

THE Hi-Fi HANDBOOK

An introduction to the terms and technology
of serious sound reproduction

 PIONEER®

THE HI-FI IDEA

A TYPICAL STEREO SYSTEM LAYOUT

CABINET

Construction should be level, stable, vibration-free and able to bear heavy loads.

HEADPHONES

Disconnect when not in use. Store in dust-free location.

THICK CURTAINS

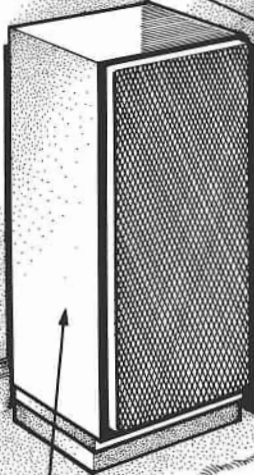
To shield equipment from direct sunlight. To reduce listening room reflections and resonance.

TAPE DECK

Instal reel clamps when operating vertically. Protect with dust cover when not in use.

AMPLIFIER

Do not use longer connecting leads than necessary. Don't locate unit in direct sunlight, near radiators or other heat sources. Avoid poorly ventilated, humid or dusty locations. Don't locate where alcohol, insect spray or inflammable material is used or stored.



SPEAKER CONNECTIONS

Check that speaker leads have sufficient current handling capacity. Don't use longer lead lengths than necessary.

SPEAKER SYSTEM

Rear and side panels of left and right speakers should be at the same conditions. (Placing rear panel against a wall improves bass.) Instal speakers so that vibrations are not transferred directly to the floor. Use stands or concrete blocks with bookshelf type speakers.

TURNTABLE

Protect from vibration. Keep dust cover closed whenever possible.

TUNER

Do not use longer connecting leads than necessary. Instal antenna in direction providing best reception. An outdoor FM antenna is recommended for FM reception.

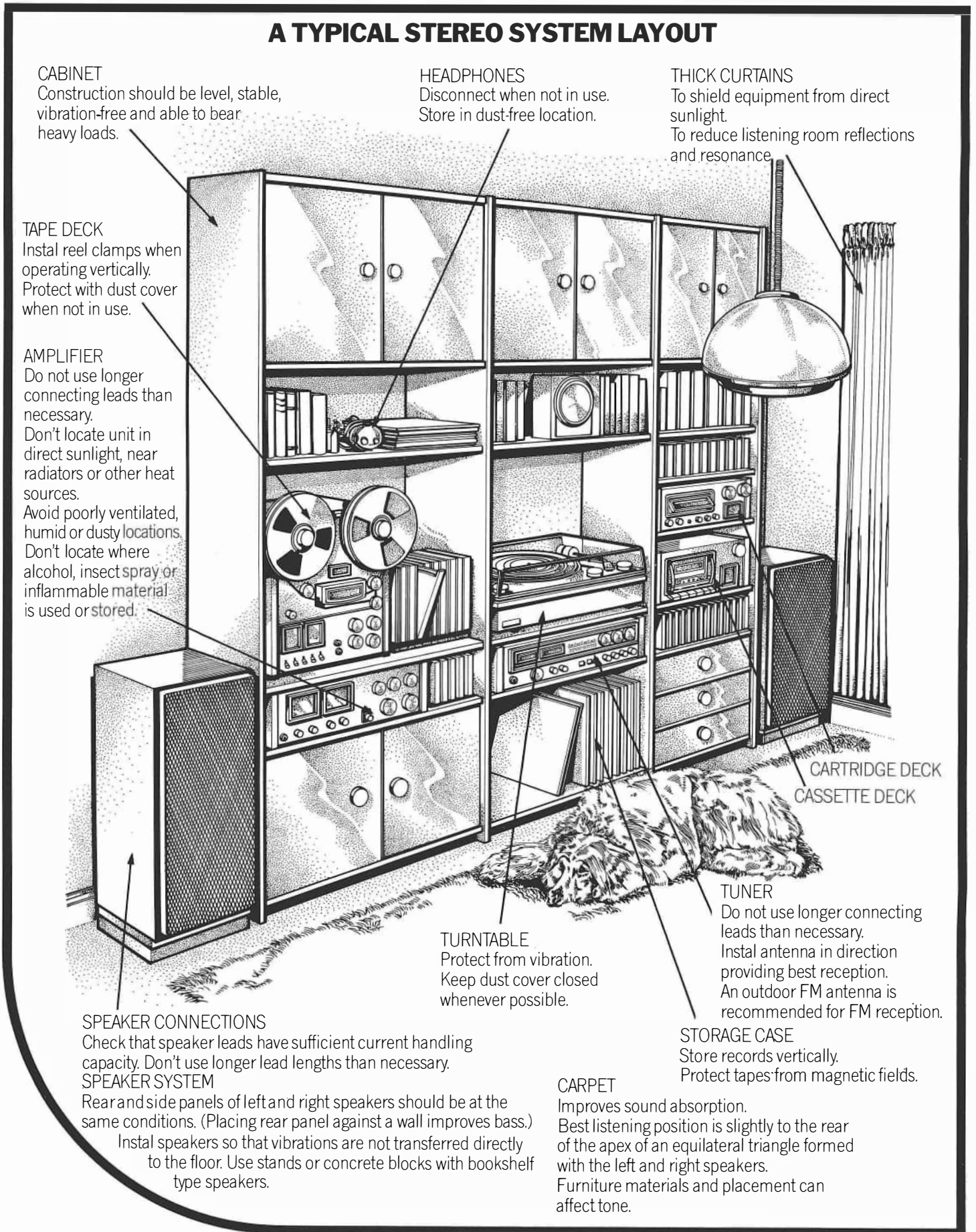
STORAGE CASE

Store records vertically. Protect tapes from magnetic fields.

CARPET

Improves sound absorption. Best listening position is slightly to the rear of the apex of an equilateral triangle formed with the left and right speakers. Furniture materials and placement can affect tone.

CARTRIDGE DECK
CASSETTE DECK



Ever since the first gramophones began squeaking their scratchy messages into the ears of a surprised world around the turn of the century, there has been an uninterrupted search for better, clearer, finer methods of recording and relaying sounds. Yet all the major breakthroughs were made in the last two decades – the first LP records, the invention of stereophonic recording and reproduction, the beginning of FM and FM stereo broadcasting, the development of noise-free amplifiers, the introduction of solid state engineering into audio equipment and, quite recently, the advent of four-channel stereo sound.

Hi-Fi and Stereo

Here we must make a distinction between two terms that are often confused: high fidelity and stereo. High fidelity or hi-fi makes very much the same demands as an oath sworn in a court of law “... the truth, and nothing but the truth.” The goal of hi-fi is to record, preserve and reproduce sound, chiefly music, exactly the way it was played by the performers, without adding anything, without leaving anything out.

Stereo, on the other hand, means that sound is recorded and reproduced via two channels in order to add a directional, spatial sensation. Stereo creates the same effect to the ear that 3-D photography offers to the eye. To continue the visual analogy: hi-fi demands sharp, clear, undistorted pictures; stereo creates an illusion of space, width, and location.

Unfortunately, the two don't always go together. There are stereo sets on the market which do not deserve the name “hi-fi” because their sound quality is inferior. Conversely, there are hi-fi systems with only one channel (“monophonic”).

The ideal hi-fi system has never and will never be built, simply because the demand to speak the whole truth and nothing but the truth cannot be realized. Between the performers in the recording studio and the listener in his living room, there is a long and complicated chain of equipment and transformations, tearing off something from the original sound here, adding something there. These unwanted fiddlings and diddlings can be reduced to a minimum – and this is what hi-fi technology is all about – but they can never be totally eliminated.

Hi-Fi and Live Sound

On the other hand, hi-fi has things to offer that no live performance can match. Freedom. You, the listener, choose what you want to hear, when you want to hear it and with whom, how many times. From this point of view, hi-fi isn't a substitute for live concert performances; it's an addition or, if you want, an alternative.

This brings us to another common misconception, one that you find repeated in too many advertisements: that a good hi-fi system will bring the concert hall into your home. It's simply not true and it would be bad if it were. First, much of the world's great music, both old and modern, wasn't composed to be played in that 19th century invention, the concert hall.

Chamber music, first of all the string quartet, calls for the intimacy of small rooms. Bach wrote his organ works as an integral part of church liturgy. Brass bands are for outdoors, and much modern music is composed especially for the recording studio and electronic reproduction.

But even those works which were intended for concert hall performance, such as the classical and romantic symphonies, can't and shouldn't be played over a living-room hi-fi system with concert hall volume – the cannon in Tchaikovsky's 1812 overture would shatter the window panes. No, hi-fi isn't supposed to put the concert hall in your living-room; it's supposed to put the music in your living-room, and that it does remarkably well.

Hearing Errors

While we're dealing with the subject of misconceptions, there's another that says “when choosing hi-fi equipment, trust your ears rather than the manufacturer's specifications.” Several reservations must be voiced against this advice. The most important is that a person who buys a hi-fi system for the first time in his life is in most cases ill-equipped to judge the quality of musical sound reproduction.

Subjected from morning to night to the ceaseless trickle of non hi-fi sounds – the traffic din, the bass-heavy bellowing of jukeboxes, the babble and squall of cheap transistor radios and TV sets – the hi-fi novice hasn't had opportunity to train and hone his sense of hearing. Experience shows that, left to his own resources, he will usually choose an “impressive sounding” system, mostly with exaggerated bass response and poor brilliance.

As his hearing becomes sharper, as he gets used to the possibilities of his system and compares it with his friends' hi-fi installations, he will gradually outgrow his original purchase and will want to upgrade his system. But since every replacement costs money, he will finally end up with a good system that has cost him much more than if he had bought it in the first place.

We feel that our – and every conscientious hi-fi salesman's – advice to the hi-fi novice should be to trust a reputable manufacturer and store, more than his own as yet underdeveloped hearing.

Even though, his first encounter with hi-fi sound may not be pleasant at all, simply because he is unaccustomed to hearing musical sounds as they really are.

With these remarks we have finally touched upon the main theme of this booklet: How to choose a hi-fi system. In the following paragraphs, you will find concise explanations of the function of each component in a stereo hi-fi installation, as well as hints concerning listening rooms, component matching, and a glossary of technical terms.

We have tried to limit the technical explanations to the bare essentials (and even these have come out quite voluminous) while avoiding over-simplifications and pure advertising.

THE TURNTABLE

TURNTABLE DRIVE SYSTEMS

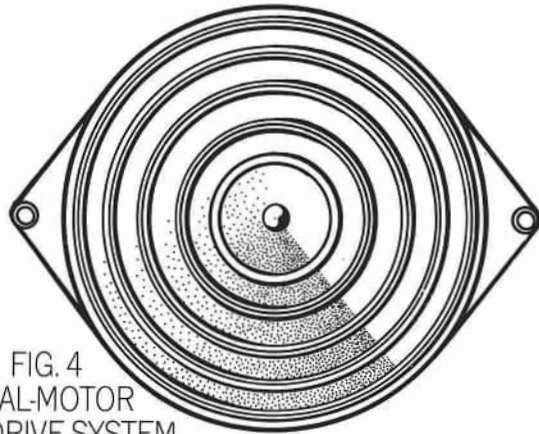


FIG. 4
DUAL-MOTOR
BELT-DRIVE SYSTEM

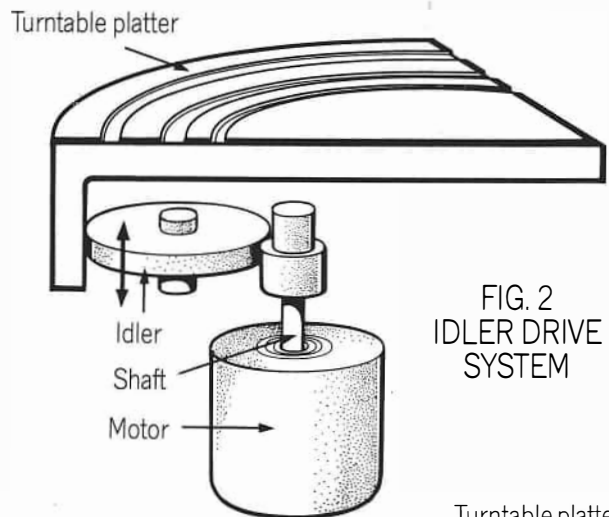


FIG. 2
IDLER DRIVE SYSTEM

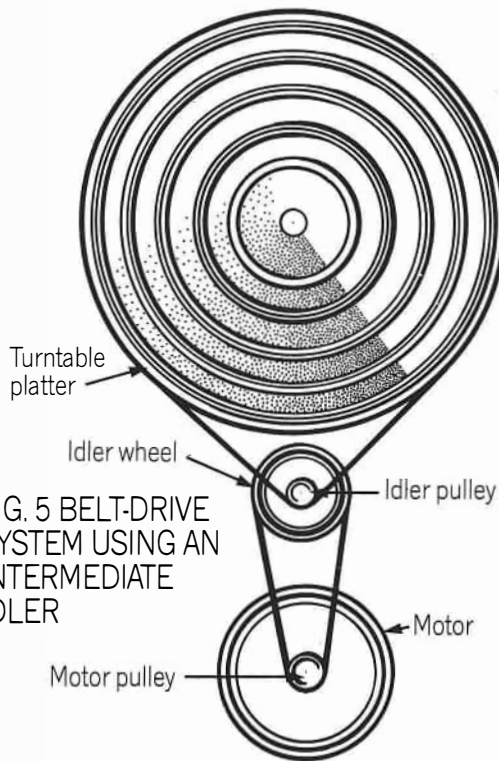


FIG. 5 BELT-DRIVE
SYSTEM USING AN
INTERMEDIATE
IDLER

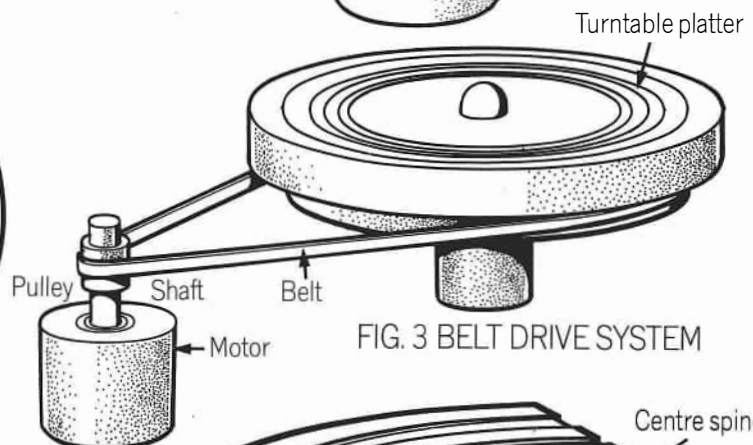


FIG. 3 BELT DRIVE SYSTEM

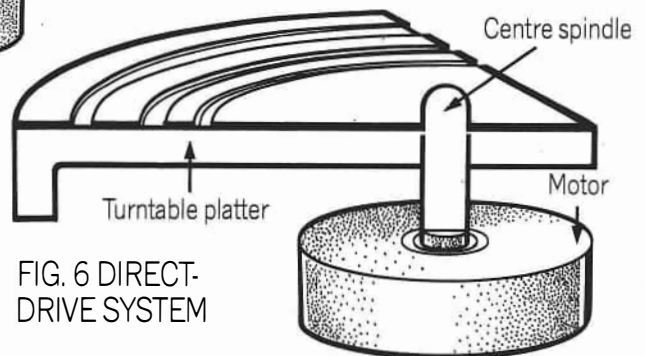


FIG. 6 DIRECT-
DRIVE SYSTEM

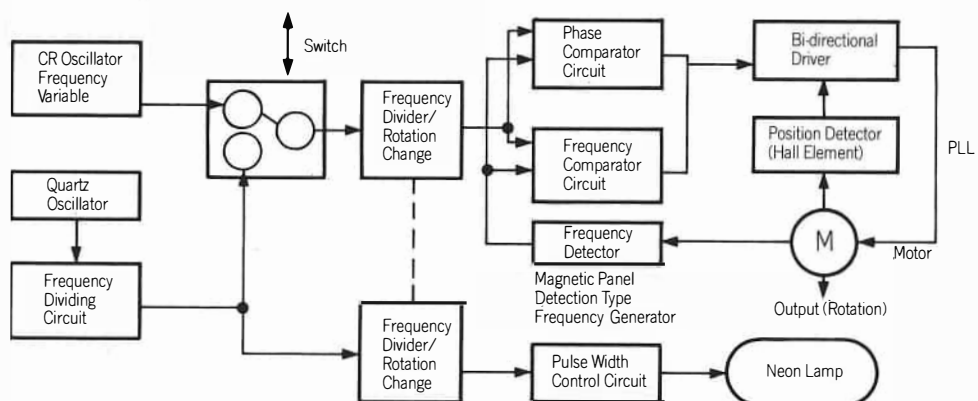
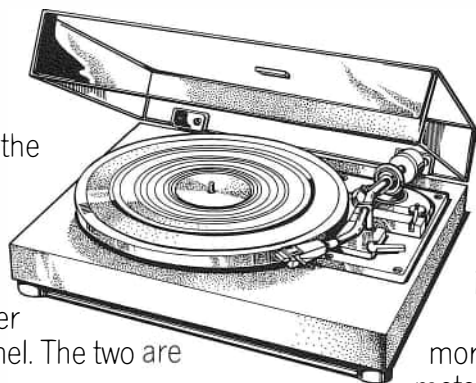


FIG. 7 QUARTZ PLL BLOCK DIAGRAM

On a stereo record, the sound information is contained in the microscopic grooves or rather, to be exact, in the undulations of the two walls of the groove. One (the inner) wall bears the information for the left stereo channel, the other (outer) wall that for the right channel. The two are at an angle of 90 degrees (Fig. 1).



As all of these motors run at higher rpm's than the 33½ and 45 rpm required for turning the record, they must be coupled to the turntable platter via a speed reducing linkage. This is usually done in one of two ways.

In the rim drive system (Fig. 2), one or more idler wheels transfer the power from the motor pulley to the turntable platter by way of friction of their rubber rims. Advantages are relatively low cost, easy speed changes and the possibility to use motors of comparatively low torque. On the other hand as the rubber rims wear out, a certain amount of slippage is bound to occur.

The other common drive system employs a rubber or plastic belt to convey the motor power to the turntable (Figs. 3, 4, 5). Speed changes are made by guiding the belt around either of two motor pulley sections with different diameters. The belt prevents motor vibrations from reaching the turntable, thus reducing noise and rumble. In precision engineered hi-fi turntables, the belt is often made of polyurethane because of its resistance to heat, humidity and oil, and its low elasticity.

Direct Drive Turntables

More recently, direct drive turntables have been introduced. In this design (Fig. 6), the motor and centre shaft form a common axis. The motor now turns slowly at exactly the same speed as the turntable platter, i.e. 33½ or 45 rpm. Vibration is thus reduced to an amount hitherto unattainable by hysteresis or synchronous belt drive turntables. Mechanical linkages between the motor and platter are completely eliminated, further aiding in an improved wow and flutter performance and reducing maintenance intervals. The direct drive motor can mean long term speed stability completely independent of variations in power line voltage or frequency fluctuations. Introduction of quartz crystal controlled direct drive motors (Fig. 7) has resulted in near laboratory standards of speed accuracy.

Turntable Platter

The turntable platter must be large enough to accommodate 30 cm LP's, but there is another reason for making it as large and heavy as the motor torque will permit. A large, heavy platter, once in motion, serves as a flywheel and keeps the speed constant by its own inertia. Turntable platters are usually made of pressed steel plate or die-cast aluminium alloys. The latter is preferable for high quality systems because of its better electrical properties (anti-magnetism) and greater machining precision. To work as an efficient flywheel, the turntable platter must be balanced just like a car wheel.

Motor and Drive System

The task of the turntable drive system seems simple at first glance: to do nothing but rotate the disc at a specified, constant speed. The difficulties are hidden behind those innocent looking words, "nothing but," "specified" and "constant." Any mechanical system with moving parts generates vibrations, inconsistencies and wobble, and in a turntable these would cause noise, unevenness of pitch ("wow") and rumble. Let us examine what is being done by turntable designers to counteract these troubles.

First, the motor in a hi-fi turntable has to meet six conditions: (a) it must revolve at a specified, constant rotational speed: (b) its speed must remain constant even if the supply voltage fluctuates: (c) it must operate almost without vibrations: (d) it must have powerful torque (rotational force) to overcome changes in its load and to bring the turntable quickly up to the specified speed: (e) it must not produce any magnetic flux leakage as this would induce noise ("hum") in the pickup system: and (f) it must be able to operate continuously for many hours.

Only two types of AC motors can fulfil all these conditions—the induction motor and the synchronous motor. Of these, the induction motor is used mainly for low priced turntables because its speed is liable to fluctuate with load variations and it would need a regulatory mechanism. This leaves the synchronous motor. Its speed is locked to the AC line frequency and therefore very constant even if the power voltage fluctuates. A special and rather expensive variety is the hysteresis synchronous motor, whose features include freedom from vibrations and noise-free operation. Most top class turntables are equipped with hysteresis motors although other designs (servo-controlled DC and AC motors etc.) have been tried with good results.

