fig. 4.1) where signals for the rear loudspeaker or loudspeakers are obtained from the difference signal contained in a stereo pair of signals (see also under *Synthesizing*).

'DISCRETE' TECHNIQUE

This refers to a four-channel system where the signals for each of the four channels are presented in isolation, as distinct from the matrix technique where there is intrinsic crosstalk between channels. An isolated channel system in perfect form would be a tape carrying four separate channels in perfect isolation and the playback machine having four heads, one for each track, each feeding a separate amplifier and coupled loudspeaker.

There are reel-to-reel four-channel tape machines for recording and replay, and JVC (the CD-4 people) have developed a four-channel 'discrete' cassette system, there being four tracks on each half of the narrow cassette tape, making a total of eight tracks in all. This makes it possible to use both halves of the tape as with stereo, the cassette merely being turned over to secure response (or recording) of the second half.

The very narrow tracks produce S/N ratio and dynamic range problems, but aided by noise reduction systems (Dolby and JVC's own Automatic Noise Reduction System—ANRS) these problems are being overcome.

DM-4 TECHNIQUE

A technique developed by the Japanese Sanyo organisation to enhance inter-channel separation of its matrix decoder.

DORREN TECHNIQUE

A discrete four-channel carrier-type f.m. broadcasting scheme evolved by a Louis Dorren of America.

DYNACO TECHNIQUE

An example of a simple type of loudspeaker matrixing for deriving rear channel signals (see under *Matrixing* and Fig. 4.1).

ENCODER

A matrix network for processing four source signals corresponding to a 360^0 sound field into two transmission channels. Sometimes called 'coder', it is the reciprocal of the decoder used at the reproducing end to retrieve the four lots of signals from the two channels. For the best results, the equation of the decoder must match that of the encoder.

EV TECHNIQUE

EV stands for Electro-Voice Inc. of America, and the technique refers to a 'quadraphonic' matrix of specific mathematical equation. The basic encoding equations are:

$$L_T = L_F + 0{\cdot}3R_F + L_B - 0{\cdot}5R_B$$
$$R_T = 0{\cdot}3L_F + R_F - 0{\cdot}5L_B + R_B$$

and there are two sets of decoding equations, one set referred to as *EVX-4* which is:

$$L_F = L_T + 0{\cdot}2R_T \qquad R_F = 0{\cdot}2L_F + R_T$$
$$L_B = 0{\cdot}76(L_T - 0{\cdot}8R_T) \qquad R_B = 0{\cdot}76(-0{\cdot}8L_T + R_T)$$

and the other set referred to as *EVX–44* which is:

$$L_F = L_T + 0{\cdot}2R_T \qquad R_F = 0{\cdot}2L_T + R_T$$
$$L_B = 0{\cdot}63(0{\cdot}4L_T + jL_T) \qquad R_B = 0{\cdot}63(L_T + j0{\cdot}4L_T) - 0{\cdot}63(0{\cdot}4R_T + jR_T)$$

where j refers to 90^0 phase shift.

GAIN-CONTROL LOGIC

A technique whereby the gain of one or more of the four circuits of a matrix decoder is adjusted automatically by specific parameters of the signal as a means of boosting a wanted signal and suppressing an unwanted one such that the effective inter-channel separation of the decoder is improved. Such logic control is incorporated in some SQ decoders to give 'full logic' and front-rear logic (see under *SQ Technique*).

LOGIC CIRCUIT

This is a circuit artifice employed with certain types of matrix decoders which controls of the gain of the output channels in accordance with the information carried in the two encoded channels in a manner subjectively to enhance the channel separation. The CBS SQ matrix and Scheiber matrix employ such circuitry, sometimes referred to as 'gain riding'.

LOUDSPEAKER MATRIX

See under *Derived Rear Channel* and Fig. 4.1 and under *Dynaco Technique*.

LOUDSPEAKER PLACEMENT

'Quadraphonic' systems mostly have the four loudspeakers positioned one approximately at each corner of the listening area, but as with stereo reproduction there is a wide scope for experimentation, with the nature of the loudspeakers, the room dimensions and the number of listeners to be taken into account (also see Chapter 7).

MASKING

A hearing phenomenon whereby when two sounds are present simultaneously only the stronger of the two is heard, the weaker being pushed below the hearing threshold, which is known as masking. Masking plays a significant role in four-loudspeaker reproduction, and the logic and gain riding circuits of some matrix decoders exploit the phenomenon.

MATRIX

A passive or sometimes active electrical network which encodes four lots of information corresponding to a 360° sound field into two signal channels for recording or transmission. Simple passive encoder and decoder matrices are given respectively at (a) and (b) in Fig. 4.4.

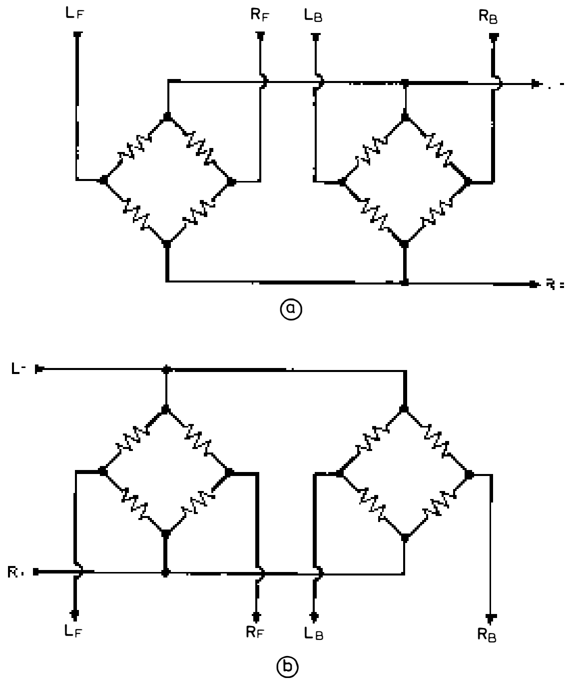


Fig. 4.4 Basic passive encoding and decoding matrices at (a) and (b) respectively, where L_F, R_F, L_B and R_B are the left front, right front, left back and right back signals and L_T and R_T the left total and right total signals.

For enhancement of the four-channel effect, phase-shifting circuits (denoted in the equations by the symbol 'j') are commonly employed in the matrix design as, for example, in the EV, SQ and QS matrix networks.

MULTIPLEXING

When two or more signals are introduced to a common transmission or recording channel and when some of the information is modulated on to a subcarrier in a specific manner for ease of ultimate demodulation and hence retrieval, the technique is referred to as multiplexing.

Typical in this respect is the subcarrier-based stereo information frequency-modulated with the mono signal on to a common v.h.f. carrier wave. A discrete four-channel broadcasting system when using the subcarrier technique could also be classified as multiplexing.

PANTAPHONIC

A term suggested* to replace the term 'quadraphonic' (avoiding the Latin-Greek hybrid) where the four-loudspeaker reproduction lies essentially in the horizontal plane.

PERIPHONIC

A term suggested to aurgment 'pantaphonic' (see footnote on this page) when the reproduction is in three directions—plane plus height.

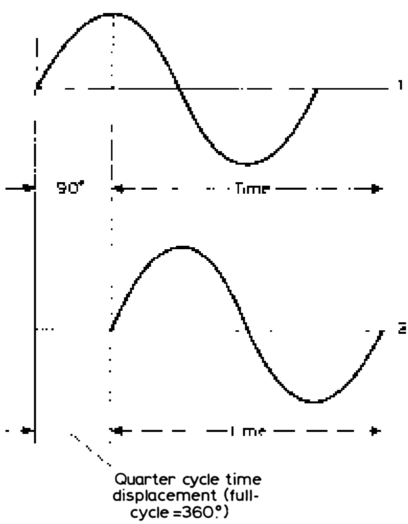


Fig. 4.5 Illustration of phase difference. Here wave 1 starts 90° or a quarter of a cycle before wave 2, so that there is a quarter of a cycle or 90° phase difference between them.

**Fellgett, P. B., Perspectives for Surround Sound, Hi-Fi Sound Annual, 1974.*

PHASE

The difference in time, referred to as an angle, between a common point of two periodic waves, as shown in Fig. 4.5. Phasing plays a significant role in a number of matrix decoders, and for this reason the phasing of the input and output signals—from source to amplifier inputs, and from amplifier outputs to loudspeakers—of a 'quadraphonic' system must be accurately determined and correctly controlled to secure the best four-loudspeaker effect.

QS TECHNIQUE

A type of regular matrix technique developed by Sansui Electric Company of Japan. It is a rotationally symmetrical matrix which includes $\pm 90°$ phase shifters, and is one of the recently RIAA (American) approved matrix techniques. A feature is the provision of uniform separation over a complete 360° compass.

The basic encoding equations are:

$$L_T = L_F + 0{\cdot}414R_F + jL_B + j0{\cdot}414R_B$$
$$R_T = 0{\cdot}414L_F + R_F = j0{\cdot}414L_B - jR_B$$

and the decoding equations:

$$L_F = L_T + 0{\cdot}414R_T \qquad R_F = 0{\cdot}414L_T + R_T$$
$$L_B = -j(L_T - 0{\cdot}414R_T) \qquad R_B = j(-0{\cdot}414L_T + R_T)$$

where j denotes 90° phase shifts.

Another expression of this type of matrix (designated Type 2 in the RIAA specification) for complete 360° circumference is:

$$\begin{bmatrix} L_T \\ R_T \end{bmatrix} = \begin{bmatrix} \sin\theta/2 & j\sin\theta/2 \\ \cos\theta/2 & j\cos\theta/2 \end{bmatrix} \cdot \begin{bmatrix} F(\theta) \\ B(\theta) \end{bmatrix}$$

where θ is the counterclockwise angle of the sound source from the centre right position in the original sound field, $F(\theta)$ the sound source located in the front half of the original sound field ($0 \leqq \theta \leqq \pi$) and $B(\theta)$ a sound source located in the rear half of the original sound field ($\pi < \theta < 2\pi$). Again, j denotes 90° phase shifts.

Software under various labels is available for the QS technique, and the scheme has been adopted by Japanese and some American f.m. stations.

QS VARIO-MATRIX

An enhancement of the basic QS matrix, Vario-Matrix is a technique developed by Sansui in 1972 to improve the inter-channel separation

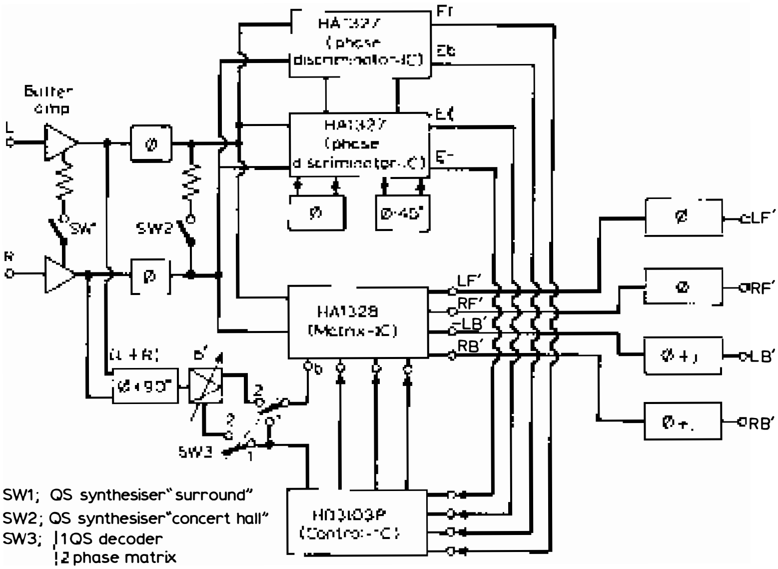


Fig. 4.6 Sophisticated QS Vario-Matrix decoder schematic, based on integrated circuits.

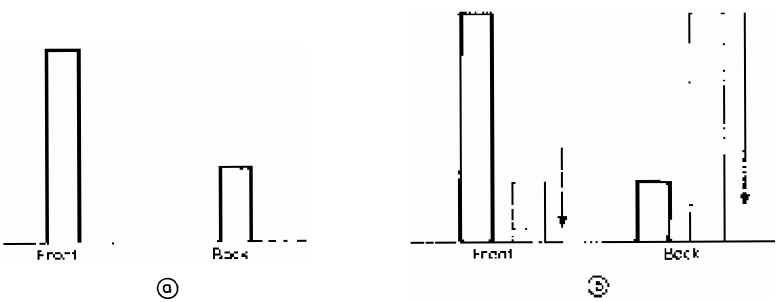


Fig. 4.7 (a) composite original signal in which the front channel sound is stronger than the back channel sound. (b) shows how the QS Vario-Matrix decoder suppresses front and back crosstalk without affecting the original signal content.

by suppressing the crosstalk in the front and back channels. It consists of a phase discriminator, phase shifters and a variable gain matrix, which controls the blending of the four signals without altering the gain of the decoder amplifier. The schematic of a sophisticated QS Vario-Matrix decoder using an 1C is given in Fig. 4.6

The technique exploits the directional masking phenomenon (see under *Masking*) and differs in principle from the gain-controlling techniques of the SQ matrix, though is similar in role. Fig. 4.7 shows at (a) an original composite signal in which the front channel sound is stronger than the back channel sound, and at (b) the effect that the QS Vario-Matrix decoder has on the crosstalk signals.

This shows how the front and back crosstalk is suppressed while the original content of both signals remains, the net result being enhancement of inter-channel separation without impairment of the original content. Inter-channel separation of 20dB or more is claimed for the QS Vario-Matrix technique.

QUASI-QUADRAPHONY

The kind of four-loudspeaker reproduction that results when a regular stereo disc or a matrix disc is played through a system using a simple Hafler type loudspeaker matrix or a similar matrix between the two outputs of a stereo control unit and four power amplifiers, one for each loudspeaker (see also under *Synthesizing*).

QUADRADISC

A discrete four-channel disc using the carrier principle (see under *CD-4 Technique*). Such records are marketed by RCA of America and by Warner-Elektra-Atlantic group labels.

'QUADRAPHONIC' SOFTWARE COMPATIBILITY

For optimum results, the decoder matrix should correspond to the encoder matrix used originally for disc (or tape) recording. However, it is possible to play any type of matrix disc through any type of decoder with varying degrees of success. Some observations in this respect are given below.

A *QS-encoded disc* played through an *SQ decoder* is somewhat lacking in directionality, allied with a reduction in front-centre to back-centre separation; played through a *Dynaco decoder* the reproduction is satisfactory but there is some loss of image accuracy; played through an EV (EVX-44) decoder, the front signals are fairly well localised, but not the back signals.

An *SQ-encoded disc* played through a *QS encoder* has good inter-channel separation, and is intimated by Sansui to be virtually compatible with a QS disc played through a QS Vario-Matrix decoder; played through a *Dynaco decoder*, the front left and right signals are angularly correct, but the positions of the other signals are put in error; played through an *EVX-44 decoder*, there is generally acceptable separation, but the positions of the back signals are put in error.

REGULAR MATRIX

This is the basic matrix decoder, which is one of three standards established by the Japan Phonograph Record Association. A typical regular matrix (RM) decoder is the QS decoder without QS Vario-

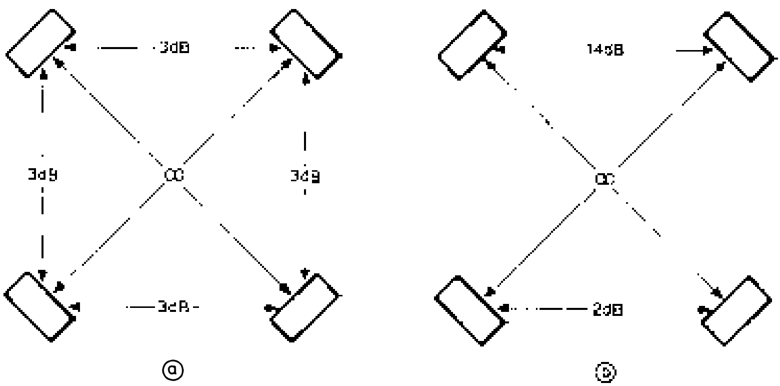


Fig. 4.8 (a) QS play through regular matrix decoder and (b) QS play through EVX-44 decoder.

Matrix enhancement. Fig. 4.8 shows at (a) the RM matrix playing a QS-encoded disc (revealing the symmetry) and at (b) the EVX-44 matrix playing a similar disc.

SCHEIBER TECHNIQUE

A matrix system proposed by Scheiber of America, from which the RM stemmed.

SHIBATA STYLUS

To define accurately the f.m. carrier modulations of CD-4 (Quadradisc) records, extending to 45kHz, the active radius of the stylus must be as small as possible and the high-frequency resonance when in contact with the groove walls as high as possible. These require-

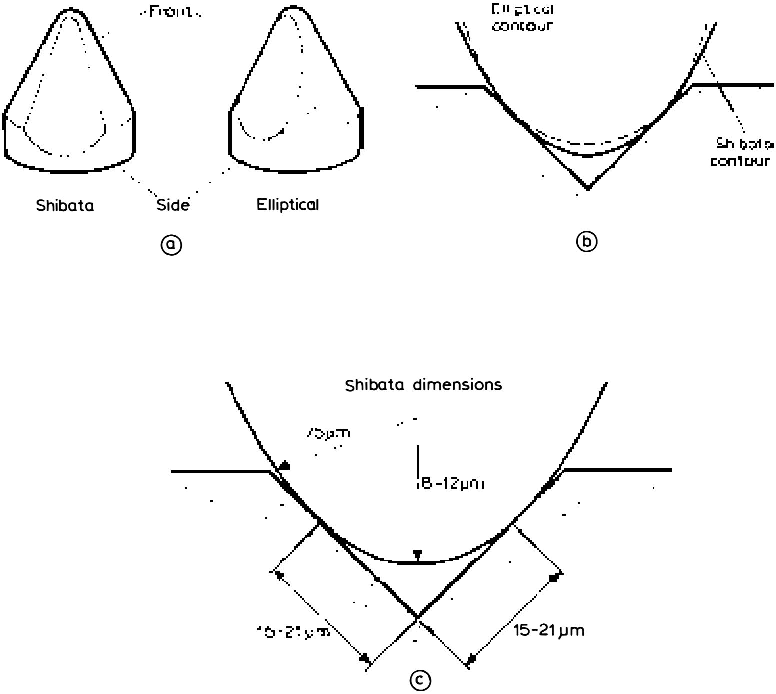


Fig. 4.9 Aspects of the Shibata CD-4 stylus. See text for details.

ments are achieved with the multiradius tip of the Shibata stylus, which is shown in comparison with an elliptical stylus at (a) in Fig. 4.9.

While the active radius is smaller than that of the elliptical, the sides of the Shibata are less rounded such that there is about four times more contact area with the groove walls, as shown at (b). This results in the distribution of the tracking force over a greater surface area and hence an increase in the high-frequency resonance. The tip dimensions are given at (c).

An impression of the improvement in high-frequency response and separation of the Shibata over an elliptical stylus in a given cartridge can be gleaned from the curves in Fig. 4.10. Owing to the greater contact areas, a pickup system incorporating a Shibata stylus can be tracked at a greater force than a system incorporating an elliptical stylus for a given record wear. The Shibata stylus can also be used to play regular stereo and matrix discs. The ordinary stylus used for playing stereo records, however, is also suitable for matrix discs.

The Shibata stylus is under continuous improvement, and at the time of writing about 200,000 are made each month. As 'square pole'

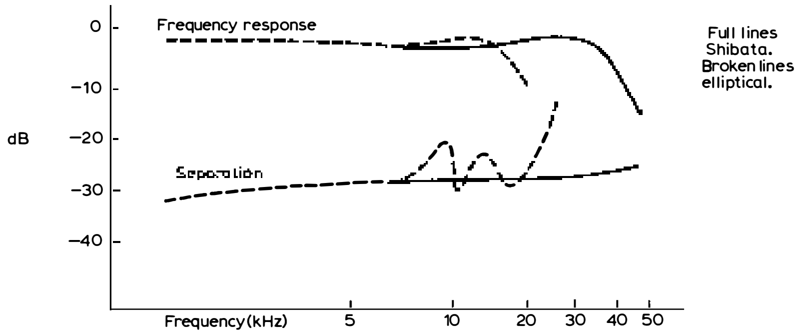


Fig. 4.10 Frequency response and separation characteristics of elliptical and Shibata styli compared.

naked diamonds will obviously rise in price and become increasingly difficult to obtain, the Namaki Precision Jewel Company of Tokyo are making preparations for the mass-production of the Shibata type of stylus based on bonding titanium to diamond in a composite stylus. These titanium-bonded tips are said to provide the required performance but with reduced wastage and good material utilisation.

