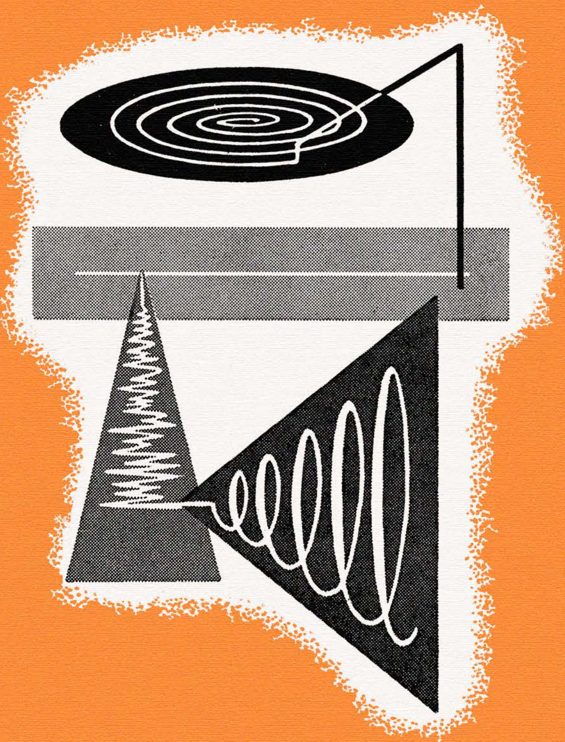
No. 226

fundamentals of HIGH FIDELITY

by H. BURSTEIN



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1. contents of a high-fidelity system ■

High fidelity is often associated with the purchase of a number of separate audio components and their assembly by the owner. This does not necessarily have to be the case, although it was true in the early days of high fidelity (prior to the 1950's). At that time, the really good equipment available at reasonable prices existed in the form of separate components, such as an FM tuner, a power amplifier, a speaker, etc. Today it is possible to buy the complete electronic portion of the system on one chassis, so that it is merely necessary to run two wires to the speaker in order to have an operating high-fidelity setup. If a phonograph, tape recorder, or TV set is to be incorporated, one cable goes from each of these units to the electronic center.

For those reluctant even to connect a few wires and mount the equipment in a cabinet, there are audio establishments that will connect all the components of the customer's choice and ensconce them in a cabinet for him. It is even possible to avoid choosing separate components by buying a *packaged unit* containing equipment preselected by the manufacturer and housed in a specially designed cabinet. The purchaser selects according to how the *entire* assembly sounds to his ear. For this reason, the following discussion does not seek to imply that purchase of separate components is the only way to go about bringing high fidelity into the home. However, in seeking to describe the contents of a high-fidelity system, it is helpful to approach the subject in terms of the separate components and what they do.

A high-fidelity system comprises four basic building blocks, as shown in Fig. 1-1. They are (1) the signal sources that produce an audio signal in electrical form; (2) the control amplifier (sometimes called a pre-

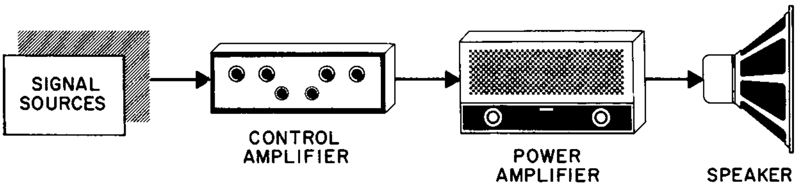


Fig. 1-1. Basic building-blocks of a home high-fidelity system.

amplifier), which amplifies and modifies the signal; (3) the power amplifier, which boosts the signal until it is strong enough to power (4) the loudspeaker, which converts the electrical signal into sound.

SIGNAL SOURCES

So far as high-fidelity systems are concerned, signal sources are essentially of three kinds: radio tuner (including TV sound), phonograph, and tape machine (Fig. 1-2). These units produce the electrical signals corresponding to audio information that are supplied to other components and ultimately drive the speaker. Other possible signal sources are microphones and such electronic instruments as an electric guitar or organ. (The use of a microphone in conjunction with a tape recorder is discussed later.)

At this point a distinction should be made between *signal* sources (tuner, phonograph, or tape recorder), which supply an electrical signal to the rest of the audio system, and *program* sources. The latter are external to the high-fidelity system and include the broadcast station, the record disc, and the prerecorded tape.

THE RADIO TUNER

The radio tuner picks up a program from the air. It may be an FM tuner, and AM tuner, or the sound portion of a TV set. FM and AM are

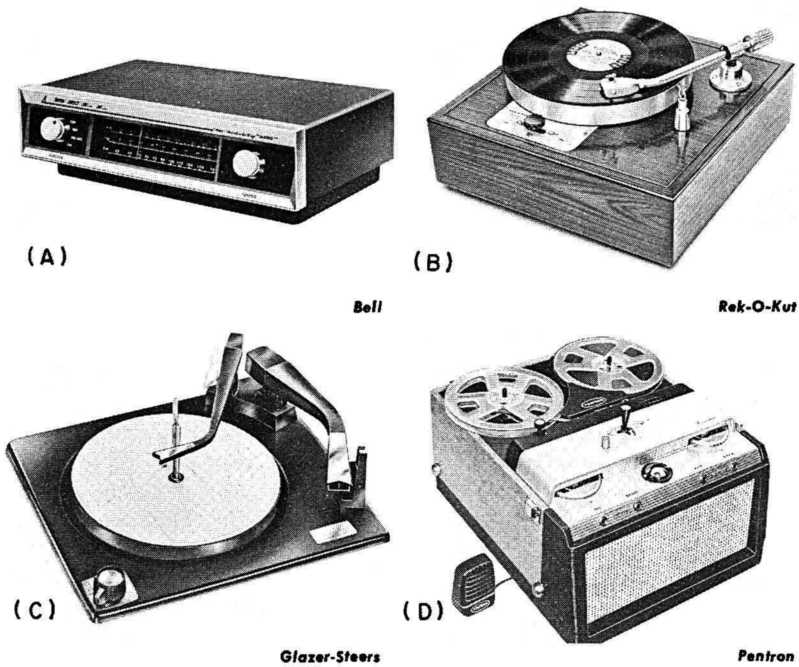


Fig. 1-2. Signal sources: (A) AM-FM tuner, (B) transcription turntable, cartridge and tone arm, (C) record changer, (D) tape recorder.

the two basic techniques used in putting information onto the radio wave that is propagated through the skies, intercepted by an antenna, and fed into the tuner. (The sound portion of a TV set uses FM, the picture portion uses AM.)

In FM (frequency modulation), the audio signal varies the *frequency* of the radio wave, that is, the number of times the wave vibrates per second. The tuner converts the variations in frequency to electrical signals corresponding to audio information. In AM (amplitude modulation), the audio signal varies the *amplitude*, or strength, of the radio waves. The AM tuner translates these amplitude variations into an audio signal.

FM can provide better-quality sound than AM. It is sensitive to a wider range of sound and is less subject to noise, fading, and interference from other radio stations. Theoretically, AM is capable of meeting high-fidelity requirements, although it seldom actually does so. High-fidelity AM requires proper atmospheric conditions, a relatively short distance between the station and the listener, the absence of interfering AM stations (ones that transmit radio waves at about the same frequency), and a carefully designed and rather expensive tuner. As a result, FM is considered the principal means of high-fidelity broadcasting, and our discussion of tuners shall for the most part be limited to the FM type. (Because TV sound is transmitted by FM, what we say of FM tuners applies to TV sets as well.)

Although FM and AM tuners may be purchased separately, it is common practice for the individual who wants both to purchase a single AM-FM unit. In equipment for stereophonic reproduction (discussed at the end of this chapter), it is often possible to operate both tuners at once. Usually, however, the FM and AM tuners share parts in common, so that only one can be operated at a time.

THE PHONOGRAPH

The phonograph has three principal subcomponents, as shown in Fig. 1-3: the turntable, tone arm, and cartridge. The turntable spins the record at a prescribed speed, and the grooves of the record are traced by a stylus (needle), which is part of the cartridge. As the stylus vibrates in accordance with the undulations of the groove, the cartridge produces an electrical signal that contains the audio information. The tone arm holds the cartridge and is constructed and mounted to allow the stylus of the cartridge to bear upon the record with the proper force as it traces the spiraling groove.

The manual type of phonograph or transcription turntable (Fig. 1-2B) takes only one record at a time, whereas the automatic type or changer (Fig. 1-2C) plays several records one after the other. The changer features convenience of operation and frequently provides acceptable performance. Transcription turntables are usually more accurate and steady in speed, as well as quieter and more durable.

Records are currently designed to operate at one of four speeds: 78 rpm (revolutions per minute), 45 rpm, 33⅓ rpm, or 16⅔ rpm. The 78 speed is declining in use, while 16⅔ is gradually coming into use. The slower the speed, the longer the playing time for a record of given size. Virtually all record changers provide at least 78, 45, and 33⅓ rpm, and the newer ones usually include 16⅔ rpm as well. (In transcription turntables it is less common to find all four speeds.)

Transcription turntables generally do not come equipped with tone arms and cartridges. The changer, on the other hand, always comes with a tone arm and sometimes with a cartridge as well. In a few cases, the tone arm and cartridge are an integral assembly, each designed for the other. In some cartridges, the stylus may be replaced by the user, whereas other cartridges must be returned to the factory when the stylus is worn.

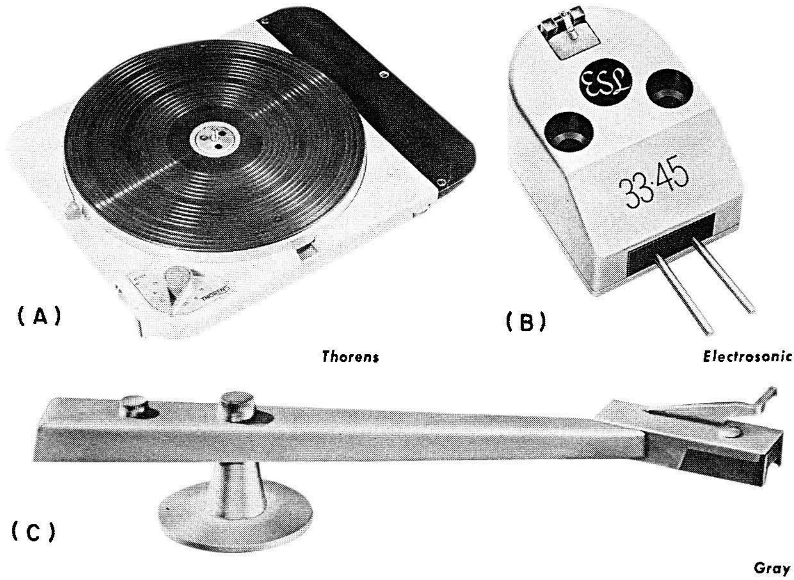
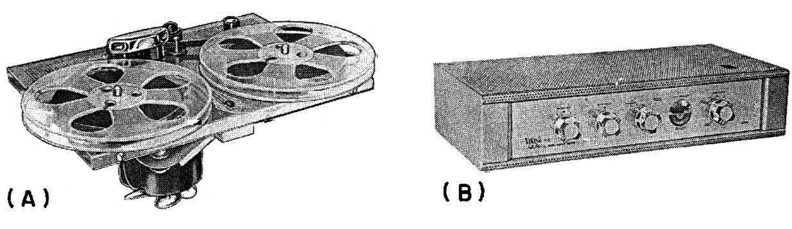


Fig. 1-3. Parts of the phonograph: (A) turntable, (B) tone arm, (C) cartridge.

Most of these cartridges, however, have a diamond stylus, with a life of 1,000 hours or more, so that replacement is quite infrequent. (What often happens is that by the time the stylus requires replacement, a better cartridge has appeared on the market, and the audiophile puts his money into a new cartridge instead of just a new stylus.)