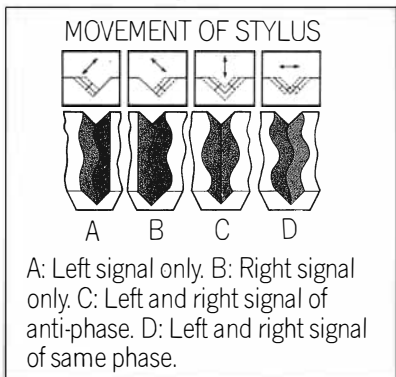


FIG. 1
THE GROOVE OF STEREO
DISC AND STYLUS MOVEMENT

THE TURNTABLE (contd)

PICK-UP SYSTEM DESIGNS

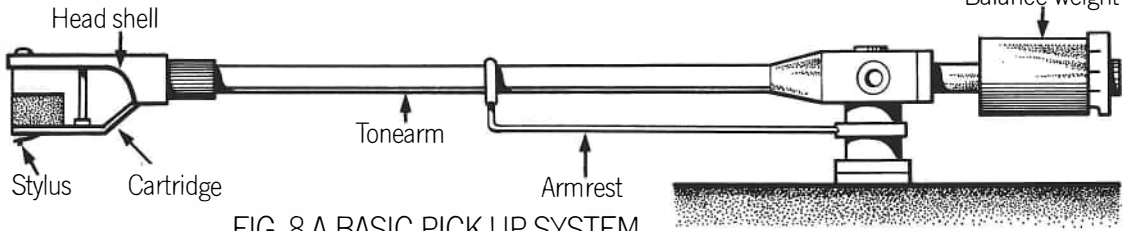


FIG. 8 A BASIC PICK-UP SYSTEM

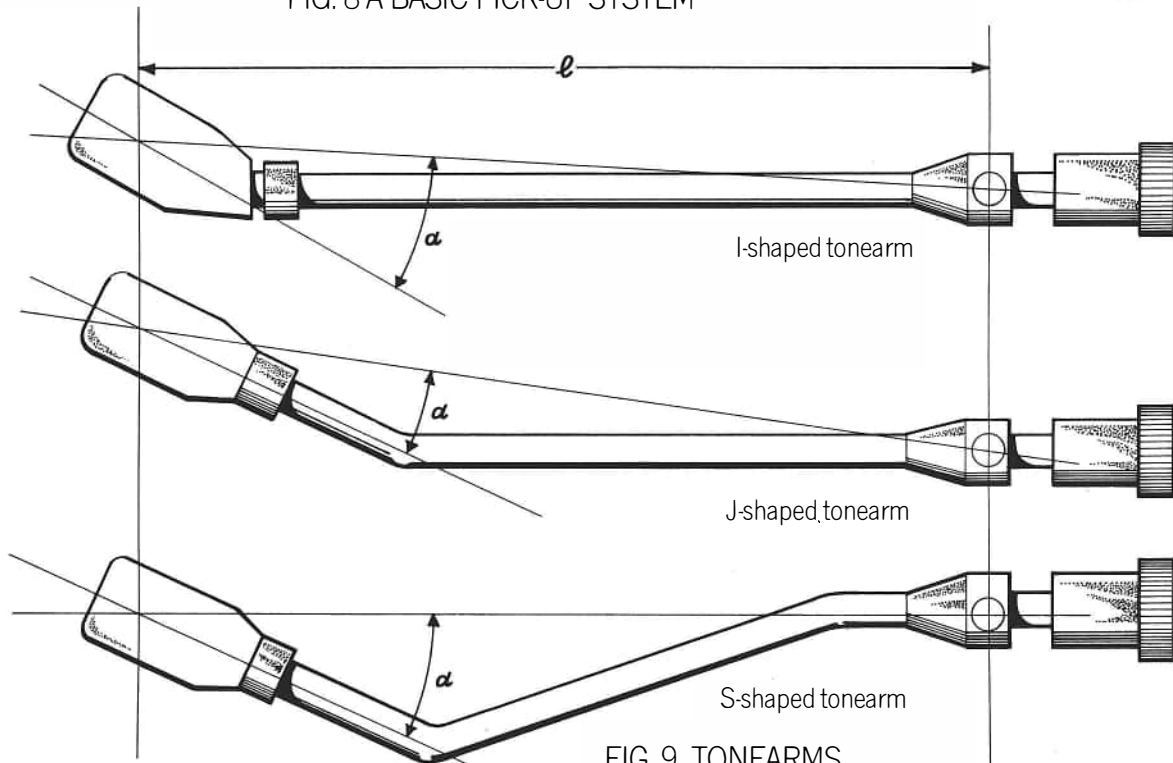


FIG. 9 TONEARMS

ℓ - effective length
 α - offset angle

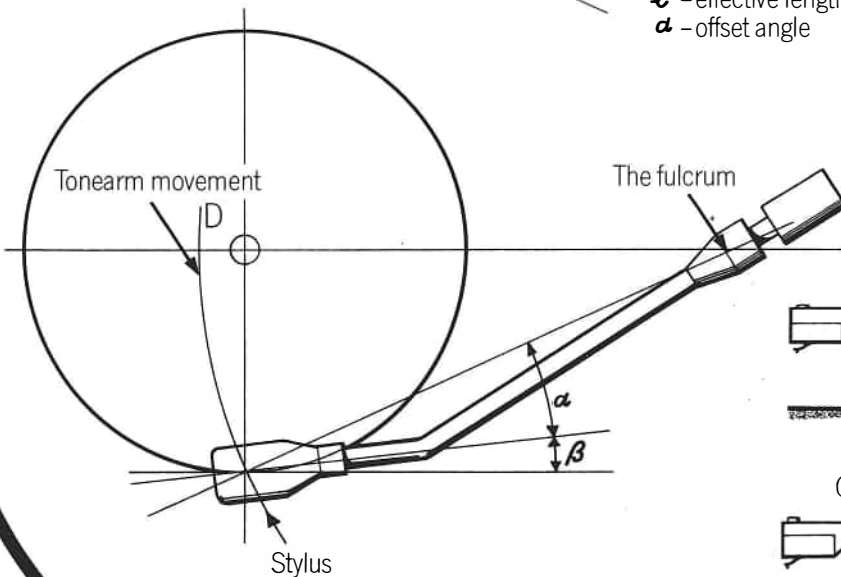


FIG. 10 OVER-HANG

α - offset angle β - tracking error angle
 D - over-hang

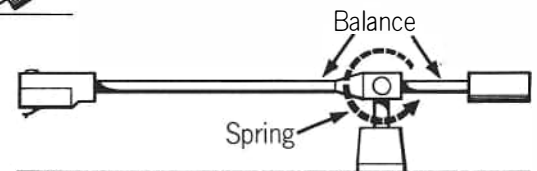


FIG. 11 DYNAMIC BALANCE

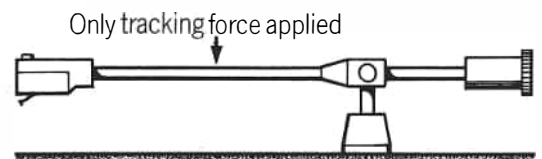


FIG. 12 STATIC BALANCE

Pick-up System

Most efforts in the design of hi-fi record playing equipment have been concentrated on the pick-up system (Fig. 8), i.e. tonearms and pick-up cartridges. The difficulties are caused by the great number of – often conflicting – factors that have to be taken into consideration. The theory of phono pick-up systems alone could fill a book.

Pick-up Cartridge

The cartridge, mounted in a shell at the tip of the tonearm, has the task of tracking the record grooves and transforming mechanical vibrations into an electric signal. As the stylus (“needle”) rests in the rotating record groove, it is deflected laterally and vertically in accordance with the undulations in the groove walls (Fig. 1). This creates one of the greatest problems: the stylus must follow (“track”) the complicated meandering of the groove instantly and without losing contact with the walls or jumping out of the groove. To be able to do this, the mass of the stylus and all moving parts must be kept as little as possible because the greater the moving mass, the higher its inertia, i.e. resistance to quick changes in movement.

Other conditions for good tracking are: a sufficient downward force called “tracking force” must be applied to the stylus to keep it in the groove – which if too great, will cause stylus and records to wear out more quickly than necessary; and the stylus assembly must have enough freedom of movement, technically called “compliance” or sometimes “trackability”.

Tonearm

A cartridge can only deliver its full performance if it is mounted on a tonearm of equally high quality. The tonearm’s main function is to hold the cartridge in its path while it travels across the record, and to apply the necessary tracking force while compensating for other, unwanted forces.

Most high quality tonearms are made of light metal alloy because of its light weight, stability, and easy machining. By their shape, (Fig. 9) arms can be grouped into three types: straight types, J-types and S-types. In every case, the shell cartridge assembly is at an angle (α) to the arm axis, the line connecting stylus tip to arm pivot. This angle is called offset angle, and its purpose is to minimize the arm’s tracking error (Fig. 10). Tracking error is the angle that the cartridge axis deviates from the record tangential at any given point on the record: the smaller the tracking error, the better. On some equipment, mainly the economy type, the shell cannot be removed from the tonearm (“integrated arm”). Most modern tonearms, however, have standardized plug-in connections for one cartridge-shell assembly, so that all cartridges following these standards can be freely interchanged.

Since different cartridges require different tracking forces, the tonearm should have a movable balance weight on its rear end, preferably with a graduated scale of tracking forces. The tracking force can then be adjusted by first balancing the arm

horizontally, then applying the desired force by moving the balance weight (or a sub-weight) further toward the arm pivot (“static balance”). On some arms tracking force is applied by the force of a spring (“dynamic balance”). See Figs. 11 and 12.

On most high quality turntables, the tonearms are also equipped with several auxiliary controls to improve their performance and ease of operation.

It requires a certain dexterity to set the stylus down on a record and to lift it off again, and countless records and cartridges have been damaged by careless handling. Most hi-fi tonearms are therefore provided with a lifter or cueing device, mostly a hydraulically damped piston which gently floats the tonearm down on the initial record grooves and lifts it up again at the end.

Anti-Skating Device

One other device must be mentioned here – the anti-skating (bias) device provided on many high quality tonearms, also called sidethrust compensator or inside force canceller. To understand its function, we must realize that a tricky combination of forces act upon stylus and tonearm as they track the record. Among these, the most disturbing is a force which tries to pull the tonearm inward, toward the center of the disc, thereby pushing the stylus against the inner wall of the record groove, or in extreme cases, causing the stylus to jump out of the groove. To assure perfect tracking of both groove walls (both stereo channels!), this force must be compensated.

Usually, a little weight affixed to a string applies a corresponding counter-force on the tonearm, although magnetic and sprung systems are also used. As the skating force varies with the stylus tracking force, the anti-skating device should also be re-adjusted each time the tracking force is changed.

TURNTABLE (contd)

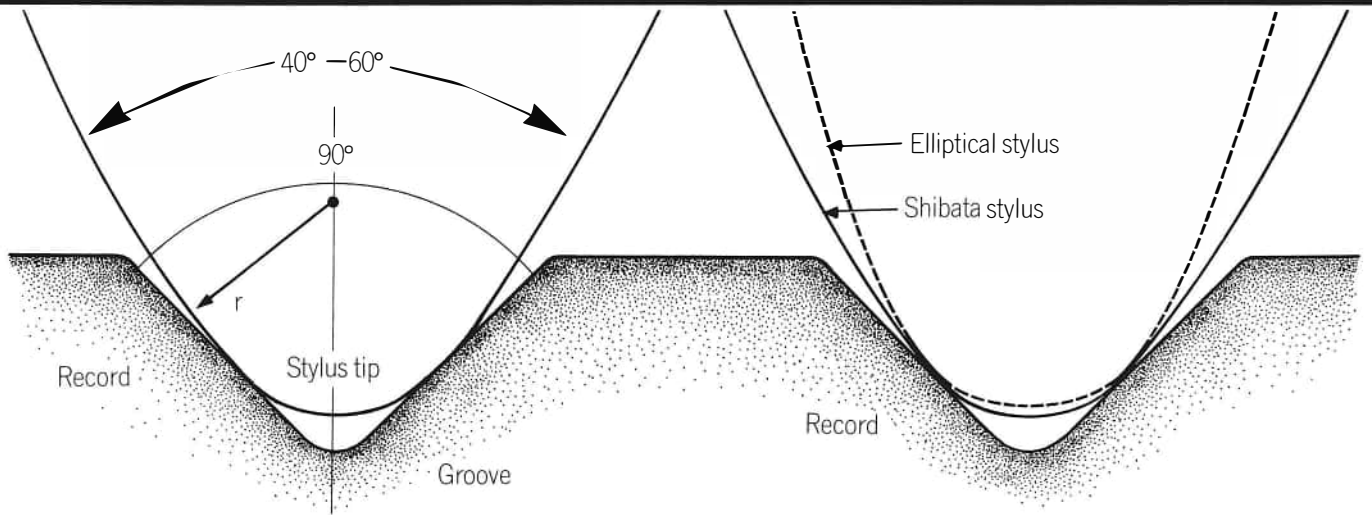


FIG. 13 STYLUS TIP RADII

For mono LPs 1 mil (0.025mm)
 Mono/stereo 0.7 mil (0.017mm)
 Stereo 0.5 mil (0.012mm)

FIG. 14 SHIBATA AND ELLIPTICAL TIPS

STYLUS CODES	
LP	= Long playing Mono 33 $\frac{1}{3}$ and 45 r.p.m. stylus tip
SR	= Long playing Stereo 33 $\frac{1}{3}$ and 45 r.p.m. stylus tip
Std	= Standard for 78 r.p.m. stylus tip
D	= Diamond stylus tip
S	= Sapphire stylus tip

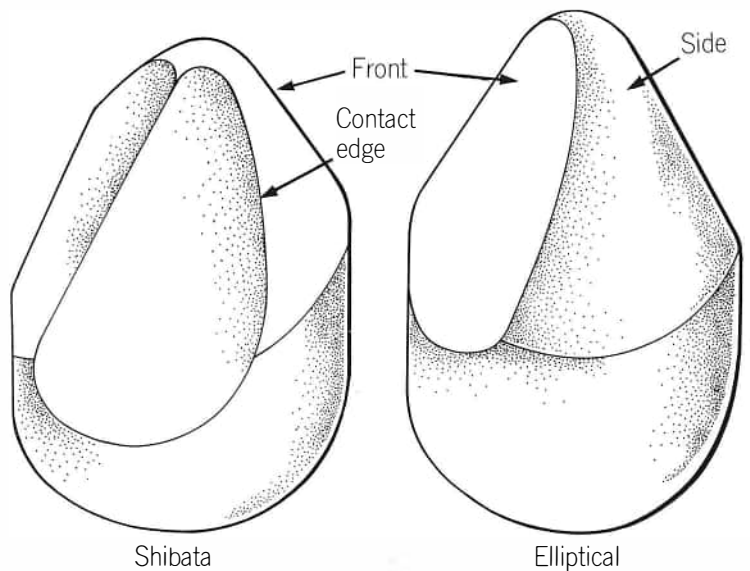


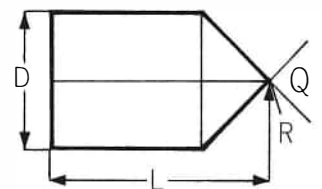
FIG. 15 SHIBATA AND ELLIPTICAL CONTACT EDGES

TYPICAL SPECIFICATION FOR SAPPHIRE AND DIAMOND STYLUS TIPS

TYPE	R	Q	D	L
SR	$\pm 0.001''$	$\pm 5^\circ$	$\pm 0.002''$	$\pm 0.002''$
LP	.0007''	55°	.016''	.050''
Std	.0010''	55°	.016''	.050''
	.0038''	55°	.016''	.060''
	($\pm 0.002''$)			

MAGNETIC STYLI (USUALLY DIAMONDS ONLY, SR or SPECIAL Std)

Spherical 1	.0007''	55°	.012''	.030''
Spherical 2	.0005''	55°	.012''	.030''
Elliptical	.0007''	55°	.012''	.030''
	.0003''	65°		
Bi-radial	.0007''	55°	.012''	.030''
	.0003''	55°		
Nude or Shibata	Dimensions as for Spherical, Elliptical - Bi-radial.		.010'' Square	.024''
	Special Shibata Shape		.008'' Square	.020''



Stylus

The stylus in a hi-fi cartridge is made either of sapphire or diamond. Diamond is preferable because of its longer service life which averages 400–800 hours of operation.

Its durability depends largely on the tracking force, i.e. the weight with which the stylus is pushed against the walls of the record groove. The tip of the stylus (Fig. 13) is machined to extreme precision. It can be spherical, with a tip radius of about 0.5 mil (1 mil = 1/1000 inch) for a stereo cartridge, or elliptical, with its greater width across the record groove. Elliptical styli usually deliver better response to high audio frequencies, i.e. they track high notes better.

The Shibata Stylus

With the introduction of “discrete” 4 channel discs it has become necessary for a cartridge and stylus to respond to frequencies as high as 45–50 kHz. To do this consistently a new stylus design was introduced – the Shibata.

The Shibata tip is shaped to conform better to the V shaped groove, and contact area between record and stylus is increased about 4 times over an elliptical stylus (Fig. 14).

Behind this long contact edge the Shibata stylus is cut away in a shallow “V” to permit a very small radius at the contact edge (Fig. 15). It is this small radius that permits proper tracing of frequencies as high as 50,000 Hz.

Another benefit of the reduced record pressure of the Shibata stylus is the raised effective stiffness of the disc, which in turn raises the resonant frequency of the record/stylus/cartridge system. This extends high frequency response and reduces peaks in the audible spectrum and beyond. This benefit can be heard with all records, especially at the inner grooves where velocity is lowest yet modulation tends to be high as the music comes to its climax.

How to Read Turntable Specifications

When comparing and evaluating turntables on the strength of their catalogue specifications, the following items deserve the greatest attention.

a) WOW AND FLUTTER

Wow is a slow cyclic speed inconsistency causing an up-and-down in the playback sound. Flutter is a high-speed quiver especially noticeable in long-held notes. Both are caused by deficiencies in the drive system and expressed as a percentage in the specifications. The smaller the figure, the better. Maximum for hi-fi turntables is around 0.2%.

b) SIGNAL-TO-NOISE RATIO

A measure for the amount of low-pitch “rumble” of a turntable in relation to the desired signal. The larger the figure, the better. Should be above 55 dB (DIN B).

c) TURNTABLE PLATTER

The shape, size and weight of the platter is determined by the type and drive system and design philosophy used. Therefore, this parameter is mainly of personal preference providing it satisfies the specification

points covered in paragraphs (a) and (b) above.

d) TONEARM TRACKING ERROR

Maximum deviation of the tracking angle from the record tangential. The smaller, the better. Should be under 2°–3°.

e) COMPLIANCE (OF THE CARTRIDGE)

A measure of the stylus' ability to follow violent undulations in the record groove. Expressed as the amount of deflection (in cm) that a force of 1 dyne upon the stylus will cause. The higher the figure, the better. Depends also upon the tracking force, should be at least 20×10^{-6} cm/dyne.

f) CARTRIDGE TRACKING FORCE

The required downward force that will make a cartridge stay in the record groove and track it. Directly related to compliance: the higher the compliance, the smaller the required tracking force. As light tracking forces save record and stylus wear, lower figures are preferable. Should not exceed 4 grams.

g) CARTRIDGE STEREO CHANNEL SEPARATION

The ability of the cartridge to separate the two channels of a stereo record. Expressed in dB, should be above 20dB.

h) CARTRIDGE OUTPUT VOLTAGE

The voltage generated by the cartridge at a certain stylus-to-groove velocity and at 1000 Hz signal. Differs according to cartridge system but should be greater than the sensitivity (in mV) of the amplifier's phono inputs.

i) CARTRIDGE FREQUENCY RESPONSE

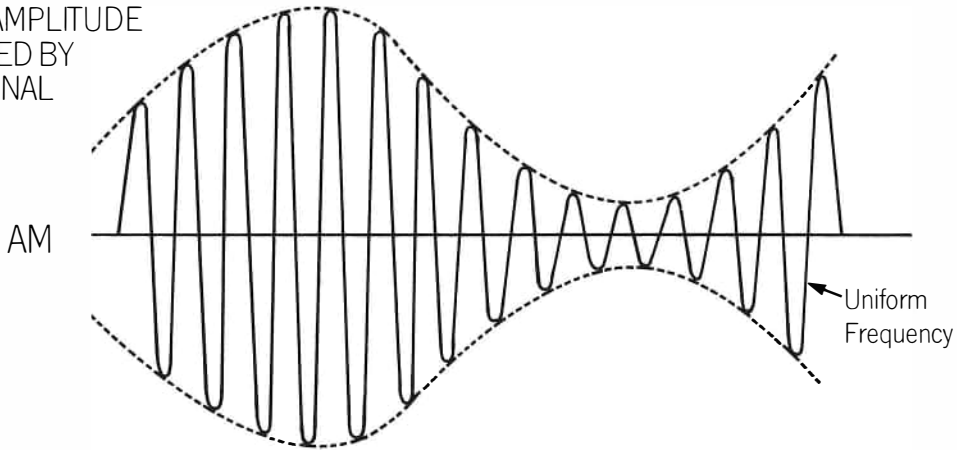
The lowest and highest audio frequencies that the cartridge will track and deliver. The wider the range, the better. Also, there should be no peaks or dips within this range. Hi-fi cartridges should have at least 20–16,000 Hz \pm 3dB.

j) CARTRIDGE LOAD IMPEDANCE

The amplifier's input impedance as seen from the cartridge. For MM and IM pickups, the standard value is about 47–50 kohm, which is what most amplifier phono inputs are designed for. MC cartridges have low load impedances (2–10 ohm), and the amplifier must have special MC phono inputs. Otherwise, a step-up transformer or booster amp will be required between MC cartridge and amplifier.

THE TUNER

CARRIER AMPLITUDE
MODULATED BY
AUDIO SIGNAL



CARRIER FREQUENCY
MODULATED BY
AUDIO SIGNAL

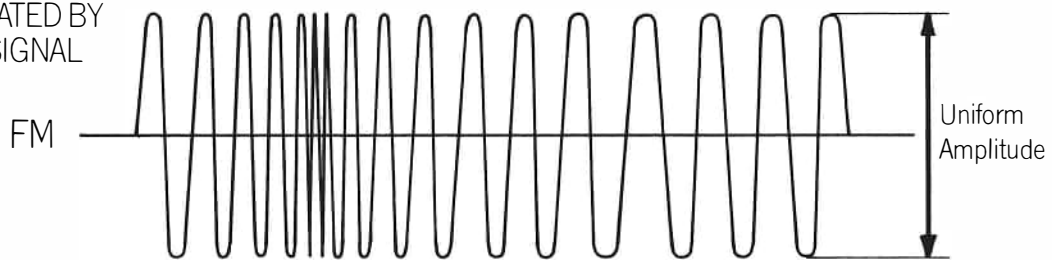


FIG. 16 THE DIFFERENCE BETWEEN AM AND FM.