It is also noteworthy that JVC now have improved CD-4 disc cutting equipment with a claimed 30Hz–15kHz frequency response, 65dB S/N ratio, 85dB minimum dynamic range and 50dB separation.

Another stylus engineered for CD-4 playing is the *Ichikawa*, which also provides good high-frequency resolution. A more recent one is the Band O *Premanik* stylus.

SOUND FIELD

The sound pattern yielded by a sounding source or sources in a given environment due to direct and reflected sounds and the acoustical nature of the environment.

SQ TECHNIQUE

Another matrix technique developed by the CBS Laboratories Incorporated of America and under promotion by CBS Records and Sony of Japan. It exhibits good frontal separation, though less front-to-back separation, depending on the nature of the decoder. The rear channels are recorded with 90° phase shifts in counter directions, which imparts a roughly helical motion to the stylus—clockwise for the left channel, anti-clockwise for the right channel.

The frontal separation ensures good stereo compatibility of the software, but in mono play some centre-stage sounds may cancel, depending on the method of recording. QS has lesser frontal separation in stereo play, the effect being a widening of the front sound stage, and in mono play cancellation appears to be somewhat less troublesome.

Phase shifters are also used, of course, in the decoder, and one design adopts a blend coefficient (i.e., SQ 10–40 Blend Decoder), and the replay of SQ software through this decoder is shown in Fig. 4.11 at (a).

Another design uses gain-controlled logic, there are various modes

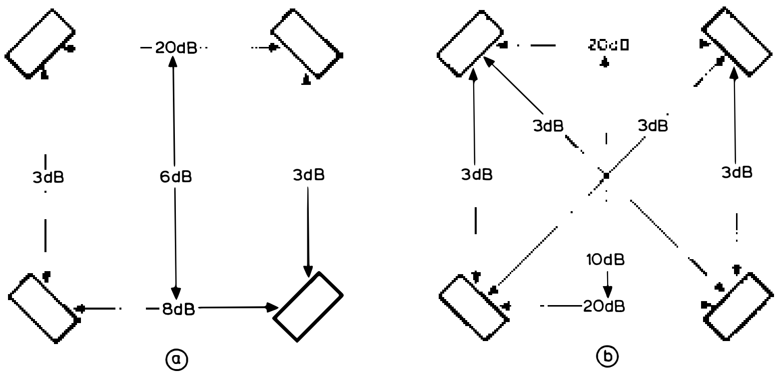


Fig 4.11 (a) SQ play through blend decoder and (b) SQ play through decoder with front-back logic.

of this, one termed wave-matching logic and another front-back logic. Gain-controlled stages are adopted such that the gain is automatically regulated in accordance with the channel information to improve the separation under dynamic conditions (see under *Gain-Control Logic*).

An impression of the difference between SQ play through a blend decoder and through a decoder with front-back logic and 10–40 blend can be gleaned by comparing (a) with (b) in Fig. 4.11. Excellent separation between the two front and the two back channels is secured and acceptable front-to-back separation, while the diagonal separation is down to the 3dB mark. An SQ 'full-logic' decoder achieves inter-channel separation of 20dB or so, depending on its class.

The basic encoding equations are:

$$L_T = L_F + 0R_F - j0{\cdot}707L_B + 0{\cdot}707R_B$$
$$R_T = 0L_F + R_F - 0{\cdot}707L_B + j0{\cdot}707R_B$$

and the decoding equations:

$$L_F = L_T \qquad R_F = R_T$$
$$L_B = 0{\cdot}707(-jL_T + R_T) \qquad R_B = 0{\cdot}707(L_T - jR_T)$$

where j denotes 90° phase shifts.

Further information on SQ decoder circuitry can be found in H. W. Hellyer's book entitled *Stereo Sound*, by the publishers of this book.

SURROUND SOUND

A general term for four-loudspeaker reproduction, applicable to the various 'quadraphonic' arrangements as well to the simple matrix and synthesizing arrangements, but is more generally adopted to describe quasi-quadraphony.

SYNTHESIZING

A term sometimes used (particularly by Sansui) to describe quasi-quadraphony. A synthesizer is a form of matrix which translates the regular left and right channel stereo signals into four signals (albeit, with significant redundancy) for four-loudspeaker reproduction.

Some four-channel amplifiers are equipped with a switch in the matrix circuitry to provide such synthesizing.

UD-4 TECHNIQUE

A phasor-matrix/carrier combination devised by a Dr. Duane Cooper of the University of Illinois and developed by the Nippon Columbia Company Limited. The phasor-matrix (sometimes called phase-matrix, which is a technique that encodes four source signals into two circuits by placing the former at certain phase relationships, such as QS, etc.) is combined with two supplementary channels using a 30kHz carrier of restricted bandwidth and frequency modulation. The maximum frequency is about 40kHz which is less than that of CD-4.

The technique is engineered so that the supplementary channels need not be used, but when they are 'directivity' is enhanced and listener placement rendered less critical. The scheme is highly compatible with mono without directional anomalies, with two-channel stereo, with four-loudspeaker surround sound and with discrete software.

The phasor-matrix yields symmetrical 45° distribution so that unwanted emissions either side of the wanted ones are phased by $\pm 45°$ which is a factor in favour of localization of sound images.

At the time of writing UD-4 software is being released, and a complementary scheme for f.m. broadcasting has been evolved.

2-2-2

Shorthand sometimes used to describe a two-channel stereo system. The first digit corresponds to the number of source channels, the second to the number of transmission or amplification channels and the third to the number of loudspeakers.

2-2-4

Shorthand describing quasi-quadraphony or synthesizing, where there are two source channels (i.e., stereo), two amplification channels and four loudspeakers.

4-2-4

Shorthand describing the matrix technique, where there are four source signals condensed to two amplification channels and retrieved by decoding for application to four loudspeakers.

4-4-4

Shorthand describing the discrete technique, such as CD-4 or, more accurately, four-channel tape, where there are four source channels, four amplification channels and four loudspeakers.

CHAPTER FIVE

RADIO TUNERS AND AERIALS

AERIAL MATCHING

THE SIGNAL ENERGY abstracted from a passing radio wave by the aerial needs to be transferred to the aerial input of the tuner with the least possible loss. The v.h.f. aerial used for f.m. (and some television channels) is coupled to the tuner through feeder cable (sometimes called downlead), and to ensure the least signal loss the feeder needs to be matched both to the terminal impedance of the aerial and to the nominal aerial input impedance of the tuner.

In the UK the feeder is commonly coaxial cable (tube of flexible outer braid through which an inner conductor is supported concentrically by a low-loss insulating material) whose characteristic impedance is around 75Ω. The value of 75Ω was chosen for this sort of coaxial cable because this is the natural impedance at the centre point of a tuned dipole aerial. Thus, correct matching at the aerial is achieved by connecting such feeder direct to the dipole.

The centre impedance of a dipole falls when additional elements are used, but various artifices are used by aerial manufacturers so that the terminal impedance of the aerial array remains essentially at 75Ω irrespective of the total number of elements comprising the array. This impedance correction back to 75Ω (or thereabouts) may be applied by a transformer or coaxial stubs.

Since a dipole is a balanced device and coaxial cable is unbalanced, the aerial matching transformer may also reflect a balanced coupling of the correct impedance to the aerial terminals from the unbalanced feeder, but all these things are (or should be) taken care of by the aerial manufacturer.

Tuners destined for the British market are mostly equipped with aerial input terminals of also 75Ω impedance, unbalanced. Thus by connecting 75Ω coaxial cable to the aerial and connecting the other end of the cable to the 75Ω unbalanced aerial input of the tuner

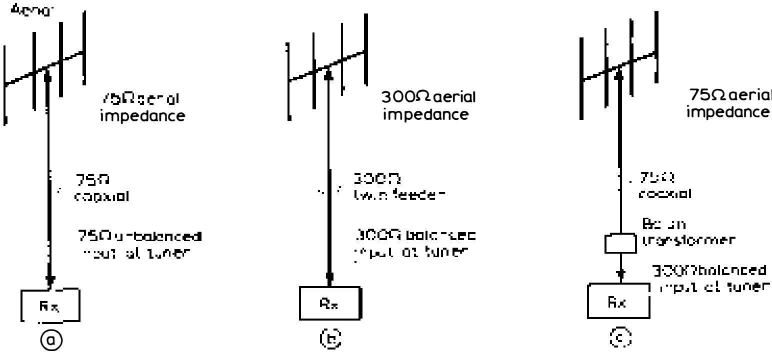


Fig. 5.1 Aerial couplings. (a) using coaxial feeder between aerial and tuner. (b) using 300Ω balanced feeder between aerial and tuner. (c) using a balun transformer for matching a 75Ω unbalanced aerial system to a tuner with 300Ω balanced aerial input terminals.

the requirements for correct aerial matching are achieved, as shown at (a) in Fig. 5.1.

In America, balanced feeder cable of 300Ω characteristic impedance is commonly adopted and since aerials in that country are designed for 300Ω terminal impedance, correct matching is again achieved by connecting the feeder to the 300Ω balanced aerial inputs of the tuner, as shown at (b) in Fig. 5.1. Hi-fi tuners are often made with both 300Ω balanced and 75Ω unbalanced aerial inputs.

In European countries the standard is 240Ω balanced, but this is close enough to the American 300Ω balanced condition for the mismatch to be insignificant when a 300Ω aerial system is adopted with tuners of 240Ω balanced input.

A more serious mismatch is likely when a tuner with 300Ω (or 240Ω) balanced aerial input is used with a 75Ω unbalanced British type aerial system. Considerable loss of signal transfer can occur by connecting 75Ω coaxial direct to 300 or 240Ω aerial terminals.

There are two ways round this problem. The best one is to use an unbalanced-to-balanced transformer (called 'balun' for short) between the coaxial cable and the 300Ω aerial terminals, as shown at (c) in Fig. 5.1. Such transformers with low-loss characteristics are available for such applications. They have a step-up ratio of 2:1 which yields an impedance step-up of 4:1, while changing the unbalanced input to a balanced output. Thus the feeder impedance is stepped up from 75Ω to 300Ω and is changed from the unbalanced to the balanced condition.

Balanced feeder differs from coaxial feeder in that the two conductors are often supported side-by-side by a plastics dielectric, which gives the appearance of a ribbon, the name sometimes used to describe it. In a different variety the two conductors are screened by an outer metallic braid, this then looking more like coaxial cable.

AERIAL ROTATOR

F.m. aerials are directional, the directivity becoming more apparent as the number of elements used for the array is increased. For optimum response of a required signal, therefore, the aerial needs to be beamed on to the station. Most reception conditions are satisfied by carefully orientating the aerial at the time of erection for the best results from the regional group of f.m. stations.

Some enthusiasts, however, endeavour to receive distant f.m. stations, and to avoid having to turn the aerial by hand each time it is required to receive a station on a different bearing an aerial rotator is employed. This is an electric motor which turns the aerial through 360° by remote control from the receiver location. The remote control unit generally incorporates a compass scale so that the enthusiast can see which way the aerial is pointing.

AMPLITUDE MODULATION

Amplitude modulation (a.m.) means that the audio information is superimposed on the carrier wave in the form of amplitude variations. The carrier wave is a constant amplitude and constant frequency radio-frequency (r.f.) wave which is radiated by the transmitter and picked up by the receiving aerial. Such a wave is shown in Fig. 5.2 at

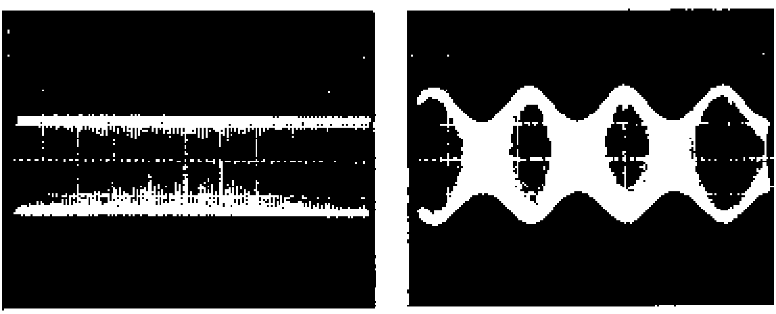


Fig. 5.2 (a) unmodulated carrier wave. (b) carrier wave which is amplitude modulated with 1kHz sinewave signal.

(a). This carries no audio information, so when tuned in by a receiver it would not be heard.

When the wave is amplitude modulated its envelope varies in accordance with the information the modulation contains, as shown by wave (b) in Fig. 5.2, which is modulated by 1kHz sinewave (tone) signal. The detector or demodulator circuit of the tuner removes the carrier wave and processes the modulation waveform, and it is this which is then reproduced by the loudspeaker.

The radio signals of the long, medium and short wavebands are amplitude modulated, so to receive these one needs an a.m. tuner covering the wavebands of interest. Long, medium and short waveband signals are generally propagated over greater distances than v.h.f. signals used for f.m. broadcasting. This leads to over-crowding of the bands, particularly of the medium wave band after dusk; so, to prevent adjacent channel interference, a.m. tuners are designed with a limited i.f. bandwidth so that the side stations are attenuated.

The resulting suppression of the higher-order sidebands cuts the treble modulation frequencies, which is one reason why the reception quality of a.m. tuners is below that of f.m. tuners, since in the v.h.f. band the distance of signal propagation is curtailed so the bandwidth restriction does not apply. There are other reasons why the f.m. system is capable of better fidelity than the a.m. system (see also under *Frequency Modulation*).

A.M. TUNER

A tuner designed specifically for the reception of a.m. signals. Such a tuner is not suitable for stereo or hi-fi reproduction.

A.M./F.M. TUNER

A tuner which is capable of receiving a.m. signals as well as f.m. signals. The a.m. section may tune over the long, medium and short wavebands, but there are many a.m./f.m. tuners whose a.m. section covers only the medium waveband, and the signals in this band are often picked up by a rear ferrite rod aerial (see under *Ferrite Rod Aerial*).

AUDIO OUTPUT OF TUNERS

The audio output of a tuner must suit the tuner or auxiliary input sensitivity of the partnering amplifier. This is because a tuner does not contain its own power amplifier section; it is rather like a receiver devoid of the output stages. A tuner, in fact, is a programme signal source.

The signal output is typically 1V r.m.s. for 100% modulation, which puts the 'average' music signal output around 300mV. The tuner or auxiliary input of most amplifiers is able to cater for this level of signal with adequate overload margin and for full amplifier drive with the amplifier's volume control set between 12 and 1 o'clock.

A low tuner output impedance is to be preferred since then the capacitance of the screened signal connecting cables is less likely to suppress the treble frequencies. If long, high-capacitance screened leads are used to connect a tuner with a high output impedance to the amplifier the time-constant of the de-emphasis filter in the tuner might be increased, resulting in early treble roll-off.

'BIRDIES' INTERFERENCE

A colloquial term describing the kind of effect caused by certain types of interference, notably that on stereo transmissions due to the interaction of adjacent channel signals. One cause of the disturbance is described by Fig. 5.3. Here the tuner is adjusted to a required

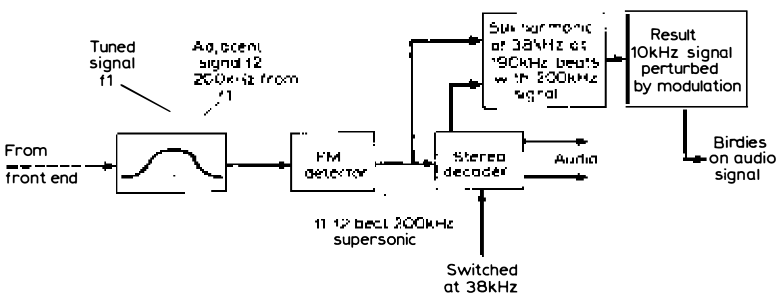


Fig. 5.3 Showing the cause of one kind of birdies interference (see text).

signal f_1 while an unwanted adjacent channel signal f_2, 200kHz from f_1, is also causing a small response. A beat at 200kHz is thus present at the output of the f.m. detector, but this is obviously supersonic and hence inaudible.

The stereo decoder is switched continuously at 38kHz, the stereo subcarrier frequency, and resulting from the squarewave switching signal odd-order harmonics at 114, 190, 266kHz, etc. are produced.

In this case the fifth harmonic at 190kHz beats with the 200kHz detector signal such that a 10kHz signal is generated in the audio circuits.

Since this signal is perturbed by the modulation, it manifests as a high-pitched twittering noise, called birdies. Other combination of signals can also produce the effect, such as a 100kHz adjacent channel beat itself beating with the third harmonic of the switching frequency at 114kHz, giving rise to a perturbed signal centred on 14kHz.

Good i.f. selectivity (steep side skirts to the response characteristic, as shown in Fig. 5.4) and low-pass filtering of turnover *circa* 53kHz between the f.m. detector and stereo decoder significantly reduce the tendency for this kind of interference.

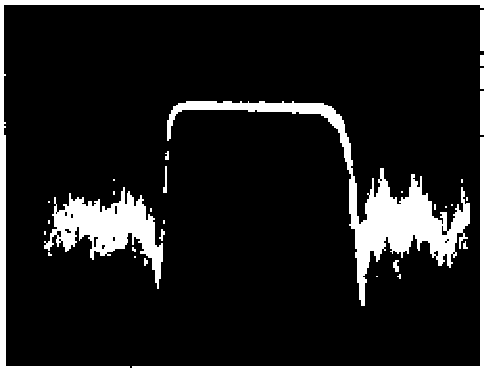


Fig. 5.4 I.F. response characteristic of high quality f.m. receiver, showing the very steep side skirts.

CAPTURE RATIO

An attribute to f.m. receivers whereby an unwanted signal is suppressed by a slightly stronger wanted signal at the same frequency is called the *capture effect*. The capture ratio indicates by how much stronger the wanted signal needs to be than the unwanted signal for the disturbance on the wanted signal to be 30dB below the normal output resulting from 100% modulation. Full details of the procedure for measuring capture ratio are given in the *Audio Technician's Bench Manual*.

This important f.m. tuner parameter is expressed in decibels, the average capture ratio being between 2 and 3dB. Some top-flight tuners, however, possess a capture ratio of less than 1dB.

The capture effect results from the insensitivity of an f.m. tuner to amplitude modulation (see Fig. 5.2). When two signals are present simultaneously in the same circuit the tendency is for one to amplitude modulate the other. Now, because an f.m. tuner incorporates an amplitude limiter and detector with de-emphasis whose output is proportional to input frequency (not amplitude), amplitude variations of the signal cause little or no output from the detector.

In general, the more stages of amplitude limiting there are in the tuner's i.f. channel and the greater the bandwidth of the f.m. detector, the smaller the capture ratio.

DE-EMPHASIS

To enhance the S/N ratio of f.m. reception the treble modulation frequencies are progressively boosted at the transmitter from a certain frequency determined by a simple time-constant circuit. To secure a flat response at the output of the tuner complementary de-emphasis is applied after the f.m. detector or stereo decoder, which also automatically reduces the high-frequency noise signals, which are subjectively more undesirable than lower frequency noise signals.

A simple time-constant de-emphasis circuit is given in Fig. 5.5.