THE TAPE RECORDER

The tape recorder has two basic subcomponents: the transport mechanism (Fig. 1-4A) and the tape amplifier (Fig. 1-4B). Most commonly, the two are sold as a unit, although they are available separately. (One manufacturer's transport cannot necessarily be coupled with another's



Viking Viking

Fig. 1-4. Parts of the tape recorder: (A) transport mechanism, (B) tape amplifier.

amplifier; in certain important respects, one must be fashioned to meet the needs of the other.) From a practical point of view, the tape recorder should be viewed as an integral unit; for discussion, however, we will consider the transport and the amplifier separately.

The transport is essentially a device that moves the magnetically coated tape from one reel onto a second reel at a specified speed. For home high-fidelity applications, the speed is generally 7.5 ips (inches per second). Where fidelity is a secondary consideration, 3.75 ips is also used. (The higher the speed, the greater the fidelity.) The tape moves past a series of devices called tape heads. When making a recording, the erase head first removes any previous signal on the tape. Then the record head places the audio signal on the tape in the form of magnetic patterns. (Technically, the audio signal causes the tape to be magnetized to varying degrees from one instant to the next.) In playing the tape, a playback head picks up the magnetic patterns, converting them back to an electrical audio signal. It is common practice in less expensive machines to employ the same head for playback and recording. Semiprofessional and professional machines generally have separate record and playback heads, which permit better performance because the technical requirements of each type of head are somewhat different. Where separate heads are used, the tape passes the playback head a fraction of a second after it passes the record head, permitting one to hear what has just been put on the tape. This enables the operator to monitor the tape as it is being recorded, thus making sure that he is obtaining a reasonably good facsimile of the original sound.

When recording a tape, an audio signal is fed into the tape amplifier from a microphone or from a high-fidelity signal source. If, for example, the signal comes from an FM tuner, he can connect the tuner directly to the tape recorder by a suitable cable, or connect the tuner to the control amplifier, which in turn, supplies the signal to the tape recorder (Fig. 1-5A; at the same time, the control amplifier continues to feed the audio signal to the power amplifier and speaker.) A third way to apply the audio to the tape recorder is to take the signal from across the speaker leads, as shown in Fig. 1-5B. This technique is the least satisfactory, because the signal available at the speaker may not be the same as that originally emanating from the tuner. (It may be less clean-sounding, and the balance of low, middle, and high notes may have been altered.)

Before feeding the signal to the record head, the tape amplifier increases the signal level, so that it can impress magnetic patterns corresponding to the audio information on the tape. The amplifier contains a record-level indicator, in the form of a meter, lamp, or other visual device,

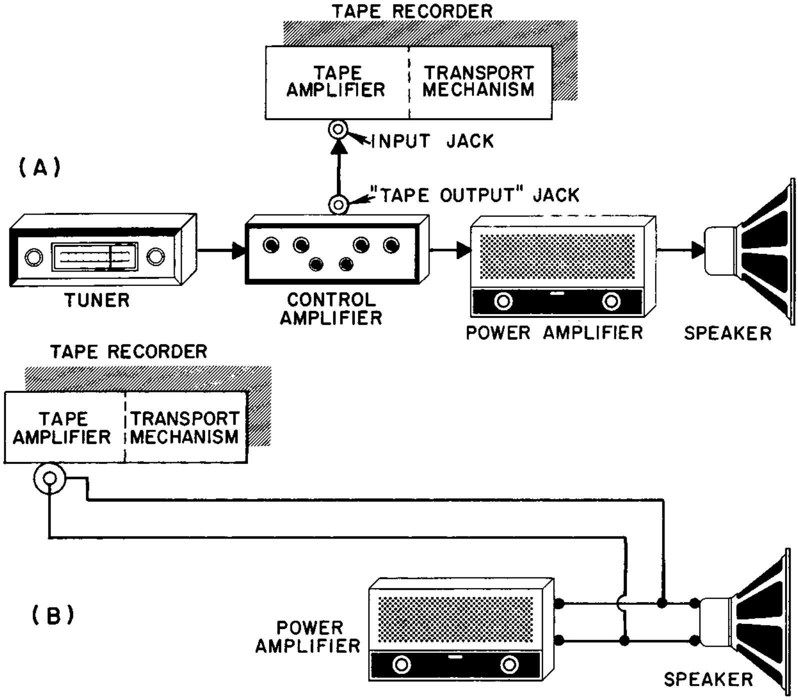


Fig. 1-5. Methods of connecting a tuner to a tape recorder: (A) conventional method, (B) least desirable method.

that warns the operator when he is applying too little or too much signal to the tape. Too little signal on the tape causes the amount of noise in the tape recorder and in the rest of the high-fidelity system (every system contains some noise) to be relatively loud compared with the recorded audio signal; too high a level causes overloading and coarse, grainy, or broken sound. The recording level is adjusted by a gain control.

In playback of the tape, the playback head develops a minute electrical signal corresponding to the audio signal on the tape. This very weak signal is built up by the tape amplifier to a level suitable for operating the following components of the audio system. Another important function of both the recording and playback amplifiers is compensation for certain inherent characteristics of the tape recording process. (This is discussed in Chap. 8.) A number of high-fidelity systems do not permit recording a tape, but only allow playback of prerecorded tapes. These tape machines fill a role analogous to that of the phonograph. The playback tape machine has become increasingly popular as commercially prerecorded tapes have appeared in growing numbers, particularly for stereophonic reproduction.

Many home recorders contain their own small speaker and power amplifier. Strictly speaking, however, these portions of the home tape recorder should not be considered a part of the high-fidelity system, since they seldom meet high-fidelity requirements. It is usually possible, however, to take the audio signal from a suitable point in the tape amplifier in order to feed it to the high-fidelity control amplifier.

THE CONTROL AMPLIFIER

The control amplifier (Fig. 1-6) can be considered the nerve center of a high-fidelity system because it controls all of the system's functions. A primary purpose of the control amplifier is selection of the signal source to be reproduced. In addition, the control amplifier raises the level of the electrical signal produced by the tuner, phonograph, etc. Amplification is particularly necessary for the signal delivered by the phonograph, because the type of cartridge most frequently employed in high-fidelity systems, the magnetic cartridge, puts out a very weak signal.

The signals from the various sources require not only amplification, but also *equalization*, the restoring of the balance between lows, midrange, and highs that is required due to natural characteristics of the sources themselves. An increasing number of control amplifiers are designed to accommodate a signal directly from the tape head and provide the necessary amplification and equalization. Thus, if one is interested solely in playing tapes and not in recording them, it is possible to get along with only a transport mechanism, at a considerable saving in cost.

An essential function of the control amplifier is to enable the user to vary the relative strength of the bass and treble tones with respect to the middle tones. This is done by bass and treble controls, often referred to as *tone controls*. The tone controls also enable the listener to compensate for the characteristics of his speaker or his room. Depending upon the particular speaker employed and the size, shape, and furnishings of the room in which it is used, there is apt to be an exaggeration of some tones and a reduction of others. These effects can be at least partially offset by the tone controls, which also allow the listener to suit his preferences in musical balance or to compensate for deficiences in his hearing.

Virtually every high-fidelity control amplifier makes provision for supplying the audio signal to a tape recorder. A jack marked "tape output" permits connection of a cable from the control unit to the input jack of a tape machine so that the signal going through the control amplifier can be

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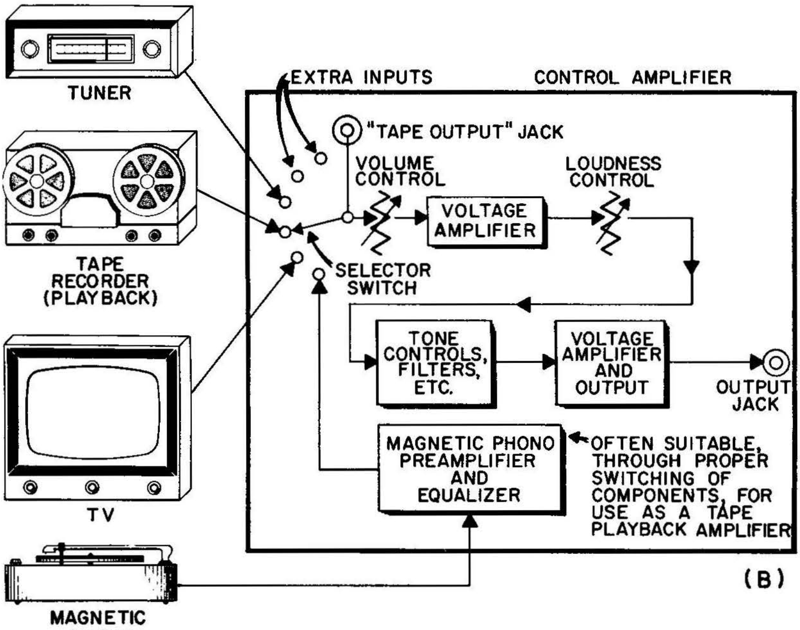

Fig. 1-6. A typical control amplifier (A) and its basic functions (B).

simultaneously recorded on tape. A separate cable connects the tape machine to the control unit when the tape is played back. Sometimes, one finds a "tape monitor" switch in a high-quality control amplifier. One position of the switch allows listening to the signal sources through the high-fidelity power amplifier and speaker system; the other position enables the listener to transfer to the signal coming from the tape recorder. (This is primarily intended for a tape recorder with separate record and playback heads, which, as previously discussed, can simultaneously make a recording and play back the recorded tape.)

Many high-fidelity control amplifiers contain a loudness control, discussed in detail in Chap. 7. This control automatically boosts the bass (and sometimes the treble) relative to the rest of the music as one reduces the volume, compensating for a natural characteristic of the human ear. The remaining features of the control amplifier are found in some, but not all, high-fidelity systems. The bass filter sharply attenuates very low notes, thereby suppressing undesirable sounds such as turntable rumble. The treble filter sharply attenuates the very high notes, thereby suppressing undesirable sounds such as scratches and ticks on a record. A presence control permits emphasis of the music that lies in the upper middle portion of the audio range, thus enhancing the illusion that the performers are in the same room with the listener.

A control amplifier sometimes permits the level of each signal source to be controlled separately. This is accomplished by one *level set* for each source, located at the input of the control amplifier. Level sets are usually accessible from the back of the control unit, because they are meant to be adjusted only infrequently. They have a threefold purpose: to prevent the signal source from presenting too much signal to the control amplifier, which might overload it; to permit equating the volume of sound obtained from each signal source; and to adjust the level of the incoming signal so that the loudness control will introduce automatic bass boost at the appropriate listening level.

THE POWER AMPLIFIER

In order to operate the loudspeaker, a substantial amount of electrical power is required. The signal from the control amplifier, however, is relatively weak in terms of the speaker's needs. It is the role of the power amplifier (Fig. 1-7) to boost the minute quantity of power emanating from the control unit to an amount sufficient to drive the loudspeaker. In every way aside from power, the electrical signal produced by the power amplifier should be an exact replica of that issuing from the control unit.

The power amplifier usually has few controls. Frequently, there are none at all, although many have at least one—an input-level set located at the very beginning of the amplifier. Its purpose is to cut down the amount of signal supplied by the control unit to the power amplifier. There are several good reasons for this, which will be discussed in Chap. 6.

An increasingly popular control in power amplifiers is the damping control. It regulates the relationship between the amplifier and the speaker

Fig. 1-7. A typical power amplifier.