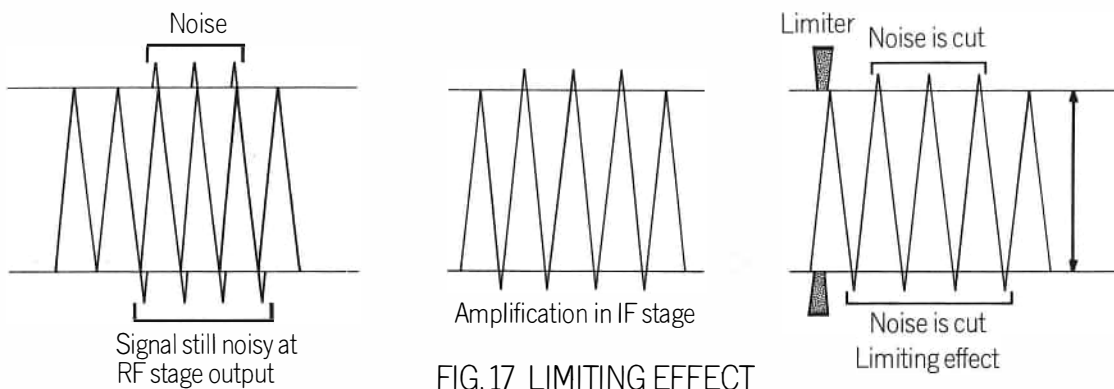


FIG. 17 LIMITING EFFECT

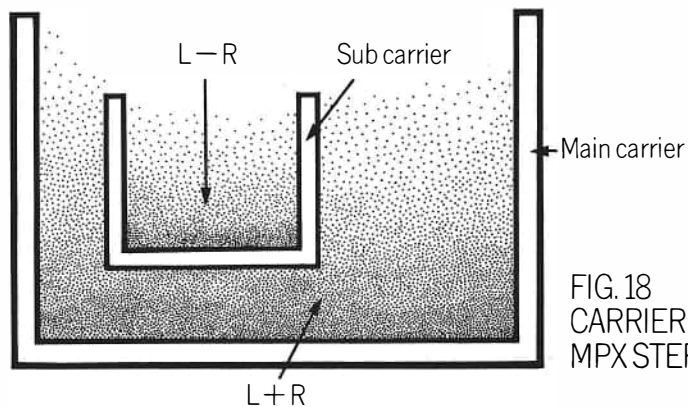


FIG. 18
CARRIER OF FM
MPX STEREO

Next to the turntable, the tuner is the most common sound source in hi-fi installations. In many cases, tuner and stereo amplifier are combined into one unit, called a “tuner amplifier,” or “receiver”. In these instances, the tuner section of the receiver should fulfil the demands outlined below.

AM (LW/MW/SW) and FM (VHF)

Most hi-fi tuners are AM/FM models which means that they receive AM broadcasts as well as FM and FM Stereo programs. As few exclusive FM models exist, but since there is hardly any difference in cost, the AM/FM tuner with its greater flexibility has advantages. (Fig. 16).

AM reception in the 535–1605 kHz radio band has been around since the earliest days of the “wireless”. Although there are built-in limitations, which relegate AM to the realm of low-fi, it doesn't hurt to have it available for news, sports programs, etc.

When choosing a hi-fi tuner, its FM – and FM stereo – performance is of incomparably greater importance. A good FM tuner will deliver the same degree of hi-fi sound purity as a turntable or tape-deck – in many cases its sound will be superior simply because FM stations use top-grade professional reproduction equipment outside the range of the hi-fi amateur. Add to this the fact that FM music programs can be tape-recorded without difficulty, and you will understand why for many budget-minded hi-fi fans the FM tuner has become the principal sound source.

Function of FM Tuner

The demands made upon a hi-fi tuner will become clear if we examine its functions step by step. The FM antenna supplies the radio signal (FM occupies the 88–108 MHz band) to the tuner's input, i.e. the RF amplifier stage. There the signal is amplified and, more importantly, separated from those of other FM stations and from random noise signals. This outlines the first two tasks of an FM hi-fi tuner: sensitivity, i.e. the ability to pick up weak signals; and selectivity, which is its ability to “slice out” only the desired station while suppressing adjacent signals and noise. Another item found in tuner specifications and determined at the tuner's RF stage is cross-modulation resistance, which describes a tuner's ability to prevent the desired signal from being influenced (“modulated”) by strong unwanted stations. In modern tuner designs, cross-modulation resistance has been improved considerably by the use of field effect transistors (FET's) because of their higher linearity. Since cross-modulation depends largely on the strength of these unwanted signals, it can be further reduced by correct, precise orientation of the antenna towards the wanted station.

As the RF stage employs an oscillator circuit, it must be designed in such a way that the oscillator frequency does not “leak” from the tuner, causing interference in nearby TV sets, radios, etc.

The RF stage delivers an intermediate frequency signal of 10.7 MHz, which is then amplified and again selected in the IF stage. The problem is to

amplify only the desired, sharply defined section of the band that contains the wanted sound information. This is done by a series of filters, of very precise electrical dimensions, preferably ceramic filters. These are a major factor in determining an FM tuner's selectivity.

The limiting circuit, also part of the IF stage, fulfils one of the most important functions – it suppresses the atmospheric noise (Fig. 17). This is one of the special features of FM broadcasting and explains the high sound quality attainable. The limiter begins to work only when a signal of sufficient strength is present, and its function is therefore closely related to the tuner's sensitivity and antenna signal strength.

The muting circuit, usually switchable with a control on the tuner's front panel, has a similar purpose – it cancels the noise encountered on unused FM channels, that irritating hiss that you hear between stations. The muting control can be either a simple on/off switch or, on expensive models, a continuously adjustable control. As the muting switch cancels out weak stations along with the noise, it must be turned off or turned down when such a weak signal is to be received.

After passing through the IF stage, the signal is demodulated, i.e. the audio signal is extracted from it. This signal is still too weak, however to drive the loudspeakers; it must first be augmented in the amplifier.

FM MPX (Stereo)

The greatest attraction of today's FM tuners, lies in the possibility of stereophonic reception. (Of course, this is possible only in areas where FM stereo stations are operating.) When broadcasting a stereophonic or multiplex (MPX) program, the station transmits a sub-carrier wave alongside the main carrier (Fig. 18) and removed from it by 38 kHz, and a pilot frequency of half the sub-carrier i.e. 19 kHz.

A pilot detector circuit in the tuner detects the presence of this pilot signal and activates the so-called FM MPX decoder circuit.

Let us call the sound information for the left channel L, that for the right channel R. In a stereophonic FM broadcast, the main carrier contains the sum of both signals, i.e. $L + R$. A conventional FM radio will receive only this signal and deliver monophonic sound. The sub-carrier transports a difference signal, $L - R$. In the FM MPX decoder, these two $L + R$ and $L - R$ signals are added and subtracted to re-gain the signal for each channel:

$$\begin{aligned}(L + R) + (L - R) &= 2L \dots \text{left channel sound} \\ (L + R) - (L - R) &= 2R \dots \text{right channel sound.}\end{aligned}$$

For reasons too complicated to explain here an FM stereo broadcast has only about half the reach of a monophonic program transmitted with the same power. This makes it easy to understand why a good FM outdoor antenna is needed in most cases to assure good stereo reception.

THE TUNER (contd)

ANTENNA CONNECTION FACILITIES AT REAR OF A RECEIVER/TUNER

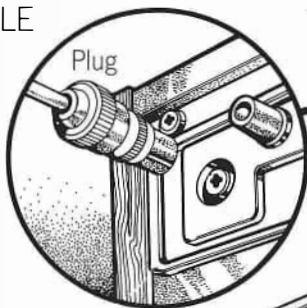
FIG. 19A OUTDOOR FM ANTENNAE

Six-element antenna. The minimum number of elements needed for good stereo reception in difficult situations.

Four-element antenna. For secondary service areas or where reception is spoilt by reflected signals.

Two-element antenna. For areas close to the transmitter.

FIG. 19B 75 OHM COAXIAL CABLE



FM 75 OHM UNBAL

FM 300 OHM BAL

FIG. 19C T-TYPE FM ANTENNA

FIG. 19E AM ANTENNA (indoor or outdoor)

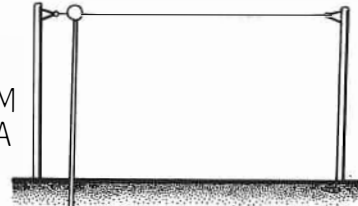
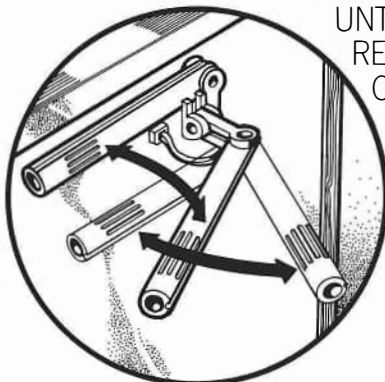
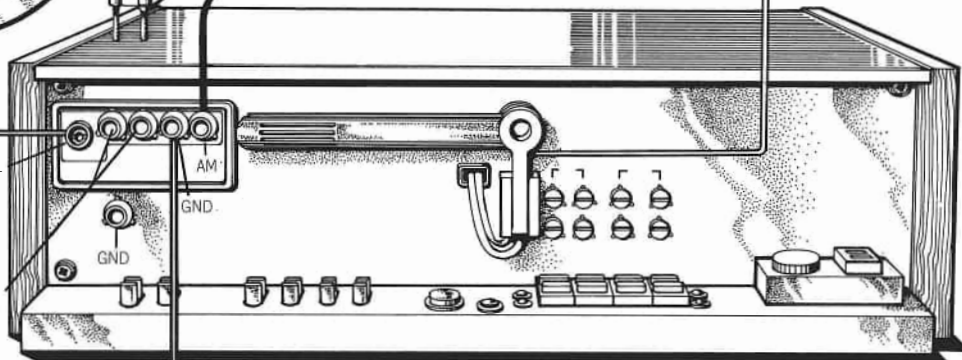


FIG. 19D TURN BAR ANTENNA UNTIL BEST RECEPTION IS OBTAINED



300 ohm feeder cable

ANTENNA



Soil

FIG. 19F GROUNDING

Antenna should face direction of radio signals.

Don't place antenna where it's subject to electrical interference from motor vehicles.



ANTENNA PLACEMENT

Antenna and Ground (Earth) Connections

FM ANTENNA

Depending on the strength and quality of the FM transmission itself, and possible interference by high ground and steel girder buildings etc., the FM signal received may be weak. You will need to set up an antenna appropriate to strength of signals received and conditions of the surroundings.

FM outdoor antenna: Normally connect the antenna feeder wire to the 300 ohm terminals of the tuner (Fig. 19A). You then install the antenna and determine the location and mounting providing optimum signal strength.

Connections using coaxial cable: It is preferable to use coaxial cable of 75 ohm impedance to connect the antenna to the tuner (Fig. 19B). Particularly if you live near heavy traffic, factories or high tension power lines.

Indoor antenna: When you receive strong FM signals (in wooden frame buildings or near transmitters) a T-type antenna can be used. This is connected to the 300 ohm terminals of the tuner (Fig. 19C). Once you've determined which orientation provides the best signal, the antenna should be secured in the determined direction to a wall or ceiling.

AM ANTENNA

A ferrite bar antenna is usually provided on the rear panel of the tuner (Fig. 19D). You simply move the ferrite bar until it provides the best reception. This type is less susceptible to external interference.

If reception with the ferrite bar antenna is poor, a vinyl insulated wire antenna may be connected at the tuner's AM antenna terminal.

AM indoor antenna: A single strand vinyl insulated wire 6-8m long should be connected to the tuner's AM antenna terminal, the other end being suspended at an elevated point, e.g. ceiling.

AM outdoor antenna: If reception is still poor with use of the indoor antenna, an outdoor antenna may be erected using single strand vinyl insulated wire (Fig. 19E).

GROUNDING (Earthing)

While reception is possible without use of grounding, for the sake of safety and reduced electrical noise it's advisable to ground the tuner by means of the GND terminal (Fig. 19F).

How to read Tuner Specifications

Compared to a turntable, a tuner's spec sheet presents greater difficulties to the layman. Indeed, a close familiarity with electronic circuits is ideally required. Nevertheless, the following brief explanations should be sufficient.

a) SENSITIVITY

The minimum antenna signal which the tuner can convert into a satisfactory sound signal. Expressed in microvolts (1/1,000,000 of a volt) and defined by the Institute of High Fidelity Manufacturers (IHF) as the signal strength at which the tuner will suppress noise by 30 dB. Sensitivity ratings of 3 microvolts are respectable, values below 2 microvolts, outstanding. Don't trust

figures which do not include the noise limiting ratio (expressed in dB, usually 30 dB).

b) SIGNAL-TO-NOISE RATIO

The ratio, expressed in dB, between a 400 Hz fully modulated signal and the noise component. 60 dB means that the signal is 1000 times stronger than the noise. The higher the value, the better. 50 dB is about minimum requirement for hi-fi.

c) CAPTURE RATIO

A tuner's ability to suppress the weaker of two stations on the same radio frequency, and to receive only the stronger. Also related to the tuner's suppression of random noise. Expressed in dB, smaller values are better. 4.5 dB is usually sufficient.

d) SELECTIVITY

The tuner's ability to slice only the desired station from all the radio waves reaching it from the antenna. Also its ability to receive one of two closely spaced stations. Expressed in dB, higher values are better. 50 dB.

e) IMAGE REJECTION

"Image" is a tuner's undesirable reception of the same signal at two or more points on the dial, of which only one is the true station signal. It's caused by interaction of the oscillator frequency and the RF frequency. Image rejection is the tuner's ability to suppress this frequency. Bigger values are better, giving less problems in areas with many FM stations broadcasting.

f) SPURIOUS RESPONSE OR REJECTION

An FM tuner can create unwanted signals which it then (a) emits through the antenna, causing interference in the neighbour's TV or radio, and (b) picks up itself, affecting its own reception because this self-generated unwanted signal can interfere with the true wanted station signal. Expressed in dB, higher values are preferable.

g) AM SUPPRESSION

Nothing to do with AM broadcasting. AM here means the amplitude modulated noise signals from fluorescent lamps, motors, car ignitions, etc., which cause irritating crackle during FM reception. AM suppression is expressed in dB, higher values are better, 40 dB is about minimum.

h) FM STEREO SEPARATION

The FM MPX decoder's ability to separate the left and right channel signals of FM stereo broadcasts. For stereo effect, channel separation in the medium audio frequency range (400-1,000 Hz) is most important. A good tuner should have 40 dB separation in this range, or 30 dB between 40 Hz and 8 kHz, or 25 dB over the total 20 Hz-15 kHz range. Beware of specifications that don't mention the frequency range.

i) OUTPUT LEVEL

Usually in the 0.5V range, adjustable on high quality FM tuners. Must be suitable with the sensitivity of the amplifier's TUNER or AUX inputs.

TAPE EQUIPMENT

TAPE AND TAPE RECORDER MECHANISMS

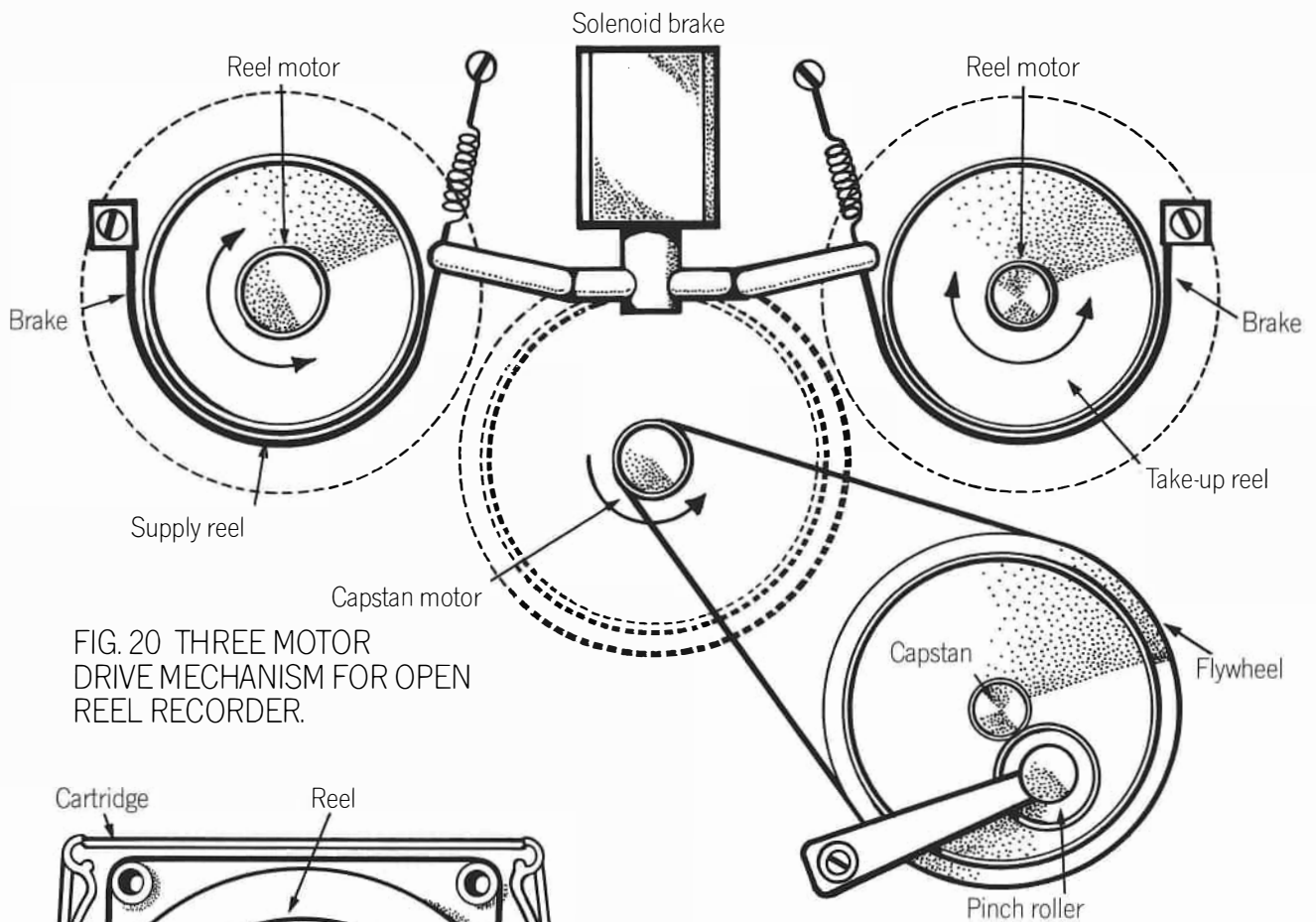


FIG. 20 THREE MOTOR DRIVE MECHANISM FOR OPEN REEL RECORDER.

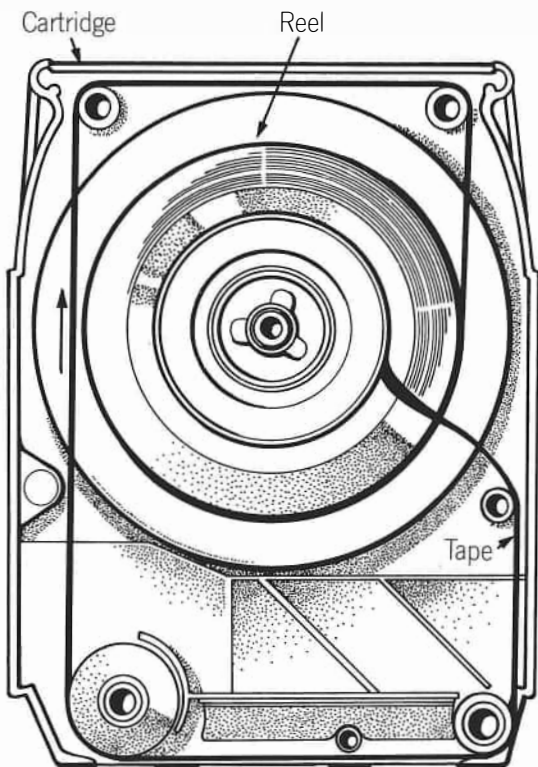


FIG. 22 ENDLESS LOOP TYPE TAPE CARTRIDGE

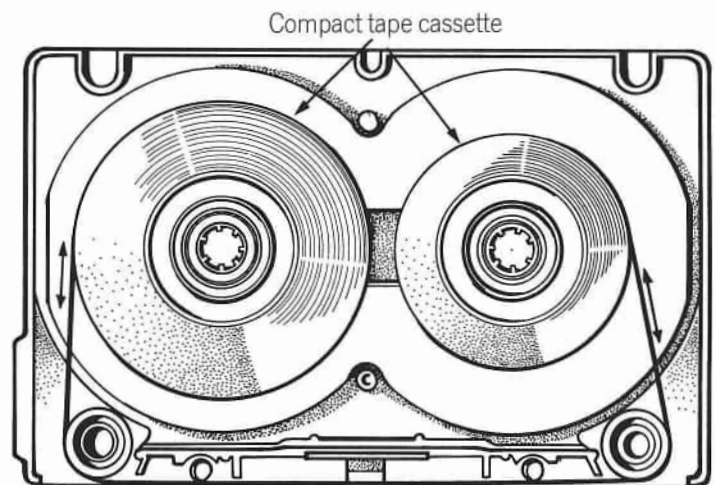


FIG. 21 SCHEMATIC VIEW OF CASSETTE TAPE

The turntable and tuner are “passive” sound sources—they can only reproduce commercially recorded or broadcast sound. Tape equipment, with the exception of so-called tape players, lets the listener participate “actively” in the selection and even creation of music programmes. This great advantage, together with the operational ease and relatively low price of modern tape equipment may explain the tremendous tape recorder boom that has been sweeping around the world.

According to the type of magnetic tape used, tape equipment can be grouped into open-reel, cassette and cartridge. The recording tape itself is basically the same for all three types, and so are the essential functions of the machines (Figs. 20, 21, 22).

Magnetic recording tape consists of a thin (15 to 35 micron) polyester or acetate film with a layer of fine, crystalline iron oxide particles imbedded in an adhesive. These iron oxide particles can be magnetized in a magnetic field and will retain this magnetism.