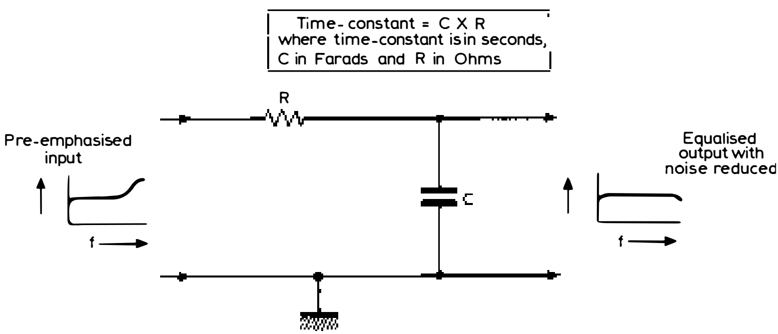


Fig. 5.5 De-emphasis time-constant circuit which is present in each channel of a stereo tuner. A time-constant of 50μS would result from a resistance of 10kΩ and a capacitor of 5nF. The de-emphasis yields treble cut which restores the frequency balance of the treble lift of the transmitter pre-emphasis.

The frequency at which the 3dB boost or cut occurs is related to the time-constant by the expression $T=1/2\pi f$ or $f=1/2\pi T$, where T is the time-constant in seconds and f the -3dB frequency. The British (European) time-constant is 50μS and the American one 75μS, these putting the 3dB frequencies at 3184Hz and 2123Hz respectively.

The American 75μS time-constant means that the treble boost and hence the treble cut comes on slightly earlier than those of the European 50μS time-constant. In other words, the pre-emphasis is greater in the American system than the European system. Thus if a tuner with the American 75μS time-constant is used on a European transmission the treble rolls off sooner than it should do.

DIRECTIONAL AERIAL

This is an aerial which favours the signals arriving at one angle more than those arriving at different angles. See under *Polar Diagram of Aerials* and *Yagi Aerial.*

DIPOLE AERIAL

A metal conductor whose length is related to the wavelength or frequency of the signal to be received and which is connected to the feeder or downlead usually at the centre point where the impedance is low and represents an acceptable match to 75Ω coaxial cable, as shown in Fig. 5.6. The dipole is generally cut to approximately half

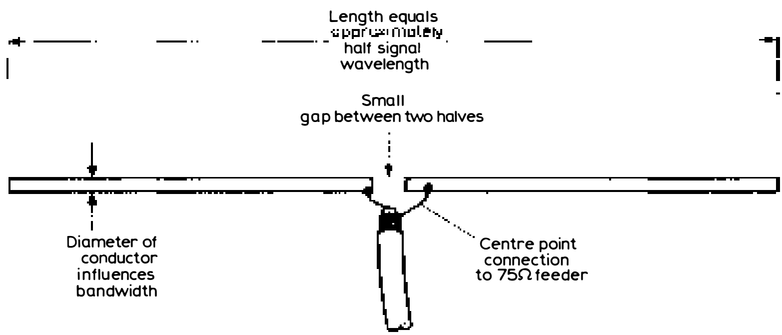


Fig. 5.6 Primary features of a simple dipole aerial. See text for details.

the wavelength of the signal frequency, but it will respond to signals either side because its tuning is not very sharp, the sharpness of tuning being determined by the length/diameter ratio of the conductor.

The wavelength of a signal is equal to the velocity of radio waves through space, which is virtually 300×10^6 metres per second, divided by the frequency of the signal in Hz. Thus the wavelength in metres when the frequency is in MHz is equal to 300/MHz.

Band II, the part of the v.h.f. spectrum which carries f.m. transmissions, extends from about 88MHz to 108MHz, so if it is required for the dipole aerial to peak somewhere towards the middle of the band, say at 95MHz, the dipole would have a length of approximately $\frac{1}{2}$300/95, which works out to 1·579 metres or 5·18ft.

In practice, the length used for the dipole is slightly less than this because the wave velocity is reduced slightly on passing from space through the dipole, a typical mid-band length being 5ft or a little less, depending on the *velocity factor* of the dipole, which is a function of the length/diameter ratio.

A simple dipole has a figure-of-eight polar diagram, which means that it picks up most signal when facing broadside to the station, and much less when either end of the dipole is pointing to the station.

DISTORTION

Radio tuners are plagued by non-linear and frequency distortion just as any other item in the hi-fi chain. Total harmonic distortion, however, is generally low, and referred to 100% modulation should not be much more than 0·5% mono and 1% stereo at 1kHz. Top-flight tuners have distortion factors lower than this.

The main cause of frequency distortion is inaccuracy of the de-emphasis either due to insufficient design detail or to increase of the de-emphasis time-constant by the use of long screened leads connecting the tuner to the amplifier.

DOLBY NOISE REDUCTION ON F.M.

The Dolby noise reduction system is an artifice, used for making gramophone records (Type A) and for home tape and cassette recording and replay (Type B, which is less elaborate than Type A), which results in a significant (*circa* 10dB) improvement in S/N ratio. It is a complementary technique whereby the recording signal needs to be subjected to Dolby encoding and the replay signal to complementary Dolby decoding.

It has been found that the kind of noise which can plague f.m. reception when the aerial signal is insufficiently strong, particularly when a not-too-strong stereo signal is being decoded by a stereo tuner, can also be processed by Dolby B such that a similar S/N ratio improvement results, which is tantamount to an increase in transmitter power (from the S/N ratio point of view) or an increase in local signal field or aerial signal fed to the tuner, which could otherwise be achieved only by the use of a higher gain aerial possibly at higher elevation.

In America the Federal Communications Commission (FCC) has already proclaimed that f.m. stations in that country are now free to use Dolby B noise reduction combined with a reduction in pre-emphasis from 75μS to 25μS.

Dolby B encoding *progressively* boosts the higher audio frequencies as the level of the audio signal reduces. The decoding works in a complementary manner such that the higher audio frequencies are progressively cut as the level of the audio signal reduces. Thus, at low signal level there is maximum treble lift and maximum treble cut, but since the encoding lift and decoding cut are always of the same complementary order at all signal levels the balance of the audio output signal is continuously maintained (i.e., the system does not result in frequency distortion).

Now, since noise is more troublesome on low level signals, particularly at the upper end of the spectrum, the boosted signals tend to mask the noise, so that during decoding not only is the signal balance restored, but the noise output is reduced, the noise reduction being greater as the signal level reduces. In other words, the technique is rather like variable pre- and de-emphasis, but unlike ordinary pre- and de-emphasis, which is fixed, there is far less danger of the various stages being overloaded by high amplitude, high-frequency signals.

In America, where the f.m. time-constant is 75μS, it has been found that by combining Dolby B encoding at the transmitter with 25μS pre-emphasis, an acceptable frequency balance is secured from tuners and receivers not equipped with Dolby B decoding, but which are endowed with the normal (for America) 75μS de-emphasis.

However, when the tuner or receiver is equipped with Dolby B decoding and the de-emphasis is reduced to 25μS not only is the frequency balanced still maintained, but the S/N ratio is also improved at the same time. In this manner the scheme is said to be compatible.

In the UK and Europe there is likely to be lesser compatibility owing to the time-constant of tuners and receivers being 50μS, against the American 75μS. The technique could thus possibly result in a somewhat 'brighter' treble output on receivers not equipped with Dolby B decoding and 25μS de-emphasis. This, of course, is frequency distortion, though improvement in frequency balance could possibly be achieved by turning down the treble tone control slightly.

At the time of writing no plans have been finalised for the use of the scheme in the UK, though it is understood that f.m. stations of the

IBA might experiment with it.

It is claimed that an added advantage of reducing the transmitter pre-emphasis from 75μS or 50μS to 25μS is that less peak limiting becomes necessary to prevent overmodulation of the transmitters on high amplitude treble signal, but the BBC overcome this problem by the use of variable de-emphasis limiters (which are not the same as signal clippers) which automatically reduce the pre-emphasis momentarily when there is very large amplitude, high-frequency content in the modulation signal.

The transmitters carrying stereo transmissions are mostly equipped with this type of limiter, but for most of the time the pre-emphasis is 50μS since the high-frequency signal amplitude is not large enough to bring them into action, which would tend to imply that with 50μS pre-emphasis the danger of overmodulation is not very great.

DOWNLEAD

The coaxial cable or twin feeder connecting the aerial to the tuner. On a long-wire aerial used for a.m.-band reception, the lead which connects the aerial to the receiver or tuner is also called a downlead.

DX RECEPTION

Another name for long-distance reception. For example, a DX enthusiast is a listener who delights in receiving distant stations on the f.m. or a.m. bands. Such enthusiasts often employ very elaborate receivers or tuners and aerial systems with rotators.

EARTHING

For optimum safety every separate item of the hi-fi chain when it is separately mains powered should be earthed by way of a conductor large enough to carry a fault current at least 2·4 times the rating of the protective fuse without overheating melting or showing undue distress. Thus, if the mains fuse is, say, 13A, the earthing lead should be able to accommodate at least 31·2A.

However, to avoid hum loops (see under *Hum Loop* in Chapter 1, for example) it is commonly accepted that the hi-fi system should be earthed only at one point. This is generally at the amplifier, the radio tuner then picking up the same earth via the braids of the signal connecting leads.

As the tuner and amplifier are separately fused, the latter often at less than 1A and the former not greater than 2A on 240V mains supplies, the fault current through the braids should never rise to a dangerously high value; the equipment fuse should certainly 'blow' before the braids show distress.

As a further aid to safety, the fuse in the domestic circuit powering the hi-fi system (or that in the plug used on a ring-mains system) should not be greater than 2A. Certainly never 13A!

FEEDER

Another name for downlead. See under *Downlead* and *Dipole Aerial*.

FEEDER LOSS

The loss in decibels that a signal of specified frequency suffers in passing through a given length of feeder cable. Special low-loss feeders are available for use in weak signal field areas where every microvolt of signal at the tuner counts.

FERRITE ROD AERIAL

A ferrite rod of some 150mm in length carrying a coil or coils of wire of inductance, in conjunction with the ferrite rod, which is suitable for tuning over the medium-frequency bands of interest. This sort of aerial is suitable essentially for the long and medium wavebands, and is commonly found at the rear of tuners and receivers which are equipped with a medium wave band (or bands) in addition to Band II, the f.m. band.

The inductance of the ferrite rod usually constitutes the high Q-factor front-end tuning of the a.m. section, so it is not usually possible to disconnect the aerial. Such tuners, however, often incorporate a terminal for connecting an external long-wire a.m. aerial, the signal from this being inductively or capacitively coupled to the ferrite rod aerial. For best results with an external aerial a good earth should also be connected to the tuner.

As a ferrite rod aerial rarely operates on the short wavebands, an external aerial (as with f.m.) is required for reception in these bands when the tuner or receiver is so equipped.

In certain models it is, in fact, possible to switch out the ferrite rod aerial when an external aerial is used. A ferrite rod aerial is also unsuitable for v.h.f. reception in Band II, but one or two models are equipped with some kind of internal f.m. aerial which is suitable only for very favourable reception conditions. A simple folded dipole may be fitted inside the cabinet or enclosure or the mains lead itself may be a resonant length and coupled to the input stage through a suitable mains isolating filter.

The ferrite rod aerial is the modern rendering of the old style frame aerial, but it is much smaller because the ferrite rod concentrates the magnetic component of the radio wave through the coil or coils. The aerial thus has to be lined-up with the magnetic component, which makes the aerial somewhat directional.

In this respect it is like the dipole aerial, giving maximum pick up when facing broadside to the station and the least response when either end is pointing to the station. This feature is useful in securing the best signal-to-interference ratio, particularly when TVI (television interference) is troublesome.

Unfortunately, on some tuners or receivers the aerial can only be

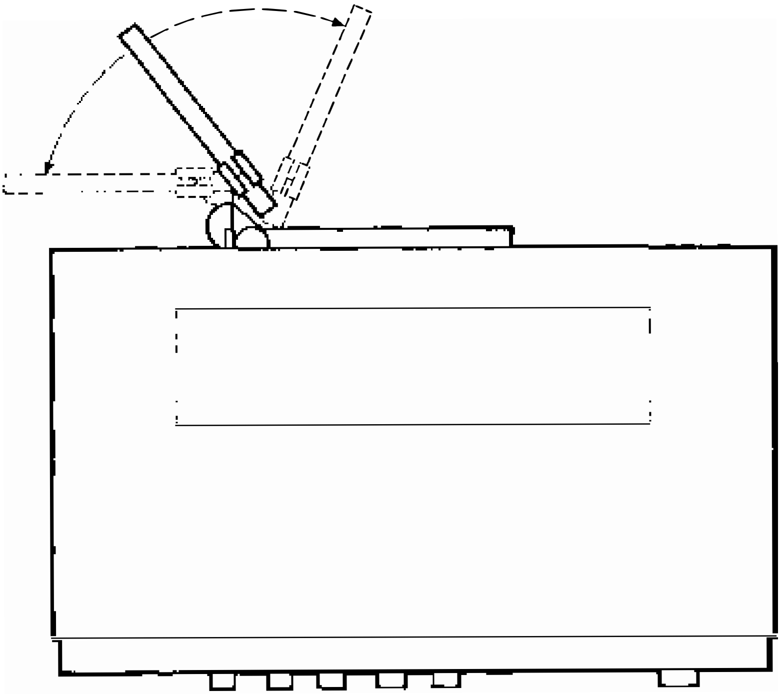


Fig. 5.7 Some ferrite rod aerials can be swivelled over 180° as shown, and the orientation for maximum response or best signal/interference ratio should be used.

hinged down away from the metal rear panel, which means that for optimum orientation the whole of the receiver or tuner needs to be turned. Other designers, however, have overcome this problem by making the aerial swivel on the horizontal plane, as shown in Fig. 5.7. The aerial is most efficient when clear of metal objects.

F.M. TUNER

A tuner which is designed solely for f.m. reception in the Band II part of the v.h.f. spectrum. The kind of tuner adopted by enthusiasts whose prime interest lies in securing the highest possible quality programme signals via radio. Design compromises are often fewer in f.m.-only tuners, though there are exceptions.

FREQUENCY MODULATION

Frequency modulation (f.m.) means that the audio information causes the frequency of the carrier wave to deviate either side of its nominal frequency, the amplitude remaining constant. Fig. 5.8 shows at (a) an unmodulated carrier wave, while at (b) is shown the same

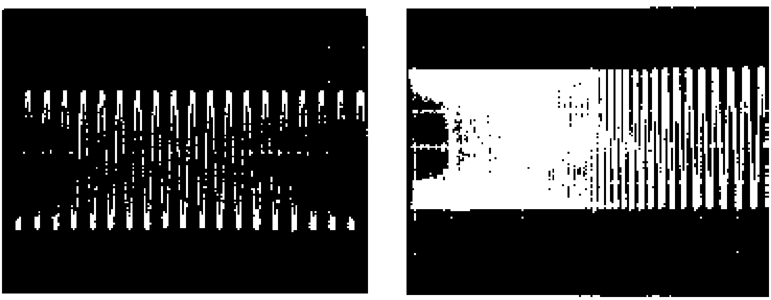


Fig. 5.8 Unmodulated carrier wave (a) and frequency modulated wave (b). In this illustration the modulation consists of a ramp waveform which progressively increases the frequency of the wave over the period of the ramp waveform. With f.m. the amplitude remains constant (compare with a.m. in Fig. 5.2) and the frequency deviates below and above the nominal carrier frequency at a rate dependent on the modulation frequency and by an amount dependent on the amplitude of the modulation signal. See text.

wave which is frequency modulated by a ramp waveform, where it will be seen that the frequency of the wave progressively increases.

On steady-state or music signal, of course, there are plus and minus changes in nominal carrier frequency. The *rate* at which the frequency changes is governed by the frequency of the modulation signal and the amount of the frequency change, or *deviation* as it is called, by the level of the modulation signal. Thus, the louder the sound in the studio, the greater the deviation.

The maximum deviation of f.m. 'entertainment' broadcasting stations is $\pm$75kHz, corresponding to 100% modulation level. This means that on the loudest sounds the carrier wave increases by 75kHz and decreases by the same amount, depending on the nature of the modulation waveform.

Unwanted signals and various kinds of interference tend to vary the amplitude of the modulated wave, but since a well designed f.m. tuner has minimal sensitivity to such variations (see under *Capture Ratio* and *Limiting*) the unwanted signals and interference cause very little response in the audio circuits following the detector, so the reproduction (provided the aerial signal is sufficiently strong) is unaffected by the spurious information, which is a major attribute of the f.m. system over the a.m. system of broadcasting.

Frequency modulation yields a large number of side-frequencies whose frequencies differ from the carrier frequency by multiples of the frequency of the modulation signal, but the high-order ones are of very small amplitude. Nevertheless, the passband of an f.m. tuner needs to be about 240kHz to the −6dB points, particularly on stereo, to avoid attenuation of the more significant, larger amplitude side-frequencies.

This is another area where f.m. differs from a.m., for with a.m. a single tone modulation signal results in the production of one pair only of side-frequencies (or sidebands), these being located below and above the carrier frequency by a frequency corresponding to the modulation frequency. Hence an a.m. tuner does not require a passband as large as that of an f.m. tuner.

Indeed, in practice the tuner passband is deliberately restricted to less than that required to accommodate the high modulation frequencies as a means of minimising adjacent channel interference—whistles and 'monkey chatter'—which might otherwise result due to the crowding of the a.m. bands, particularly after dusk when medium-frequency signals are propagated over significant distances. This, of course, reduces the treble response, which is another reason why contemporary a.m. broadcasting is incapable of yielding the high quality of f.m.

With f.m., the number of side-frequencies is related to the *modulation index*, which is the carrier deviation frequency divided by the frequency of the modulation signal.

FRINGE AREA

The areas which falls outside the service area of an f.m. station in particular and where consistant reception cannot be guaranteed. Usually upwards of 50 miles from an f.m. station. See also under *Service Area* and *Troposphere.*

FRONT-TO-BACK RATIO OF AERIAL

The ratio of the response of the front of a directional aerial to that at the back. For example, if the response or pickup at the front of a directional aerial is given as 0dB and the response at the back as —25dB, then the front-to-back ratio is 25dB.

INPUT DYNAMIC RANGE OF TUNER

If the least useable sensitivity of an f.m. tuner is, say, 2μV, which is a very low signal level, and front-end overloading produces spurious responses corresponding to about —30dB when the input signal level is raised to, say, 10mV, then the input dynamic range can be expressed as 10mV/2μV, which is 5,000:1 or about 74dB. The input dynamic range of f.m. tuners should be as large as possible since the total signal input could well be from a multiplicity of stations.

Overloading of the front-end manifests as intermodulation, 'birdies' interference and the presence of spurious signals over the tuning range.

LIMITING

In f.m. parlance, this generally refers to amplitude limiting in the tuner which shaves off amplitude variations of the f.m. signal, thereby decreasing the sensitivity of the tuner to amplitude modulation and interference, while also enhancing the capture ratio. See Fig. 5.9 and under *Capture Ratio.*

MONO RECEPTION

When the modulation at the transmitter consists of audio signal in one channel only and when the tuner is responding to one-channel modulation only. Not all the BBC transmitters are as yet stereo-

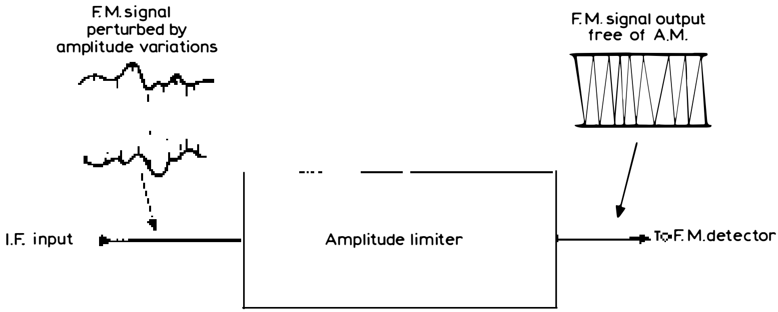


Fig. 5.9 Amplitude limiting used in f.m. tuner. Sometimes more than one stage of limiting is used (based on integrated circuits) which results in a very small capture ratio and high a.m. rejection ratio.

encoded and not all programmes from those transmitters which are stereo-encoded are in stereo. A stereo tuner operates in the mono mode when the signal it receives is not stereo-encoded.

MULTIPATH DISTORTION

A result of multipath reception which occurs when the v.h.f.–f.m. signal is reflected by large buildings or hills such that the receiving aerial picks up first the direct signal and then a small fraction of a second later the reflected signal or signals, as shown in Fig. 5.10.