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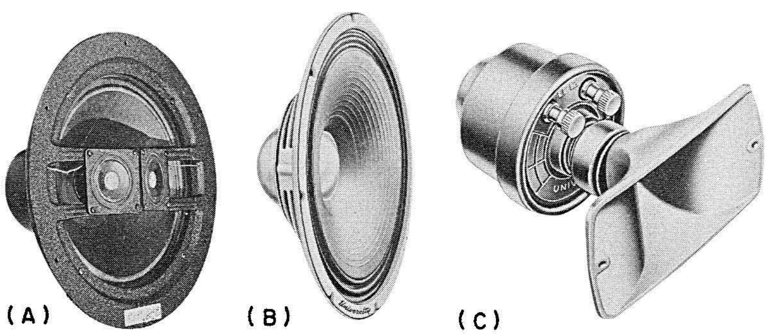
so as to achieve the fullest and most satisfactory reproduction of bass notes. Frequently, there are one or two other controls for adjusting the amplifier to produce its maximum power with the cleanest sound. These controls are usually preadjusted at the factory and meant to be touched only by the service technician. A few power amplifiers, however, include a meter or other device that makes it very simple for the audiophile himself to adjust the controls correctly.

THE LOUDSPEAKER

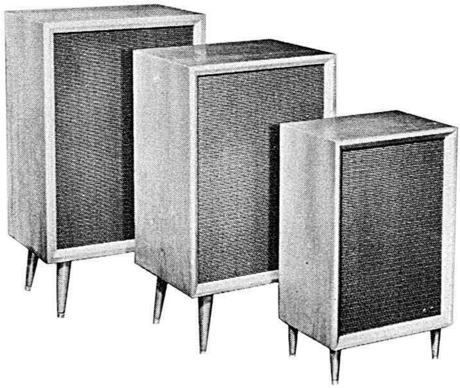
The task of the loudspeaker is to convert the electrical signal from the power amplifier into sound. The type of speaker in common use (Fig. 1-8) is essentially an electric motor that moves back and forth; it is called a *dynamic speaker*. It has a cone or diaphragm whose vibration moves the

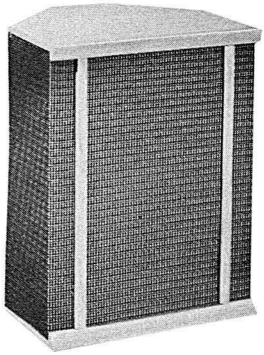


(A) (B) (C)

Bozak University University

Fig. 1-8. Typical loudspeakers: (A) coaxial, (B) single-cone full range, (C) tweeter.





Cabinart *Cabinart*

Fig. 1-9. Typical loudspeaker enclosures.

air in a manner that our ears recognize as sound. A relatively new form of speaker, the electrostatic, has a much larger sheet of material that is moved back and forth by the changing electrical signal. A still newer form of speaker, as yet quite rare, is the Ionophone, in which the air is moved directly by an electrical signal without an intervening cone, diaphragm, or other material. At the time of this writing, the electrostatic speaker is used principally for reproduction of the high and middle notes; the Ionophone for the high notes.

An essential part of a speaker intended for reproduction of the bass notes is its enclosure, or cabinet (Fig. 1-9). When we speak of the performance of a speaker, we must do so in conjunction with an enclosure, which makes a great difference so far as its quality is concerned. Although speakers are sold separately from enclosures, that is not how they are

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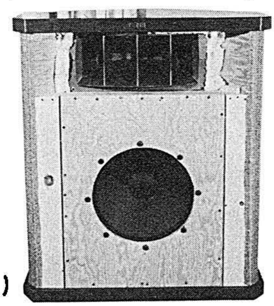

(B)

Acoustic Research *United Speaker Systems*

Fig. 1-10. Integral speaker units: (A) enclosure using a speaker specifically designed for it, (B) enclosure designed for commercially available speakers selected by the manufacturer (front panel removed).

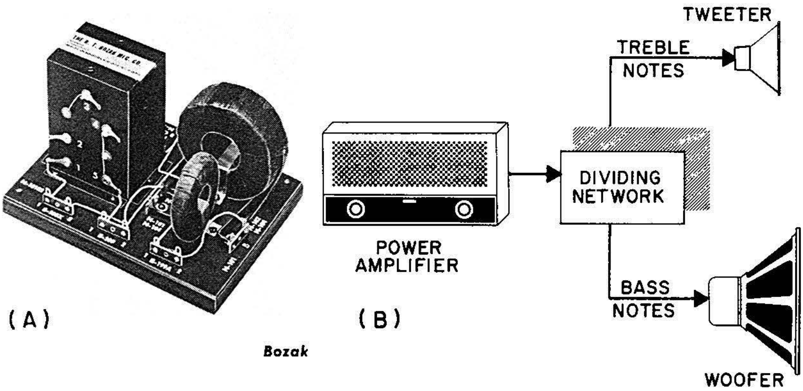

Fig. 1-11. A typical dividing network (A) and its location in the high-fidelity system (B).

meant to be used. There are a variety of forms of enclosure, some of which work better with some speakers than with others. In the last analysis, the ear of the individual purchaser must decide what is the best combination of speaker and cabinet. Because of the important relationship between speaker and enclosure, a number of manufacturers produce integral speaker systems (Fig. 1-10), selling the speaker and cabinet as a unit. Most of these integral systems are produced by speaker manufacturers, although some are made by independent producers who employ commercially available speakers in cabinets of their own design.

The enclosure may contain one speaker or several. Sometimes it houses several speakers of the same kind as a means of increasing the system's ability to deliver a large volume of clean sound, particularly of low tones. More often, it contains two or more different kinds of speakers, each designed to handle only a portion of the audio range. (A speaker designed to reproduce only a limited range of notes can ordinarily do a better job than one intended to reproduce the full audio range.) In two-way systems, *woofers* handle the low to middle notes, and *tweeters* handle the middle and high notes. It is also common, particularly in the most expensive speaker systems, to find a third unit called a *midrange* speaker (or *squawker*) that reproduces the notes between the bass and treble. In a few speaker systems, the audio range is divided among four speakers. The high notes are apportioned among two tweeters, one of which is a *super-tweeter* that reproduces only the very top notes. (Depending on the range that each speaker in a system is to handle, this nomenclature may vary; some three-way systems use a woofer, tweeter, and super-tweeter.)

In order to divide the music among two, three, or four speakers—low notes to one, middle notes to another, high notes to a third, and very high notes to a fourth—it is necessary to insert a dividing network or crossover (Fig. 1-11) between the power amplifier and the loudspeaker system. Although the crossover is electrical in nature, it should be thought of as part of the loudspeaker system because it has to be designed for the particular range of sounds to be covered by each speaker. (When one pur-

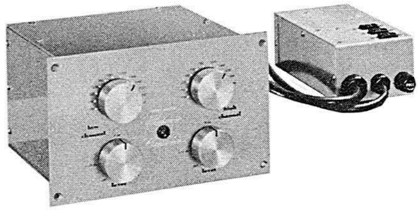

Fig. 1-12. A typical electronic crossover (A) and its location in the high-fidelity system (B).

(A)

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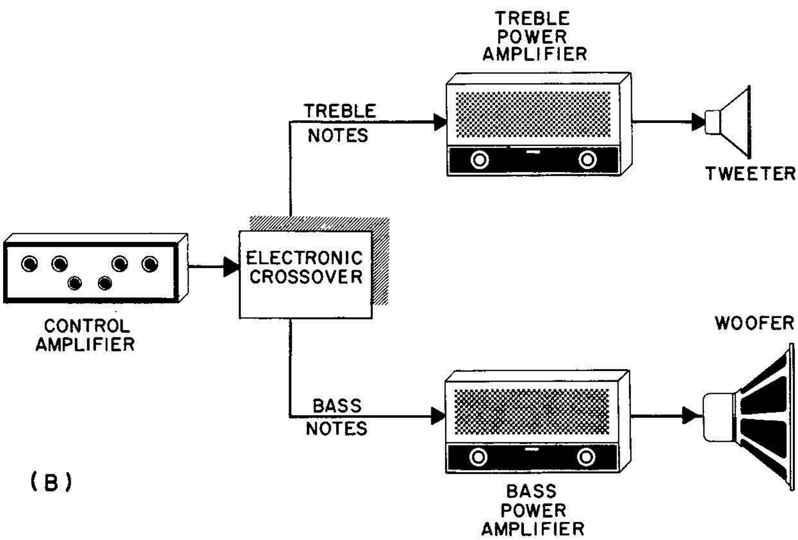

(B)

chases an integral speaker system containing two or more speakers, it contains its own dividing network.) In some high-fidelity systems, the sound ranges are divided between two amplifiers, using an electronic crossover (Fig. 1-12). The benefits gained at the power amplifier stage are similar to those obtained from restricting a speaker to a particular range.

Where more than one speaker, each covering a different part of the audio spectrum is employed, each *additional* speaker is often given a separate level control. The controls are ordinarily mounted on the enclo-

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Fig. 1-13. Combination components: (A) tuner-control amplifier, (B) control amplifier-power amplifier, (C) tuner-control amplifier-power amplifier.

sure, since they are connected between the dividing network and the speakers. They enable the user to balance one speaker against the other to obtain uniform reproduction of sound throughout the audio range. (This problem is discussed in Chap. 6.) The manufacturers of some speaker systems deliberately omit level controls because the speakers are balanced against each other at the factory, and because they feel compensation for characteristics of the room or the listener's ear can be achieved by the bass and treble controls on the control amplifier.

COMPONENT COMBINATIONS

For explanatory purposes, we have been discussing electronic high-fidelity components as separate entities. The audiophile, however, can purchase them in various combinations, which facilitates the assembly of a complete music system and may lead to a saving in cost.

Frequently, one finds an FM (or AM-FM) tuner combined with a control amplifier, as in Fig. 1-13A. The combination of a control amplifier and power amplifier (Fig. 1-13B) is even more common. It is even possible to purchase the complete electronic section of a high-fidelity system

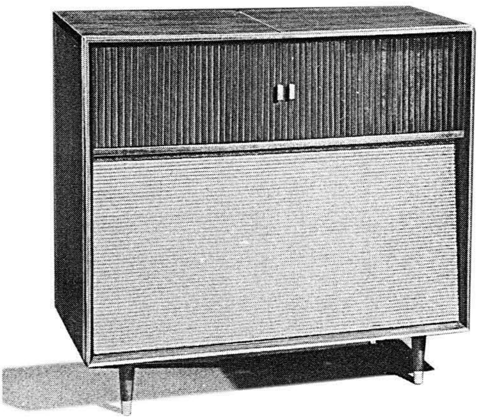

Fig. 1-14. A high-fidelity radio-phonograph.

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(Fig. 1-13C), combining the tuner, control amplifier, and power amplifier in one unit. (This combination is called either a *receiver* or an *electronics center.*) *Packaged units* contain a phonograph and speaker in addition to the receiver (Fig. 1-14), or a tape recorder as well; sometimes, they even include a TV set.

STEREOPHONIC EQUIPMENT

Stereophonic equipment is intended to handle two (or more) audio signals *(channels)* simultaneously. Each signal or channel differs slightly in its characteristics, and is fed into a separate speaker system, as shown

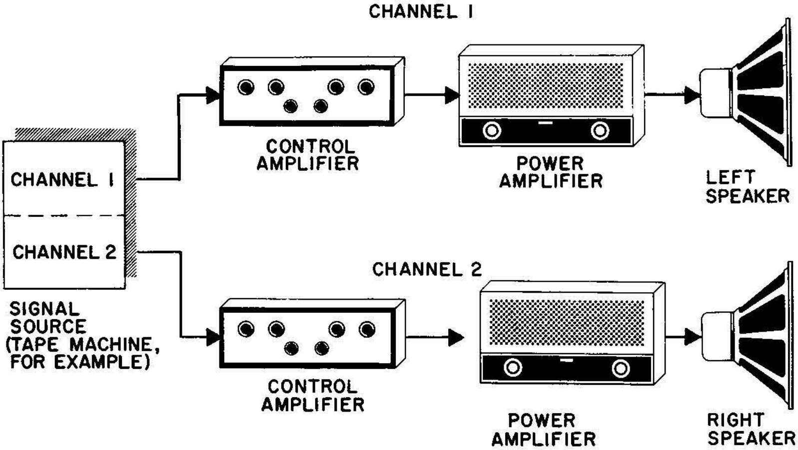


Fig. 1-15. Basic elements of a two-channel stereo system.

in Fig. 1-15. Stereo seeks to enhance the illusion of hearing the original performance by providing a wide spread of sound, giving spatial direction to the sound (violins on the left and woodwinds on the right, for example), and imparting greater *definition* to each instrument, making it stand out more clearly from the rest.