Recording and Playback Process

The recording process is briefly, as follows. Electric signals supplied by a sound source are amplified and then applied to the coil of a small electro-magnet called a “recording head”. This causes a magnetic field, alternating in accordance with the sound’s rhythm and intensity, in and around a tiny gap between the poles of the magnet. As the recording tape travels past the recording head, its iron oxide particles are magnetized by the alternating magnetic field. The tape thus stores the sound information in the form of magnetism (Fig. 23).

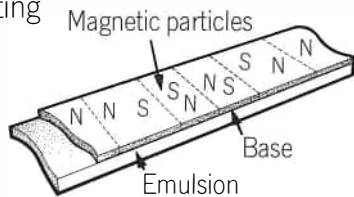


FIG. 23 SCHEMATIC VIEW OF RECORDED TAPE

To retrieve this information, the reverse process is used. The tape transported past the so-called playback head—another small electro-magnet with a similar structure to the recording head—induces a small, alternating current in the magnetic coil. As this current contains the same sound information as the original signal—provided that the tape travels at the same speed at recording and playback—it can be amplified and used to drive a loudspeaker.

Tracks

The first tape recorders, built during and after World War II, used up the full tape width for a single passage. Later, as tape recorder components, recording tape and techniques became more refined, the tape width was split into two and then four “tracks”. This provides greater tape economy and, moreover, the possibility to store two (or even four) stereo channels on

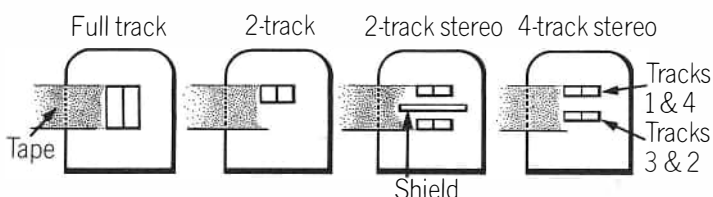


FIG. 24 TRACK LOCATIONS IN TAPE HEADS

the tape. Most modern open reel and cassette stereo tape recorders for amateur use are usually 2-track or 4-track models, whereas in cartridge recorders and players the same tape width is divided into eight tracks, i.e. four stereo channels (Fig. 24).

Tape Speeds

Another tendency in tape recorder design has brought about slower and slower tape speeds. Although professional studio machines transport the tape at 15 or at least 7½ inches per second (ips), most amateur recorders offer a choice between 7½ and 3¾ ips, and sometimes 1⅞ ips. (Corresponding decimal values: 19, 9.5 and 4.75 cm/sec). Cartridge players always run at 3¾ ips (9.5 cm/sec) while cassette tape is transported at 1⅞ ips (4.75 cm/sec). This explains the comparatively long recording time of cassette tapes.

Tape speed is of vital importance for the sound quality, especially in the high sound range, because a tape recorder’s upper frequency response limit is directly and mathematically related to the tape speed and the gap width of its recording and playback heads. The higher the tape speed, and the narrower the head gap, the better the high-range frequency response. Most hi-fi fans with open-reel tape recorders therefore prefer the highest available speed for recordings they consider important. Recent advances in the technology of recording heads have made it possible to manufacture ultra-narrow head gaps so that excellent frequency response up to the limits of the audible sound range can now be obtained even at the slower tape speeds (Ferrite head, etc.).

Tape speed is of course directly connected with recording time. The table shows recording times (per track and per return trip) for common types of reel tapes at 7½ ips (19 cm/sec) recording speed. 3¾ ips provides twice the recording time, 1⅞ ips four times as much.

TYPE	BASE	THICKNESS (μ)	LENGTH (m) (ft.)	RECORDING TIME (min)	
				ONE WAY	BOTH WAYS
Standard 100	Acetate	52	370 1200	32	64
”	Polyester	”	” ”	”	”
Low noise high output 100	Acetate	”	” ”	”	”
”	Polyester	”	” ”	”	”
Long play 150	Acetate	35	550 1800	48	96
”	Polyester	”	” ”	”	”
Double play	”	25	740 2400	64	128
Triple play	”	20	1100 3600	96	192

TAPE EQUIPMENT (contd)

Tape Quality

The quality of the recording tape is equally as important as that of the equipment itself. The poor quality tape wears out your tape recorder faster than necessary. Some of the demands on quality tape are obvious: it must not break or stretch easily, its edges must not curl, and its base must retain its strength and flexibility for many years. Also, the emulsion must not separate from the base or wear off, because this fouls up the heads and tape path and causes "drop-out" i.e. holes in sound. As the tape constantly rubs against the recording and playback heads, its surface should be as smooth and even as possible to assure long head service life. In addition, there are numerous requirements regarding the tape's electrical characteristics – its dynamic range, frequency response and freedom from noise.

Low-Noise High-Output Ferric Tapes

These so called "low noise high output" tapes, available on open reels and as compact cassettes, possess higher magnetic particle density and therefore a wider dynamic range – they can record and reproduce a greater span of sound intensities from a whisper to a roar – and are used with standard bias and equaliser settings.

TYPE	BASE	THICKNESS (μ)	LENGTH (m) (ft.)	RECORDING TIME (min.)	
				ONE WAY	BOTH WAYS
C-30	Polyester	18	45 150	15	30
C-45	"	18	68 223	22.5	45
C-60	"	18	90 300	30	60
C-90	"	13	135 450	45	90
C-120	"	9	180 600	60	120

Chromium Dioxide Tape

This tape uses chromium dioxide particles instead of the conventional iron oxide ones and is distinguished from other tapes by its very high output as well as its excellent frequency response and wide dynamic range.

Ferrichrome Tape

Ferrichrome tape is a further development in tape technology. Here, a layer of Ferric-Oxide is covered (or blended) with Chromium Dioxide. This gives a much improved response in the midrange, plus the high frequency response of Chromium Dioxide.

Special Tapes: Super Ferric

Recent advances in tape technology have now produced a new generation of Super Ferric Tapes which can provide performance similar to Chrome and Ferrichrome tapes. Bias and equaliser settings used are the same as for Chrome tapes.

Audio frequency signals are never recorded on magnetic tape as it is: the tape develops its optimum characteristics only after it has been "printed" with a rather high frequency bias current, usually applied through the recording head together with the audio signal. To get the most from these tapes, your tape deck should allow various bias and equaliser settings for Chromium, Ferrichrome tapes as well as LH tapes.

Chrome, Ferrichrome and Super Ferric tape technology has given cassette stereo increased performance, which permits the user to reach sound quality quite similar to many expensive reel to reel tape decks.

Noise Reduction Systems

There are several types of noise reduction systems in use with reel to reel and cassette tape decks, such as Dolby, A.N.R.S., D.N.L., D.B.X. etc. The most widely used at the present time is the Dolby system.

The Dolby System

Dolby system is a unique noise reduction system, which electronically eliminates the irritating noise (tape hiss, circuit noise, etc.) without sacrificing the original tonal quality (Fig. 25).

- 1) Noise level is not disturbing (Signal-Fortissimo)
- 2) Noise level is disturbing (Signal-Pianissimo)
- 3) Fortissimo signals are recorded naturally
- 4) Pianissimo signals are up-levelled and recorded
- 5) Pianissimo signals are down-levelled and played-back
- 6) Since noise level is down-levelled by same degree, it is not disturbing

'Dolby' is a trademark of Dolby Laboratories Inc.

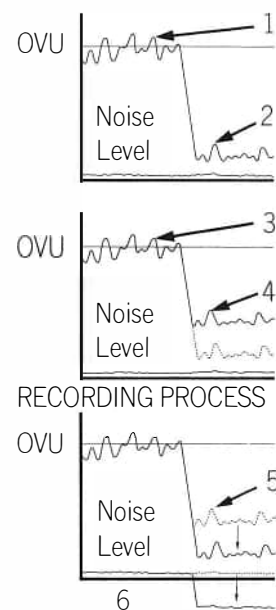


FIG. 25 THE DOLBY SYSTEM

The Care of the Tape

The rules for the care of tapes are much the same as for records, and really come down to no more than common sense. Dust is one of the chief enemies. It can be just as abrasive to tapes as it is to LPs. So always keep cassettes and tapes in their plastic library cases, and after use put cartridges back in their sleeves or replace the plastic clip-on shields which protect the exposed tape.

For storing tapes at home, nothing could be better than ordinary bookshelves, but, as with LPs, beware any heat much above normal room temperature.

For storage of cassettes and cartridges in cars, it is a very good investment to buy one of the numerous carrying cases made to hold a number of cassettes or cartridges, and to keep it in a cool place in the car. Above all else, do not put tapes, in whatever form, on the shelf behind the rear seats where the sun can stream in and in time ruin them. Even the glove compartment can be too warm, especially if it is enclosed.

One more word about playing tapes outside the home. There are few things more calculated to damage them and their playing equipment than sand and salt, so if you must take tapes onto the beach, you have been warned.

A less obvious hazard is the risk of accidental erasure from magnetic fields. The most likely risk in the average household is that created by the magnet of a loudspeaker. However convenient it may be to put a cassette or cartridge on top of such a speaker for a moment – DON'T. It will not damage the tape as such but may well cause partial erasure of its contents or induce increased background hiss, as also may putting the telephone on top of the tape.

Finally, if after a time the quality of sound falls below the standard you expect, do not be too ready to blame the tape, or the equipment for that matter. A certain amount of the recording oxide material with which tapes are coated inevitably rubs off in use, and any accumulation of this on the tape heads impairs the quality of the sound; and a substantial accumulation can cause actual damage.

Regular use of a cleaning tape both for cassette and cartridge players will ensure consistent performance. Cleaning tapes consist of a length of suitably impregnated ribbon made up in cassette or cartridge form. This can be run through the player just like a recorded tape but instead of producing sound it cleans the heads of the deposit. Occasionally, more elaborate cleaning may be required. Kits for this purpose are available but when using them take care to follow the instructions strictly, and use only the cleaning fluid provided.

How to Read Tape Deck Specifications

a) WOW AND FLUTTER

Inconsistencies in the tape speed which can cause pitch variations and quivering sound. Expressed in percentages, with different values for each tape speed. Look for the lowest values. 0.25% is about the permissible maximum.

b) SIGNAL-TO-NOISE RATIO

Denotes the tape deck's freedom from self-created noise. Expressed in dB, higher values are better, 50 dB is about minimum.

c) FREQUENCY RESPONSE

The lowest and highest audio frequencies that the tape deck will record and reproduce. Should be as flat as possible and cover at least 50 – 13,000 Hz at 7½ ips or 50 – 10,000 Hz at 3¾ ips.

Parameter	Excellent	Good	Fair	Unit	
Signal to Noise	54-56	50-53	45-50	dB	
Frequency response	7½	30-16,000	50-13,000	50-10,000	Hz
	3¾	50-12,000	50-10,000	50-8,000	Hz
	1⅞	50-8,000	50-7,000	50-4,000	Hz

d) CROSSTALK

The amount of signal that leaks (i) from one stereo channel into the other and (ii) from one tape track into another. Expressed in dB, the higher the value, the better. Channel-to-channel crosstalk of a 4-track machine should be above 30 dB.

e) CHANNEL SEPARATION

Another word for channel-to-channel crosstalk (above).

f) MAXIMUM REEL DIAMETER

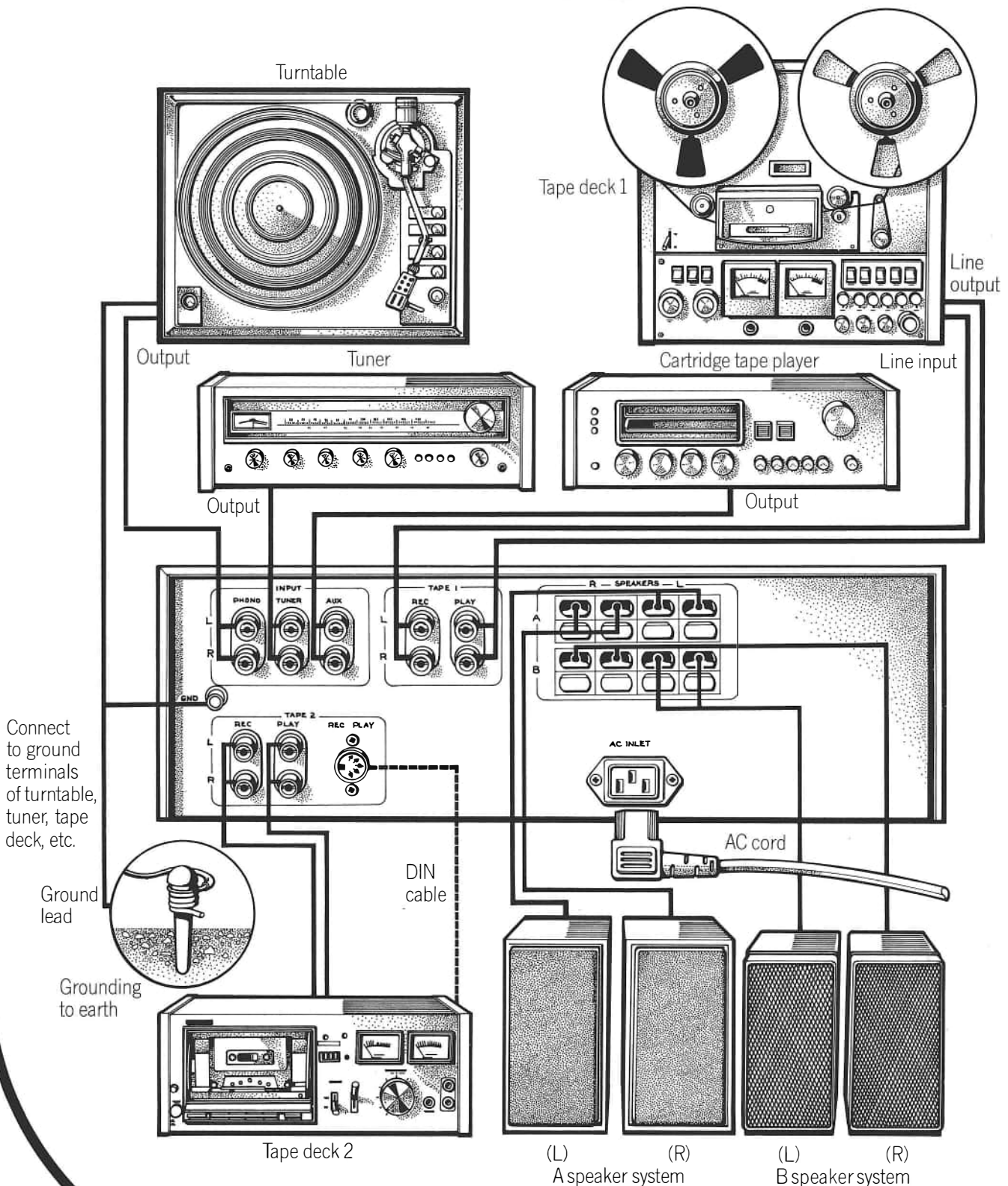
The largest reel size that the tape deck will accept. Mostly 7 inches, a few semi-professional models accept 10 inch reels.

g) OUTPUT LEVEL

Must be suitable with the sensitivity of the amplifier's tape inputs.

THE AMPLIFIER

FIG. 26 TYPICAL CONNECTION FACILITIES AT REAR OF AN INTEGRATED AMPLIFIER



As the audio signals supplied by the turntable, tuner or tape deck are much too weak to be converted into audible sound directly, they must first be raised to a higher level. This is done by the amplifier which thereby becomes the heart of every hi-fi installation. Before we examine the fundamentals, functions and features of hi-fi amplifiers, we must first clarify a few distinctions in terminology.

Pre-Amplifier, Power Amplifier, Integrated Amplifier, Receiver

An amplifier system in a hi-fi installation consists of two parts: the pre-amplifier which includes all the controls for regulating tone, volume and channel balance; and the power amplifier which, as its name implies supplies a sufficiently powerful signal to drive the loudspeakers.

Manufacturers provide the audio enthusiast with a multiple choice: he can purchase separate pre-amplifier and power amplifier units, he can select a combination of both (Fig. 26) which is commonly called "integrated amplifier", "control amplifier" or just simply "amplifier", or he can opt for a receiver, i.e. a combination of tuner, pre- and power amplifier (Fig. 27).

Since units built to very high performance standards can be found in each group, the question is not so much one of sound quality but rather a question of flexibility versus economy and convenience.

It stands to reason that a combination of two or three functions in one unit should be more economical than separate sets – just think of the cost of chassis and enclosures and power supply units (i.e. the transformer-rectifier assembly in each unit which supplies the needed operating current to the circuits). What's more, an integrated amplifier or receiver saves space and eliminates the need for – sometimes troublesome – cable connections.

On the other hand, the purchaser of a separate tuner plus pre-amplifier plus power amplifier system may gain a little bit in performance and may obtain a more individually "selected" kind of sound. He also gains in flexibility... he can easily add an electronic crossover network and one or two more power amplifiers to upgrade his system into a multi-amplifier installation... he can move on to a four-channel system without sacrificing any of his present equipment... and he gets, last but not least, the pride of owning his own custom-assembled personalized system.

Some integrated amplifiers have the possibility to use the pre-amplifier and power amplifier stages separately. This is done by removing a short cable connection on the back panel and can permit a system to be enlarged into a multi-amplifier configuration at a later date.

The following explanations deal with pre-amplifier and power amplifier functions, in that order,

but they apply to integrated amplifiers and receivers, too; only the remarks about connections and matching of components may be disregarded.

Pre-amplifier Functions

To the hi-fi novice, the need for and functions of the pre-amplifier are usually more difficult to understand than those of the power amplifier which, after all, clearly produces power. The pre-amplifier can be thought of as a sort of super switchboard – it selects the sound source to be played, it determines the mode of sound (stereo or monophonic), it influences the sound colour with its tone controls, it regulates the volume and the channel balance, and it brings the signals up to voltage levels which the power amplifier can then work with. The different sound sources are connected to the pre-amplifier inputs: the turntable to the phono inputs, the tuner to the tuner inputs, the tape deck to the tape (or sometimes "monitor") inputs, and an auxiliary source such as a cassette tape deck to the AUX inputs. Some amplifiers also have microphone inputs.

Phono Equalization

Let us observe the phono signal on its way through the pre-amplifier. First it enters a circuit called an equalizer amp. To understand its function, we must be aware of a peculiarity of phonograph records. When the master disc is cut, the high sound ranges are over-emphasized ("boosted") while the bass range is somewhat suppressed. This must be done in order to obtain a record groove with sufficient modulations in the high range for the stylus to "feel" and, on the other hand, without excessively wild bass undulations (because these would require a wider groove). Technically, cutting of a disc is not "flat" but follows a frequency response curve as shown in Fig. 28 called the RIAA curve. Therefore, the audio signal delivered by the pickup cartridge would result in equally disfigured sound if it were passed on as it is. The RIAA curve must be "equalized" first to obtain natural sound – and this is the function of the equalizer amp (Fig. 29 overleaf).

Inputs

Because the phono inputs have to be equalized, they are called "non-linear" inputs as opposed to "linear inputs" such as those for tape, tuner, microphone or auxiliary signals. Some high grade amplifiers provide two or even three pairs of phono inputs, permitting two or more turntables or cartridges to be connected simultaneously for easy comparison. Also, the

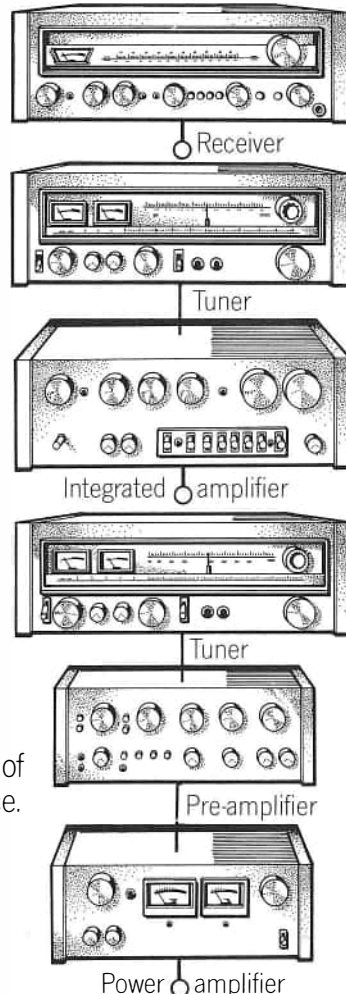


FIG. 27 CLASSIFICATION OF AMPLIFIERS

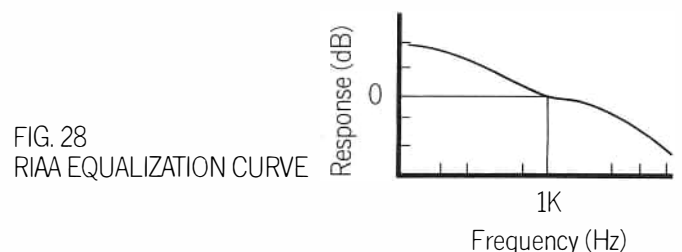
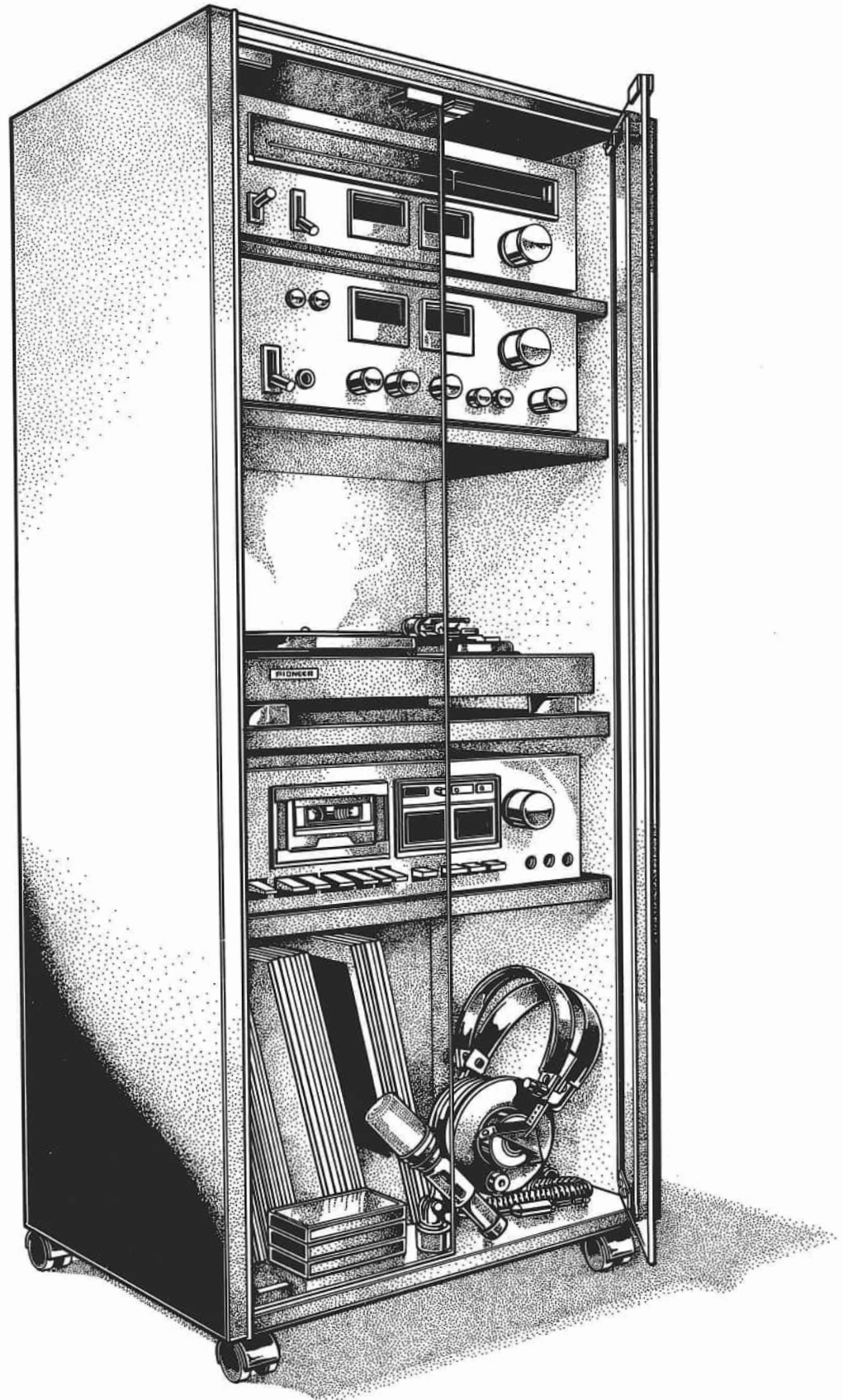