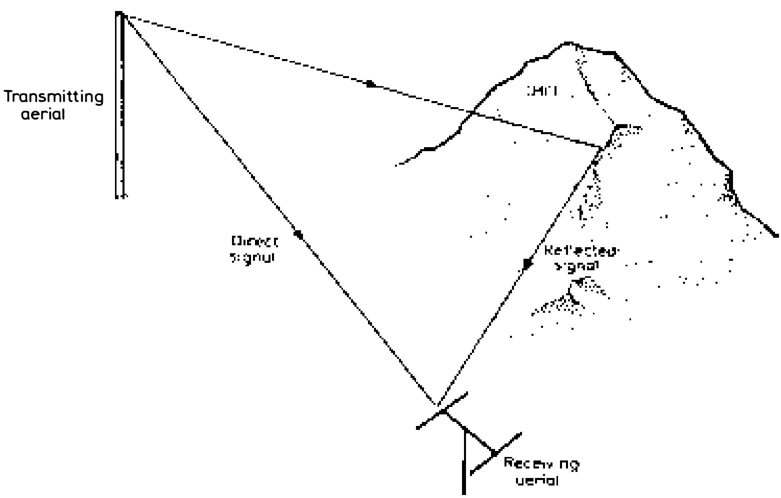


Fig. 5.10 Multipath reception as illustrated is responsible for multipath distortion. See text.

On television, the reflected signal produces a ghost image slightly to the right of the main image, but on f.m. it tends to cause an unpleasant kind of harmonic distortion, particularly at peak modulation levels, which is called multipath distortion. It can also result in a loss of stereo separation and sometimes a form of 'birdies' interference.

Since the effect results from a kind of amplitude modulation, and some tuners have a meter or built-in oscilloscope (or sockets to connect an external oscilloscope) to detect it, an f.m. tuner with a very small capture ratio and hence very good amplitude limiting suffers less from the trouble.

However, in areas subjected to signal reflections it pays to employ a directional aerial carefully orientated to provide the greatest discrimination against the reflected signal or signals, see Fig. 5.11.

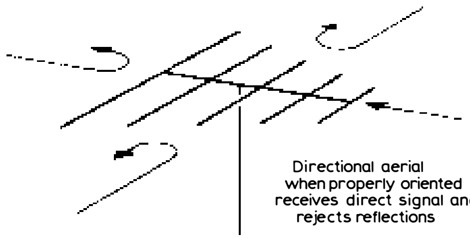


Fig. 5.11 A directional aerial should be used to combat multipath reception and hence multipath distortion.

OUTPUT IMPEDANCE

The source impedance of the audio outputs of a tuner. The lower this impedance (within reason!) the better, for then the de-emphasis is less likely to be affected by the capacitance of the screened connecting leads, particularly if they need to be extra long.

PLANE OF POLARISATION OF SIGNAL

A radio wave is sustained in space and propagated by two components, an electric component and magnetic component, these being at right-angles to each other and also at right-angles to their direction of travel through space. The radio wave is said to be polarised in the plane of its electric component. Thus, as with f.m., the wave is horizontally polarised when the lines of electric field lie in a horizontal plane. Some v.h.f. and u.h.f. television signals adopt vertical polarisation.

For maximum response by way of a dipole type of aerial or its derivatives, the aerial has to lie in the same plane as the electric field. Hence the reason for f.m. aerials being horizontally disposed and some telvision aerials being vertically disposed.

Some f.m. stations are using so-called 'tilt' polarisation, which permits reasonable response from both vertically disposed car aerials and horizontally disposed domestic aerials.

POLAR DIAGRAM OF AERIALS

A graph which reveals the degree of response of an aerial round the compass of 360° or 180°, as shown, for example, in Fig. 5.12, which

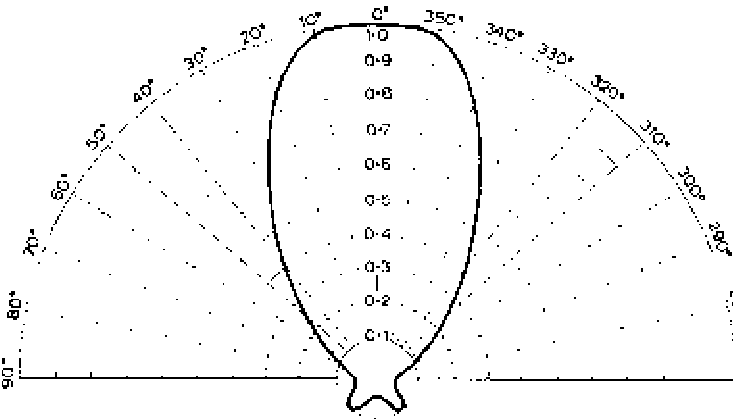


Fig. 5.12 Polar diagram as measured by the author of the Antiference FM264 six-element 'Mushkiller' aerial. This tunes the f.m. band over 88–100MHz.

is the measured polar response of the Antiference FM264 six-element *Mushkiller* f.m. aerial. The Y scale indicates the relative response from maximum (on beam) referred to unity, which puts the −3dB points at about 0·7 on the scale, so that relative to the −3dB points the overall acceptance angle of the aerial is around 57°.

Relative to a single dipole, the aerial has a forward gain of between 7·9 and 8·5dB, an average front/back ratio of 30dB and a bandwidth of 88–100MHz.

A simple dipole aerial has a figure-of-eight polar diagram, as shown in Fig. 5.13. The directivity is increased by the addition of a reflector and directors, and good directivity improves both the forward gain, relative to a dipole and the discrimination against unwanted and reflected signals arriving at off-beam angles.

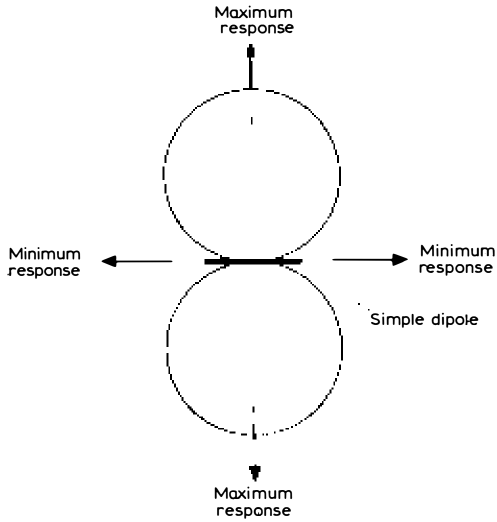


Fig. 5.13 A simple horizontally disposed dipole has a figure-of-eight polar diagram as shown.

PRE-EMPHASIS

The treble boost given to the modulation signal at an f.m. transmitter based on a time-constant, which is 50μS in Europe, including the UK, and 75μS in America. A time-constant of 25μS is used in America when Dolby B noise reduction encoding is also applied to the transmitted signal (see under *Dolby Noise Reduction on F.M.*). The BBC adopt variable de-emphasis, which momentarily reduces the pre-emphasis to avoid overmodulation when the audio signal contains large amplitude treble components.

The rising treble characteristic of pre-emphasis is corrected at the tuner by reciprocal de-emphasis (see under *De-Emphasis*). Pre- and de-emphasis are used to improve the f.m. S/N ratio.

QUADRAPHONY

In the UK no decision has as yet (late 1974) been made regarding the f.m. transmission of quadraphony, though various stations in other countries, notably Japan and America, are said to be transmitting matrix signal sources over two-channel stereo networks (see Chapter 4).

On the night of July 5/6 the BBC conducted a quadraphonic experiment using the stereo Radio 2 transmitters for the left and right front channels and the Radio 3 transmitters for the left and right back channels. This discrete quadraphony was extremely well received, and the equipment setup employed at Alan's home (the author's son-in-law—that up and coming hi-fi chap again!) is illustrated in Fig. 5.14. A second experiment was staged on Dec. 23/24.

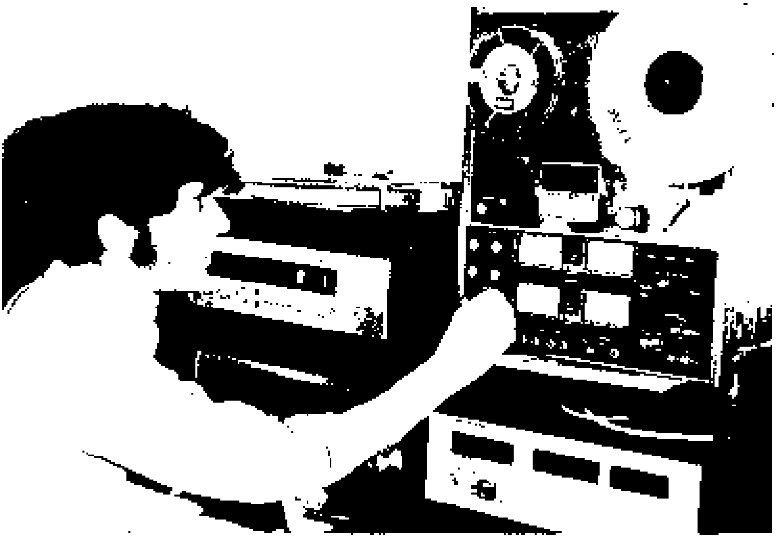


Fig. 5.14 Alan (the author's son-in-law) at the controls during reception of the BBC's experiment in quadraphony. See text for details.

The equipment included the Fuba Uka Stereo-8 aerial, Transistor Devices Limited f.m. aerial preamplifier with dual outlets, one for each receiver, the New Acoustic Dimension 160 receiver, the JVC four-channel 4VR-5426X receiver, the TEAC A-3340 four-channel tape recorder for recording the results for the archives, the Scott T33S digital f.m. tuner and four Tannoy loudspeakers. The author can recall making a similar test of one of the first experimental stereo transmissions, way back in May, 1958.

REJECTION RATIOS OF TUNERS

Owing to the superhet principle of radio tuners, various spurious responses can arise, a typical one being the image or 'second channel' response, which falls at a frequency two times the intermediate-frequency from the tuned frequency. Another is 'repeat spot' which falls at half the intermediate-frequency from the signal frequency. A strong signal at the intermediate-frequency may also pass through the front-end and give an output response.

The degree of rejection to which such responses are subjected depends essentially on the design and selectivity of the front-end, which constitutes the r.f. stage and the frequency changer. The simplest of front-end uses a solitary variable-tuned circuit between the aerial and the mixer, so the discrimination against unwanted signals falling either side of the tuned frequency is not very great.

Most hi-fi tuners use at least two variable-tuned circuits (always excluding the tuned circuit of the local oscillator which does not come into the selectivity equation); indeed, some top-flight models boast four such circuits.

The image rejection ratio is a fair measure of front-end selectivity (though a recently devised test by Gordon J. King and called front-end selectivity 'figure of merit' may also be found in reviews of tuners and tuner-amplifiers) and a value of 50–60dB is good, 60–80dB very good and 80dB upwards excellent. Models with only one variable-tuned circuit between aerial input and mixer will generally have a ratio below 50dB.

The ratio refers the aerial input signal voltage required for the particular spurious response to manifest at 30dB S/N ratio (or at the least IHF usable sensitivity readout—this being the datum adopted for the 'figure of merit' measurement) to the signal voltage corresponding to the normal 30dB S/N ratio sensitivity. This is used rather than the IHF least usable sensitivity because some spurious responses yield modulation signal distortion and harmonics.

The large-signal handling capability of the r.f. and mixer transistors is another factor in the spurious response equation since early overloading of these stages from the aerial signals results in intermodulation components which may beat with the signals elsewhere in the front-end (see under *Input Dynamic Range of Tuner*). The 'half i.f.' response ratio gives some indication in this respect since it arises from a harmonic of the aerial signal beating with a harmonic of the local oscillator signal.

Field effect transistors are now often found in f.m. front-ends, they being particularly desirable owing to their essentially square-law transfer characteristic, which means that they produce very little third-harmonic components. In a well designed circuit they also have good dynamic range, though this is not to say that bipolar transistors cannot be engineered for similar dynamic range—they can.

However the FET is useful for low-noise mixing and for the suppression of third-order intermodulation, and the dual-gate variety facilitate local oscillator signal injection (at the mixer) and, perhaps, automatic gain control (at the r.f. stage).

SELECTIVITY OF TUNER

This generally refers to the selectivity provided by the i.f. channel, and is often expressed in the specification in terms of *IHF alternate channel selectivity*. F.M. channels are 200kHz wide, so a signal 200kHz from the tuned signal would be in the adjacent channel and one 400kHz from the tuned signal in the alternate channel.

Thus two input signals are involved in the measurement, one 400kHz (exactly) from the other. That representing the alternate channel signal is increased in strength until 30dB disturbance ratio (referred to 100% 400Hz modulation level) occurs at the output when that representing the tuned, wanted signal is applied at a level of 100μV (for the IHF *rated* alternate channel selectivity).

The ratio of 100μV to the input required for the disturbance ratio, expressed in decibels, is the parameter. More details about the measurement can be found in the companion book *Audio Technician's Bench Manual*.

A ratio of 30dB is fair, up to about 55dB good, 55 to 70dB very good and higher than this excellent. The average ratio of hi-fi tuners is 55–65dB, but real state-of-the-art models have a ratio as high as 100dB.

High ratios are being achieved by the use of ceramic filters in the i.f. channel instead of the earlier i.f. transformers, but for the least harmonic distortion and optimum stereo performance the filters must have a reasonably linear phase characteristic over the passband. Ceramic filters provide sharply falling side skirts to the response characteristic, as shown in Fig. 5.4.

SENSITIVITY OF TUNER

A commonly specified sensitivity parameter is that by the American IHF—the least usable sensitivity (sometimes referred to merely as the usable sensitivity)—which is the signal p.d. required at a specified input frequency to provide 30dB distortion factor (i.e., noise plus total harmonic distortion) when the signal is modulated 100% at 400Hz. An average least usable IHF sensitivity is 2 to 2·5μV, and a state-of-art value 1·4μV or slightly better, depending on the impedance of the aerial input (see later).

Another expression is 30dB S/N ratio sensitivity, this taking account of the noise only—not the total distortion. As the aerial input is increased, so the noise falls, and reviews commonly evaluate the sensitivity by curves, as shown in Fig. 5.15.

Here the output refers to 100% modulation, where at around 2·3μV the output no longer increases with increasing input signal. This is the input required for ultimate limiting (see under *Limiting*). The other full-line curve shows how the noise output falls with increasing input signal, and that in this example the 30dB S/N ratio is around 1·2μV, indicative of a very sensitive tuner.

A more realistic S/N ratio is 50dB, which is shown to fall just

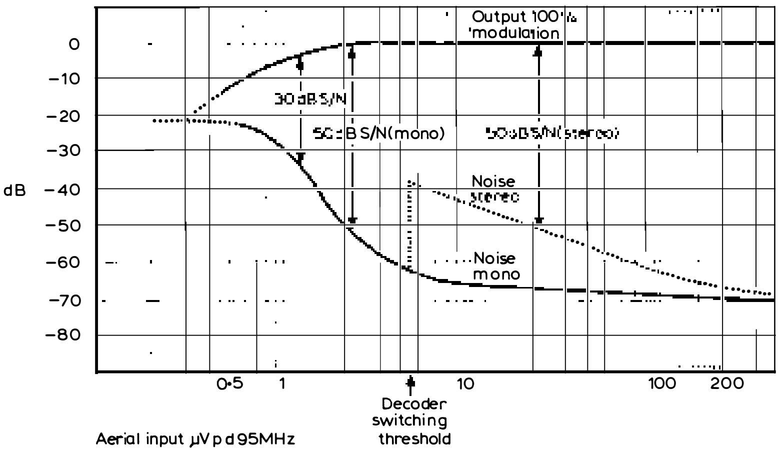


Fig. 5.15 Curves showing mono and stereo S/N ratios of sensitive f.m. tuner. See text for more details.

above 2μV. The background noise is just perceptible at 50dB S/N ratio, but obtrusive at 30dB S/N ratio.

The action of stereo encoding and decoding increases the noise level (see under *Stereo Reception*), and in this example (broken-line curve) the stereo decoder switches in at about 4·5μV, where the S/N ratio is about 38dB. The input required on stereo for 50dB S/N ratio is about 20μV, ten times that required for the same ratio on mono, which is fairly typical.

For stereo reception, therefore, a stronger aerial signal is required than on mono for an acceptable S/N ratio at the lower end of the signal strength scale. The difference between the mono and stereo S/N ratios diminishes as the aerial signal is increased, as revealed by the full- and broken-line noise curves in Fig. 5.15.

The aerial input impedance of the tuner affects the apparent sensitivity (that is, the sensitivity readout) because the p.d. required, say, for 30dB S/N ratio at 300Ω is (under perfect conditions) double that required at 75Ω since there is a 2:1 difference in input transformer ratio between the two conditions.

SERVICE AREA

The area round a transmitter where, under normal conditions, virtually consistent reception can be guaranteed. With f.m., the maximum range is between 30 and 50 miles, depending on the power of the transmitter and the height of the transmitting and receiving aerials.

The directivity of the transmitting aerial may, however, modify the range in certain directions, and even in the service area natural or man-made objects can attenuate the signal, particularly where the receiving site is low-lying. Fig. 5.16 shows how the v.h.f.–f.m. signal can be attenuated by a wooded area.

In general, less complicated aerials are required for service area reception than for fringe area reception (see under *Fringe Area*), but multipath reception and its resulting distortion (see under *Multipath Distortion*) may demand the use of highly directional arrays, even in the service area.

Fig. 5.16 Even in the service area the f.m. signal can be attenuated by having to pass through a wooded area as this diagram shows.

STEREO DECODER

This is the part of the f.m. stereo tuner or receiver which effectively 'unscrambles' the stereo information which is transmitted along with the mono information. The term multiplex decoder (or merely MPX) is sometimes adopted to describe the stereo decoder or decoding action.

STEREO RECEPTION

Stereo reception requires a tuner equipped with stereo decoder. Most contemporary tuners are so-equipped and thus stereo-ready.

Early tuners, however, generally require the inclusion of a decoder. Some are designed to accept a decoder module, though with others an external decoder may have to be used. Not all f.m. stations are transmitting in stereo. Certain parts of the country, notably the South-West at the time of writing, are still without stereo coverage.

In some such areas DX stereo reception (see under *DX Reception*) is possible with a highly sensitive and selective tuner and elaborate aerial system, but since noise can be troublesome on stereo unless the aerial signal is sufficiently strong, this can be regarded only as more of an experiment than a consistent service (also see under *Troposphere*).

STEREO SEPARATION

The effective isolation between the two stereo channels. A separation of 40dB is generally possible from well designed stereo decoders at 1kHz, falling at higher and possibly lower modulation frequencies.

TROPOSPHERE

The lower part of the Earth's atmosphere extending upwards from the Earth's surface, in which temperature decreases with height except in local layers of temperature inversion. Outside the service area, the troposphere tends more to influence the propagation of v.h.f. f.m. waves owing to the waves being refracted by the progressive change of the atmosphere through which they pass.

In the service area, the waves travel a more or less line-of-sight path from transmitting to receiving aerial, but a greater 'radio distance' than this is provided by tropospheric refraction, and also to some extent by diffraction of the waves round the curved surface of the Earth (see Fig. 5.17). As the troposphere tends to change with changing weather conditions, the reception conditions outside the service area are also so affected and hence less consistent.

Sometimes abnormal conditions occur in the troposphere such that the normal range of v.h.f. signals is increased, sometimes more than ten times. This is called a 'tropospheric opening'. Conditions like this are exploited by DX enthusiasts for long-distance f.m. reception (see under *DX Reception*).