The program source is a stereo tape, a stereo record, an FM and an AM station (usually jointly owned) working together, two FM stations working in cooperation, or a single FM station employing the multiplex technique. At the original performance, two microphones spaced some distance apart are employed to pick up the sound; the electrical signal

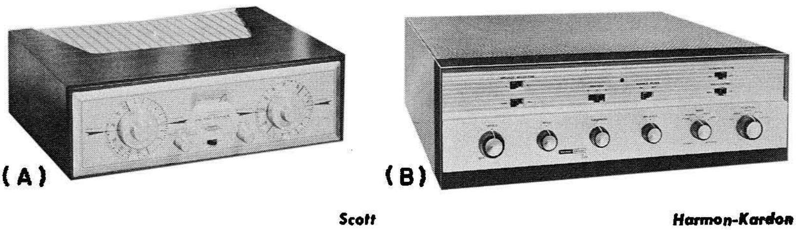


Fig. 1-16. Stereo components: (A) AM-FM tuner, (B) control amplifier-power amplifier, (C) speaker system.

produced by the microphone on the left becomes one channel, and that produced by the microphone on the right becomes the second channel. In the home, the positioning of the two speaker systems corresponds more or less to the positioning of the microphones at the original performance.

Stereo signal sources in the home consist of an FM and an AM tuner, (sometimes on one chassis, as in Fig. 1-16A), two FM tuners, an FM multiplex tuner, a stereo tape machine, or a stereo phono cartridge. Each of these sources is capable of reproducing two channels simultaneously. The rest of the electronic components may consist of two of each type or of dual equipment. For example, one might employ two control amplifiers and two power amplifiers terminating in two speakers. Or, one may use one control amplifier capable of handling two channels and one power amplifier capable of doing the same. There are also control amplifier-

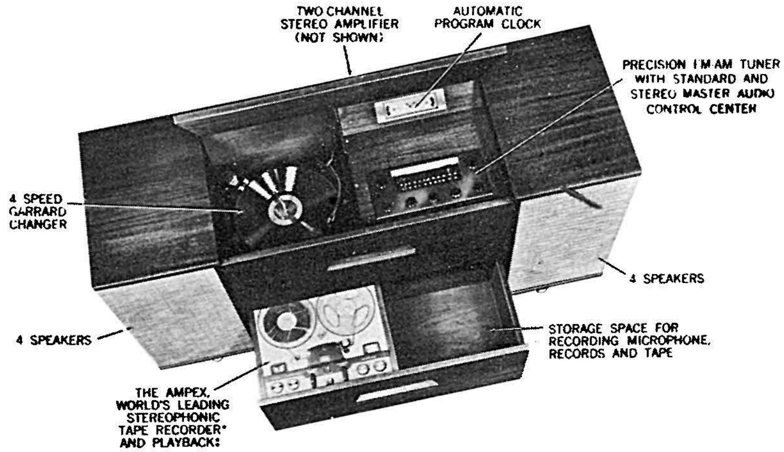


Fig. 1-17. A complete stereo package unit. *Fisher*

power amplifier combinations capable of handling two channels (Fig. 1-16B). They perform virtually all the functions of a monaural control amplifier and power amplifier, but for two channels. Some of these functions are coordinated, so that turning a single control produces the desired result in both channels at once. Finally, it may be pointed out that although two separate speakers systems are required for stereo, it is possible to obtain them in one housing, as shown in Fig. 1-16C. It is even possible to purchase a complete stereo package (Fig. 1-16D).

2. the meaning and significance of the decibel ■

Before attempting to describe the attributes of high-fidelity performance, it is necessary to explain a few key terms that are constantly employed in discussions of high fidelity. A clear understanding of such expressions as *frequency response, distortion, decibel,* and *signal-to-noise ratio* etc., is as important to an appreciation of high fidelity as heat is to cooking. The decibel receives priority of treatment partly because it is so frequently used to describe various aspects of audio, and partly because a grasp of its meaning facilitates comprehension of other terms.

GENERAL MEANING OF THE DECIBEL

The decibel (db) is a measure of the loudness of sound. It is not an absolute measure, such as a yard, a pound, or a degree of temperature; rather, it is a relative thing — a comparison between quantities. To be exact, the *decibel is a ratio between two amounts of power.*

The decibel is associated with acoustic power, that is, the amount of sound energy generated by a source of sound. In speaking of audio equipment, for example, one could compare the volume of sound produced by two different speakers, as in Fig. 2-1. If the first generates twice as much acoustic power as the second, one could say that the ratio of one to the other is 2. A different way of saying the same thing is to state that the power produced by one speaker is a certain number of decibels higher than the other (in this case 3 db, as we shall soon see); or, one can state that there is a difference of 3 db between the two speakers. Both state-

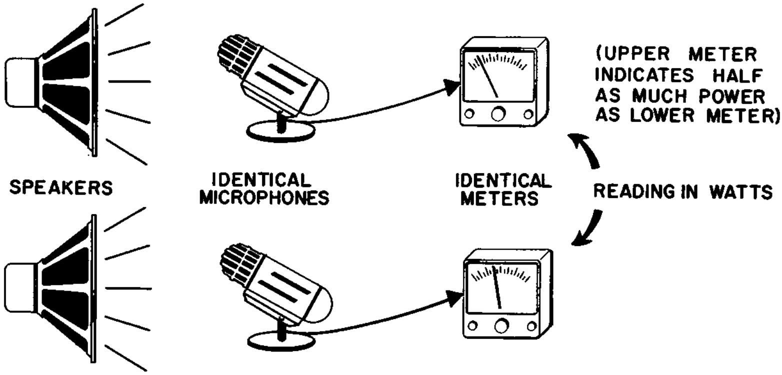


Fig. 2-1. Comparing the acoustic power output of two speakers.

ments signify a ratio of 2 between the amounts of power involved. The higher the ratio, the greater number of decibels difference.

Acoustic power—vibration of the air—is ordinarily generated by means of electrical power. Both forms of energy are measured in watts. (When dealing with very small quantities of power, the milliwatt—one-thousandth of a watt—is commonly used.) The power amplifier supplies electrical power to the loudspeaker, which converts it into acoustic form. The conversion is far from 100% efficient, because a large part of the electrical power is wasted in the generation of heat instead of being converted to acoustic energy. Nonetheless, there is a direct relationship between *variations* in electrical power and those in acoustic power. For example, if 10 watts of electricity are converted by the loudspeaker into 1 watt of acoustic power, doubling the electrical-power input would double the acoustic-power output. Thus the ratio between the two values of acoustic power (2 watts to 1 watt) is the same as the ratio between the two values of electrical power (20 watts to 10 watts). For this reason, in comparing the performance of two amplifiers it is customary to speak in terms of decibels when no loudspeakers are connected (Fig. 2-2).

A comparison of power does not necessarily have to involve two amplifiers or speakers. To illustrate, an amplifier may be compared with itself at different times or for different notes. Thus an amplifier may produce 1 watt when one signal is fed into it and 2 watts when another signal is fed into it. One can then say that the amplifier output has changed 3 db or, that its output is 3 db greater for one input signal than for the other.

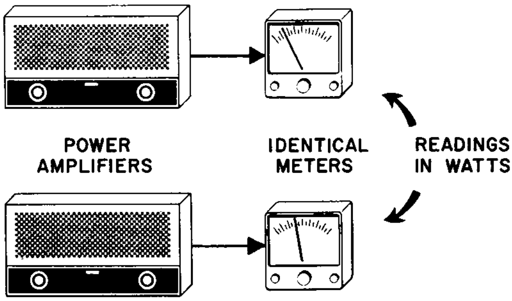


Fig. 2-2. Comparing the electrical power output of two power amplifiers.

Either of the input signals may differ in intensity or one may be a low note while the other is a high note, with the amplifier responding differently to each of them.

PRECISE MEANING OF THE DECIBEL

Now we can consider the precise, numerical meaning of 1 db. *Given two amounts of power, 1 db denotes a ratio of 1.259 between them.* (Very likely it seems strange that 1 db should correspond to such an odd ratio, but it will soon be found that the underlying reason makes complete sense.)

To illustrate the meaning of 1 db: if speaker A produces 1.259 times as much power as speaker B, the difference between them is 1 db. We could also say that the output of speaker A is 1 db higher than that of speaker B. Sometimes decibels are expressed as a negative rather than positive quantity. For example, speaker B may be referred to as having an output of —1 db, using speaker A as a standard of comparison. The minus sign simply means that B has the smaller output compared with the reference speaker. If the two amounts of power being compared are the same, their ratio of course is 1. Since no difference exists between these quantities, this ratio is called 0 db.

Let us investigate the meaning of successive additions of 1 db. We already know that 0 db denotes no difference between two amounts of power, and that 1 db indicates that one power is 1.259 times as great as the second. A further increase of 1 db means that the power has again in-

TABLE 2-1

DECIBELS AND THEIR CORRESPONDING POWER RATIOS

Number of Decibels	*Ratio Between Two Powers**	*Number of Decibels*	*Ratio Between Two Powers**
0	1.000	6	3.981
1	1.259	7	5.012
2	1.585	8	6.310
3	1.995	9	7.943
4	2.512	10	10.000
5	3.162		

*Each ratio, except the first, is obtained by multiplying the preceding ratio by 1.259, which is the equivalent of adding 1 db.

creased to 1.259 times its previous value. We multiply twice by 1.259: 1:259 $\times$ 1.259 = 1.585; therefore 1.585 is the ratio corresponding to 2 db. Continuing, 3 db means that we are adding 1 db to 2 db. Since 2 db represents a ratio of 1.585, we multiply this figure by 1.259 in order to add 1 db to it. This gives us 1.585 $\times$ 1.259 = 1.995, which is the ratio corresponding to 3 db. Since 1.995 is very close to 2, it is commonly stated that a ratio of 2 corresponds to 3 db. To get to 4 db, we multiply 1.995 by 1.259 yielding 2.512.

This process can be carried on indefinitely. But for our purposes we can learn all we have to know about the numerical meaning of the decibel simply by carrying the process out to 10 db. This is done in Table 2-1. The last step in the table brings us to a key point: 10 db corresponds to a ratio of 10 between two amounts of power. In fact, the definition of the decibel is based upon this relationship. Some time ago it was agreed by engineers that a ratio of 10 between two amounts of power should be called a bel. However, it was found that a bel was too large a unit for many applications, just as a mile is too large for measuring small distances. Therefore it was decided to divide the bel into 10 units, each called a decibel, meaning one-tenth of a bel. Table 2-1 shows that ten additions of 1 db, each denoting successive multiplication by 1.259, give a ratio of 10, corresponding to 10 db.

The vital characteristic of the decibel, is that it permits the simple process of addition to be substituted for the more complex process of multiplication. Moreover, the use of db avoids dealing with large cumbersome figures. We have seen that every time we add 1 db we are multiplying by 1.259. It doesn't take very long for these successive multiplications to rise to big numbers. Thus 30 db represent a ratio of 1000, 60 db a

TABLE 2-2

POWER RATIOS CORRESPONDING TO VARIOUS MEDIUM AND LARGE DECIBEL VALUES

Number of Decibels	*Ratio Between Two Powers*	*Number of Decibels*	*Ratio Between Two Powers*
20	100	80	100,000,000
30	1,000	90	1,000,000,000
40	10,000	100	10,000,000,000
50	100,000	110	100,000,000,000
60	1,000,000	120	1,000,000,000,000
70	10,000,000		

ratio of 1,000,000, and 100 db a ratio of 10,000,000,000. Table 2-2 lists the power ratios corresponding to some of the medium and large decibel values encountered in audio.