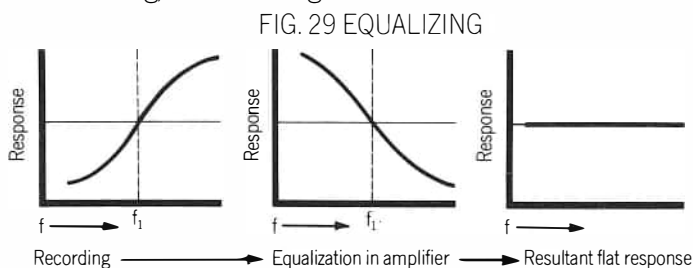
FIG. 28 RIAA EQUALIZATION CURVE

THE AMPLIFIER (contd)

A TYPICAL DOMESTIC HI-FI RACK SYSTEM



input sensitivity and impedance of the phono inputs is sometimes switchable to accommodate cartridges of different characteristics. In this context, another value often found in pre-amplifier specifications is of importance: maximum input voltage of the phono units. This describes the highest signal level that the amplifier can accept through the phono inputs without overloading, i.e. distorting.



It should be at least 10 times higher than the output level of the pickup cartridge.

Tape deck and tuner connections usually do not pose much difficulty because the electrical characteristics are pretty much standardized. With microphones, however, it is important to choose the right type. Also, microphones must correspond in their impedance to the amp's mic input impedance, within a tolerance of about ± 20 per cent.

Tape Monitoring

Switching between the different connected sound sources is usually done with a program selector, mostly a rotary knob with five or more positions. Tape sound, however, is switched on and off with a so-called tape monitor switch with two positions: source and play (or tape). For tape playback, this switch is set in position "play," for all other program sources it must be in position "source." When making a recording on a tape deck with independent recording and playback heads (a 3-head deck), this switch provides two ways of monitoring the recording quality. In position "source," the original before-tape sound is heard from the loudspeakers; in position "play," however, the already recorded after-tape sound is reproduced. By switching back and forth between these positions, the source sound can be easily compared with the taped sound and readjustments made if required.

The Tone Controls

It is usually not difficult to tell the hi-fi greenhorn from the old pro – just watch him operate the tone controls of his amplifier. The beginner will show great delight in his ability to turn up the bass and suppress the treble or vice versa, he will play around endlessly with these knobs and will usually wind up with a setting of heavy, boomy bass accentuation. The more advanced hi-fi listener, however, will use the tone controls sparingly and near the neutral level, except for test purposes. The true purpose of the bass and treble controls on a hi-fi amplifier lies indeed not in the dictatorial power play of the novice, but in the ability to compensate for certain weaknesses in the program material, the listening room, and, to some degree,

the loudspeakers. With good, new LP records or tapes played on good equipment in a normal family living room, however, extreme tone control settings are almost never required.

How do tone controls – and their cousins, the high and low filters and the loudness switch provided on most amplifiers – work?

In its simplest, bargain basement form – as found on portable radio etc. – a tone control is simply a combination of a capacitor with a resistance, and it cuts off the high parts of the sound spectrum. More suitable to hi-fi applications is the passive CR type which boosts or reduces frequencies above and below a "turnover" point in a way as illustrated in Fig. 30. For even more precise, active control, negative feedback techniques (NFB) have to be introduced which require rather elaborate circuits with transistors etc. The obtainable frequency response characteristics are illustrated in Fig. 31.

On some amplifiers, the tone controls work on the left and right stereo channel simultaneously, which is quite sufficient for most home applications; on others, there are separate knobs (usually concentric double knobs) for controlling the left and right channel separately, which is useful in difficult listening room conditions.

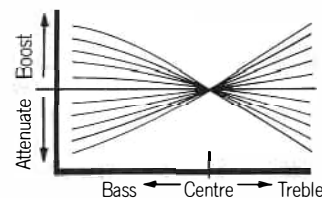


FIG. 30 CR TYPE TONE CONTROL

Filters, Loudness Contour

Low and high filters (also referred to as rumble or bass filters and noise or scratch filters, respectively) have functions as shown in Fig. 32 overleaf. The bass filter is designed to cut off (or, more precisely, to attenuate by so many decibels) the sound spectrum below a certain point, usually somewhere between 100 and 50 Hz. This has the effect of cancelling out very low hum and rumble generated by turntables. The high filter, on the other hand, attenuates frequencies above 8,000 or 10,000 Hz, as the case may be, to reduce scratching and hissing noise from records, tapes, FM, etc. It is bad practice, however, to use these filters

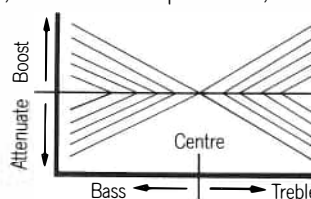


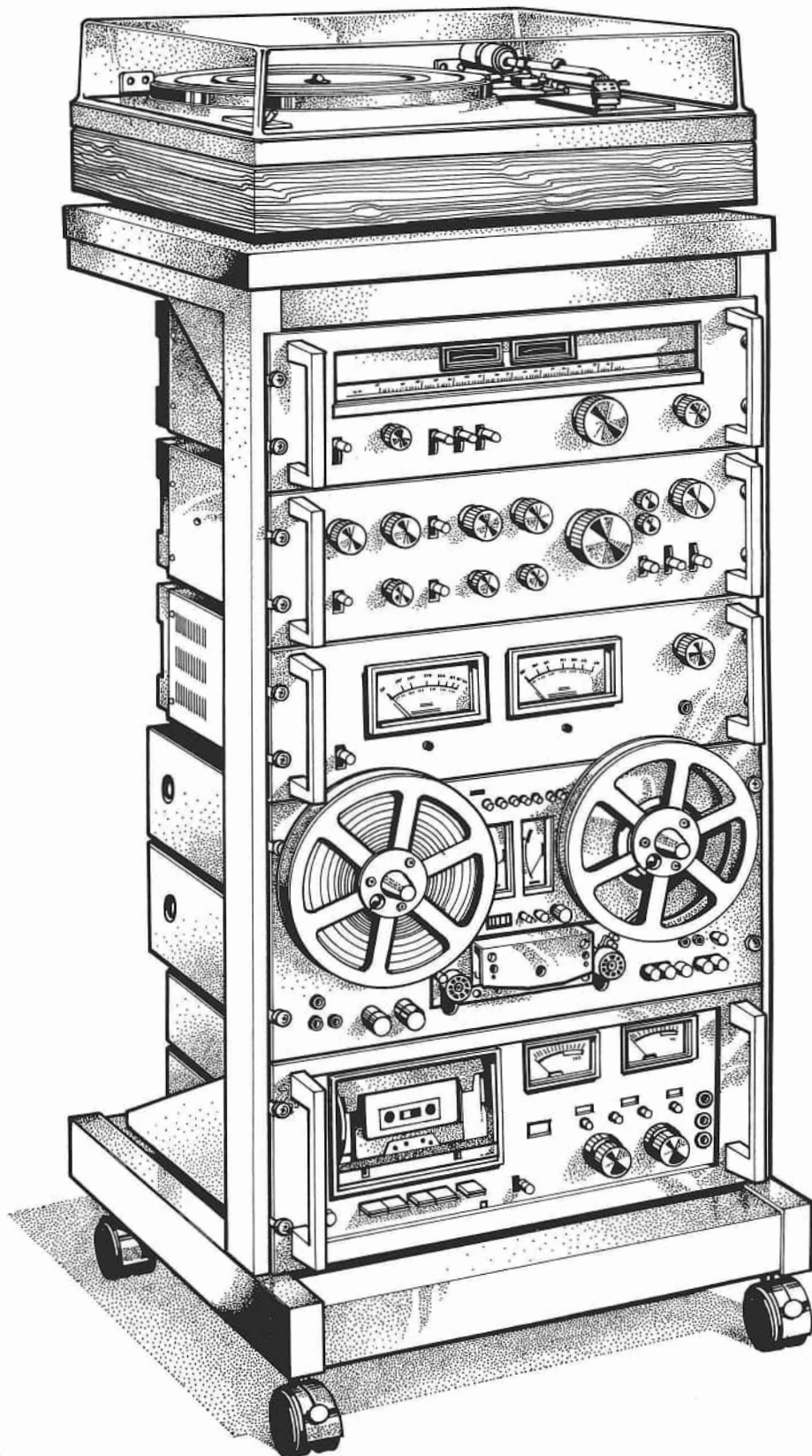
FIG. 31 NFB TYPE TONE CONTROL

indiscriminately, because together with the noise they also cancel part of the desirable musical spectrum. Use them only when necessary.

The loudness switch on an amplifier must not be confused with the volume control. A more accurate description of its function is found in the name "physiologically correct loudness contour compensation." This name already hints at a deficiency of the human sense of hearing. Indeed, measurements demonstrate clearly that the human hearing apparatus is, even at its healthy best, a far from perfect hi-fi device. One of its short-comings is that it is rather insensitive to extremely

THE AMPLIFIER (contd)

A TYPICAL AUDIOPHILE'S SEMI-PROFESSIONAL
HI-FI RACK SYSTEM.



low and very high tones when the overall volume is soft. The loudness switch (Fig. 33), a special kind of tone control, overcomes this human error by boosting these extreme ranges at low volume settings. It should therefore be turned off at high listening volumes, although many amplifiers automatically override this control at these levels.

Power Amplifier Function

From the pre-amplifier, the signal enters the power (or main) amplifier. There, its power is raised several hundred times to a value which drives the loudspeakers.

Much design-effort has gone into the development of better and better hi-fi power amplifiers. Even the vacuum tube types of years ago – only very few are now on the amateur hi-fi market – produced some excellent results. To this day, hi-fi enthusiasts brought up in that age swear by the vacuum tube amplifier, and in terms of sound quality it does possess a certain mellow charm. Its bad habits – vacuum tube amplifiers guzzle electricity, create tropical heat waves in their vicinity and break down or burn out at all too frequent intervals – made it easy for solid state designs to take over their role. Today, the transistor power amp has emerged as the almost undisputed victor.

Protector Circuits

The expensive output stage of a transistor amplifier can be easily damaged by a short circuit in the speaker leads. To prevent this, designers have invented a variety of protector circuits and combinations thereof which shut down the output stage at the first sign of trouble. These can be simple heat fuses or, more elaborately, circuits which detect a drop in output load impedance (in other words, a short circuit). A few top quality amplifiers have protector circuits which detect the presence of DC current at the output stage. As DC current could also lead to loudspeaker damage, these protection circuits safeguard not only the amplifier itself but also the loudspeakers.

How to read Amplifier Specifications

a) CONTINUOUS POWER OUTPUT (RMS)

The maximum power (in Watts) that the amplifier will deliver from each channel (with both channels operating) without exceeding its rated harmonic distortion (T.H.D.). Power ratings without clarification of the harmonic distortion are meaningless. Also, the load impedance (4,8 or 16 ohms) enters into the picture. Responsible manufacturers list output powers as in the following example: Continuous power output is 80 watts per channel min. RMS, at 8 ohms or 100 watts per channel at 4 ohms from 20 to 20,000 Hertz with no more than 0.5% total harmonic distortion.

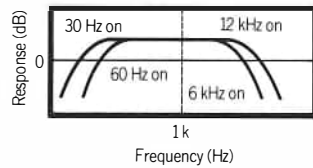


FIG. 32

FILTER CHARACTERISTICS

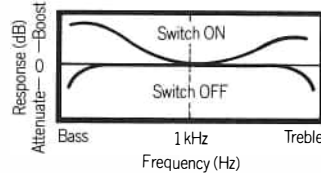


FIG. 33

FUNCTION OF LOUDNESS SWITCH

b) FREQUENCY RESPONSE

The lowest and highest audio frequencies that the amplifier will reproduce from input signals supplied to its AUX input and at one watt output power. To be meaningful, the frequency response rating must be accompanied by a \pm dB figure which indicates by how much the response varies from the standard level (0 dB at 1 kHz) over the given frequency range. Smaller dB values are better – the same amp may have a frequency response of 20 – 18,000 Hz \pm 1 dB or 18 – 25,000 Hz \pm 3 dB. Example of an outstanding amplifier rating: Frequency... 15 – 40,000 Hz \pm 1 dB response (through AUX inputs).

c) POWER BANDWIDTH

An important value that puts output power and frequency response in a relation. Power bandwidth signifies the audio frequency range (in Hz) over which the amplifier will deliver at least half of its rated output power without exceeding its rated harmonic distortion. For example, if the continuous output power of an amp is listed as 20W per channel at 1% harmonic distortion, then a power bandwidth rating of 20 – 30,000 Hz means that the amp will deliver 10 watts at 20 Hz and at 30,000 Hz without distorting more than 1% (see Fig. 34).

d) SIGNAL-TO-NOISE RATIO

The ratio between (wanted) signal and (unwanted) noise delivered by the amplifier at maximum volume setting and with its treble and bass tone controls set at flat position. The higher the value, the less hum and noise there is. Signal-to-noise ratios should be listed individually for each input; for instance:

PHONO (MAG) 65 dB, AUX 70 dB. A 65 dB rating means that only 1/2000 of the signal will be noise, 75 dB means 1/6000.

e) DAMPING FACTOR

Damping factor is a measure of an amplifier's ability to damp, i.e. control unwanted residual speaker movements caused by forces other than the audio signal. High values are better, but anything over 20 is not really very meaningful.

f) INPUT LEVEL, INPUT IMPEDANCE

Input level: the minimum level of the input signal supplied to the power amplifier from which it will deliver its full output power. Should be around 0.1V – 1V, must be close to the pre-amplifier's output level. (Only important if separate pre- and power amps are used.)

Input impedance: The power amplifier's input impedance as seen from the pre-amplifier output. Should be at least 50 k-ohms when using a modern transistorized pre-amplifier.

g) RESIDUAL NOISE

Found in power amplifier specifications. Denotes the amount of noise constantly generated by the amplifier itself regardless of its volume or gain control setting. Look for the lower values – a good hi-fi amplifier should not exceed 5 millivolts or 3 microvolts.

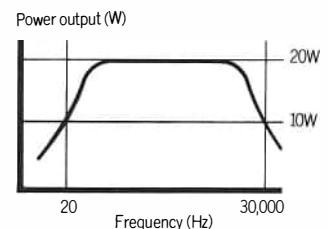


FIG. 34 POWER BANDWIDTH

LOUDSPEAKERS

LOUDSPEAKER DRIVE UNITS

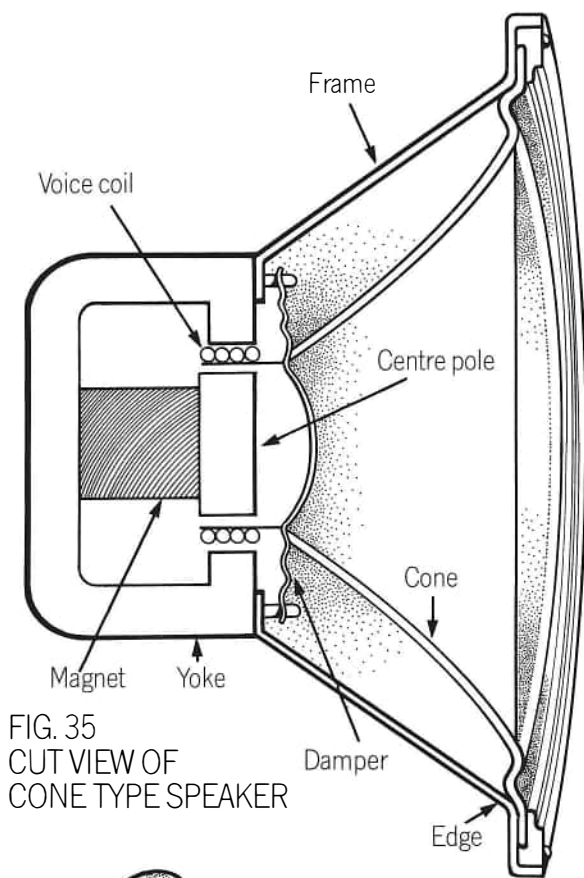


FIG. 35
CUT VIEW OF
CONE TYPE SPEAKER

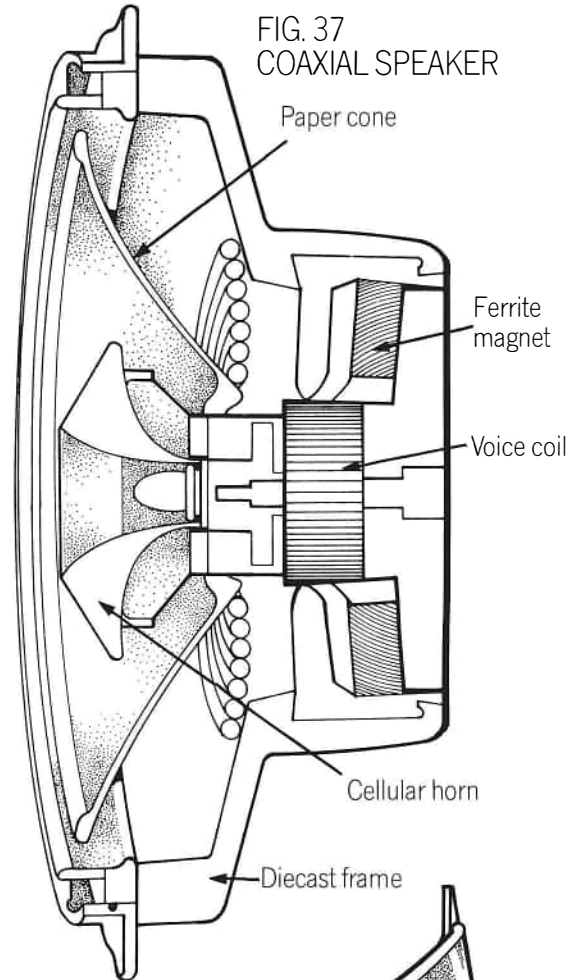


FIG. 37
COAXIAL SPEAKER

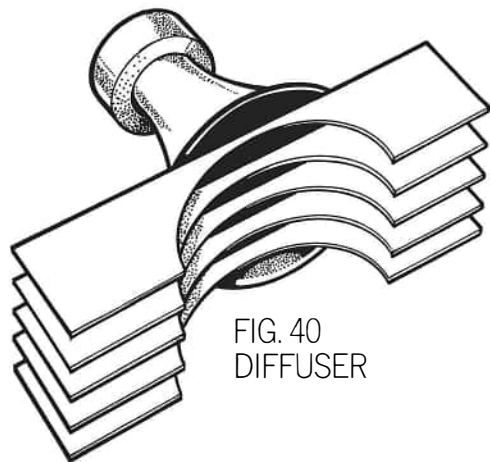


FIG. 40
DIFFUSER

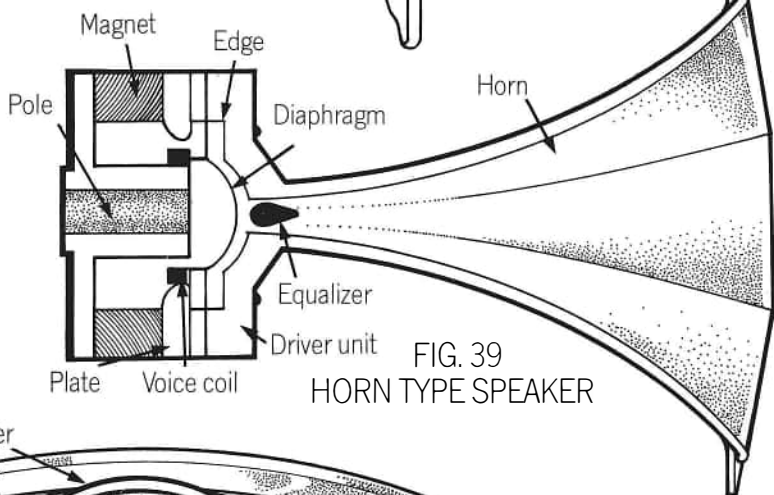


FIG. 39
HORN TYPE SPEAKER

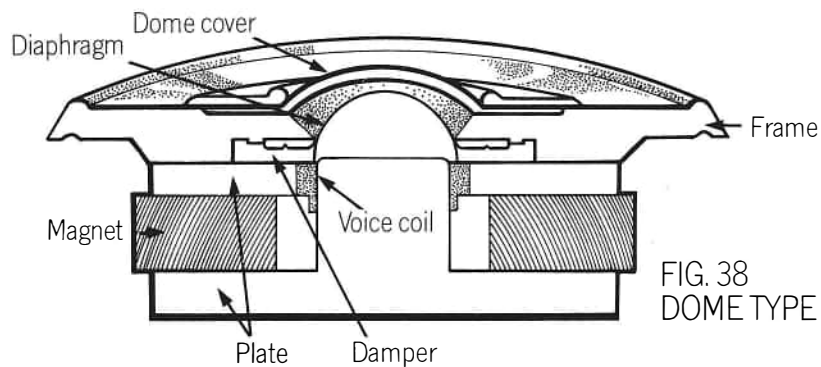


FIG. 38
DOME TYPE SPEAKER

Loudspeakers in a hi-fi system are somehow like the last pages of a novel. The plot gradually thickens, the parts of the puzzle fall into place, and from them emerges the solution and the revelation of the writer's intents. But, as in some stories, loudspeakers are not always a happy ending, sometimes leaving questions unanswered, wishes unfulfilled.