It has been noted from experience that when the atmospheric pressure is above 30in Hg, and then rises a little more, a tropospheric opening is almost certain to result when the pressure *starts*

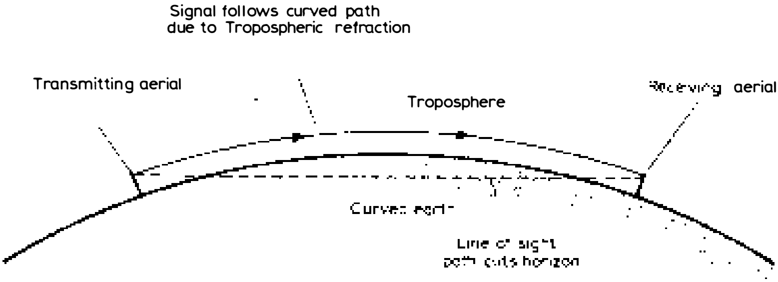


Fig. 5.17 F.M. v.h.f. reception beyond the line-of-sight path is possible owing to refraction of the wave in passing through more dense regions of the troposphere. Beyond the service area reception becomes more influenced by the state of the troposphere and hence by the weather conditions. See text.

to decline from its peak. F.m. reception over significant distances is then possible with a suitable receiver and aerial installation.

TUNER

The radio-frequency sections, including detector, stereo decoder (in the case of a stereo f.m. tuner) and low-level audio stages of a receiver. Since there are no power amplifier stages, the low-level audio signals need to be applied to a separate amplifier system for loudspeaker reproduction. Some tuners, however, incorporate facilities for operating a headphone set. Also see under *A.M. Tuner*, *F.M. Tuner* and *F.M./A.M. Tuner*. A tuner is essentially a programme signal source.

TUNER-AMPLIFIER (HI-FI RECEIVER)

An integration of a hi-fi amplifier (2- or 4-channel) and an f.m.-only or a.m./f.m. tuner.

V.H.F.

Standing for very high-frequency, and corresponding to a frequency spectrum from 30MHz to 300MHz. Band II, in which f.m. broadcasting occurs, is from a maximum of about 88MHz to 108MHz, though only a part of this is fully exploited for entertainment radio, parts being used for communications radio services, police, ambulance, taxis, etc.

WIRE AERIAL

A length of wire, usually from 25ft upwards, used for a.m. reception. Most a.m. tuners, though, are equipped with a ferrite rod aerial (see under *Ferrite Rod Aerial*) for LW and MW reception, but for SW (LW, MW and SW refer respectively to long-wave, medium-wave and short-wave) reception an external wire aerial is required. This may be of random length (i.e., untuned) or of specific length cut to resonate at the frequency or range of frequencies of interest. The aerial may also be of a particular configuration, such as dipole, Marconi, Franklin, etc.

YAGI AERIAL

A dipole aerial (see under *Dipole Aerial*) with a director and/or reflector system and named after the Japanese engineer who first published the principle from work by colleague Professor Uda way back in 1926/28. The f.m. 'H' aerial is a Yagi in simple form, where a separate rod is supported in parallel with the dipole. When the rod is to act as a reflector it is slightly longer than the dipole, and when as a direction it is slightly shorter.

No specific electrical connection exists between the reflector or director(s) to the dipole, but the spacing of these elements from the dipole is important. When more than one director is used, they may gradually reduce in length. In elaborate arrays a system of reflectors may be adopted, consisting of two or more rods (see Fig. 5.18).

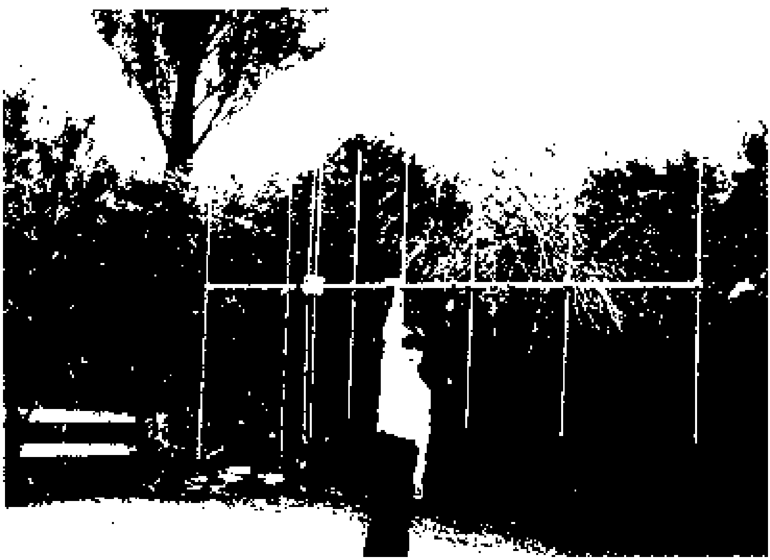


Fig. 5.18 Michael, the author's son, displaying the Fubu Uka Stereo-8 f.m. aerial.

Because the rods are not in electrical connection with the dipole they are sometimes called parasitic elements.

When they are correctly spaced from each other and the dipole, and each is cut to the correct associated length, they pick up the wanted signal and re-radiate it so as to reinforce the signal picked up by the dipole. In this way the array favours signal arriving in line with the director(s) and discriminates against signals arriving from other bearings. Moreover, the forward gain of the aerial is increased over a dipole, the forward acceptance angle reduced and the front-to-back ratio increased (see Fig. 5.12).

Fig. 5.18 shows the author's son displaying the acclaimed Fuba Uka Stereo-8 f.m. aerial. This consists of a folded dipole (helping with the feeder matching), five in-line directors and a two-element reflector system. The junction box at the centre of the folded dipole contains a transformer to optimise the feeder matching and featuring couplings to 75Ω coaxial cable or 300Ω balanced twin feeder. This is the aerial which was used for the Quadraphonic f.m. tests (see under *Quadraphony*).

CHAPTER SIX

RECORDING AND REPLAY

AUTOMATIC NOISE REDUCTION SYSTEM

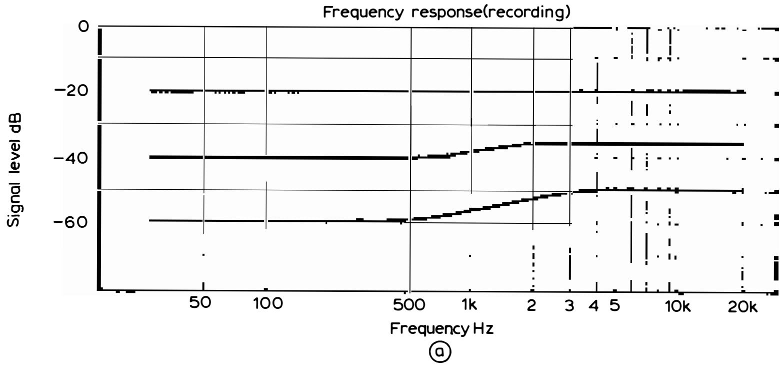
A NOISE REDUCTION SYSTEM developed by the Japanese Victor Company (JVC) for use with disc and tape recording and replay. Commonly referred to as ANRS for short. Like the Dolby method of noise reduction (see under *Dolby Noise Reduction*), the signal is processed both during recording *and* replay, being first encoded and then decoded.

In fact, a fair degree of compatibility exists between the two systems; that is, a Dolby B encoded recording can be replayed through an ANRS decoder, and *vice versa*, with acceptable results. For absolute performance, of course, the encoding and decoding need to be complementary.

The systems exploit the masking ability of the human ear, such that when two sounds occur at the same time and one is louder than the other, the listener is aware only of the louder sound. During recording the signal is automatically boosted in a special way progressively as its intensity decreases, and during replay complementary processing is applied, as shown at (a) and (b) in Fig. 6.1.

The curves show that it is the higher frequencies only which are subjected to the boost and complementary cut. With the ANRS, signals below about 500Hz are unaffected; it being only the higher frequency low-level signals which are so processed. The net result is that the noise produced during recording and replay is reduced by about 5dB at 1kHz and 10dB between 5kHz and 10kHz.

It must be stressed that the system does not combat noise present on the source signal or in the associated circuits. It is primarily a method for reducing the noise actually resulting from recording and replay, and is thus of considerable value for reducing the overall tape noise of cassette recorders and playing units.



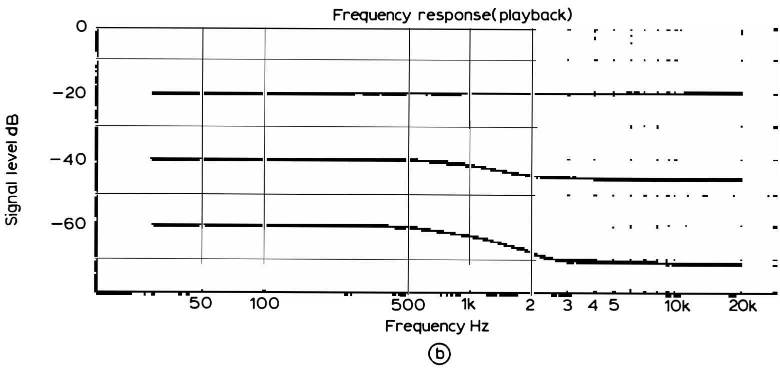


Fig. 6.1 Encoding (a) and decoding (b) characteristics of the JVC Automatic Noise Reduction System (ANRS).

BIAS (PICKUP ARM)

A small force applied mechanically or magnetically to a pickup arm to combat or neutralise the inward bias created by the friction of the pickup stylus tracing the groove and overhang geometry (see under *Side-thrust Correction* and *Overhang*).

BIAS (TAPE RECORDER)

A high-frequency signal applied to the recording head of a tape recorder during recording to reduce the non-linearity of the transfer characteristic and hence to minimise harmonic and intermodulation distortion.

CARTRIDGE (TAPE)

A spool of tape forming a continuous loop and contained in a special housing which facilitates swift and uncomplicated application (without threading, etc.) to a tape machine specially designed for cartridge working. A cartridge is thicker than a cassette because $\frac{1}{4}$-in tape is used, which is *approximately* twice the width of the tape used in cassettes.

Multiple-track cartridges facilitate 'quadraphonic' replay, but at the time of writing the tape cartridge has most application in car music systems where, in some cases, the play unit automatically switches to the next track when the previous one is played out.

CARTRIDGE (PICKUP)

The electromagnetic or piezo-electric transducer which is fitted to the playing end of a pickup arm and which carries the stylus. The cartridge is fitted to a head-shell which may be detachable from the arm.

CARTRIDGE TAPE MACHINE

A tape machine designed to accommodate a tape cartridge. Such machines are mostly designed for the playing of pre-recorded cartridges (i.e., cartridge records), but one or two machines are now appearing on the market with recording facilities as well, also with Dolby B encoding and decoding.

CASSETTE

A plastics container with tape and spools for use with a cassette tape machine. Approximate dimensions of the tape and tracks are given in Fig. 6.2 when four tracks are recorded to provide two-channel stereo on each half of the tape width, the cassette being turned over to

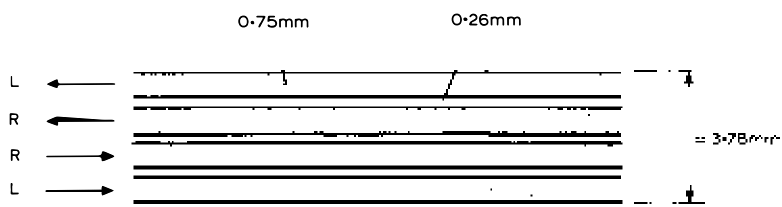


Fig. 6.2 Cassette tape and stereo tracks.

utilise the second pair of tracks when the first pair is played (or recorded).

Cassette machines now run at the standard speed (pioneered by Philips) of 4·75cm/S ($1\frac{7}{8}$in/S), and the amount of playing time provided by the tape at this standard speed is indicated on the cassette such as C60 30 minutes each side (1 hour in all), C90 45 minutes each side ($1\frac{1}{2}$ hours in all), and C120 1 hour each side (2 hours in all). The extra tape required for the longer-play cassettes is accommodated by a reduction in thickness, these being of 26μm and 18μm in thickness.

One problem, particularly with the thinner tapes, has been for the tape to loosen or unwind slightly in the cassette; or even to jam. The resulting variations in tension can precipitate minor speed changes of the tape passing the heads, and hence wow and flutter.

To overcome this sort of thing BASF, the well known tape people, have developed a new cassette which they call *Special Mechanics*, which consists of a finger-like tension control arranged to hold the tape on to its spool at all times, thereby providing a constant tension and reducing the possibility of jamming.

CASSETTE MACHINE

A tape machine designed to accommodate a tape cassette and transport the tape at the standard speed of 4·75cm/S. Such machines are mostly suitable for both the replay of cassette records (called *Musicassettes*) and also for the recording and replay of one's own material. Many hi-fi versions are equipped with a noise reduction system, such as Dolby B, ANRS or the Philips DNL (Dynamic Noise Limiter).

Musicassettes are recorded in stereo and arranged to replay in correctly balanced mono on a mono-only machine. This happens because the left and right channel tracks are laid side-by-side, as shown in Fig. 6.2. The head of a mono-only machine scans both left and right tracks together so that the signal is the sum of the left and right channel signals, which is mono.

A stereo machine has a double head system whereby each track is scanned separately so that separate left and right channel signals on yielded on replay.

A four-track machine has been developed by JVC for discrete quadraphony, the width of the tape then carrying eight tracks, four on one half and four on the other half. The tracks are arranged for mono and stereo compatibility. As the track width is reduced so the S/N ratio diminishes. The reduction in inherent S/N ratio thus needs to be combated by a noise reduction system.

CERAMIC PICKUP

A pickup whose cartridge operates on the piezo-electric principle (see under *Piezo-Electric Cartridge*).

CHROMIUM DIOXIDE TAPE

Iron compounds are used for most tapes, but of recent times a new tape has been developed based on chromium dioxide (CrO_2). Although more costly than iron-based tape, CrO_2 tape has certain attributes which stem essentially from its greater resistance to demagnetisation. This means that CrO_2 tape is better able to accommodate and retain higher frequency information. The net result is better high-frequency performance and S/N ratio when the machine is suitable for handling the tape.

CrO_2 tape will work after a fashion with almost any machine, but a greater erase field is generally required completely to clear a previous recording, and a greater recording current for full modulation than can be adequately provided by machines not equipped for CrO_2 application. Moreover, to achieve the full attributes of extended high-frequency response and S/N ratio both the recording pre-emphasis and the replay equalisation need to be modified.

Hi-fi machines are equipped with a switch allowing the circuits to be changed easily to the requirements of CrO_2 tape. Sometimes this switching is performed automatically when a CrO_2 cassette is inserted into the machine by means of a switch operated by suitable identification at the back edge of the cassette.

This works in a similar way to the lug which makes it possible to set the machine to the recording mode when a cassette is inserted. If this lug is removed (there is one for each side of the cassette) it is not possible to set the machine for recording, which is a good protection for previously recorded material one requires to retain.

Musicassettes are so protected against accidential erasure (by over-recording). A piece of Sellotape or a special lug (available from Bib) can be applied to the cassette if it is required to eliminate a previous recording or to record new information on a *Musicassette*.

COMPLIANCE

Yielding quality; the reciprocal of stiffness. A parameter of a pickup cartridge previously expressed in cm/dyne such that a compliance of 1×10^{-6}cm/dyne implies a deflection of 10^{-6}cm due to a force of

1 dyne, now expressed in SI units (international system of units) as m/N, where N is the Newton, corresponding to 10^5 dynes.

The compliance of a cartridge can resonate with the effective mass (due to inertia) of the associated arm and give an undesirable peak in the low-frequency output unless suitably damped. Another resonance, but this time at the high-frequency end, is caused by the effective mass of the stylus and the compliance of the record material or the compliance of the cantilever.

COUNTERWEIGHT

The weight at the end of a pickup arm which is adjusted to balance out the weight of the cartridge, headshell and arm to facilitate the application of the required playing weight by means of a separate rider weight, spring or other device.

CROSSFIELD BIAS

The application of the high-frequency bias required for tape recording by a separate head which is positioned in relation to the recording head so that the bias signal has less demagnetising influence on the tape. The net result is improved high-frequency response.

CRYSTAL PICKUP

A pickup whose cartridge operates on the piezo-electric principle (see under *Piezo-Electric Cartridge*).

DEMAGNETISER

A device emanating an alternating magnetic field whose purpose is to demagnetise a tape (such as in a bulk eraser where a whole tape is demagnetised), a recording/replay head or item of metal in proximity to the tape transport of a tape machine to prevent the spurious field from affecting a tape recording or implanting undue noise on the recording. Also known as degausser or defluxer.

'DISCRETE' QUADRAPHONY

A method of four-channel recording or reproduction where the four channels are recorded or replayed via electrically isolated circuits. See under *Cassette Machine.*

DOLBY NOISE REDUCTION

A noise reduction system whereby the recording signal is subjected to increasing treble boost with decreasing signal level and whose replay signal is subjected to increasing treble cut with decreasing signal level, the net result being a correctly balanced frequency response of lower noise level than if such processing had not been applied. The term 'stretched' is sometimes used in connection with this technique as an implication that the gap between the signal and noise has been increased—or stretched.

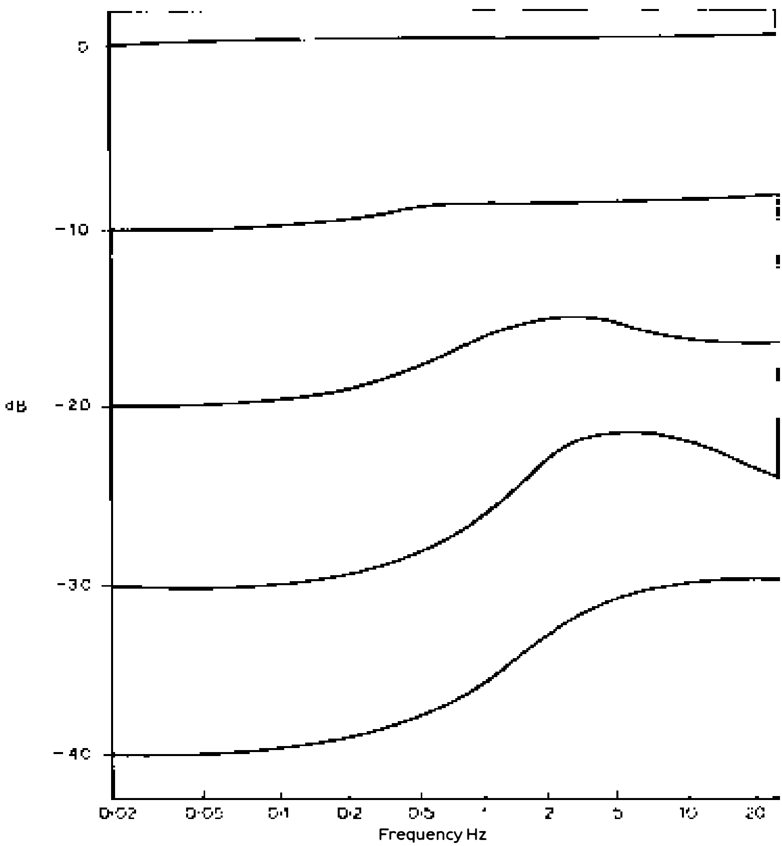


Fig. 6.3 Dolby Type B encoding characteristics at different signal levels.

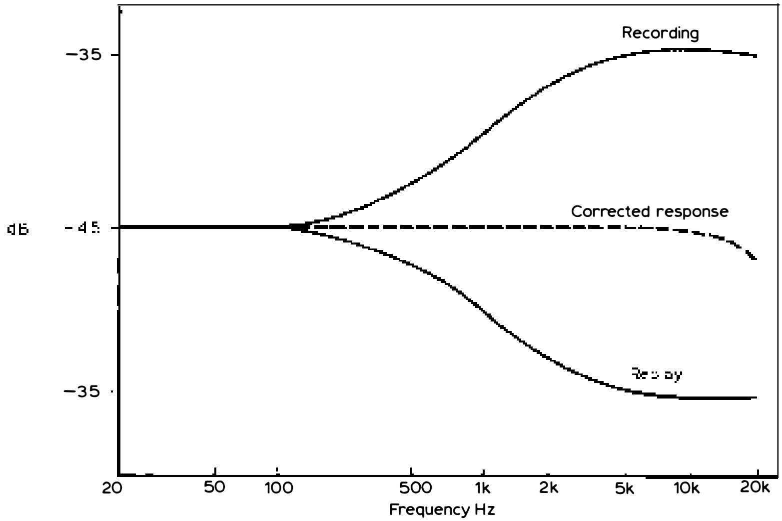


Fig. 6.4 Dolby Type B encoding and decoding relative to a specific signal level, showing the correct overall frequency response.