Table 2-1 permits us to convert any number of decibels into the equivalent ratio between two amounts of power. All we have to do is remember that adding decibels is equivalent to multiplying ratios. Let us consider the meaning of 25 db. The table tells us that 10 db corresponds to a ratio of 10. This accounts for the first 10 db in our problem. The next 10 db similarly correspond to a ratio of 10. Since adding decibels means multiplying ratios, 20 db therefore signifies 10 $\times$ 10, which equals 100. There are now 5 db left to account for. Table 2-1 shows that 5 db correspond to a ratio of 3.162. So to add 5 db to 20 db, we multiply 3.162 by 100, which yields 316.2. Thus 25 db denote a ratio of 316.2 between two amounts of power.

In solving this problem, a time-saving and possibly error-saving step is available. Twenty db, as we have seen, represents two multiplications by 10, yielding 100. Note that 100 consists of 1 followed by two zeros, one zero for each 10 db. Therefore, instead of going through the process of multiplying by 10, we may simply attach two zeros to the number 1. This principle may be illustrated by another example. What is the power ratio corresponding to 60 db? We add six zeros—one for each 10 db—to the number 1, and thus obtain 1,000,000 as the answer, as may be checked against Table 2-2.

THE MEANING OF THE DECIBEL APPLIED TO VOLTAGE

The concept of the decibel stems from a relationship between amounts of *power*—acoustic or electrical; but decibels can also be used when referring to a ratio between voltages. Electrical power requires a combination of current (a supply of electrons) and voltage (analogous to a pressure that causes the electrons to flow and perform work). In audio components, such as the control amplifier, tuner, and tape recorder—actually, in all components but the power amplifier—only very small

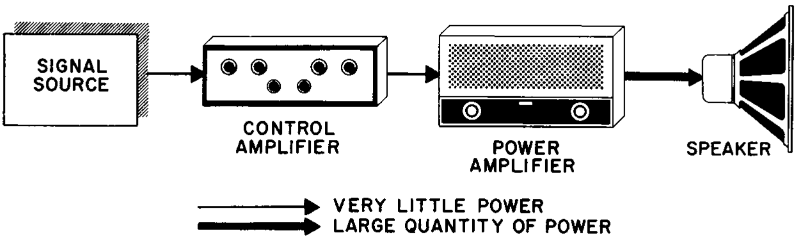


Fig. 2-3. Power levels in a high-fidelity system.

signal currents and small amounts of power are involved (Fig. 2-3). With components other than the power amplifier, we think in terms of *voltage* rather than power.

Voltage and power are related, as implied in the last paragraph. To be precise, power is proportional to the square of voltage. Thus as voltage doubles, power is increased (2×2) or four times. This is the case because every change in voltage causes a corresponding change in current. When the voltage doubles, current doubles, and power (voltage $\times$ current) quadruples.

If power varies with the square of the voltage, it may be said that voltage varies with the *square root* of the change in power. (The square root of a number is the quantity which is multiplied by itself to produce the number in question; the square root of 9 is three, because 3×3 equals 9.) To illustrate the voltage-power relationship, suppose that power increases to 16 times its previous value. This implies that voltage has increased to 4 times its former value, because 4 is the square root of 16. Accordingly, a given number of decibels implies a certain power ratio; and in turn it implies a voltage ratio that is the square root of the power ratio. In our example, the power ratio is 16 and the voltage ratio is 4.

If 1 db represents a power ratio of 1.259, then 1 db represents a voltage ratio that is the square root of 1.259, which is 1.122. In terms of a voltage ratio, 2 db signifies 1.122×1.122, which is 1.259. A voltage ratio of 3 db signifies multiplying again by 1.122, yielding 1.259×1.122, which is 1.413. To get to 4 db, we multiply 1.413×1.22, which is 1.585. Table 2-3 carries this process up to 20 db, which is all that we need to convert

TABLE 2-3
DECIBELS AND THEIR CORRESPONDING VOLTAGE RATIOS

Number of Decibels	*Ratio Between Two Voltages**	*Number of Decibels*	*Ratio Between Two Voltages**
0	1.000	11	3.548
1	1.122	12	3.981
2	1.259	13	4.467
3	1.413	14	5.012
4	1.585	15	5.623
5	1.778	16	6.310
6	1.995	17	7.079
7	2.238	18	7.943
8	2.512	19	8.913
9	2.818	20	10.000
10	3.162		

*Each ratio, except the first, is obtained by multiplying the preceding ratio by 1.122 which is the equivalent of adding 1 db.

any number of decibels into a voltage ratio. (The reason for carrying this table to 20 db is that 20 db is equivalent to a *voltage* ratio of 10.)

Table 2-3 is used in the same manner as Table 2-1,except that every 20 db, instead of every 10 db as in the case of power, corresponds to a voltage ratio of 10. As an example, let us find the meaning of 48 db in terms of a voltage ratio. We add two zeros—one for each 20 db—to the number 1 in order to represent the first 40 db. This gives us a voltage ratio of 100 as equivalent to 40 db. Table 2-3 shows that 8 db correspond to a voltage ratio of 2.512. Therefore 48 db correspond to 2.512×100, which is a voltage ratio of 251.2.

SIGNIFICANCE OF THE DECIBEL

It has already been pointed out that decibels are less cumbersome than the corresponding power or voltage ratios, which often become very large. Another and more compelling reason for use of the decibel is because it corresponds to the way the human ear reacts to changes in the volume of sound. As the sound level increases, each unit of increase has a different effect upon the ear. Starting from a low sound level, at first an additional unit of sound seems to make an appreciable difference in apparent loudness. But successive additions of a given unit of sound become less and less effective, and eventually a point is reached where the ear senses no increase in loudness despite the physically measurable increase in acoustic power.

The behavior of the ear may be described by saying that it tends to hear changes in loudness as being equal when the acoustic power is increased by the same *ratio* rather than by the same *amount*. Assume, for example, that a sound seems "a little louder" when it increases from 1 milliwatt to 2 milliwatts. In order for the sound to become still louder by

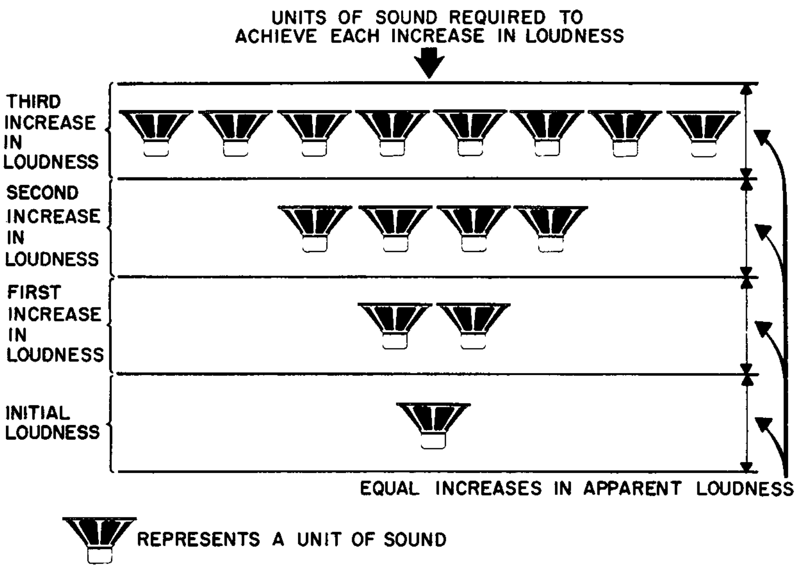


Fig. 2-4. Increases in acoustic power required to cause equal increases in apparent loudness.

the same amount as before another increase of 1 milliwatt is insufficient. Now it is necessary that, as before, the acoustic power double, which is an increase to 4 milliwatts. If the apparent loudness is once more increased "a little," it will be found that the acoustic power has doubled again. From Fig. 2-4 it is clear how quickly the units of sound must multiply in order for this same "little" increase in loudness to take place three more times.

We have already seen that the decibel is used to describe a ratio between two amounts of power. Inasmuch as the ear tends to interpret equal ratios between acoustic levels as equal differences in apparent loudness, the decibel is a most useful and realistic device for comparing different sound levels or for measuring changes in sound level. A given increase in number of decibels generally describes the same increase in loudness, regardless whether we start with a soft or loud sound. It must be understood that this is an approximation rather than a precise rule,

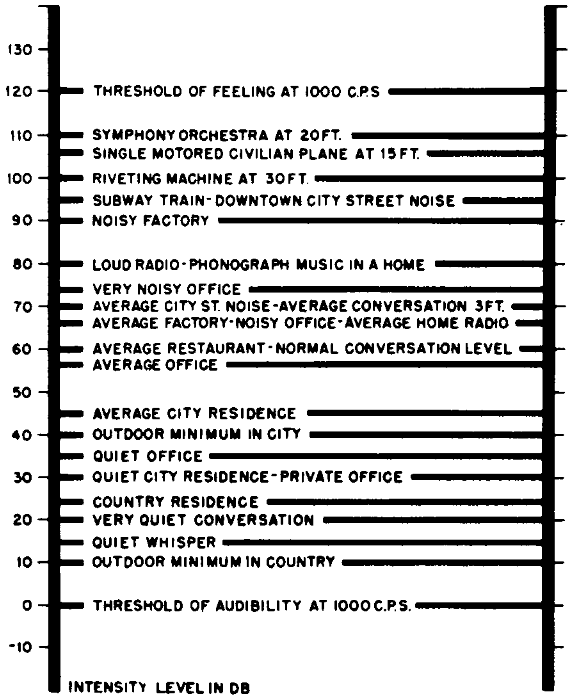


Fig. 2-5. Approximate db levels of various sounds relative to the threshold of hearing.

but it is accurate enough to make the decibel the preferred unit for measuring changes in loudness as apparent to the human ear.

How many decibels of increase or decrease in acoustic power are required to produce an appreciable effect upon the ear? As a general rule, a difference of about 1 db—a ratio of 1.259 between acoustic-power levels—is barely perceptible if one is listening to a single note. Where mixed sounds are involved, for example in music, 3 db—a ratio of 2 between power levels—is customarily considered to be the first truly noticeable difference. To the ear, an increase of 3 db appears to be a very slight change in loudness, certainly nowhere near a doubling of sound.

In evaluating the performance of audio equipment, it is often desired to compare the relative magnitudes at which different sounds or notes are produced. These comparisons of magnitude are commonly made in terms of decibels. For example, it may be stated that a control amplifier can reduce rumble from a phonograph 10 db below the rest of the audio spectrum.

Often a sound is identified as having a level of a certain number of db. To illustrate, a symphony orchestra, as heard from a distance of 20 feet, is said to produce a level of about 110 db during the loudest passages. But we have learned that decibels denote a ratio between *two* things. What, then, is the sound of the orchestra being compared with? The comparison quantity, or standard of reference as it should be termed, is an implied one; it is a specific, agreed-upon volume of sound, approximately the lowest level audible to the average human ear. This threshold of audibility is arbitrarily called 0 db, and all sounds are related to it. Thus a symphony orchestra in full force produces a sound level 110 db above—100,000,000,000 times as great as—the minimum audible level. Other sounds are similarly rated with respect to the 0 db standard. For example, a solo voice might be said to produce a level of 60 db, noise in the home might be at a level of 30 db, and so forth. Figure 2-5 shows the approximate levels of various familiar sounds relative to the threshold of hearing.

3. frequency response, distortion, and noise ■

This chapter deals with the three pivotal aspects of high-fidelity performance: frequency response, distortion, and noise (including hum). Although there are other criteria of performance, as will be revealed in later chapters, these supplementary factors for the most part boil down to the principal ones discussed here.