As simple as loudspeakers are in theory, as quizzical can they be in practice. Let's look at their structure and functions first.

Structure and Functions

The signal from the power amplifier passes through a coil (called the "voice coil") which is suspended between the poles and round the centre pole of a large, powerful permanent magnet. The interaction between this permanent magnetic field and the alternating magnetic field generated around the coil by the signal makes the coil vibrate in rhythm with the signal. A cone shaped membrane attached to the coil passes these vibrations on to the surrounding air, creating sound waves. A cross-section of such a loudspeaker is shown in Fig. 35.

The speaker cone, being a mechanically vibrating mass has a resonance point, that is a certain sound frequency at which it "likes" to vibrate more than at others and in the vicinity of which it would therefore over-emphasize the sound. One major task of loudspeaker designers, therefore, is to eliminate this resonance or at least to bring it under control or lower it beyond the lowest audible sound frequency. This is done by means of specially selected cone materials and cone designs.

Double Cone and Coaxial Speakers

On the other hand, a loudspeaker can reproduce only a limited sound range – large-diameter loudspeakers are best at delivering the low sound spectrum, small ones are most suitable for the high range. Though most portable and economy class audio equipment has only one full-range loudspeaker, various ways have been conceived to divide the sound spectrum into two or three ranges and feed each portion to a cone of the proper dimensions. The next logical step is the coaxial speaker shown in Fig. 37 which has a larger, outer cone for the low range and a smaller horn speaker for the medium and high frequency spectrum. Designs with three coaxially arranged low, medium and high range speakers usually have a crossover network which divides the signal into the suitable portions, and level controls (attenuators) for each range.

Two-Way, Three-Way, Four-Way Systems

By logical extension of this principle, we arrive at speaker systems with two, three or even four separate speakers mounted in one box. According to the number of sections into which the audio spectrum is divided,

such systems are called 2-way, 3-way or 4-way.

The number of speakers in such a box is not always the same as the number of ways, because the high range "tweeters" and mid-range speakers are sometimes doubled, so that we get 3-way 4-speaker or 3-way 5-speaker systems.

In such systems, each speaker unit can be designed to reproduce, as ideally as possible, its apportioned sound range.

The low-range "woofer" is usually of large diameter (20,25, or 30 cm) and of considerable weight because of its big, heavy magnet. The mid-range speaker, about 10 cm or so in diameter, reproduces the sound range from approximately 500 to 2,000 or 3,000 Hz. The top two or three octaves – up to about 20,000 Hz in excellent hi-fi speakers – are the domain of the tweeter, either a small paper cone speaker or, increasingly, a dome-type

(Fig. 38) with a dome-shaped membrane or a horn type (Fig. 39) with a metal horn instead of a cone. (Horns are more efficient in coupling the sound vibrations to the air, like the bell of a trumpet.) A relatively new design is the multi-cellular horn used for mid-range and tweeter applications which is partitioned into a dozen or so cells. The purpose of this is to diffuse the sound more evenly throughout the room.

Directionality

This may require some explanation. Speakers have the sometimes desirable, sometimes unwanted, characteristic of radiating their sound in a certain, clearly defined angle which becomes narrower and narrower as the sound frequency becomes higher, so that tweeters show a rather strong directionality. As this would affect the sound diffusion in the listening room and limit the usable listening positions, speaker designers have come up with various methods of overcoming this deficiency. One of these is the multi-cellular horn design, others include the above-mentioned dome tweeter or a small reflector called "diffuser" mounted before the tweeter (Fig. 40).

What complicates matters for the audio amateur is that the speaker box with five speakers does not always sound better than the one with four – a number of other factors enter into the picture, including the quality of each speaker and the way they are matched, the size, shape and weight of the speaker enclosure, and the listening room.

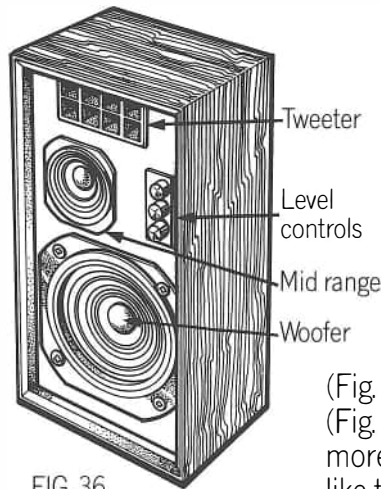


FIG. 36

LOUDSPEAKERS

LOUDSPEAKER ENCLOSURES

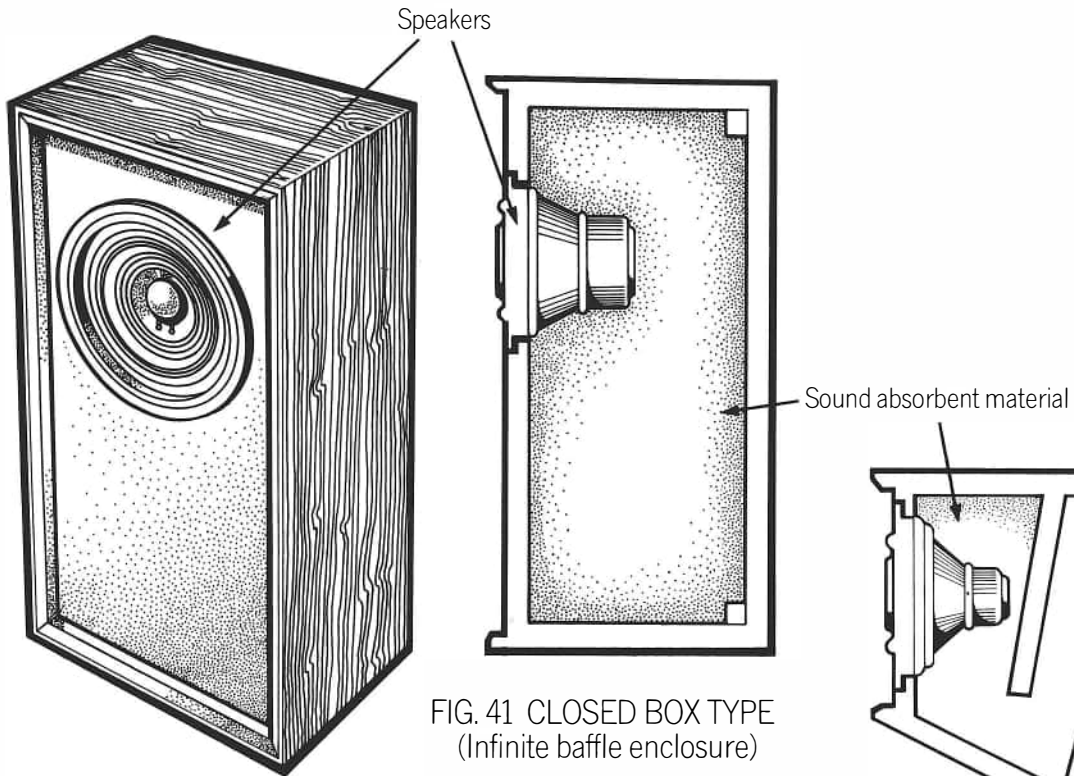


FIG. 41 CLOSED BOX TYPE
(Infinite baffle enclosure)

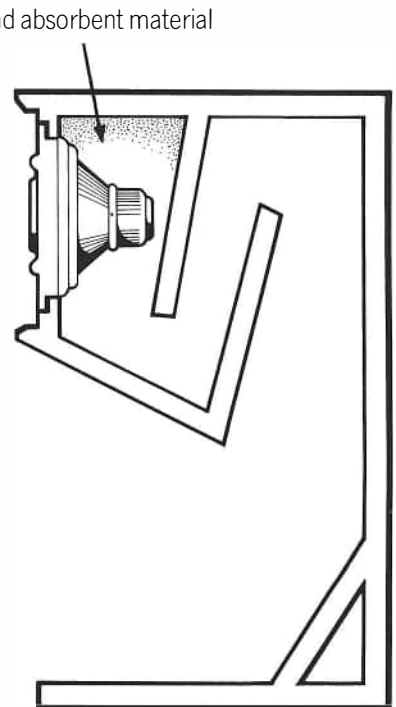


FIG. 43 SPECIAL ACOUSTIC
LABYRINTH ENCLOSURE
(Back loaded horn enclosure.)

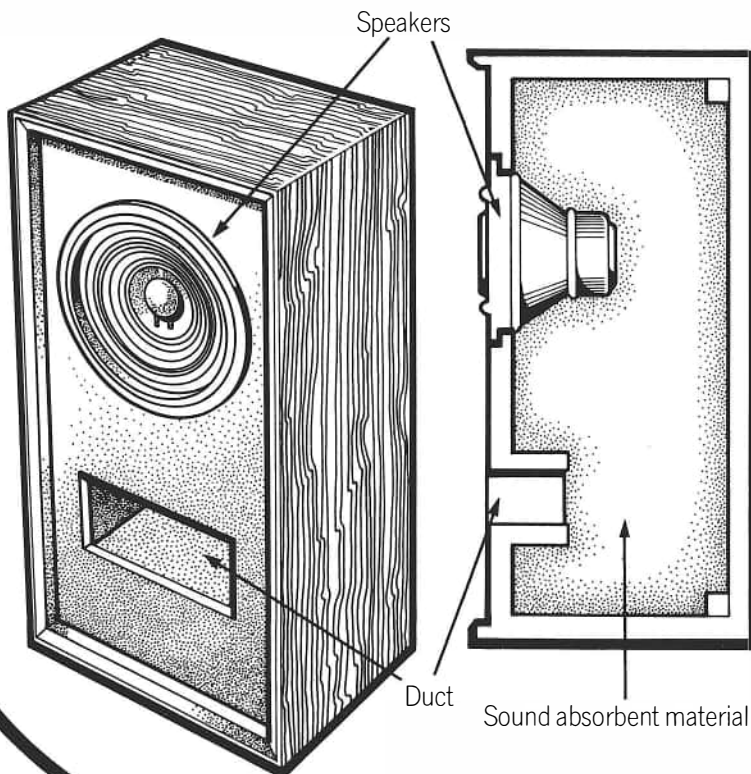


FIG. 42 BASS REFLEX BOX

Speaker Enclosure

Enclosures can be roughly divided into three types: closed boxes (Fig. 41), bass reflex types (Fig. 42) and special acoustic labyrinth (Fig. 43). A closed box is hermetically sealed at the back so that sound emanating from the rear of the speakers is absorbed in the box (which is usually clad with sound absorbent glass wool or similar material). Only the sound from the face of the speakers is radiated into the room. Closed boxes usually excel in sound clarity and subtleness, but they require rather voluminous dimensions to obtain satisfactory bass response. A modern variety of the closed box is the air suspension type often used as compact bookshelf speaker systems, in which the woofer is mounted in a soft baffle suspension so that it can perform longer piston movements for better bass response.

The bass reflex type, on the other hand, boosts the bass sound by channeling part of the rear sound back to the front through a duct or port. In other words, it behaves as if two woofers were installed. The key to successful designs of this type lies in the phase inversion which the box must perform, because otherwise the sound from the speaker and from the duct would be out of phase and might cancel each other out. Bass reflex boxes are being used widely because of their smaller volume in comparison to closed boxes, but their drawback is that the bass range sometimes sounds a little boomy.

A combination type of enclosure is a folded tuned column enclosure or Acoustic Labyrinth. The term "labyrinth" arises from the fact that a tube or column is folded to preserve physical space and therefore appears as a maze or labyrinth in the enclosure. The tuning of the tube or pipe is similar to that of an organ in principle and is determined by its cross section, length and loading, usually with sound absorbent material. In some units, the physical size of the labyrinth can be readily adjusted by the user for personal and listening room variations.

How to Choose Speakers

The choice of speaker systems thus becomes rather tricky because the manufacturer's specifications say very little about the character of the sound. The ideal speaker, of course, would reproduce the entire sound range without arbitrarily adding or subtracting anything, but such a loudspeaker has never and will never be designed—all speakers have some individual habits and idiosyncracies which are called "colouration" or "transparency" or other such abstract words. When purchasing speaker systems, the specifications should be used to determine the general size, type and power range, but the final choice must be made either after comparative listening tests, preferably with one's own amplifier and in one's own listening room or if that is impractical by trusting a well reputed manufacturer. Ensuring that they are suitable for the system they have to operate with, especially regarding the power parameters.

Crossover Network, Level Controls

Auxiliary parts in a speaker system include the crossover network and sometimes one or more level controls (Fig. 44). The crossover network is an electronic circuit which splits up the sound spectrum into the required number of ways—low and mid/high range in a 2-way system, low, mid and high in a 3-way. It consists of a combination of coils and capacitors, often equipped with ferrite coil cores which permit more compact design. The border frequencies between one range and another are called "crossover points". The level control or brilliance control is a simple attenuator which permits the medium or high sound ranges to be suppressed or boosted to an extent—a helpful way of matching the speaker response to the characteristics of the listening room.

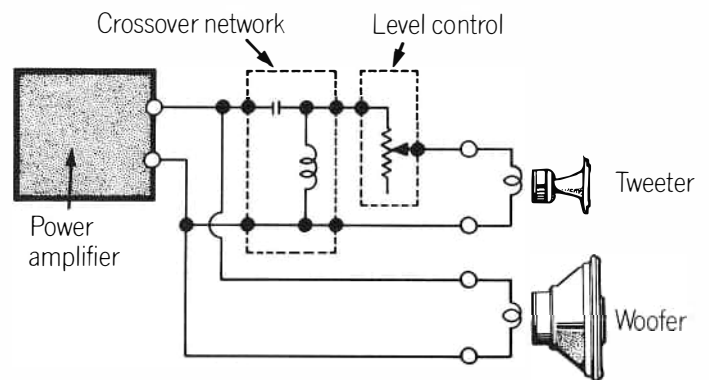


FIG. 44 WIRING IN A CROSSOVER NETWORK AND LEVEL CONTROL.

LOUDSPEAKERS AND HEADPHONES

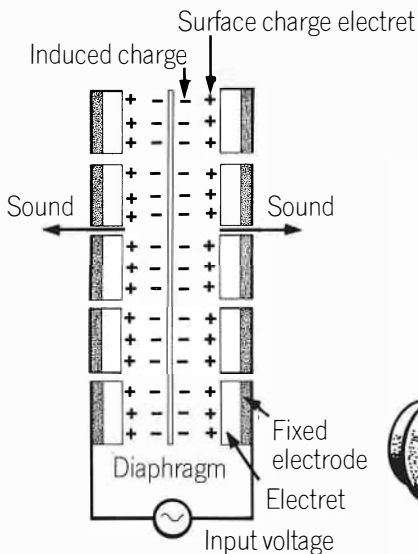


FIG. 46

FIG. 47 THE PRINCIPLE OF ELECTROSTATIC HEADPHONES (push-pull type)

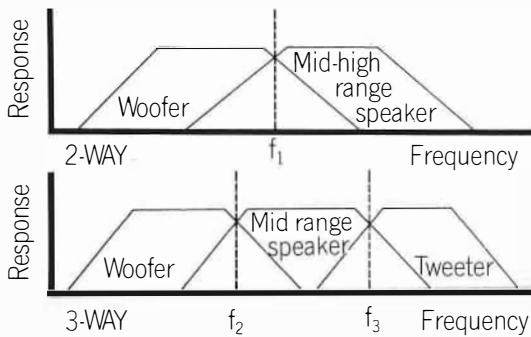
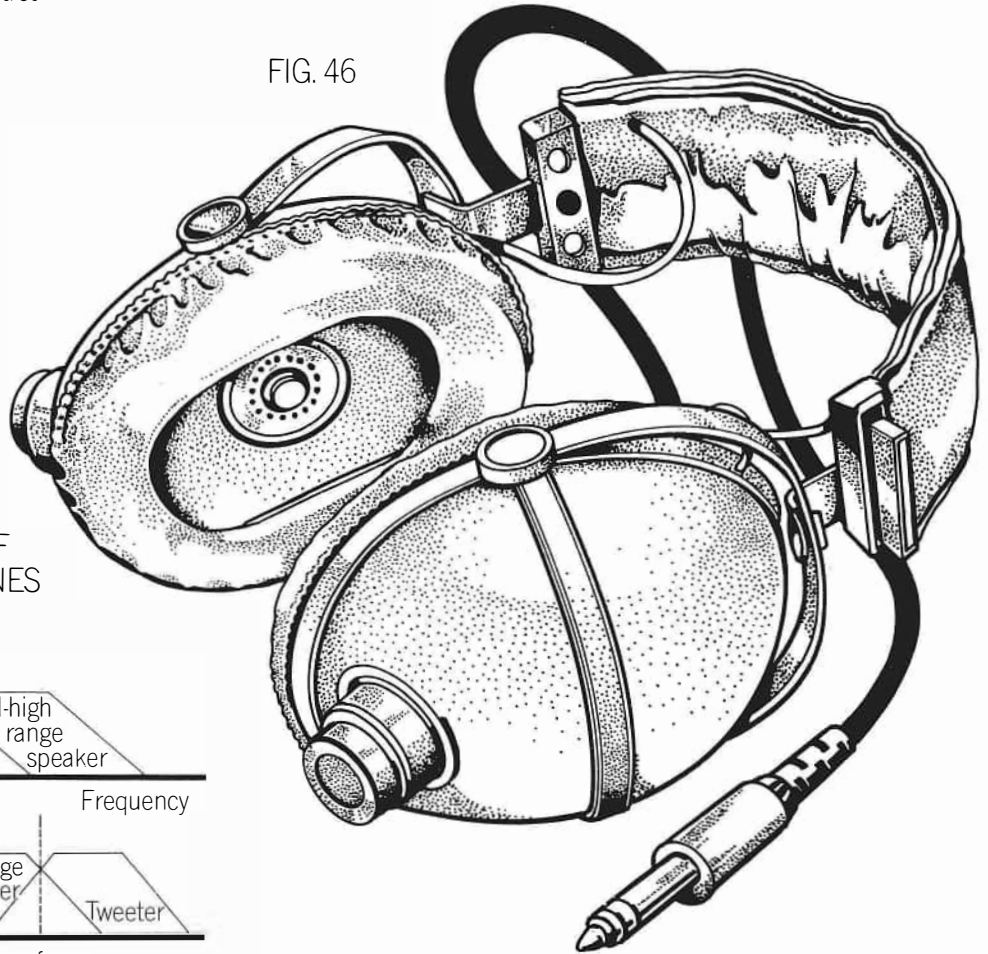


FIG. 45 CROSSOVER FREQUENCIES (f_1, f_2, f_3)

Fig. 48 Principle
When AC voltage is applied from Direction 3, the element expands and contracts in the amplitude direction (Direction 1) according to the alternating pulses of the voltage. Conversely, when an external mechanical force is applied from Direction 1, voltage is generated toward Direction 3.

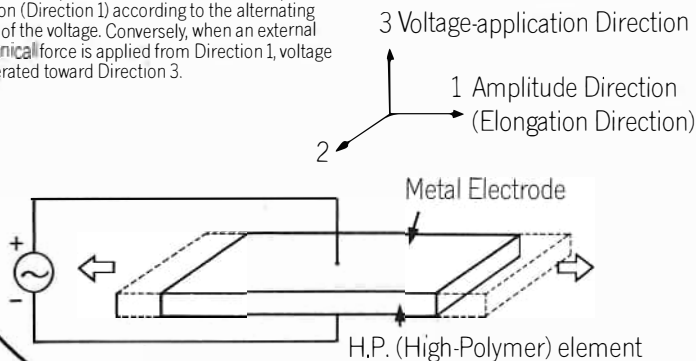
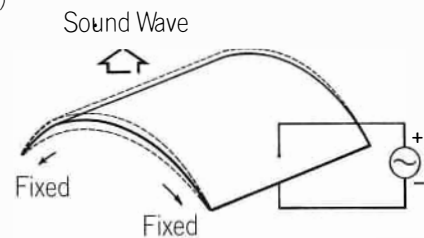


Fig. 49 Application
The ultra-thin film high-polymer element is bent to form a curved surface. Both edges of the film are fixed, allowing it to generate sound waves when AC voltage is applied from the amplifier. It should be noted that the action of the diaphragm is not the "piston motion" of conventional cone-shaped diaphragms but rather an ideal "breathing motion" for natural audio clarity.



FIGS. 48, 49. HIGH-POLYMER PIEZO ELECTRIC HEADPHONES.