In domestically-orientated equipment, like cassette recorders, the so-called Dolby Type B technique is used. This is less complicated, and hence less costly, than the Dolby Type A technique which is adopted by the professionals, such as the recording studios, where turnover frequencies as well as signal levels come into play.

Like the JVC ANRS, both the recording and replay signals are processed in a complementary manner, but the relative lifts and cuts and the turnover frequencies involved differ between the two systems, which is why they cannot be entirely compatible (though in practice the degree of error on interchange is small—subjectively).

Fig. 6.3 shows how Dolby Type B yields increasing treble pre-emphasis during recording. The opposite effect occurs during replay. Fig. 6.4 shows together the recording pre-emphasis and the complementary de-emphasis characteristics of specific signal level, and the resulting 'flat' response characteristic.

A graphical description of the overall process as provided by the Dolby Laboratories is given in Fig. 6.5. Also see in Chapter 5 under *Dolby Noise Reduction on F.M.*

DROP-OUT

Not a turned off youngster, but a momentary disappearance of

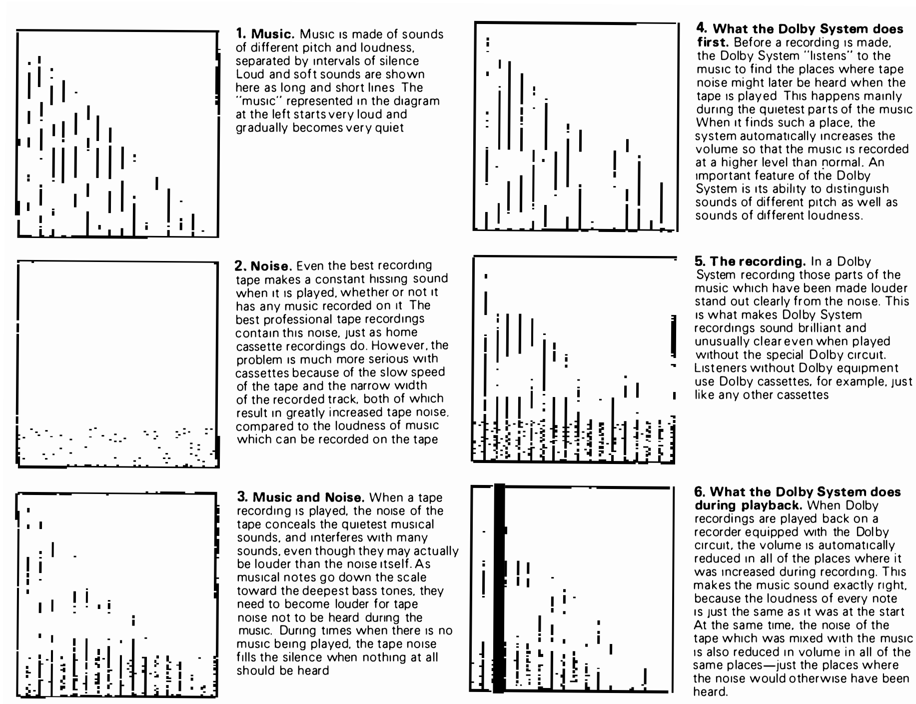


Fig. 6.5 Graphical appraisal of the Dolby Type B technique, as provided by Dolby Laboratories.

signal information during tape relay. Caused in the main by inconsistency of the tape coating. Far less troublesome these days than hitherto.

DUBBING

The copying of one tape recording on to another tape or a disc, or the copying of a disc record on to tape. The process with regard to tape-to-tape dubbing can be by direct transfer or superimposition. Some of the more costly hi-fi amplifiers and receivers are equipped with dubbing facilities in addition to tape monitoring (see under *Tape Monitoring*).

DYNAMIC NOISE LIMITER

A Philips-engineered system for suppressing high-frequency noise, which differs from Dolby and JVC ANRS in that it is effective during replay only. It can thus be used with reel-to-reel and cassette recorders without affecting their compatibility; and can also be used with non-Dolbyised or even with Dolbyised tapes and cassettes.

The DNL splits the replay signal into two paths, both of which carry all the signal information and noise. One path goes via an all-pass filter and phase inverter. The other path includes a high-pass filter which suppresses the low and middle frequencies, so that signals only above about 4kHz emerge. These signals are subjected to pre-emphasis which increases with decreasing signal level.

The signals from the two paths are then added, and because they are antipath the high-frequency signals from the pre-emphasis circuit tend to cancel those of corresponding frequencies which arrive via the all-pass filter. The result is that signals up to about 4kHz are unaffected, irrespective of signal level, while the higher frequency signals including the noise are reduced in level by an amount depending on the overall level of the signal at the high-frequency end of the spectrum.

DYNAMIC RANGE

With tape, this is the range, usually expressed in decibels, between tape overload at the top and tape noise at the bottom. Thus the dynamic range is effectively increased by a noise reduction system.

With disc, it is the range between the level of modulation that can be recorded and ultimately replayed by the pickup at the top and the intrinsic noise of the medium under dynamic conditions at the bottom. A dynamic range of *circa* 60dB is possible with today's discs and good quality replay equipment.

With Dolby Type B encoding and decoding the dynamic range of cassette machines has been improved dramatically since this software was first introduced.

EFFECTIVE ARM MASS

Mass or moment of inertia reflected to the tip of the stylus due to the mass of the arm after counterbalancing and application of the correct playing weight. This can resonate with the compliance of the cartridge at low-frequency, and damping is adopted to prevent groove jumping due to this cause.

EFFECTIVE TIP MASS

Mass or moment of inertia of the moving elements of the pickup cartridge reflected to the tip of the stylus. A very low effective tip mass is necessary for optimum tracing of high-frequency, high level signal components (see *How to Choose and Use Pickups and Loudspeakers*).

The effective tip mass can resonate with the compliance of the record material or cantilever at high-frequency, and damping in the cartridge is adopted to prevent the response rising unduly due to this cause.

ELLIPTICAL STYLUS

A pickup stylus of two radii, one small (*circa* 7·5μm) for optimum tracing of high-frequency modulation (i.e., small wavelength) with the least tracing distortion, and the other larger (*circa* 17·5μm) to prevent the stylus from 'bottoming' in the groove, which would otherwise cause an undue level of background noise.

ERASE SIGNAL

High-frequency signal, usually from the same source as the bias signal, which is applied to the erase head for demagnetising the tape just prior to recording.

ERASE HEAD

The head in a tape machine to which the erase signal is applied. It is placed just before the recording/replay head so that previously recorded information is erased before a new recording is made.

FLUTTER

Fast warbling of a signal due to cyclic speed variations above about 10Hz, caused by irregularities in tape transport or turntable. It is a form of frequency-modulation and its peak percentage is given by $f_{max} - f_{min} \times 100/f_{av}$, where f_{max} is the maximum frequency, f_{min} the minimum frequency and f_{av} the average frequency. It is sometimes converted into an r.m.s. value or weighted in accordance with a particular standard, such as DIN B.

FLUX DENSITY (AND OTHER MAGNETIC PROPERTIES)

The signal level recorded on to tape is referred to a level of magnetic flux, which in the SI system of units is expressed in nano-Webers per meter of track width (nWb/m), where nano (n) is 10^{-9}. It was formerly (and still is sometimes) expressed in milli-Maxwells per millimetre of tape width (mM/mm).

For conversion, 1 Weber equals 10^8 Maxwells, where the Weber corresponds to the magnetic flux which, linking a circuit of one turn, yields an electromotive force of 1 volt as it is reduced to zero at a uniform rate in 1 second.

The flux level, then, is a function of the width of the recorded tape track; that is, the flux that is scanned by the relay head. Test tapes are recorded to a specific level, sometimes to provide a reference level for setting up, head alignment, etc., and one such reference is 320nWb/m (corresponding to 32mM/mm) at 1kHz. The reference levels for setting Dolby Type B encoding and decoding are 200nWb/m for cassettes and 180nWb/m for reel-to-reel tape.

Other magnetic properties are *magnetic induction*, symbol B, expressed by Tesla (T) in the SI system of units (formerly Gauss, G), and *magnetic field strength*, symbol H, expressed in ampere turns per metre (A.t/m) in the SI system of units (formerly Oersted, Oe).

FOUR TRACK

A reel-to-reel tape machine designed to record four tracks on magnetic tape and to replay the tracks so recorded. For stereo, two tracks are used simultaneously as shown at (a) in Fig. 6.6. The heads are arranged so that the second pair of tracks comes into operation when the tape is turned over. Sometimes called quarter-track stereo.

For mono, each track is used separately. The machine usually incorporates a dual head assembly so that tracks 1 and 3 can be switched without turning the tape over. When the tape is turned over tracks 2 and 4 become available by head switching, as shown at (b) in Fig. 6.6. Sometimes called quarter-track mono.

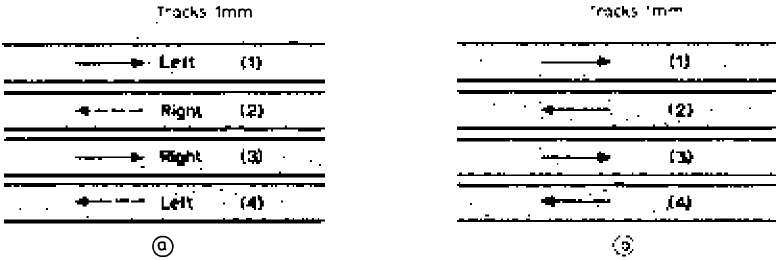


Fig. 6.6 Four-track (or quarter-track) stereo (a) and four-track (or quarter-track) mono (b).

FULL-TRACK

A reel-to-reel tape machine designed to record one track only across the full width of the tape and to replay the track so recorded. Rarely used domestically.

HEAD GAP

The interface separating the pole pieces of a recording, replay and erase head across which a magnetic field develops. The field at the recording head results from signal current passing through the associated winding, that at the erase head from the high-frequency signal (usually from the bias oscillator) passing through the winding, and that at the replay head from the flux of the recorded tape, which results in the production of a signal e.m.f. in the associated winding.

For optimum definition of high-frequency small wavelength signals, the gap has to be as small as possible. When the recorded wavelength corresponds to the *effective* gap dimension, which is larger than the physical gap owing to 'spreading' of the field and to the lack of perfect contact between tape coating and head pole pieces, the output from the replay head falls to zero. The frequency corresponding to this wavelength is called the *extinction frequency*.

Since the recorded wavelength depends on the speed of the tape passing the head, the upper frequency response is thus a function both of replay head gap and tape speed. At the lower tape speeds, particularly at the cassette speed of 4·75cm/S ($1\frac{7}{8}$in/S) the gap has to be diminutive (in the order of 1μm) to maintain a reasonable high-frequency response. The −3dB response is approximately half the extinction frequency.

Treble pre-emphasis is often applied to hold up the treble output, and *replay equalisation* is also necessary because with a constant tape flux the head output increases by 6dB each time the frequency doubles.

The gaps of the recording head (when the machine features separate heads for recording and replay) and the erase head are less critical in this respect.

LATERAL TRACKING ERROR

The error of lateral tracking due to the disc being made with the cutter following a true radial path and the pickup arm being pivoted such that the stylus follows an arc across the disc. This error is minimised by geometric artifices such as angling the arm (see under *Offset Angle*) and arranging for the stylus to pass the centre point of the disc with an overhang (see under *Overhang*).

Lateral tracking error produces harmonic distortion, but the distortion so resulting is very small from a correctly designed and adjusted cartridge/arm system.

LINE INPUT

Tape input circuit (or circuit on other audio equipment) designed for 600Ω working such that 0dB corresponds to 1mV. That is, when the r.m.s. signal voltage across 600Ω is 0·774mV. Such an input (or output) is used essentially for professional applications.

MAGNETIC PICKUP

Pickup or cartridge based on the electromagnetic principle and whose output is proportional to the velocity of the recording, for which reason the term velocity pickup or cartridge is sometimes used. Examples are variable reluctance, moving-magnet and moving-coil. For further information refer to the companion book *How to Choose and Use Pickups and Loudspeakers.*

MAGNETIC TAPE

A plastics ribbon which is evenly coated on one side with very small particles of magnetic material, such as gamma ferric oxide (γFe_2O_3) or chromium dioxide (CrO_2). Used for the magnetic recording and subsequent replaying of audio, video or binary information. Some recent tapes have ferric and chrome layers.

MONITOR HEAD

When a tape machine features separate recording and replay heads, the replay head can be used, via suitable amplification, to monitor the signal directly following its recording. Contemporary amplifiers and tuner-amplifiers are equipped with an input and suitable switching to receive such a monitor signal. The switching allows the recorded signal to be compared immediately with the source signal (see Chapter 1).

OFFSET ANGLE

Part of the geometry used on a pickup to combat laterial tracking error (see under *Lateral Tracking Error*). It takes the form of either arm curvature or an angled headshell and hence cartridge on a straight arm. Offset angle is illustrated in Fig. 6.7. See also *Overhang* below.

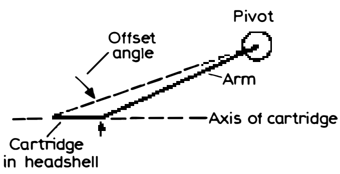


Fig. 6.7 Definition of offset angle.

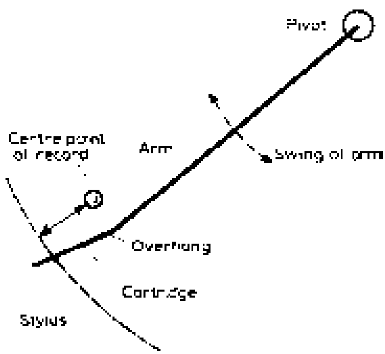


Fig. 6.8 Definition of overhang.

OVERHANG

The other part of the geometry used on a pickup to combat lateral tracking error, as illustrated in Fig. 6.8. Offset angle combined with

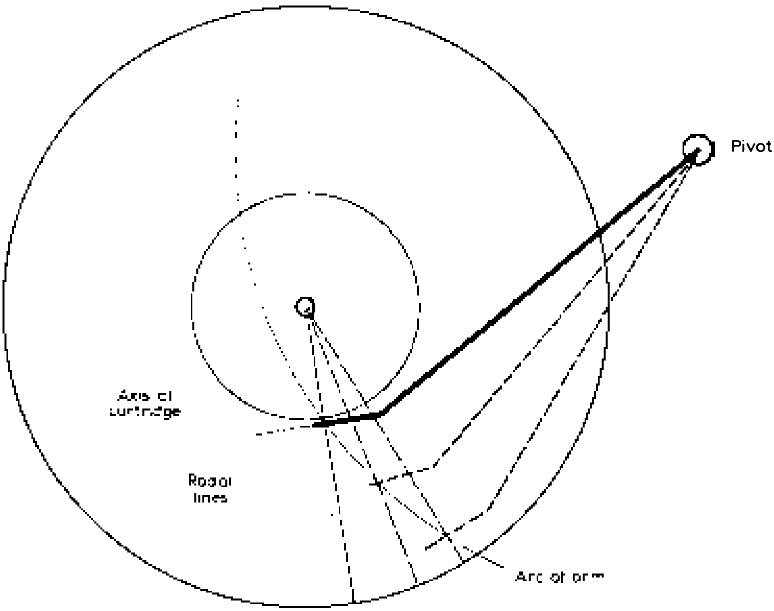


Fig. 6.9 The combined effects of offset and overhang minimise lateral tracking error by keeping the angle between a radial line and the axis of the cartridge essentially at 90° at all positions of the arm as shown.

overhang keeps the angle between a radical line and the axis of the cartridge essentially at 90° at all positions of the arm, as shown in Fig. 6.9, which is the angle corresponding to the least lateral tracking error.

The arm/cartridge combination is usually adjusted with a special protractor for zero lateral tracking error at the smallest recorded diameter, and with a well designed and correctly adjusted arm the maximum tracking error over the active swing of the arm should then not rise much above 1·5°.

PEAK PROGRAMME METER (PPM)

A recording or signal level meter which is designed, in conjunction with its circuitry, to indicate amplitude peaks of the audio information. Such meters have a fast rise time and a slow decay time. See also under *VU Meter*.

PLAYING WEIGHT

The weight (force) at which the pickup plays a record. Sometimes called tracking weight. As the playing weight is increased up to the maximum stipulated by the manufacturer, so higher recording levels become trackable within the capabilities of the cartridge/arm combination.

Mistracking, particularly of high level, high-frequency modulation, occurs when there is insufficient playing weight or if the arm is unsuitable for the cartridge. However, the playing weight should not normally be increased beyond the maximum specified in an endeavour to improve the tracking performance. Tracking failure at the maximum playing weight is indicative of a worn stylus or an unsuitable arm (insufficiently low bearing friction, for example).

There is a maximum modulation level/frequency limit of tracking for all cartridges, but the more costly ones are generally capable of tracking higher modulation and frequency than the budget variety at the maximum specified playing weight. Fig. 6.10 oscillogram shows mild mistracking of a 1kHz plus 1·5kHz signal of 40cm/S peak velocity.

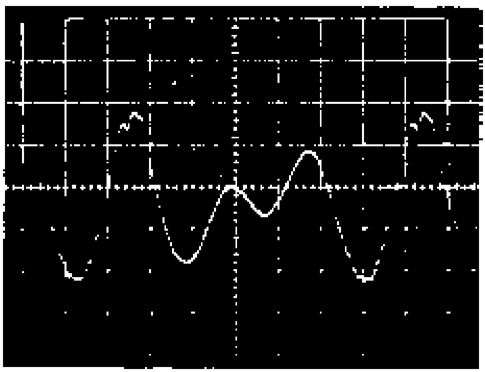


Fig. 6.10 Oscillogram showing mild mistracking of combined 1kHz/1·5kHz 40cm/S peak recording.

PICKUP ARM

The pivoted arm which carries the pickup cartridge, the latter is generally fitted into a shell which is detachable from the arm. To accommodate top-flight cartridges of low tracking weight, the arm should have low mass to avoid a critical low-frequency resonance in conjunction with the compliance of the cartridge, should be suitably damped to restrict resonance peaks, should have very low bearing friction (30mg or less lateral and vertical), should counterbalance a low-mass cartridge and apply a tracking weight over a range of 1g or less to 3g or more. There should also be a facility for the application of side-thrust correction (see under *Arm Bias* and *Side-Thrust Correction*).

PICKUP SYSTEM

The arm/cartridge combination. A pickup (generally) is a device which translates the modulation implanted in the groove of a gramophone record to electric programme signal.

PIEZO-ELECTRIC CARTRIDGE

A pickup cartridge which works on the piezo-electric principle. The active element is either natural crystal (see under *Crystal Pickup*), such as Rochelle salt, or electrically polarised ceramic. When such an element is stressed an electric charge develops across its surfaces (sometimes called 'piezo electricity').

This type of cartridge, therefore, produces an output proportional to the *amplitude* of the recorded signal, for which reason it is sometimes termed an amplitude cartridge. See also under *Magnetic Cartridge*.

PRE-DISTORTION

Distortion deliberately introduced to a disc record during recording so that the tracing distortion on replay is partly cancelled.

PRE-RECORDED

Commonly refers to a reel-to-reel or cassette tape which has been recorded, such as a musicassette or tape record.

RAKE ANGLE

The angle of the axis of the stylus referred to the surface of the disc. Different from vertical tracking angle (see under *Vertical Tracking Angle*).

RECORDED VELOCITY

The recorded level of a disc record in terms of corresponding rate of alternating stylus motion (i.e., velocity). Average recorded velocity is 5cm/S, which is the velocity sometimes used to express the output signal from a cartridge at 1kHz. The lower velocity of 1cm/S might otherwise be used.

REEL-TO-REEL TAPE MACHINE

Type of tape machine using standard $\frac{1}{4}$-in tape with supply and take-up spools, as distinct from a cartridge or cassette machine.

RUN-IN

The start of the groove spiral on a gramophone record which leads the stylus on to the groove modulation. The *run-out* is the converse at the end of a record.