ALTERNATING CURRENT

If an electric current flows first in one direction, reverses itself to flow in the opposite direction, again reverses itself to flow in the original direction, and continues to change in this manner, we call it an alternating current (ac). We think of the current as a host of electrons able to heat bodies, turn motors, produce light, etc. The flow of current in one direction through a conductor (say a wire) may be termed positive, and the flow in the opposite direction may be termed negative. One cycle of alternation consists of a flow first in one direction, then in the opposite direction, and back to the starting point (Fig. 3-1).

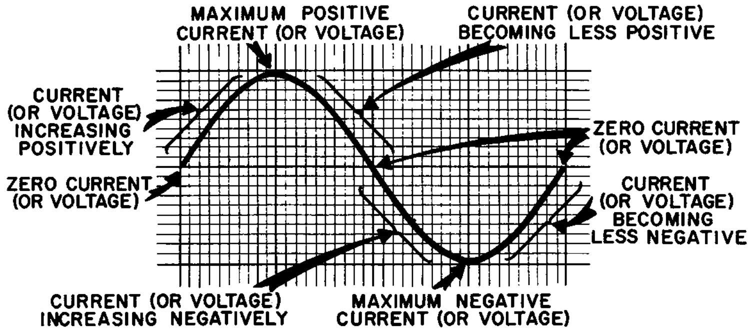


Fig. 3-1. One cycle of an alternating current or voltage.

Although Fig. 3-1 represents electric current, it may also be considered representative of voltage since the two are directly related to each other. (Voltage may be thought of as the pressure that causes current to flow through an electrical conductor, just as another type of pressure causes water to flow through pipes.) In the case of alternating current, the voltage is also constantly changing.

The starting point of Fig. 3-1 represents no current flow at all. As the curve starts upward, this represents a positive flow of current, which eventually reaches a maximum amount. When the curve begins to go downhill, it indicates a decrease in amount of current flowing in the positive direction. Again a point of no current flow is reached. Current then starts to flow in the negative direction, increasing in amount until it attains a maximum. Again it begins to decrease in amount, until zero is reached. This completes one cycle: from zero to zero, encompassing both a positive and negative current flow.

Figure 3-1 can also represent an alternating voltage. The starting point corresponds to zero voltage. The initial rise corresponds to increasing positive voltage. The crest corresponds to maximum positive voltage. The subsequent decline corresponds to decreasing positive voltage, followed by zero voltage and then increasing negative voltage. The trough corresponds to maximum negative voltage, after which there is a return to zero voltage. A cycle does not necessarily have to start at the point of

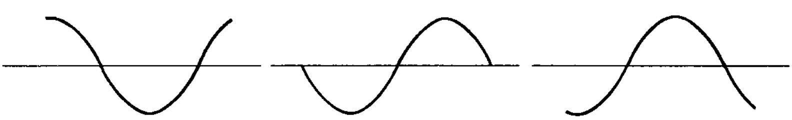


Fig. 3-2. Other representations of one a-c cycle.

zero current flow. It can start anywhere. So long as we return to the point where we started tracing current flow (or voltage) we have described one cycle. Figure 3-2 shows several other representations of one complete cycle.

The curve shown in Fig. 3-1 resembles a wave, which (for mathematical reasons) is called a *sine wave.* The sine wave is a fundamental building block in electricity and, as we shall soon learn, in the world of

Fig. 3-3 Complex waves made up of several sine waves of different frequencies.

sound. By combining different sine waves, we can form other waves that have very different shapes, such as those shown in Fig. 3-3.

FREQUENCY

How long does a cycle take? It can take a minute, a second, a fraction of a second, any amount of time. For example, let us consider the familiar electric current that we generally find in our homes. It is an alternating current that goes through 60 complete cycles every second. Another way of saying this is to state that the current in our homes has a 60-cycle frequency. The term *frequency* tells us how many cycles occur within a given timespan. (The accepted practice is to speak of frequency in terms of *cycles per second*; when no reference is made to a time period, as is often the case, it is understood to be a second.)

Electric "waves" can have many frequencies. Some involve millions or billions of cycles per second, whereas others represent thousands or hundreds or just tens of cycles per second. Waves and frequencies are not restricted to electricity; air and water can be agitated to form waves of various frequencies. Although we cannot see the waves in air as we can those in water, we hear them, provided their frequency is within the range of about 16 to 20,000 cycles per second. (Not that all people hear over so wide a range—20 to 15,000 cycles would be more typical.)

These sound waves are caused by a vibrating body. This body, as it pushes against the air, imparts motion to the molecules of air adjacent to it. These molecules transfer the motion to the molecules a little farther along, and so on, until the wave motion in the air impinges on our ears and registers in the brain as sound. Do not think that the molecules that originally were in contact with the vibrating body are the same ones that strike the ear. The molecules themselves do not travel through the air with the sound wave. Rather, their *motion* is transferred by one group of molecules pushing against the next. (This is similar to wave action in the ocean; an ocean wave may travel hundreds of miles, yet the water in the wave has traveled hardly at all in the direction of the wave.)

In the case of a sound wave, the vibrating body first pushes against the adjacent molecules of air, causing them to compress. Then as the

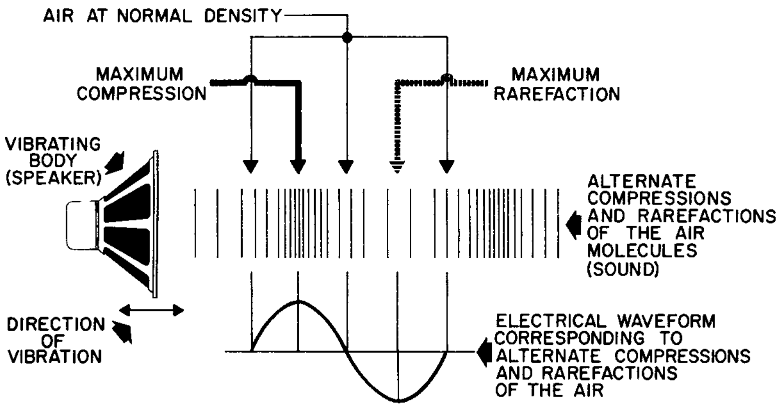


Fig. 3-4. The transmission of sound in air.

vibrating body moves in the opposite direction, it leaves behind it a partial vacuum (rarefaction), which draws the compressed air molecules back. As the body vibrates, it causes alternate compressions and rarefactions of the adjacent air, and it is this motion of the molecules that is communicated to adjoining molecules until it reaches our ears. From one compression to the next, or from one rarefaction to the next, is one cycle, as shown in Fig. 3-4.

Sound waves can be converted into electrical waves by a microphone (Fig. 3-5A). The sound wave (alternate compressions and rarefactions of the air) causes a pressure-sensitive element of the microphone to vibrate back and forth. This mechanical vibration is converted by another element in the microphone into an electrical current that alternates in accordance with the motion of the pressure-sensitive element. The electrical signal thus created, if the sound wave consists only of one frequency, is a sine wave like that shown in Fig. 3-1. Of course, a sound wave can consist of several different frequencies transmitted through the air simultaneously, in which case the resultant electrical wave is not a sine wave, but a complex wave, like those shown in Fig. 3-3. A device,

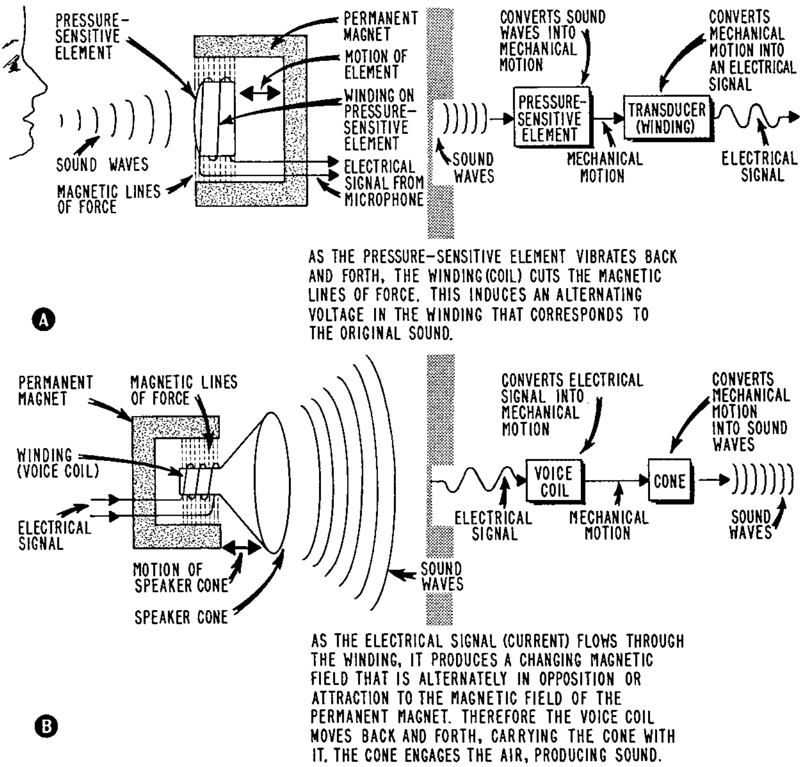


Fig. 3-5. Basic elements of a microphone (A) and a dynamic speaker (B).

such as the microphone, that converts sound into electricity is called a transducer. One can also have a transducer that works in the opposite fashion, converting electrical signals into sound—a loudspeaker (Fig. 3-5B).

In audio we are generally concerned with the range of about 16 to 20,000 cycles. Although there are acoustic vibrations which we know take place at rates lower than 16 or higher than 20,000 cycles, they are not generally considered as sound. The audio spectrum is often thought of as being divided into a bass range (low-pitched notes) and a treble range (high-pitched notes). In a general sense, the bass portion may be said to include the frequencies below 800 cycles, whereas the treble includes those above 800 cycles. Sometimes the audio range is divided into bass, mid-range,and treble. In this case the bass may be thought of as below 200 cycles, the mid-range as from 200 to 2,000 cycles, and

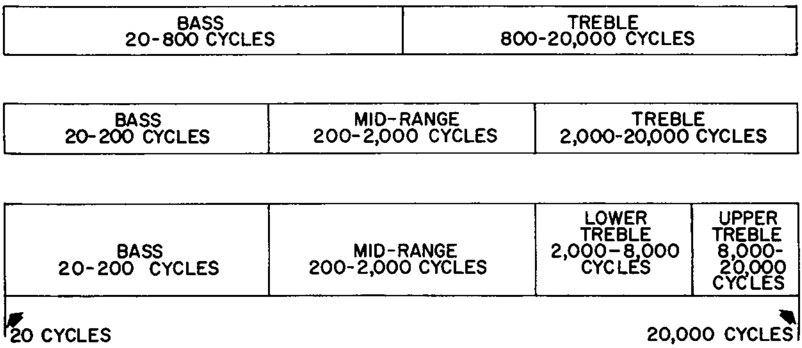


Fig. 3-6. Various methods of dividing the audio range.

the treble as above 2,000 cycles. Sometimes the treble is further divided so that one portion covers 2,000 to 8,000 cycles and the other 8,000 to 20,000 cycles. These divisions are shown in Fig. 3-6.

FREQUENCY RESPONSE

What is meant by frequency response? This term concerns the manner in which a device—say, a power amplifier—treats the relative amplitudes of various frequencies. If frequencies over a given range come out of a device at the same amplitude *relative to each other* as they went in, the device is said to have *flat response* over this range. If, for example, frequencies between 16 and 20,000 cycles of equal strength are fed into a power amplifier, and if all these frequencies are still of equal strength at the amplifier's output, response could be *described as flat within 16-20,000 cycles.* We do not always put signals of equal strength into an amplifier. Assume, for example, that at 100 and 10,000 cycles the input frequencies are 6 db below the level of 1,000 cycles (in voltage terms, a level about one-half as much, as was shown in Table 2-3). Flat response would mean that 100 and 10,000 cycles at the output of the amplifier would still be at a level 6 db below 1,000 cycles.