How to read Loudspeaker Specifications

a) NUMBER OF WAYS, NUMBER OF SPEAKERS

For example: 2-way 2-speaker, 3-way 3-speaker, 3-way 4-speaker, etc. Unfortunately the number of ways and speakers is not always directly related to the sound quality.

b) INPUT IMPEDANCE

The load impedance of the speaker system as seen from the amplifier output. 4, 8 and 16 ohm are the internationally standardized nominal values for speakers to be driven by transistor amplifiers.

c) MAXIMUM POWER HANDLING ABILITY

Also called "maximum input power".

The power (in watts) that the speaker system can handle continuously without fear of damage. Sometimes, a higher limit value is also given which describes the maximum input power of short transients and peaks. Power handling ability does not need to equal or exceed the amplifier's output power unless the amplifier is to be operated at top volume – which almost never happens in home applications.

d) FREQUENCY RESPONSE

The width of the sound spectrum (in Hz) that the speaker will reproduce. Taking 1000 Hz as the reference level, the lower and upper limits are the points at which the loudspeaker will still deliver $\frac{1}{3}$ of the sound pressure level (± 10 dB). Look for wider frequency response range.

e) EFFICIENCY

(Also variously called "sensitivity" or "sound pressure level"). Measured in dB/W or microbar/W. The speaker's loudness measured in an anechoic chamber with 1W input power and at a distance of 1m (or sometimes 50 cm) from the speaker, directly in front of it. The higher the value, the better the speaker's efficiency. Note that with values given in microbar, measurements at 50 cm distance will result in 4 times higher values than if measured at 1m. Pioneer specifications give values as measured at 1m.

As larger speaker systems, especially large bass reflex boxes, have higher efficiencies than smaller boxes, a seemingly paradoxical situation arises: smaller speakers require larger amplifier output power than big speaker systems.

f) CROSSOVER FREQUENCIES

The "borderlines" between the low, mid and high ranges in a 3-way speaker system, or between the low and medium/high ranges in a 2-way system. Usually not adjustable. Contains no information about the sound quality (Fig 45).

Headphones

A good pair of stereo headphones is a very worthwhile, small investment for the serious listener. Headphones serve two main purposes: they permit late-night listening in complete privacy and at high volume levels; and they are indispensable for monitoring tape recordings. Practically all modern stereo amplifiers, tape decks and even some tuners are equipped with standardized headphone jack. In some instances,

the headphone volume can be adjusted with a special control, independently from the master volume setting.

It must be noted that listening through headphones creates a fundamentally different sensation from listening through speakers. As the sound enters directly into the ear, it acquires much stronger "presence". A monophonic program heard through headphones will appear as if the sound source were located right in the centre of the head. Stereophonic programs seem to be spread out in "interior space" – an experience that can be startling at first.

Good headphones should be light and comfortable to wear without causing perspiration around the ears. They should also have a wide frequency response – the bass range is very critical – and should sound transparent, not stuffy. There is a wide selection of stereo headphones, including some models with built-in volume and tone controls.

Electrostatic Headphones

Electrostatic type headphones (Fig 47) are totally different in design and construction from conventional-type headphones which utilize moving coil loudspeakers. The force emanating from the amplifier drives the overall surface of the reproducer in the same phase movement, thereby improving both distortion and transient characteristics. The electret type headphone is distinguished from conventional headphones by its extended frequency response and excellent transient characteristics.

Infra-red Headphones

In the opinion of some people, conventional headphones have a slight drawback. Because they're connected to the amplifier by a long wire, you have to be careful when you trail that wire around a room. But recently, a new type of headphone has appeared, which doesn't depend on wires. Instead, it depends on infra-red signals. In this system, signals from your hi-fi are accepted by a special transmitter, which converts them into varying amounts of infra-red energy. This energy is beamed to a special pair of headphones with an infra-red detector, circuitry to convert the signals back to audio frequencies, and an amplifier to drive the phones themselves. As you can see, this is a complex substitute for a trailing wire. And at its current state of development, it's very expensive.

Four-channel Headphones

Another type is the four-channel headphone. They are usually larger and heavier than stereo phones, because they contain four reproducers, not two. At the present state of the art, no four-channel headphone reproduces four-channel sound really effectively, although, by providing a sense of ambience, it does add an extra dimension to stereo listening.

High-Polymer Piezo Electric Headphones

These headphones take advantage of the unusual principle of electrophysics by employing high-polymer driver elements with evaporated aluminium layer to produce natural high fidelity sound. (Figs 48, 49).

FOUR-CHANNEL

Four-channel stereo has been created to solve the biggest problem inherent in conventional two-channel stereo, that is the fact that all sound – both direct and indirect, and of different phases – is heard from the two loudspeakers placed in front of the listener. Four-channel stereo employs four speakers, usually two in front and two in the rear of the listener as illustrated. (Fig. 50).

The two major advantages of four-channel stereo are: (1) The listener can enjoy in his own home a musical environment often comparable to a concert hall, since the two rear loudspeakers reproduce the natural sounds of the concert hall – mainly the indirect sounds. And (2) a new world of sound, one heretofore inconceivable, is created. What one hears is music that has been composed, arranged and recorded with the effects of four-channel stereo in mind, i.e. two orchestras in position, one in front another in the rear to enable switching back and forth between the two; surrounding the listener with music; or enjoying the movement of sound.

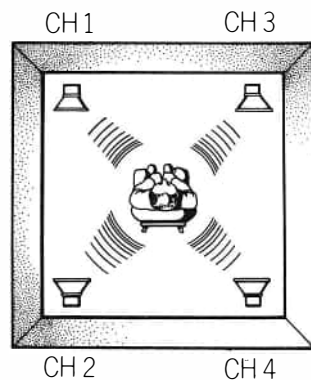


FIG. 50 4-CHANNEL STEREO

Four-channel Programme Sources

There are today a number of different forms and types of four-channel stereo program sources. But generally speaking, they can be classified in the following three categories. (Discrete, Matrix, SQ.)

1. Discrete System

This system is the most authentic in terms of sound quality, and also the most expensive, since it not only requires four-channel equipment, but also four-channel tape recordings. When the sounds are reproduced four independent playback amplifiers are used, so that optimum four-channel effects are achieved. This system is often referred to as the 4-4-4 system. (Fig. 51).

Conceivable programme sources for discrete four-channel stereo include open reel, 8-track cartridge, or cassette tapes, or disc recordings. Today open reel, 8-track cartridge tapes and discs are already on the market.

Discrete Four-channel Tape

Figure 52 shows the functions in discrete four-channel open reel tapes and 8-track cartridge tapes. They provide the best tonal quality reproduction, as well as the most distinct separation of sounds, since independent signals can be recorded on each track, so that the playback effects provide the same rich and abundant sounds of the concert hall for classical or popular music. Additionally with this system, other unusual effects are possible, such as switching back and

forth between sounds of two orchestras or the movements of a sound field at very high speeds.

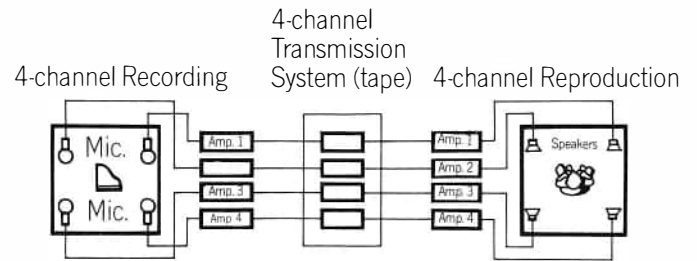


FIG. 51 FOUR-CHANNEL TAPE SYSTEM

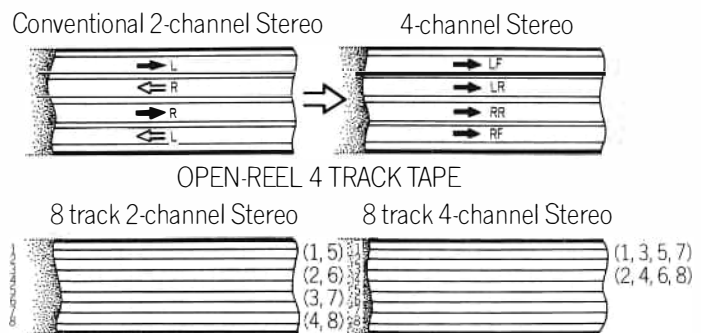
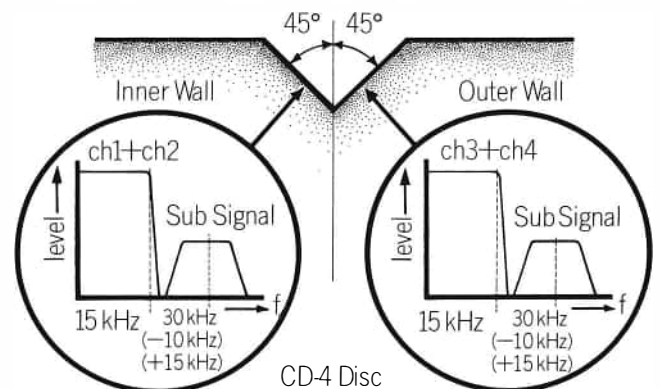


FIG. 52 OPEN-REEL AND CARTRIDGE FOUR-CHANNEL TAPE

Discrete Four-channel Disc (CD-4 Disc)



$$\begin{aligned} \frac{1}{2} (\text{ch.1} + \text{ch.2}) + (\text{ch.1} - \text{ch.2}) &= \text{ch.1} \\ \frac{1}{2} (\text{ch.1} + \text{ch.2}) - (\text{ch.1} - \text{ch.2}) &= \text{ch.2} \\ \frac{1}{2} (\text{ch.3} + \text{ch.4}) + (\text{ch.3} - \text{ch.4}) &= \text{ch.3} \\ \frac{1}{2} (\text{ch.3} + \text{ch.4}) - (\text{ch.3} - \text{ch.4}) &= \text{ch.4} \end{aligned}$$

FIG. 53 THE PRINCIPLE OF DISCRETE FOUR-CHANNEL DISC

The CD-4 disc record is a unique four-channel discrete record that has compatibility with mono or stereo records.

In the CD-4 disc, four independent signals are recorded by using special modulation techniques. This disc maintains the conventional 45°/45° record groove, but on the inner groove wall the left channel sum signals (channel 1 + channel 2) are engraved, while on the outer wall the right channel sum signals (channel 3 + channel 4) are engraved. (Fig. 53). Additionally, important signal components for the rear channels are transmitted by the inaudible subcarrier, the centre of which exists at 30 KHz. These signal components consist of the

STEREO

differential signals (channel 1 – channel 2) of the left channels as well as the differential signals (channel 3 – channel 4) of the right channels. These signals are frequency modulated and are then added to the walls of the record groove in addition to the above-mentioned sum signals. When we matrix these signals and the differential signals, four independent signals are obtained. Since this disc is the modulation type (Fig. 54), a CD-4 demodulator unit is required for exclusive playback in addition to a high-quality stereo turntable and phono cartridge.

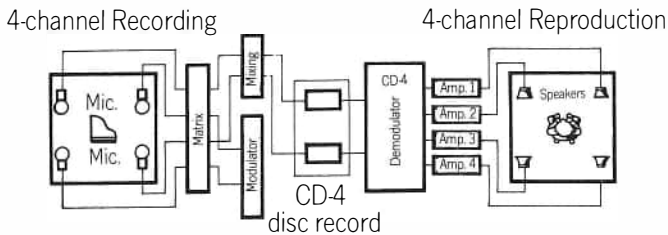


FIG. 54 FOUR-CHANNEL DISC (CD-4 DISC) SYSTEM

2. Matrix System

Four signals recorded in the form of four-channel stereo are fed through a special “encoder” and converted into two-channel signals. In the playback process, they are fed through a “decoder” to recreate the original four-channel stereo sound field. (Fig. 55).

This system is called “matrix,” or “4-2-4” system, and when used, does not require a special tape deck or turntable as is the case of discrete stereo. All the matrix systems available today (except the SQ matrix) can be classified into this group.

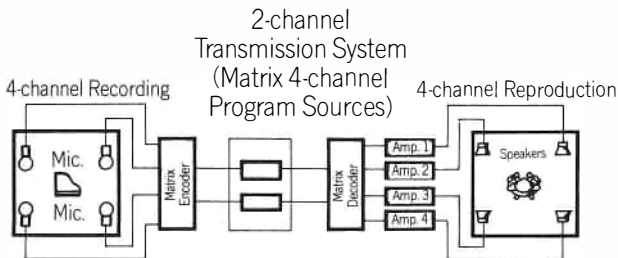


FIG. 55 THE MATRIX FOUR CHANNEL SYSTEM

3. The SQ Matrix Decoder

The matrix encoding principle of the SQ system (Fig. 56) is quite different from matrix systems where the four-channel signals are recorded from different angles against the 45°/45° conventional record groove. This inevitably results in cross talk between the front two speakers. The SQ system, on the other hand, maintains the 45°/45° principle, by introducing an additional phase difference of 90 degrees, with the left rear channel information translated into a clockwise circular motion and the right rear channel information translated into a counter-clockwise circular motion of the stylus tip. This principle is called “circular modulation.” Thus, as the record rotates, a clockwise helix is produced for the left rear channel, and a counter-clockwise helix for the right rear channel. (Fig. 57).

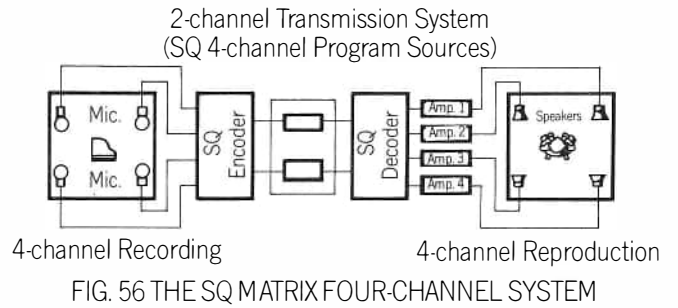
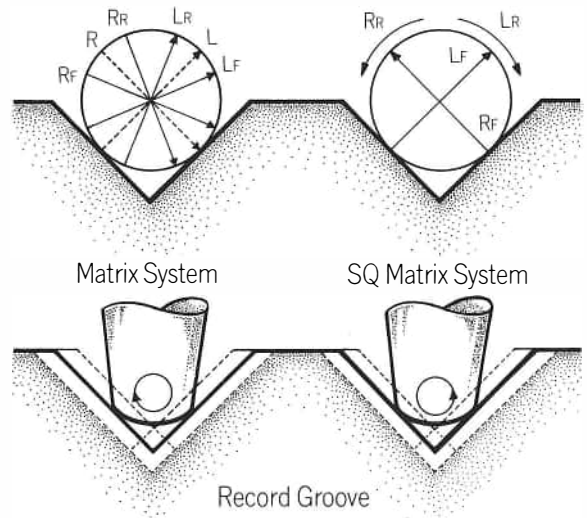


FIG. 56 THE SQ MATRIX FOUR-CHANNEL SYSTEM



Clockwise rotation of the stylus (modulation of Left Rear signal) Counter-clockwise rotation of the stylus (modulation of Right Rear signal)

FIG. 57 THE TWO CIRCULAR MODULATION OF THE SQ SYSTEM

CARE OF RECORDS

"One careless playing can damage a record irretrievably!"

When long-playing records were first introduced, the manufacturers proclaimed their indestructibility. This appears to be an overstatement. It is perfectly true that unlike the old 78 records, they are difficult indeed to break. The material of which 78s were made was not only coarser but harder and more brittle, and in those days a dropped record usually meant a broken record.

It takes much more than dropping to break an LP but to set against this is the comparative softness of the material of which it is made and the supreme ease with which it can be scratched and damaged in other ways.

Records leave the factory in clean condition and packed in both protective inner sleeves and stout cardboard covers.

The biggest enemy of the LP is dust, so when not in actual use it should always be kept in the two covers designed for its well being. Unfortunately, the material of which LP's are made holds, and persists in holding, a high charge of what is usually called static, which has the property of attracting and holding dust in much the same way that a magnet attracts and holds iron filings.

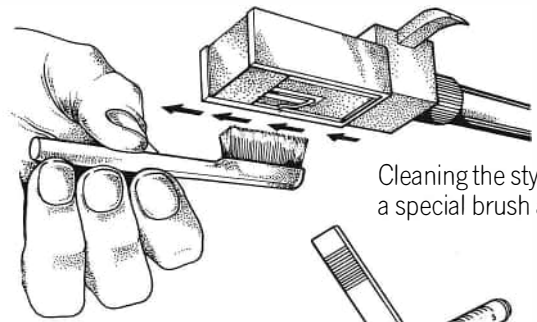
A dusty record quickly becomes a damaged record. Most dust is abrasive and gets ground into the record surface. The friction between stylus and record surface can generate enough heat momentarily to weld the specks of dust permanently into the record grooves, especially if the pick-up is over-heavy. But pick-up playing weight is another matter and is explained elsewhere.

The elimination of static has been the subject of much research, but a suitable material for records, permanently free from static, has not yet been discovered. Most records contain static charges when new and these are likely to increase with use unless countermeasures are taken.

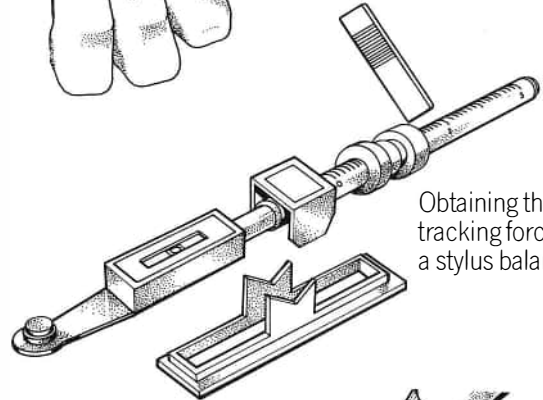
There are several ways of reducing static. A periodic light wiping in circular motion with a soft and slightly damp cloth or pad treated with a suitable record cleaning fluid or a dust remover or ioniser; and usually nothing else is necessary for a long time other than regular use of a groove cleaner (if possible, of the conductive and earthed type); plus ensuring that after playing records are returned promptly to their inner paper (not plastic) and outer sleeves.

Mishandling easily causes trouble. Records should be handled only by their groove-free edges and centre labels. A finger print on the record means some deposit of grease which will retain dust beyond the powers of most basic cleaning devices. The stylus tip also accumulates dust, dirt and other deposits; and like the record should be cleaned regularly using a suitable brush and cleaning fluid.

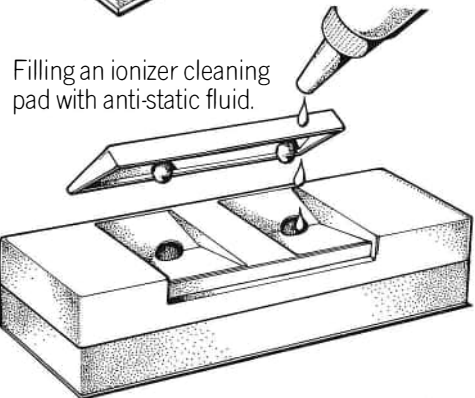
Another frequent source of unnecessary wear – the careless placing of the stylus on the record. It should be put gently on to the rim of the record so that the widely-spaced blank run-in grooves will convey it to the closer-spaced grooves which carry the program. But there may be times when one wants to play only a piece or movement which does not start at the beginning of a record. Most of the better pick-up arms are fitted with a device that will lower the stylus gently wherever desired, and where these are not built-in, a separate lowering device can be added quite cheaply.



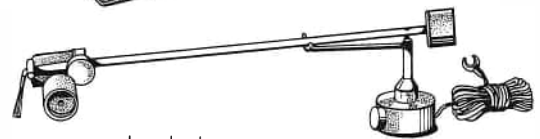
Cleaning the stylus tip with a special brush and fluid.



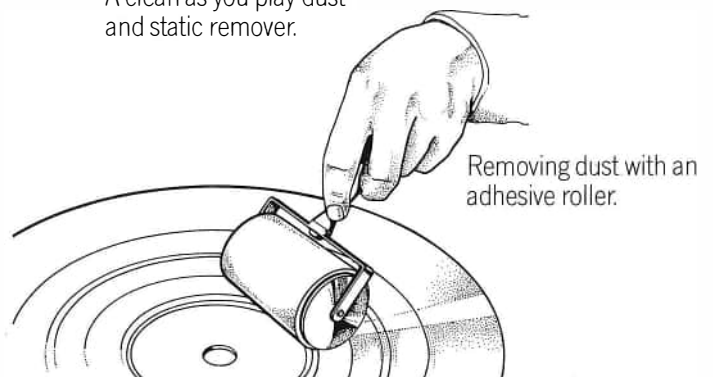
Obtaining the correct tracking force by means of a stylus balance.



Filling an ionizer cleaning pad with anti-static fluid.



A clean-as-you-play dust and static remover.



Removing dust with an adhesive roller.

Poor pick-up design, tracking errors and unsuitable styli are all prolific causes of unnecessary wear of records. The first two are to be taken into account when buying the playing equipment but a stylus can be changed, and indeed should be changed periodically. Both diamonds and sapphires have been described as "permanent" but neither of them is. The diamond, however, is much longer lasting, and its longer playing time makes it a good investment even without taking into account the potential risk of damaging a record by a worn stylus.

The best method of storage is to keep the records in their containers on their edges, bookwise, on shelves, placing them close enough together for each to exert a reasonable degree of side pressure on its

neighbours to prevent warpage but not enough to make withdrawal difficult. Many cabinets are made for the purpose, but normal bookshelves will do splendidly. The essential thing is to have uprights at reasonable intervals and these should be smooth surfaced and at right angles.