SHIBATA STYLUS

A pickup stylus of special contours designed for optimum definition of the back channel f.m. information of discrete four-channel gramophone records (i.e., CD-4 records). See under *Shibata Stylus* in Chapter 4. Other contours are also being produced for CD-4 disc replay, including the Premanik stylus used by Bang and Olufsen in the MMC 6000 four-channel cartridge.

SIDE-THRUST CORRECTION

Owing to the overhang applied to a pickup system, a part of the correction for lateral tracking error, the arm is subjected to an inward torque due to the frictional drag of the stylus in the groove (see Fig. 6.11). To maintain a correct balance of stylus assembly a counteracting torque is required, which is applied by a dangling weight, spring or magnetic field to the arm.

As the inward torque is a function of the amount of stylus/groove drag, the counteracting torque or side-thrust correction required depends on the coefficient of friction at the stylus groove interface and thus on the playing weight and radius of the stylus. As the drag varies with recording level, the correction can never be absolute.

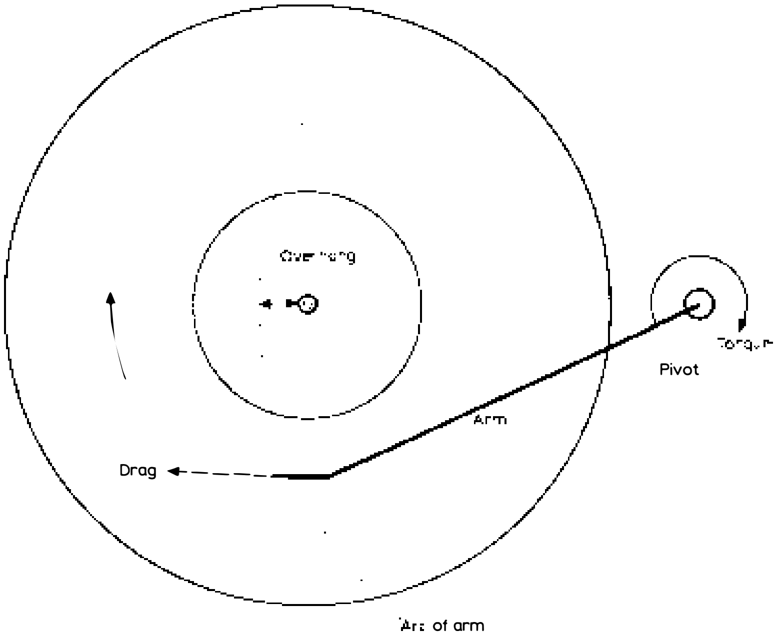


Fig. 6.11 Due to the offset and stylus/groove drag a torque is developed which tends to pull the arm towards the centre of the disc. An equal and opposite torque is applied by side-thrust compensation (see text).

SOUND-ON-SOUND

The term describing a process of dubbing from one·tape track to another while additional information is simultaneously recorded.

STYLUS

The part of a pickup cartridge which traces the groove of a gramophone record. The stylus assembly is often detachable from the main body of the cartridge for replacement, cleaning, etc. There are various contours, including elliptical (see under *Elliptical Stylus*), spherical and the special contours used for CD-4 replay (i.e., Shibata and Premanik). The radius of a spherical stylus is commonly 12·5μm, and the styli of hi-fi cartridges are made of diamond, naked or shanked.

The stylus assembly of a magnetic cartridge is clearly visible in Fig. 6.12.

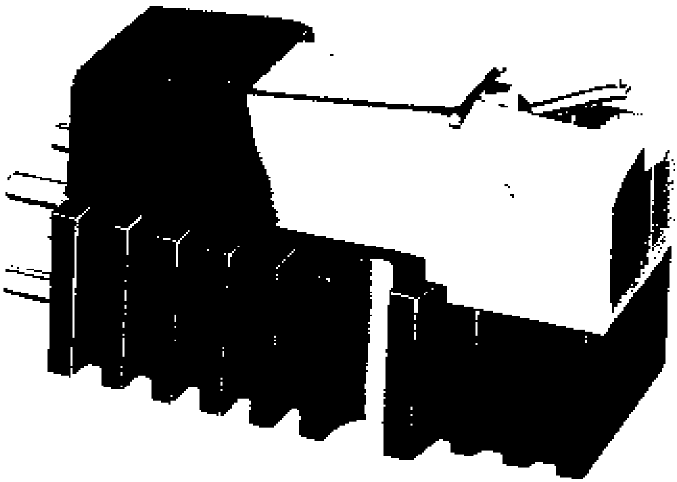


Fig. 6.12 Magnetic pickup showing stylus assembly, including cantilever.

TAPE DECK

A reel-to-reel or cassette machine which is devoid of amplification.

TAPE MONITORING

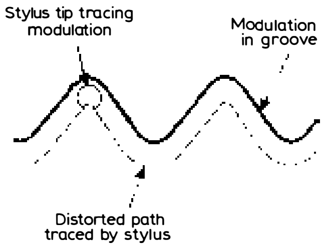
The process of monitoring with a separate replay head the recorded signal directly after it has been recorded. This facility is only possible from those machines equipped with separate recording and replay heads (in many machines one head is used for both recording and replay, via switching) or with a head specially for monitoring purposes. The signal from the monitor head needs to be correctly amplified and equalised for reproduction.

TAPE TRANSPORT

The mechanism of a tape deck concerned with transporting the tape at a very constant velocity past the recording/replay and erase heads.

TAPE SPEEDS

Standard tape speeds are 38cm/S (15in/S), 19cm/S (7½in/S), 9·5cm/S



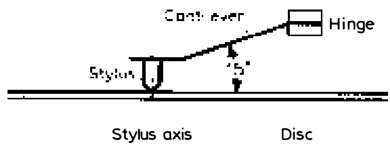


Fig. 6.13 (left) Illustration of tracing distortion which is essentially second harmonic. Fig. 6.15 (right) Illustration of vertical tracking angle (see text).

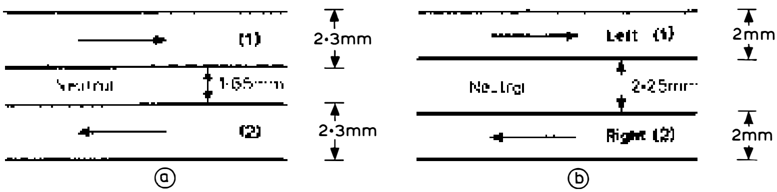


Fig. 6.14 Two-track recording. (a) mono each track being used separately by turning the tape over, and (b) stereo both tracks being used together.

($3\frac{3}{4}$in/S) and 4·75cm/S ($1\frac{7}{8}$in/S). The latter is the standard cassette velocity, and the higher reel-to-reel tape speeds are used for high quality music recording and reproduction.

TAPE UNIT

A tape deck mounted in a case (typically plinth) complete with the equalisation and preamplification required to drive a separate hi-fi amplifier.

TRACKABILITY

A term referring to the tracking ability of a cartridge or pickup system. It generally refers to the threshold of tracking (i.e., just prior to the onset of mistracking) at a given playing weight, recorded velocity and frequency.

TRACING DISTORTION

Harmonic distortion resulting from the inability of the pickup stylus faithfully to trace high frequency, high level modulation due to the finite radius of the stylus tip (see Fig. 6.13).

TWO-TRACK

A reel-to-reel tape machine designed to record two tracks on magnetic tape and to replay the tracks so recorded. On mono each track is used separately, as at (a) in Fig. 6.14, and on stereo both tracks are used together, as at (b) in Fig. 6.13. The first is sometimes called half-track mono and the second half-track stereo.

VERTICAL TRACKING ANGLE

The vertical stylus motion path relative to the true vertical. Records are made with a 15° vertical tracking angle and for the least distortion the pickup stylus assembly is arranged to have a corresponding vertical tracking angle (see Fig. 6.15). This is different from rake angle.

VARIGROOVE

A device employed during disc recording which automatically regulates the spacing of adjacent grooves depending on the level of the signal at any instant. The scheme leads to maximum utilisation of the available recording space.

VU METER

Signal level meter which indicates the amount of energy contained in the signal rather than the amplitude peaks. Often found on tape recorders and tape units. Such meters are generally adjusted so that full recording level is reached when the meter points to 0VU or 100% modulation. The distortion factor is then *circa* 5%.

WOW

Cyclic variation in pitch of constant tone recording when replayed due to irregularities in tape transport or turntable drive. Refers to fluctuations below about 10Hz. See also under *Flutter*.

CHAPTER SEVEN

SOUND AND ROOM ACOUSTICS

ACOUSTICAL FEEDBACK

A POSITIVE FEEDBACK CONDITION whereby the feedback loop is partly electrical and partly acoustical and such that when the loop gain exceeds unity sustained oscillation occurs, colloquially referred to as howlback. The frequency at which the loop gain exceeds the requirement for oscillation is often determined by a resonance or resonances in the electrical and/or acoustical feedback path, and since the fundamental (low-frequency) resonance of the loudspeaker(s) figures in such a resonance, the oscillation often occurs at a low frequency.

A typical acoustical feedback condition is established when a microphone connected to a low-level input of an amplifier is within the direct sound field of a loudspeaker connected to the output of the same amplifier. A low-frequency howlback then occurs beyond a critical setting of the volume control.

Another condition, this time involving a record playing deck, hi-fi amplifier and loudspeakers, is illustrated in Fig. 7.1. In this case it is the gramophone pickup whose stylus is in contact with the record which responds to the sound waves from the loudspeaker. Such a combination, of course, is far less sensitive than a microphone, but when the resonance of the pickup system, the resonance of the loudspeaker and possibly the resonance of the room tend towards the same frequency, a substantial peak in loop gain can occur which can break through the stability threshold when the amplifier volume control is well advanced, thereby starting the howlback which continues until the volume control is retarded.

The condition should not occur at normal, hi-fi listening level. If it does, then it means that either the loudspeaker/s is/are too close to the record playing deck or that the record playing deck has a bad resonance which is close to that of the room or loudspeaker(s).

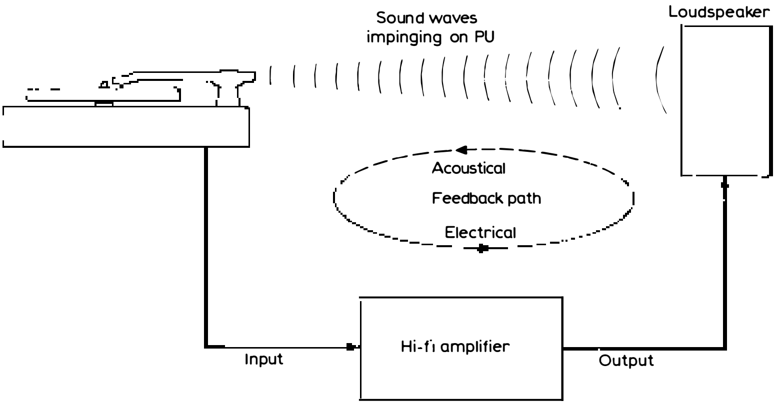


Fig. 7.1 Illustration of acoustical feedback.

Insufficient mechanical decoupling in the record playing deck is a common cause since the plinth can aggrevate the effect by acting as a (tuned) sounding board.

In a very bad case a change of loudspeakers can help, as can a change of pickup cartridge, the idea being to stagger the low-frequency resonances. There is not much that can be done to change the resonance frequencies of the listening room, but an offending standing wave (eigentone) can sometimes be reduced by changing the placement of the loudspeakers, rearranging the furniture or increasing the room damping.

In public address and sound reinforcement applications, when the feedback is via a microphone, the stability margin can be increased by shifting the frequency of the signal fed to the loudspeakers by a few Hz. A device known as a 'Frequency Shifter' (for howl reduction) which works on this principle is available (Surrey Electronics). The frequency shift is *circa* 5Hz, and the device is said to improve the stability margin by some 6–8dB.

AMBIENCE

The acoustical 'atmosphere' of a concert hall or listening room resulting from multiple signal reflections, echos, etc. Although a form of colouration, it is desirably retained in the reproduction when actually present on the programme signal.

The ambience of the listening room, however, can sometimes be obtrusive, giving an unnatural form of colouration to the reproduction. Insufficient damping in the listening room, leading to a too large reverberation time, can badly colour the reproduction.

Recent stereo recording results in some ambience information. This can be retrieved and fed to back loudspeakers by simple loudspeaker matrixing for four-loudspeaker reproduction, as explained in Chapter 4.

ANECHOIC

A free-space condition without reverberation. To avoid having to test loudspeakers out-of-doors in natural free-space conditions (necessary to avoid errors due to reflections, etc.) an artificial anechoic condition is created by specially treating room boundaries with heavily acoustically absorbent material.

ATMOSPHERIC PRESSURE

Unit pressure exerted on all objects and from all directions due to the effective weight of the atmosphere. The standard atmospheric pressure (atm) is 101,325 pascals (Pa), usually taken nominally as 1 bar, since 1 bar is equal to 10^5Pa, which corresponds to 10^5 newtons per square metre in SI units (see under *Sound Waves*).

ATTACK

Generally descriptive of amplifiers and loudspeakers to transient reproduction. A loudspeaker exhibiting good transient performance implies that the diaphragm responds immediately to a fast-rising transient component of a music signal and that the diaphragm after the passing of the transient is adequately damped acoustically and/or electromagnetically to avoid overshooting or 'ringing' at its natural resonance frequency (which is a form of transient distortion).

To preserve the transient characteristics of the music signal passing through the amplifier, the rise-time of the amplifier needs to be faster than that of corresponding to the highest audio frequencies (see Chapters 1 and 2).

AUDIO POWER REQUIREMENTS

The audio power requirements of an amplifier for hi-fi listening depend on the r.m.s. peak sound intensity required in the listening room, on the volume of the listening room and its reverberation characteristics and on the efficiency of the loudspeaker(s). Some devoted hi-fi enthusiasts favour listening at an intensity of r.m.s. peak levels approaching 100dB, corresponding to a sound pressure level of 2Pa or 20μbar, which is an incredibly loud sound in the domestic scene.

This sort of pressure occurs at 1 metre from a loudspeaker operating under free-field conditions (i.e., out-of-doors) when the acoustical power delivered by the loudspeaker lies in the order of 125mW, assuming spherical free-field radiation. Thus, if the loudspeaker is 1% efficient the electrical power input to yield this 1m sound pressure would need to be around 12·5W.

However, indoors the conditions change significantly. On the one hand, the room tends to 'magnify' the sound pressure due to reflections from walls, surfaces of objects, etc. and on the other hand it tends to 'dampen' it owing to the absorption effect of furnishings, carpets and so forth (see under *Reverberation* and *Reverberation Time*).

The situation is made even more complicated when stereo or four-channel reproduction is adopted. Moreover, some of the smaller hi-fi loudspeakers are significantly less than 1% efficient (0·3% being more typical of such loudspeakers) as quite a lot of power is used up in the divider network and in maintaining a fairly flat response down to low frequencies.

Clearly, no hard and fast ruling can be easily applied to the amplifier power requirements. Nevertheless, various approximate methods for working out amplifier power have been evolved (by this author and others). One is based on a pair of stereo loudspeakers each of 1% efficiency, a reverberant in-room sound pressure of 96dB r.m.s. peaks and a room reverberation time of between 0·3 and 0·4 second, which is fairly typical due to contemporary furnishings. The *per-channel* amplifier power than works out very approximately to 7·5W for a room of some 2,000ft^3 (average size), and increasing by a factor of 1·5 for every extra 1,000ft^3.

If 100dB r.m.s. peaks are required without amplifier clipping, then the amplifier power should be increased by a factor of 2·5. Also, of course, greater amplifier power is required when the efficiency of the loudspeakers is less than 1%. For example, for 96dB r.m.s. peaks in a room of 3,000ft^3 and 0·5% loudspeakers the amplifier should have a continuous wave power rating of 22+22W.

An amplifier of 55+55W would be required under the same conditions to produce 100dB r.m.s. without clipping, but the loudspeakers must always be capable of handling the high input power without distress (see Chapter 2).

BEAT

A difference tone produced by the non-linear mixing of two tones or signals. Such non-linear mixing can occur in the ear itself, which is one reason why a beat tone can be heard during the presence of two separate tones close in frequency. Beats can also arise from harmonics and sub-harmonics, which are sometimes used in tuning string instruments after establishing the correct tuning of one string with a tuning fork or pitch pipe.

BINAURAL

Hearing with two ears, usually on headphones. Not to be confused with monophonic or stereophonic

COLOURATION

A change in quality of sound commonly caused by room and loudspeaker resonances. Ambience is a form of colouration, and this can be undesirable when it results from listening room resonances. See also Chapter 2.

DAMPING

A method of damping resonances, electrically by resistance and acoustically by sound absorbent material. Absorbent damping is extensively used in loudspeaker enclosures for eliminating standing waves therein and hence for quelling resonance peaks.

DERIVED CENTRE CHANNEL

When it is necessary for an abnormally large spacing to exist between the two stereo loudspeakers, a so-called 'hole-in-the-middle' effect can occur, where sound midway between the loudspeakers is weak or apparently lacking. This can be overcome by reinforcing the centre-point sound by means of a third loudspeaker driven by a suitable level of left channel plus right channel (L+R) signal, as shown in Fig. 7.2.

In this illustration, the centre-channel signal is obtained by adding left and right channel signals at low-level and then applying the so obtained L+R signal to a single-channel amplifier to feed the centre

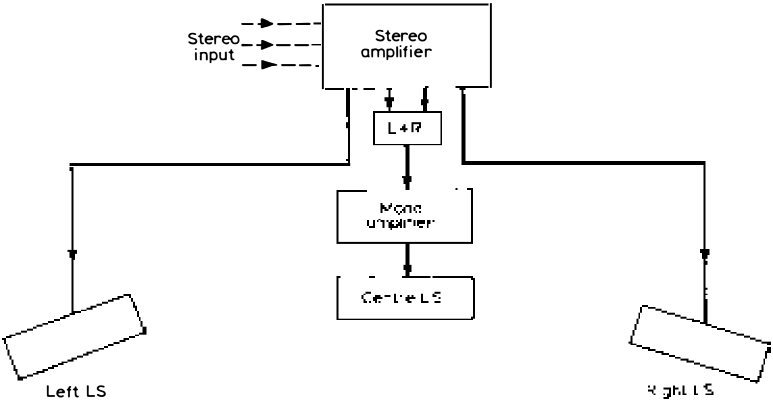


Fig. 7.2 Method of deriving a L+R signal for powering a centre-channel loudspeaker (see text).

loudspeaker. This method facilitates the addition of the signals without impairment to the stereo crosstalk, which can result when the signals are added at high level or power owing to the difficulty in obtaining adequate signal isolation in low impedance circuits.

Moreover, the separate single-channel amplifier permits easy adjustment to the level of the signal fed to the centre loudspeaker. The converse of L—R matrixing.

DYNAMIC RANGE

Sound intensity range between *fff* and *ppp*, as shown in Fig. 7.3. Acoustically, the upper *fff* limit is established by the peak overload of the amplifier and loudspeakers and the lower *ppp* limit by the intrinsic noise on the programme signal and the ambient noise in the listening room. High quality programme sources accommodate a dynamic range of at least 60dB, and most hi-fi amplifiers have a potential dynamic range capability even better than this.

Thus, if it is required accurately to reproduce a 60dB dynamic range, a r.m.s. peak output of 60dB above the total noise threshold must be established to avoid *ppp* sounds being masked by ambient noise. This, then, implies that in a room of overall ambient noise level of, say, 35dB, the amplifier and loudspeakers will need to supply r.m.s. peak sound pressure up to 95dB (see under *Audio Power Requirements*).