It is customary to use 1,000 cycles as a standard of reference in measuring frequency response and to feed signals of equal strength into a device. Its performance is measured by comparing the amplitude of each frequency relative to 1,000 cycles at the output. If no standard of reference is given, it can be assumed that the 1,000-cycle standard is implied. For example, one may find that the frequency response of a component is described merely as being 3 db down (or —3 db) at 20 and 20,000 cycles. This may be taken to mean that if signals of equal level were fed into the component at 20, 1,000, and 20,000 cycles, at the output the power in the 20- and 20,000-cycle signals would be 3 db lower than in the 1,000-cycle signal.

To show frequency response of a high-fidelity component, a *frequency chart* is often used. The individual pursuing his interest in high fidelity by reading articles, books, or just manufacturers' literature runs into these charts frequently, so this is a good time to become acquainted with the frequency chart and to learn its meaning. Figure 3-7 is the frequency chart of the power amplifier just discussed. Its frequency response was stated as being 3 db down at 20 and 20,000 cycles, relative to 1,000 cycles. The vertical scale is calibrated in decibels. The horizontal scale represents the audio frequencies, ranging from 20 to 20,000 cycles.

Because of the large number of cycles to be covered, the horizontal scale is drawn in *ratio* form. That is, equal horizontal distances represent the same ratio between frequencies rather than a certain number of cycles difference between frequencies. (Note that the distance between 1,000 and 2,000 cycles is the same as between 100 and 200 cycles; the *ratio* is the same for each pair of frequencies.) Drawing the frequency scale on a ratio basis is not unrealistic, because this corresponds to the way the human ear discerns differences in pitch. For example, the change from 100 to 200 cycles (a ratio of 1 to 2) seems to be an increase in pitch of the same magnitude as an increase from 50 to 100 cycles, or from 200 to 400 cycles.

The dark horizontal line in Fig. 3-7 is the standard of reference, which is usually 1,000 cycles and usually called 0 db. Sometimes other frequencies are used as a reference, and occasionally, when 1,000 cycles is the reference, it may be given a value other than 0 db. This does not present a problem because a change from, say, 10 db to 13 db denotes the same ratio between two amounts of power or two voltages as a change from 0 to 3 db. It is the *difference* in decibels that counts, regardless what value is given to the standard of reference. It is easy to see from Fig. 3-7 that the response of our amplifier is 3 db down at 20 and 20,000 cycles. Furthermore, it can be observed that there are slight humps in response at the lower and upper ends of the audio spectrum, reaching 2 db at 100 cycles and 4 db at 10,000 cycles. (This is not a particularly good

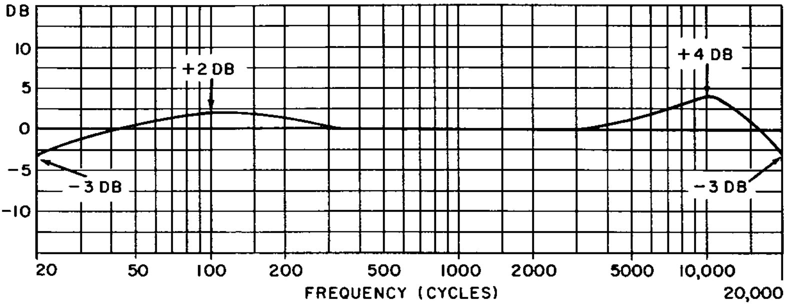


Fig. 3-7. Frequency chart showing the response of a hypothetical power amplifier.

response curve for a power amplifier, which should be virtually flat between 20 and 20,000 cycles.)

Flat frequency response is not always required or desired. In fact, some equipment is expressly designed to provide departures from flat response, either by boosting or lowering the bass and treble portions of the audio spectrum. This topic is considered in detail in Chaps. 6, 7, and 8.

DISTORTION

In Fig. 3-1 we saw the electrical waveform that corresponds to a pure sound, a single frequency. However, the music, speech, and other sounds that we ordinarily encounter do not consist of a single pure note. Rather, they are a mixture of many frequencies, resulting in a complex wave-

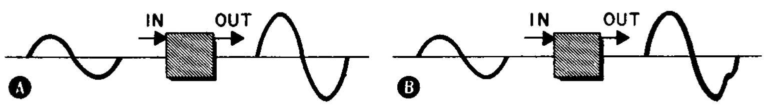


Fig. 3-8. A waveform processed (A) by a linear component and (B) by a nonlinear component.

form, such as those shown in Fig. 3-3. If the audio system is to be a perfect transducer, the waveform should pass through it without change in shape, although it must change in amplitude or signal strength. (See Fig. 3-8.)

Suppose, however, that the waveform does not issue from the audio system as it went in: that is, it may have undergone a change in shape, which we call distortion. (Figure 3-9 shows how a pure frequency might change if distorted in a simple manner.) It can be proved by mathematical methods, and more readily demonstrated by electronic instruments that dissect a complex waveform into its separate components, that this change in waveshape is equivalent to the addition of one or more new frequencies to the basic frequency. For example, the distorted 100-cycle wave shown in Fig. 3-9C actually consists of two different frequencies, 100 and 200 cycles (parts A and B of the illustration). When these two frequencies are combined—their values added at each point of the cycle—they form the complex wave. The additional frequency of 200 cycles represents a *distortion component* or *distortion product.* The ear readily hears this new component unless it is extremely small.

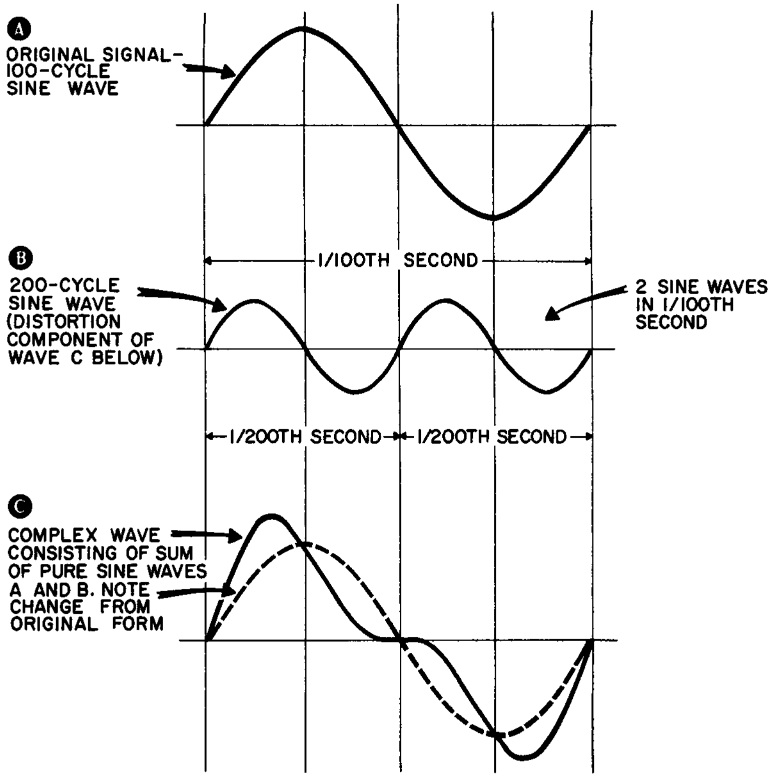


Fig. 3-9. The development of a complex wave (second-harmonic distortion).

How did the distortion product come into being? Speaking as simply as possible, distortion (which can occur in any part of an audio system) is produced because a component fails to act exactly as it should. In an amplifier, for example, the output signal should vary from instant to instant in exactly the same manner as the input signal. The term "nonlinearity" is often used to refer to that characteristic of an audio component which produces distortion. *(A linear component is one that produces an output waveform exactly like the waveform fed into it.)*

There are a number of types of distortion, of which harmonic and intermodulation distortion, particularly the latter, are probably the most important. (Other kinds of distortion, including transient distortion and mechanical wow and flutter, will be discussed shortly.)

HARMONIC DISTORTION

Harmonic distortion results when the additional frequencies produced by a change in waveform are exact multiples of the original frequency. For example, harmonic distortion of 300 cycles could produce frequencies of 600, 900, 1,200, 1,500 . . . cycles. (Figure 3-9 showed a simple case of harmonic distortion, the basic frequency being 100 cycles and the distortion product 200 cycles.) The ear is not highly sensitive to harmonic distortion, but there are definite limits to this saving aspect. If, for example, the audio system reproduces a low piano note with a good deal of harmonic distortion, the harmonics represented by the change in waveform may cause the reproduced note to sound tinny and offensive. (Perhaps the sound in itself might not be unpleasant, but it is no longer like that of a piano.)

The displeasure caused by harmonic distortion depends upon the amount of harmonic signal added by the amplifier compared to the magnitude of the original signal. (The louder the added harmonics the more unpleasant the sound.) Moreover, the effect of harmonics depends upon their *order.* (If the frequency of the harmonic is twice the fundamental frequency, it is called a second-order harmonic; if three times, it is a third-order harmonic, and so on.) Assuming a 440-cycle fundamental, the second-order harmonic would be 880 cycles, the third-order harmonic 1,320 cycles, and the fourth-order harmonic 1,760 cycles.

It has been observed that the higher the harmonic order, the more distressing it is to the ear. As a result, the same *total* harmonic distortion—the sum of all harmonic frequencies generated through distortion—can have different effects, depending upon the relative amount of harmonics of each order. A small amount of second harmonic combined with a lot of third harmonic is apt to sound more distressing than a lot of second harmonic combined with a small amount of third harmonic, assuming that the total is the same in each case.

It is frequently desirable to obtain a numerical measure of the amount of harmonic distortion in an audio component or in a complete system. What is usually done is to measure the level of the harmonics that are produced when a single frequency is fed into the system, computing the ratio of harmonic content to the single frequency at the output. This ratio is usually expressed as a percentage. For example, if measurement at the output of an amplifier by means of a wave analyzer, (which separates the various frequencies) showed that the harmonic content was 0.1 volt when the original frequency was 3 volts, the ratio expressing harmonic distortion would be 0.1/3. To convert this into a percentage, we multiply by 100; that is $0.1/3 \times 100 = 3\frac{1}{3}\%$.

INTERMODULATION DISTORTION

The characteristics of the audio system that produce harmonic distortion are also responsible for intermodulation distortion. However, intermodulation distortion (IM) can only occur when two or more frequencies pass through the system at the same time. IM refers to the effect that one frequency has upon another in terms of changing its waveform. If many frequencies are being reproduced simultaneously, the interaction among them (in terms of producing distortion products) becomes extremely complex. For purposes of discussion, and also for purposes of measurement, it is convenient to think in terms of just two frequencies.

Let us assume that two frequencies, one of 60 cycles and the other of 600 cycles, are fed into an audio component, say, a power amplifier. Let us also assume that the 60-cycle signal is appreciably stronger than the 600-cycle signal. If the amplifier is not perfectly linear (if it distorts the input waveform), the 60-cycle waveform will not be perfectly reproduced. The deformation of the waveform is greatest in the region of the positive and negative extremes. If the 60-cycle frequency is of sufficient magnitude, it will cause the amplifier to operate in nonlinear fashion when it is reproducing the peaks or troughs of the waveform. During these moments of nonlinear operation, the 600-cycle frequency is also reproduced nonlinearly, even though it is in itself too small to cause the amplifier to operate in an appreciably nonlinear fashion. This is intermodulation—the 60-cycle frequency has imposed itself on the 600-cycle wave.