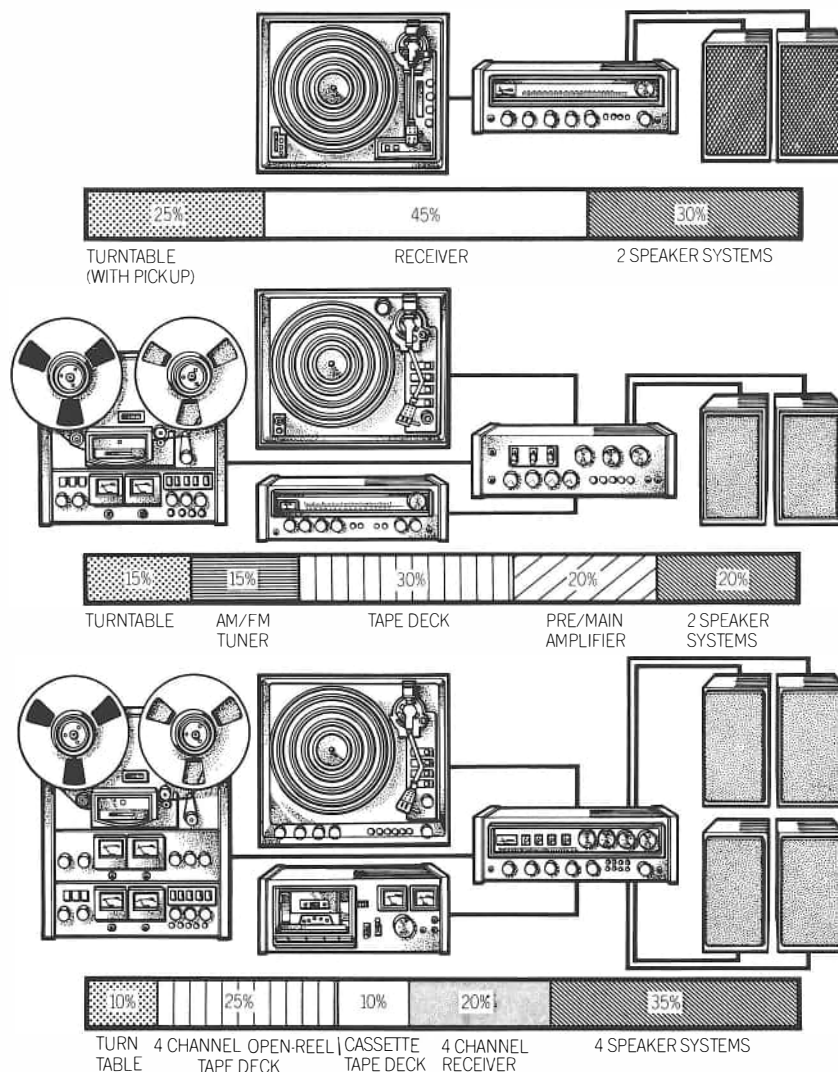
Inevitably there will be a section that is not completely filled and something to maintain the desirable side pressure will be needed. A large book or two will fill the space between the records and the next upright.

The only other thing to beware of is undue heat. Normal room temperature is ideal but do not place your shelves or cabinet over or immediately beside a radiator or in front of the chimney breast.

MONEY MATTERS

A few words of pecuniary advice to the hi-fi novice may be warranted here because an audio system represents a major investment and should be purchased with wisdom and care. As a convoy of ships can only travel as fast as its slowest vessel, so a hi-fi system will only be as good as its weakest link – the catchword is "balance". It makes little sense to match a superb amplifier with a poor speaker system, or equip a top turntable with a poor tone-arm and cartridge.

To help avoid such pitfalls we have drawn up the following illustrations which give approximate percentages of the total available budget to be spent on each component of a hi-fi system. Absolute amounts could not be quoted due to the wide variations in possible purchase prices, but these percentages should prove about right in practically all cases.



GLOSSARY OF HI-FI

AC: Alternating current. An electric current with periodically changing polarity. In popular parlance, often used as synonym for electrical house (or mains) current.

Acoustic Feedback: Unwanted acoustic interaction between output and input of an audio system usually between loudspeaker and microphone or turntable cartridge. Can lead to "howl".

Acoustics: The science of sound. Also, the acoustical character of halls and rooms.

Active Filter: See filter.

Aerial: Synonym for antenna.

AF (Audio Frequency): Frequency within the range of human hearing—approximately 20–20,000 Hz.

AFC (Automatic Frequency Control): An AFC circuit of an FM tuner corrects for an inaccuracy in tuning by locking in the station being tuned.

AM (Amplitude Modulation): Modulation accomplished by varying the amplitude (intensity) of the carrier.

Amplification: Increase in signal magnitude.

Amplifier: Unit providing amplification of signal, from low to higher voltage or current (pre-amplifier) or power (power or main amplifier).

AM Suppression: Characteristic of an FM tuner to suppress changes of amplitude in received signals thereby improving the signal-to-noise ratio by rejecting noise and interference.

Anechoic Chamber (room): A specially designed room used for testing microphones and speakers, rendered acoustically "dead" by sound absorbing material.

Antenna: Assembly of metal bars, wires or loops for picking up radio waves. Dimensions depend largely on the wavelengths to be received.

Antenna Directionality: The characteristic of certain antenna configurations, including most FM and TV antennae, to receive radio waves reaching them within certain defined angles more strongly than those coming from other directions.

Antenna Gain: An indirect measure of an antenna's output level, i.e. the strength of the signal supplied by the antenna when compared to a standard antenna and at a certain frequency. Expressed in dB.

Anti-Skating Device: Mechanism exerting a small outward force on a tonearm to compensate the inward thrust caused by groove/stylus friction on tonearm geometry.

Audibility Threshold: The minimum sound intensity that the average human ear can hear. Approximately 0.0002 microbar at a frequency of 1000 Hz.

Automatic Level Control: A circuit which automatically keeps the level of a signal within a certain range. Examples: automatic gain control in AM radios. ALC in portable tape recorders.

AUX (auxiliary) Input: Input on amplifier etc. Accepts extra signal such as cassette tape player etc.

Baffle: The board on which one or several loudspeakers are mounted. Separates the radiations from the front and back of the speaker.

Balance Control: Potentiometer used to

adjust volume difference on left and right stereo channels.

Bandpass Filter: A circuit which will pass signals above (high bandpass) or below (low bandpass) a certain frequency while attenuating others.

Bass: Low audio frequency range, below approximately 200 Hz.

Bass Reflex: Loudspeaker enclosure with an outlet permitting sound from the rear of the speaker cone to be radiated to the front.

Beat: A pulsation caused by interaction of two waves of different frequencies.

Bias: In tape recording, a high frequency signal applied to the tape to optimize its frequency response. Also, the side-thrust on a tonearm.

Bias Compensator: See anti-skating device.

Bi-Radial Stylus: See elliptical stylus.

Capacitor: A device which can store an electric charge. Permeable to AC, impermeable to DC.

Capstan: Drive spindle in a tape deck. The tape is pressed against the capstan by a pinch roller.

Capture Ratio: An FM tuner's ability to reject unwanted FM stations and interference occurring on the same frequency as the desired station.

Cardioid Microphone: A microphone with a directional characteristic that resembles the shape of a kidney.

Carrier: Main radio signal from a transmitter. Can be modulated in different ways (AM, FM) to convey sound or picture information.

Cartridge: (a) Phono pickup; (b) endless loop tape in a packaged, standardized container.

Cassette: Preloaded container with tape and spools for use on cassette tape recorder.

Centre Channel: An output terminal found on some stereo amplifiers which supplies a monophonic mixed L+R signal.

Channel Separation: Degree to which left and right channel signals are separated in a stereo pickup, FM stereo tuner, amplifier, etc.

Characteristic: Refers to a characteristic curve which conveys information about an amplifier or other item.

Chrome Tape: A tape using chromium dioxide particles, offering excellent frequency response and wide dynamic range.

Coaxial Cable: A cable consisting of an inner conductor and an outer screen. Used for antenna leads and for interconnecting audio units.

Coaxial Speaker: A loudspeaker consisting of a bass cone with a concentrically mounted tweeter.

Compatible: (a) FM MPX signal receivable as mono by a radio or tuner; (b) Stereo disc playable with a mono pickup; (c) Stereo pickup suitable for playing mono records, too; (d) The ability to connect various units together in a system without an electrical mismatch.

Compliance: The flexibility of a phono cartridge stylus assembly. The unit of compliance is 10^{-6} cm/dyne. e.g. Compliance 25×10^{-6} cm/dyne.

Condenser Microphone: A microphone utilizing the changes in capacitance caused in a condenser if one of its plates—the microphone membrane—vibrates in rhythm with sound waves. Same principle can be applied to a condenser

type phono cartridge.

Control Amplifier: A preamplifier and amplifier combined in one unit.

Counter Weight: Weight fitted at rear end of tonearm behind the pivot to counter the weight of the arm/head assembly and to permit adjustment of the tracking force.

Crossover Frequency: In loudspeaker systems and multi-amplifier audio installations, the borderline frequencies between low/medium range and medium/high range speakers or amplifiers.

Crosstalk: Leak of right channel signal into left channel, and vice versa. Expressed as level of unwanted signal in relation to wanted signal channel, measured in dB.

Cutting: The process of engraving undulating grooves in a rotating wax-coated disc. An important step in the manufacture of phonograph records.

Cutting Angle: See vertical angle.

Cutting Stylus: Stylus used for cutting of phonograph records.

Damping: Reduction of resonant effects by use of resistance or its mechanical and acoustic equivalents.

Damping Factor: Ratio of loudspeaker impedance to amplifier's impedance. Denotes amplifier's ability to damp unwanted, residual speaker movement.

DB (Decibel): A logarithmic unit used to express the ratio between two powers, voltage or current levels.

Mathematically:

$$20 \log_{10} \frac{\text{level}_1}{\text{level}_2} \text{ or } 10 \log_{10} \frac{\text{Power}_1}{\text{Power}_2}$$

Decoder: In an FM stereo tuner, the circuit that extracts the left and right channel signals from an FM MPX broadcast signal. Or matrix decoder for translating matrix 4-channel program sources.

De-Emphasis: Attenuation of high sound frequencies in an FM tuner, to counteract the boosting of these frequencies ("pre-emphasis") done at the FM station.

Demodulation: The process of "extracting" from a modulated high or intermediate frequency wave the original audio signal which it was modulated with.

Derived Centre Channel: See centre channel.

DIN (Deutsche Industrie Normen): German Industrial Standards. In audio, one of the European recognised performance measurement standards. Also the German standard for plugs, sockets, etc.

Dispersion: Distribution of sound from a speaker within a room.

Distortion: Output signals not present in the original input. See harmonic distortion, intermodulation distortion.

Dolby System: A unique noise reduction system, which electronically reduces the irritating noise (tape hiss, circuit noise, etc.) without sacrificing the original tonal quality.

Drift: Tendency of a tuner to move away from optimum adjustment as its components warm up. Compensated by AFC.

TERMS

Drop-Out: In tape recording and playback, the "holes" in the sound caused by thin or bare spots on the tape.

Dual Cone: Speaker consisting of separate bass and treble cones mounted concentrically and driven by the same coil. See also Coaxial. Mechanical 2-way speaker.

Dynamic Mass: The effective mass of the moving parts of a pickup cartridge. This is not simply the sum of the masses of each component, but rather the equivalent mass that determines the mechanics of its behaviour.

Dynamic Range: In program material, the range of signal amplitudes from highest to lowest; the range (in dB) which a device will handle.

Efficiency: Ratio of output to input power in a transducer. In loudspeakers, the percentage of electrical input available as acoustic output.

Electrostatic Speaker: A loudspeaker utilizing the principle of a membrane vibrating in a strong electric field.

Elliptical Stylus: A pickup stylus shaped so that its width across the groove is greater than the width of its sides. Claimed to have better high frequency characteristics than spherical styli.

Enclosure: The cabinet which houses one or more speaker units. Has great influence on bass response of speaker system.

Equalization: Correction for frequency non-linearity of recordings. Phonograph records are cut with low frequencies attenuated and high frequencies boosted. Equalization compensates for this, producing a flat frequency characteristic.

Equalizer Amplifier: In a pre-amplifier, the circuit which amplifies and "flattens" the phono input signal. Needed because phonograph records are not cut with flat frequency response but according to a standard "equalization" curve.

Feedback: Signal from output of amplifier or electronic network applied to input in anti-phase (hence negative feedback) to reduce distortion and noise, and to flatten or otherwise shape frequency response. Also unwanted acoustic feedback.

Ferrite Core Antenna: An antenna, used chiefly for AM reception, consisting of wire windings around a core of ferrite. Advantages are: compact size, good sensitivity and high directionality.

FET (Field Effect Transistor): Special kind of transistor. Amplifies voltage, not current. Used in audio equipment because of its good linearity and high impedance.

Field Strength: The intensity of an electrical or magnetic field.

Field Strength Meter: See signal strength meter.

Filter: A circuit which attenuates signals above, below or at a particular frequency.

Flutter: Quick waver of pitch caused by speed fluctuations in the movement of tape or discs. Heard as a sort of quiver.

Flywheel: A disc of large mass which, when rotating, has the tendency to maintain its rotational velocity. This effect is utilized in tape equipment and turntables to maintain constant speed.

FM (Frequency Modulation): Type of modulation of radio waves in which the frequency, not the amplitude of the carrier is modulated by the audio signal. FM broadcasting achieves higher sound quality.

Frequency Response: The frequency range which a unit will reproduce or respond to.

Ferrichrome Tape: This tape blends Ferric Oxide with Chromium Dioxide particles. For better mid-range response, plus the high frequency response of Chrome tape.

Front End: Section of a tuner that selects the wanted station from the radio band and converts the RF signal to IF.

Gain: Degree of signal amplification achieved in an amplifier circuit. Expressed in dB. Opposite: negative gain or loss.

Gap: Vertical slit in a tape-head. In this gap, a magnetic field occurs during recording, and a magnetic signal is induced during replay.

Ghost: In TV, the appearance of a secondary picture slightly to the right of the main picture. Similarly, ghost can be used to mean multipath FM reception. Also, the "images" of the true station frequency that an FM tuner purports to receive.

Harmonic Distortion: The sum of all signals in an output which are multiples of the input signal frequencies ("harmonics"). Their intensities are expressed as a percentage of the total output intensity.

Head: (a) the erasing, recording and playback heads in tape equipment; (b) the shell-cartridge assembly of, or attached to, a tonearm.

Head Shell: The—often detachable—part of a tonearm which carries the cartridge.

Heat Sink: A device used to remove heat from electronic components such as tubes, transistors, etc.

Hertz (Hz): Unit of frequency, equal to one cycle per second.

Horn: Speaker unit using a trumpet-bell shaped, flaring funnel to couple its sound vibrations to the surrounding air.

Hum: Unwanted low frequency tone. Usually caused by 50 Hz or 60 Hz AC and its harmonics.

Hysteresis Motor: A motor used in high quality turntables. Characterized by very constant speed regardless of power voltage fluctuations.

IC (Integrated Circuit): Solid circuit block containing the functions of numerous transistors, diodes, resistors, capacitors, etc.

IF (Intermediate Frequency): The frequency which results in a tuner when the incoming signal from the antenna is mixed with the oscillator frequency.

IF Transformer: Component in tuner or radio receiver used to couple or feed IF signal between successive amplifying transistors or tubes. Windings usually tuned with capacitors to resonate at the fixed IF frequency.

IHF (Institute of High Fidelity): Institute founded by American manufacturers of audio equipment, devoted to the improvement of audio technology, standardization of test and measuring methods, etc. "IHF" in audio specifications means that values were obtained in measurements according to IHF standards.

Image Rejection: The ability of a tuner to reject an RF signal which appears to be received but which is actually a sum or difference frequency of the tuner's oscillator and intermediate frequencies.

IM Distortion: (See Distortion). Signals in output caused by interaction of two or more input signals, but not harmonically related to them. Expressed as a percentage of total signal intensity. IM distortion is known to cause listener fatigue.

Impedance: Resistance to the flow of alternating current. Measured in ohms. It may vary with the frequency of the applied alternating current.

Induced Magnet Cartridge: A pickup cartridge in which both magnet and coils are fixed. The moving part is a tiny iron sheet.

Infinite Baffle: Type of loudspeaker mounting without an air path between front and rear of speaker cone.

Limiters: Circuits in an FM tuner that reject unwanted amplitude variations caused by atmospheric or ignition noise, producing an FM signal of constant amplitude.

Linearity: (a) Amplitude linearity, distortion of which produces harmonic distortion and intermodulation; (b) Frequency linearity referring to the straightness of a frequency response curve.

Line Output: Output terminal of a pre-amplifier, tape deck, etc., providing a signal for monitoring tape recording or supplying to a power amplifier.

Loudness Control (Contour): A circuit which counteracts the reduced sensitivity of the ear to very low and high notes at low volume levels.

Low Filter: A filter circuit designed to remove low frequency noises (rumble, hum, etc.) from the program.

Magnetic Cartridge: A cartridge which derives its electrical output signal from change effected in a magnetic circuit by means of some mechanical device such as a moving coil, moving magnet.

Monaural: One-eared. Something erroneously used to mean monophonic.

Monitoring: Listening to a program to judge or control the sound quality.

Monophonic: Recording, transmission and reproduction of sound via a single channel.

Moving Coil (MC) Cartridge: Magnetic cartridge in which the coils move and the magnet is fixed.

Moving Magnet (MM) Cartridge: Magnetic cartridge in which the magnet moves and the coils are fixed.

Multiplex: Transmission of two or more channels on a signal carrier so that they can be independently recovered by the receiver. In FM stereo: transmission of the L + R (sum) signal and L - R (difference) signal on main carrier and subcarrier, respectively.

Multipath Reception: Arrival of FM or TV signal via several paths of different length, due to obstructions, reflecting objects, etc.

Multiplex Decoder: See decoder.

Music Power: The maximum power available temporarily from a power amplifier. Also called "dynamic-power".

NAB (National Association of Broadcasters): Most widely used standard of tape recording techniques

Network: In audio, a frequency dividing network in a speaker system, or an electronic crossover network in a multi-amplifier installation.

Noise: Unwanted signal consisting of a mixture of random electrical agitations. Also, the sum of all unwanted signals such as hum, hiss, rumble, interference, distortion, etc.

GLOSSARY (contd)

Omni-Directional: Equal sensitivity or output in all directions. Said of antennae, microphones and loudspeakers.

Oscillator: An electronic circuit which generates an alternating current, e.g. the oscillator in a tuner produces the frequency used to mix with incoming radio signals.

Output Impedance: Impedance at output terminals of a device "seen" by the load.

Output Stage: Final stage of a power amplifier which supplies power to a loudspeaker.

Output Transformer: Transformer in tube power amplifier to couple output tubes to loudspeaker.

Overtone: A tone accompanying the fundamental in a musical note. May or may not be harmonic.

Phasing: Connections between power amplifier and speakers in a stereo system must be made in such a way that signals representing a central sound source cause the speakers to move equi-directional, i.e. in phase.

Phase-Shift: As a signal passes through a tuner, amplifier, etc. some frequencies may lag behind others. In a tuner, this phase shift can, in extreme cases cause loss of channel separation of FM stereo broadcasts. Phase shift in an amplifier results in blurring of stereo spatial localization and, in serious cases, can cause unstable amplifier performance and distortions.

Polarity: Positive and negative terminals of a battery of power supply, or the north and south poles of a magnet. Sometimes refers to phasing of cartridges and loudspeakers.

Power Bandwidth: The frequency range over which a power amplifier will produce at least half of its rated output power (according to IHF standard).

Power Handling Ability: Maximum amount of power that can be safely fed into a loudspeaker.

Pre-amplifier: A circuit unit which takes a small signal, e.g. from a tuner or turntable and amplifies it sufficiently to be fed into the power amplifier for further amplification.

Quietening: Term sometimes used instead of "muting" in FM tuners.

Rated Output Power: The maximum power that an amplifier will deliver continuously without exceeding its specified distortion rating. Also called continuous power output or RMS power output.

Ratio Detector: Circuit in an FM tuner for extracting audio signals from modulated radio or intermediate frequency signals.

Resistor: Circuit device which offers resistance to the flow of electric current. Resistors may be made from wire, metallic film, carbon and other materials.

Resonance: The tendency of a mechanical or electrical device to resonate at a particular frequency.

Reverberation: Reflection of sound from walls or ceilings. Echo. Can be created artificially by electronic or mechanical devices to imitate the effect of large halls.

RF (Radio Frequency): The frequency of a radio carrier wave. AM (MW) covers 535-1605 KHz, FM occupies 88-108 MHz.

RIAA (Record Industry Association of America): Usually refers to the disc recording and replay frequency response curves established as standard by this association.

Rumble: Low frequency noise resulting from vibrations in platter and motor of a turntable and from record warp.

Selectivity: The ability of a tuner to receive only the desired station while rejecting stations which are not required. Measured in decibels (dB).

Sensitivity: The input signal level required by a tuner amplifier, etc., to be able to produce a stated output. The lower the necessary input, the higher the sensitivity.

Signal-to-Noise Ratio: Ratio of desired signal voltage to unwanted noise and hum voltage. Expressed in decibels (dB).

Solid-state: Circuits using semi-conductors, e.g. transistors and integrated circuits (ICs).

Stylus: A finely machined piece of sapphire or diamond. The part of a phono pickup that traces the record groove.

Synchronous Motor: Type of AC electric motor in which motor speed is related directly to frequency of power supply.

THD (Total Harmonic Distortion): See Distortion.

Tone Control: Control circuits used to vary the proportion of bass and treble in the sound.

Trackability: Ability of phono cartridge to track record grooves of high amplitude and velocity. Also see Compliance.

Tracking Error: Deviation of centre-line of phono cartridge from tangential of record at point of stylus contact. Caused by tonearm geometry.

Transducer: Device for converting energy from one form to another; e.g. a loudspeaker converts from electrical to acoustic, a tape head converts from magnetic to electrical.

Transient: Abrupt change of state; sudden change in signal amplitude as caused by percussion instruments, "attack" of plucked strings etc.

Transient Response: The ability of an amplifier, cartridge or speaker to follow sudden changes in the level of a sound.

Tuner: The part of a receiver, or a separate unit, which receives radio broadcasts and converts them into audio frequency signals.

Tuner-Amplifier: Unit combining the functions of a tuner, pre-amplifier and power amplifier. Also called "Receiver".

Tweeter: A speaker designed to reproduce the high part of the sound spectrum.

Vent: Opening or port in a bass reflex loudspeaker enclosure.

Voice Coil: A coil of wire attached to a speaker cone. When placed in the field of a magnet, the coil responds to the alternating signal from the amplifier, moving the speaker cone back and forth in accordance with the frequency and intensity of the signal.

Woofers: A speaker designed to reproduce the low part of the sound spectrum e.g. organ, bass, etc.

Wow: Slow variation of pitch caused by speed fluctuation in tape or record movement.