This is not quite true in theory because the *overall* noise level is an integration of noise components across the full spectrum, which means that masking of any particular range of music frequencies will

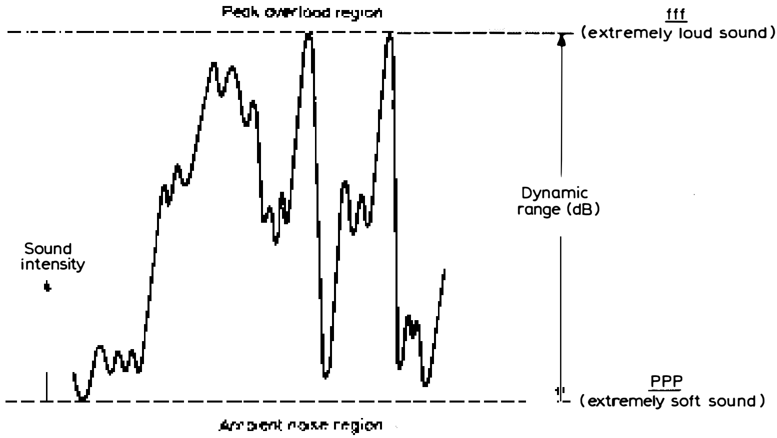


Fig. 7.3 Illustration of dynamic range.

occur only when the ambient noise components over the same range of frequencies are of greater amplitude, as shown in Fig. 7.4.

ECHO

Multiple to-and-fro reflection of sounds between walls, floor and ceiling, etc. of listening room or auditorium. A function of reverberation (see under *Reverberation*).

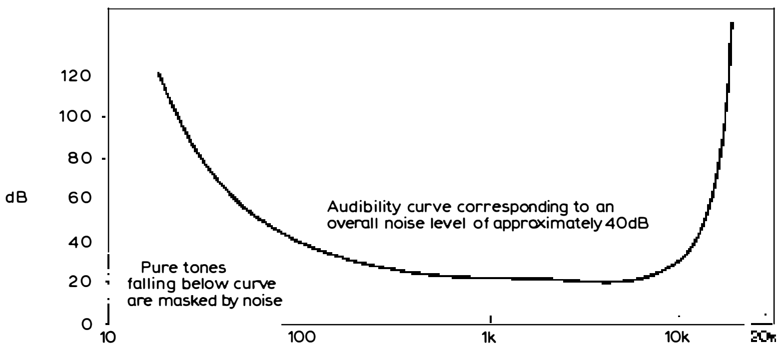


Fig. 7.4 Threshold of audibility curve combined with a noise masking curve of *circa* 40dB overall noise level. A range of frequencies or a pure tone of level below the curve is masked by the noise. For example, a pure tone of 1kHz would need to be greater than 20dB sound pressure to be audible in a listening room of 40dB overall (total) ambient noise level.

EIGENTONES

In-room resonances resulting from standing waves developed between parallel surfaces, such as between walls and ceiling and floor.

fff

Musical term corresponding to *molto fortissimo*, meaning extremely loud.

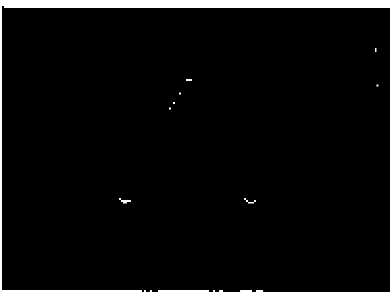
FREE-FIELD

Acoustically, meaning an environment whereby sound radiation can expand in an unrestricted manner. For example, an environment devoid of sound reflecting features. In clear space out-of-doors (see also under *Anechoic*).

FREQUENCY OF SOUND

Cyclic variation in sound pressure or vibration of a sounding source in Hz corresponding to a pure tone of specific pitch. The sinewave representation of a sound wave carrying pure tone information.

All orchestral instruments produce complex waveforms comprising the fundamentals and harmonics or overtones (though the latter may or may not be harmonically related to the fundamentals). The



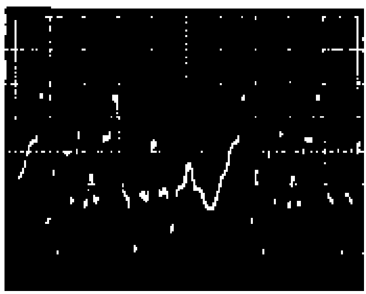


Fig. 7.5 (left) Oscillogram of the author's wife (Babs) singing middle C (the whole family joins in the conception of an audio book!). Fig. 7.6 (right) Oscillogram of sounding pitch pipe in the key of B.

harmonics and/or overtones are responsible for the timbre of 'character' of the particular instrument.

Even a tuning fork when sounding in the hand yields two main overtones at 6·27 and 17·55 times the fundamental frequency (a characteristic vibratory mode of metal bars secured at one end). However, when a sounding tuning fork is held against a resonator the fundamental frequency greatly predominates, the tone then being essentially pure (i.e., of sinewave nature).

Bottom A of a full-octave piano has a fundamental frequency of 27·5Hz and top C a fundamental frequency of 4,186Hz. The fundamental frequency of middle C is 261Hz.

Oscillogram Fig. 7.5 is of the author's wife (Babs) singing middle C, this deviating from pure sinewave owing to the presence of harmonic components. Oscillogram Fig. 7.6 is the waveform corresponding to a sounding pitch pipe in the key of B, again components other than the fundamental contributing to the wave shape.

HAAS EFFECT

A characteristic of human hearing whereby when the same sound is radiated at equal level from two loudspeakers but with a small delay between the two (such as would occur when one loudspeaker is closer to the listener than the other), the sound appears to the listener to emanate from that loudspeaker which is without the delay. Also known as the *precedence effect* because the sound which arrives at the listener's ears first takes precedence in terms of evoking the subjective response.

The effect is closely associated with the subjective effects of stereo and four-loudspeaker reproduction. The sound which is slightly delayed needs to be at a greater level than the undelayed sound to 'neutralise' the Haas effect.

HOLE-IN-THE-MIDDLE

The apparent weakness or lack of sound at the point midway between the two stereo loudspeakers when they are placed too far apart. Also see under *Derived Centre Channel.*

HOWLBACK

An oscillation (usually low-frequency) resulting from a positive feedback path from the output back to the input of an amplifier which is partly electrical and partly acoustical. Also see under *Acoustical Feedback.*

INTENSITY OF SOUND

The sound or acoustical energy or power per unit area occurring in a given sound field. The intensity is affected by the nature of the environment containing the sounding source or in which the sound field is operative, such as the listening room when the source is a loudspeaker. See also under *Audio Power Requirements.*

Sound power analysis requires two basic instrument capabilities; one to measure directly space/time averaged sound pressure level for each frequency band, and the other which takes account of the 'room effect' in reverberant and semi-reverberant environments.

It is internationally accepted that the sound power per metre of area at a sound pressure corresponding to 0dB (2×10^{-5}Pa) is a mere 10^{-12}W (i.e., 10^{-12}W/m^2). Thus at a sound pressure level of 100dB, which is a very loud sound, the acoustical power is 10^{-2}W/m^2.

LISTENING INTENSITY

Average r.m.s. peak sound pressure when listening to music is around 70dB(A), the suffix A indicating the kind of weighting applied to the sound pressure level measuring instrument which takes into account the changing sensitivity of the ear to sounds of different frequencies and levels (see under *Loudness*). The curve for A weighting is given in Fig. 7.7, but there are other weighting characteristics.

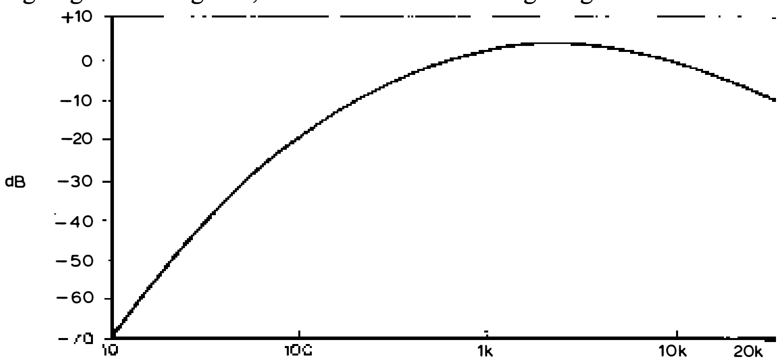


Fig. 7.7 A-weighting characteristic.

The acoustical power at 70dB sound pressure is $10^{-4}W/m^2$. If we peak our hi-fi at 100dB sound pressure level, then the acoustical power is $10^{-2}W/m^2$, or 10mW acting on a unit area of 1 metre. We need the high amplifier powers to combat the sheer inefficiency of the loudspeakers and to cater in part for the absorption characteristics of the listening room.

Horn loudspeakers of 50–60% efficiency call for much smaller amplifier powers to sustain, say, 100dB r.m.s. peaks in the listening room. This is the philosophy of Lowther Acoustics Limited (the old established British firm of hi-fi, horn-type loudspeakers and—even today—valve amplifiers).

LOUDNESS

The subjective relationship to sound intensity, which follows a logarithmic law. That is, the perception of loudness does not follow proportional changes in sound pressure or intensity, but instead logarithmic changes. Moreover, loudness is a function of the frequency of the sound, as shown by the curves of equal loudness in Fig. 7.8.

The unit of loudness is the phon, and the loudness level in phons of a sound is almost numerically equal to the intensity level in decibels of a 1kHz pure tone which is judged by listening to be equally loud.

Taking the 60 phon curve in Fig. 7.8 as an example, it will be seen

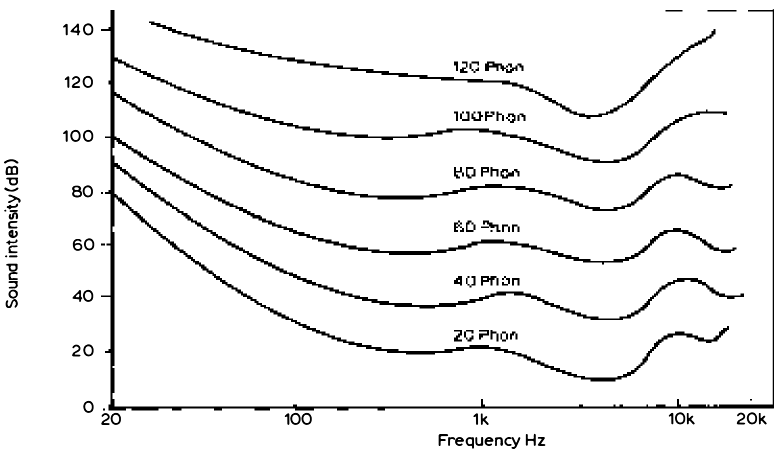


Fig. 7.8 Curves of equal loudness after Robinson and Dadson. The treble sensitivity of the ear diminishes with age $0dB = 2 \times 10^{-5}$ Pa.

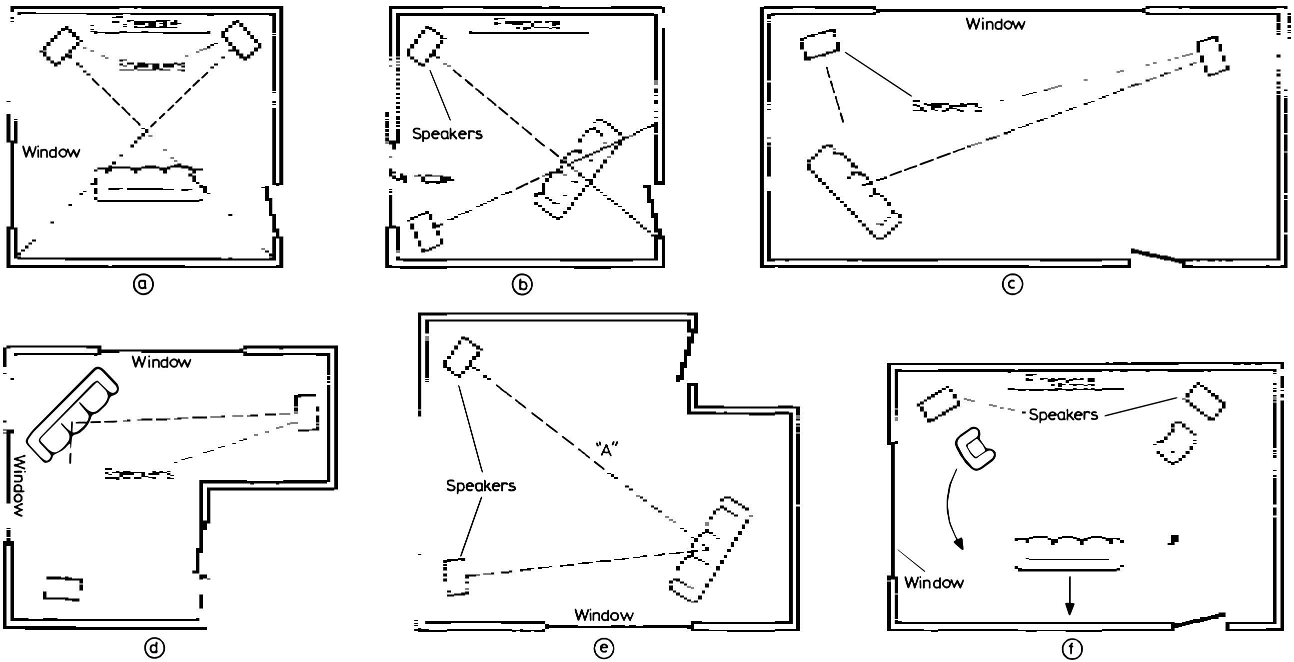


Fig. 7.9 Some loudspeaker placements suggested by Alan, the author's son-in-law. (a) ideal stereo listening position. (b) typical compromise; correct use of balance control to improve stereo image. (c) awkward (long) room needs careful adjustment of balance control. (d) fairly good position, despite shape of room and two windows. (e) reasonable postion for stereo; could be better at point A. (f) a typical situation; to avoid furniture obstructing sound path and therefore seriously affecting the main stereo area, furniture should be moved as shown by arrows.

that around 1kHz the loudness is 60 phons when the intensity level is 60dB. To obtain the same loudness at 100Hz, however, the intensity level needs to be about 67dB. A lift of about 5dB in intensity is also necessary at 10kHz to retain the 60 phon loudness sensation.

As the intensity of the sound falls, so greater bass and (to a smaller extent) treble lifts are required to maintain a given loudness in phons.

We saw in Chapter 1 that some amplifiers have a facility to combat this natural and subjective effect in the form of a so-called 'loudness' control. This is arranged so that progressively more lift in bass and a smaller lift in treble is applied as the volume control is turned down. (See also under *Threshold of Hearing* and *Threshold of Pain*).

LOUDSPEAKER PLACEMENT

Basically, for stereo reproduction, the two loudspeakers need to be separated by at least 1·8m with the listener sitting at the third point of an equilateral triangle formed by himself and the two loudspeakers for optimum stereo effect.

If the loudspeakers are too close together, the stereo sound stage will be diminished and the full impact of stereo reproduction lost, while too wide a spacing will result in a hole-in-the-middle effect. It is also desirable sometimes to angle the loudspeakers so that their axes line up with the listener.

For four-loudspeaker reproduction, the stereo loudspeaker placement can remain, while the two extra loudspeakers can be arranged in a complementary manner at the back of the listening scene. Some four-channel listeners prefer having the back loudspeakers more to the sides of the listening area, while for 'ambio' listening, when the back loudspeakers derive signals from a simple loudspeaker matrix (see Chapter 4), there is sometimes merit in facing the back loudspeakers towards the wall to diffuse the rear sounds.

There are no hard and fast rules that can be applied to loudspeaker placements. It generally requires careful and detailed experiment to establish the best positions for the loudspeakers, depending on the room size, shape and acoustical conditions. It may also be necessary to adjust carefully the amplifier balance control to centre the stereo stage when the two loudspeakers are not equal distance from the listeners.

Some ideas suggested from experience by Alan, the author's son-in-law, are given in Fig. 7.9.

MASKING

A property of human hearing whereby a sound of given frequency or range of frequencies is made inaudible by another sound of the same frequency or range of frequencies. A sound can also be masked by noise (see under *Dynamic Range* and Fig. 7.4). Some of the noise reduction systems exploit this property of human hearing.

MONAURAL

Listening with one ear. Different from monophonic.

NOISE

In hi-fi parlance, referring to the background 'hiss', and possibly hum, resulting from spurious signals generated by the equipment. Noise (excluding hum) has component frequencies covering a wide spectrum, and such unadulterated noise is called *white noise*, being

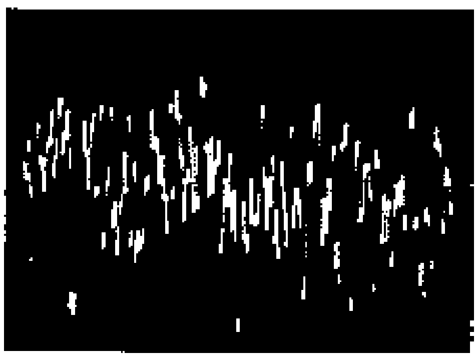


Fig. 7.10 Oscillogram of noise.

analagous to white light which contains components of all light frequencies.

White noise sounds rather like a car tyre running on a wet road and has constant energy per *unit* bandwidth. Noise is sometimes filtered, such as so-called *pink noise* which has constant energy per *octave* bandwidth. An oscillogram of white noise is given in Fig. 7.10.

PHON

The unit of loudness. Numerically equal to the sound intensity in decibels at *circa* 1kHz. See also under *Loudness.*

PITCH

The quality of sound resulting from sound waves of a particular frequency (see also under *Frequency of Sound*).

ppp

Musical term corresponding to *molto pianissimo*, meaning extremely soft.

PSYCHO-ACOUSTICAL

A term meaning the psychology of human hearing which is being used more in recent times since the advent of four-loudspeaker reproduction. One such association is the Haas or precedence effect; another is masking. These play subjective roles in stereo and four-loudspeaker reproduction.

QUADRAPHONIC

A Latin-Greek hybrid describing four-loudspeaker reproduction. Semantically offensive in some quarters.

REVERBERATION

Multiple and decaying reflections of sound waves in the listening room, auditorium, etc. (see also under *Ambience* and *Colouristion*). The quality which yields the acoustical atmosphere or ambience.

REVERBERATION TIME

The time that elapses from the cessation of a sound to the moment it reaches one-millionth of its original intensity; in other words, the time taken for a sound to fall by 60dB from its initial steady state value.

Reverberation time stems from the multiplicity of echos from reflecting surfaces. When a sound is started in an enclosed space it does not reach maximum intensity immediately; neither does it collapse immediately to zero when the sounding source ceases.

THRESHOLD OF HEARING

The lowest level of sound intensity (corresponding to 0dB) that is audible in a suitably quiet, non-masking environment to a person of normal hearing. Corresponds to a sound pressure at middle frequency of 2×10^{-5}Pa and to a sound power of 10^{-12}W/m^2. See also under *Loudness*.

THRESHOLD OF PAIN

Sound intensity of such high level that it results in pain to the listener. Corresponding to about 130dB, which is equal to a sound pressure of about 63Pa and a sound power of 10W/m^2. A jet plane at take off yields this sort of intensity close by.

Thus, from the threshold of hearing (0dB) to the threshold of pain (130dB) the sound pressure range is a little over three-million-to-one and the sound power range about ten-million-million-to-one. Small wonder, then, that Nature has endowed human hearing with a logarithmic response characteristic.

VELOCITY OF SOUND

Correctly called the velocity of propagation, it depends on the physical properties of the medium, including temperature. When the medium is atmospheric air at sea level and 20°C, the velocity of propagation is close to 344m/S (1130ft/S).

WAVELENGTH OF SOUND

The wavelength of sound is also related to the frequency and the velocity of propagation according to the expression $c=\lambda f$, where c is the propagation velocity in metres/sec., λ the wavelength in metres and f the frequency in Hz.

The propagation velocity is fixed by the medium, temperature, etc., so it is easily possible to find the wavelength merely by dividing the velocity (344) by the frequency in Hz. For example, the wavelength of the sound emanating from a piano when the middle C key is struck is 344/261, or 1·318m, corresponding to the fundamental frequency, of course.

JOHN EARL

Readers of audio literature and record magazines will be familiar with the work of the author, particularly in the field of hi-fi sound reproduction. One of his chief occupations is the detailed testing of audio equipment of all types from all parts of the world, and his wide experience in this respect is clearly illustrated in the pages of this book.

Although trained as an engineer, John Earl has gained a considerable reputation for his ability to present to his readers complex technical material in a very down-to-earth manner and this is one reason why his articles and equipment reviews are so very readable.

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