Figure 3-10 shows this process. In part A we see the effect upon a 60-cycle wave of a typical amplifier nonlinearity. The peak of the wave is somewhat compressed as compared with a pure sine wave. (The amount of amplification at the peak is less than at other portions of the

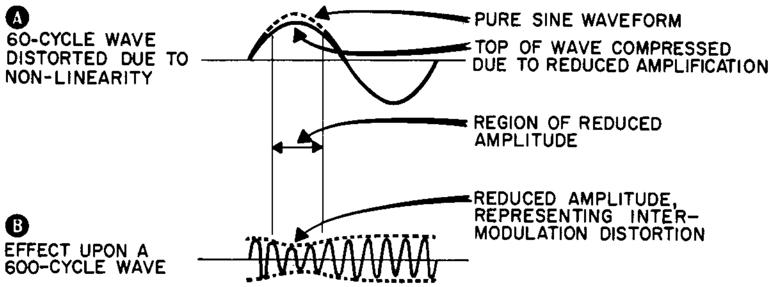


Fig. 3-10. Intermodulation distortion.

wave.) When the amplifier is reproducing the peak of the 60-cycle wave, it will therefore amplify the 600-cycle wave less than at other moments. As a result, the 600-cycle wave is compressed 60 times a second, as shown in Fig. 3-10B. Clearly the 60-cycle frequency is present in the 600-cycle frequency.

As in the case of harmonic distortion, IM also produces new audio frequencies that are distortion products. However, these products are not harmonically related to the original frequencies. Instead of being multiples of the separate frequencies, the distortion products consist of various multiples of one plus or minus multiples of the other. For example 60 and 600 cycles will form IM products of 660 cycles (the original frequencies added together) and 540 cycles (one frequency subtracted from the other). They will form 720 cycles (twice 60 cycles plus 600 cycles); 1,260 cycles (twice 600 cycles plus 60 cycles); 1,140 cycles (twice 600 cycles minus 60 cycles); and 480 cycles (600 cycles minus twice 60 cycles). Other multiples of the original frequencies will form still other distortion products. The ear is quite sensitive to such additional frequencies.

It has been stated that when IM distortion occurs, one frequency is impressed upon another. To measure IM, the usual technique is to use two widely separated frequencies and measure the extent to which the low frequency is impressed on the high frequency at the output of the component being tested, say, an amplifier; the ratio of the low frequency to the high frequency, expressed as a percentage, is the quantity of IM distortion. For example, assume that two signals whose frequencies are widely separated are fed into the amplifier. At the output, it is found by means of the IM tester that the high-frequency signal is 1 volt, with .01 volt of the low-frequency signal impressed on it. The IM distortion is .01/1 or 1%.

Not only is IM distortion much more unpleasant to the ear than harmonic distortion, but the magnitude of IM as determined by the customary method of measurement usually tends to be considerably greater—three to four times as high. Thus if harmonic distortion is 1% at some representative frequency, such as 1,000 cycles, IM will tend to be in the order of 3% or 4%. At times there is an accelerating relationship—after a given point, IM goes up much faster than harmonic distortion. Thus IM may remain three or four times as large as harmonic distortion when low-level signals are being reproduced, but may become much greater for high-level signals. For example, in a tape recorder, when harmonic distortion is 1%, IM may be 3%, but when harmonic distortion rises to 3%, IM may rise to 20%.

Tests have shown that measurements of IM distortion correspond much more closely than harmonic distortion to the results obtained by listening to audio equipment. For this reason, engineers and others place considerable reliance on IM measurements as a way of obtaining the same results as from listening tests, but more quickly. Although the ear has to be the final judge of audio quality, the use of IM tests can go a long way in saving time.

TRANSIENT DISTORTION

If you were to push a bell, it would produce no sound. But tap it with your finger and it will ring. The same sort of thing can happen in audio equipment. A sudden, sharp signal may cause a component to produce a sound of its own, which gradually dies away. This sound, initiated by a sharp impulse, is another form of distortion. It can happen in amplifiers, in speakers, and in other equipment. It is known as transient distortion, because it is initiated by a *transient,* a waveform that lasts for an extremely brief time. Since the waveform occupies a short span of time, it is equivalent to high-frequency signal.

For example, a drum beat, although essentially a low frequency sound, is started by a very brief high frequency sound of considerable magnitude, representing the sudden attack of the stick upon the drum. This attack, when converted into an electrical signal, may cause an amplifier or other component to oscillate on its own, producing a signal unrelated to the audio signal, as shown in Fig. 3-11. Transient distortion is caused not only by sudden attacks, but also by sudden releases. If you hold a postcard upright in one hand and bend the top back with the other hand, when you let go you will see the card vibrate briefly (producing a sound).

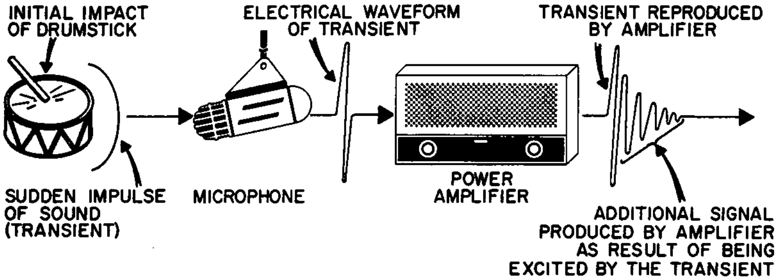


Fig. 3-11. Transient distortion.

In similar fashion, the sudden cessation of a steady signal may excite an audio component into generating an undesired signal.

Because transients are equivalent to high-frequency signals, a component that lacks good high-frequency response will not reproduce them perfectly. This results in the music lacking crisp attack. The sound of such instruments as drums and the piano suffers most from this form of distortion.

PHASE DISTORTION

Although the audio signal travels through the electronic portion of a high fidelity system extremely rapidly, it does take a finite amount of time. Depending upon the design of the equipment, different lengths of time may be required for the low- and high-frequency signals to get

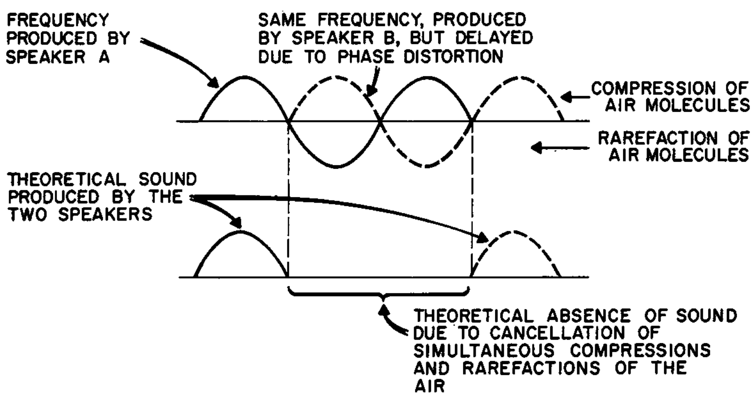


Fig. 3-12. Cancellation of sound due to phase distortion.

through, resulting in *phase distortion.* Ordinarily, this distortion is not detectable by the ear.

On the other hand, a given frequency may be reproduced by two components of the same kind, as in a two-amplifier system, which feeds the low frequencies into one power amplifier and speaker, and the high frequencies into another pair. Over a small range in the audio spectrum, the frequencies appear in both speakers at about equal strength. If phase distortion is present it may mean that the peaks of these frequencies are being reproduced by one amplifier and speaker while the valleys of the frequencies are being reproduced by the other. Simultaneous reproduction of a peak and a valley results in cancellation of the sound (Fig. 3-12). This is an extreme case; however, it is quite possible that due to phase distortion, *partial* cancellation may take place.

Phase distortion can become detectable in speaker systems, for here we are dealing with the passage of acoustic waveforms through the air, a much slower process than the passage of electrical waveforms. In some speaker systems, low frequencies may have to travel through a sufficiently long path before leaving the enclosure as to change the character of the reproduced sound. For example, the relatively low-frequency sound of the drum may become separated from its high-frequency impact. Serious phase distortion—delay in either low or high frequency sounds—can cause music to lose clarity and become somewhat fuzzy.

WOW AND FLUTTER

High-fidelity systems must depend upon mechanical as well as upon electronic components. In the phonograph and the tape recorder, a great deal depends upon accurate motion of the tape or disc. Motion must be uniform—undeviating in speed—because departure from uniformity breeds distortion, commonly called *wow and flutter.*

Wow and flutter are the same kind of thing, systematic deviations from basic speed. If a turntable that is supposed to turn 33⅓ rpm (revolutions per minute) periodically varies between 32⅓ and 34⅓ rpm, this represents a systematic deviation. Such a deviation can be fast or slow. If slow (less than about 10 times per second), it is called wow. If faster than this, perhaps up to several thousand times per second, it is termed flutter.

In the case of wow, the changes in pitch (frequency) of the sound resulting from changes in speed are slow enough so that we actually hear them as a quavering effect. (An otherwise steady note rises and falls in pitch at a rate that the ear can follow, resulting in a sour tone.) Where flutter takes place, the changes in speed are too rapid for the ear to detect as changes in the frequency of the note that is being reproduced. Instead, one hears a new frequency, which is the rate of fluctuation itself. For example, if the change in frequency takes place 100 times per second —100 times upward and 100 times downward—a 100-cycle note is created.

The ear is extremely sensitive to wow and flutter, particularly to high-frequency flutter. Amounts well below .01% are audible, particularly when the flutter takes place at the rate of several thousand times per second. In order to measure flutter, the ratio is found between the level of the flutter frequency and the level of the fundamental frequency. This ratio is then expressed as a percentage.

NOISE

Noise (including hum) is sometimes listed as a form of distortion, inasmuch as it, too, represents an undesired sound. However, the various forms of distortion discussed thus far only appear when an audio signal is present, whereas noise generally exists whether or not a signal is being reproduced. (There are one or two exceptions to this statement, but they need not concern us.)

Noise includes an infinite number of frequencies that occur within the audio range. Various electronic parts in the high-fidelity system, such as tubes and resistors, are responsible for noise. The program sources—tape, phono disc, and radio—also produce noise. Noise is even created outside the audio system—by people moving around, a paper rustling, water flowing from a tap, cars passing, and by random motion of air molecules. External or environmental noise is termed "ambient" noise. Noise due to electronic parts and program sources *(system noise)* is caused by erratic fluctuations in their characteristics or behavior. Such fluctuations produce electrical signals that become apparent as noise when fed into the loudspeaker.

The waveforms that constitute the system-noise frequencies are not steady, as is a musical note, but last for very short periods of time and are constantly changing in random fashion. System noise typically sounds high-pitched or hissy. Actually, however, the frequencies constituting noise are evenly distributed throughout the audio range, but it is in the nature of things that there are more frequencies in the treble range of the spectrum—800 to 20,000 cycles—than in the bass range—20 to 800 cycles. (Refer to Fig. 3-6.) Since there is about the same amount of acoustic power in each noise frequency, the greater number of noise frequencies in the treble range means that most of the noise power is in that part of the spectrum.

We have stated that noise frequencies are not steady, but there is an exception to this—hum. Hum refers principally to intrusion of the house-current frequency, 60 cycles, and its harmonics, 120 and 180 cycles. These frequencies tend to insinuate themselves into the sound system in various ways. Hum is radiated into the air and is picked up by various electronic components; some gets through by more direct means. In any case, by hook or by crook, it usually appears in the output, although in insignificant quantities in the best equipment.

SIGNAL-TO-NOISE RATIO

The significance of noise (including hum) depends upon its magnitude relative to the desired audio signal. The customary way of expressing this relationship is in terms of signal-to-noise ratio, which tells us how much greater the signal is than the noise. This ratio is generally stated in decibels. For example, if a certain signal level represents one million times as much acoustic power as does the noise in the audio system, the signal-to-noise ratio is 1,000,000:1 or 60 db. (Refer back to Table 2-1.)

Signal-to-noise ratios most often compare the maximum amount of signal produced by a component with the noise generated by that component. If a power amplifier has a maximum output of 20 watts (after which distortion becomes excessive for high-fidelity purposes), and a signal-to-noise ratio of 70 db, it means that when the amplifier is supplied with an input signal large enough to drive it to 20 watts, these 20 watts are 70 db (10,000,000) times greater than the noise being produced by the amplifier. If the amplifier were delivering less than 20 watts because of a smaller input signal, the output signal would be less, but the noise generated by the amplifier would still be the same. Therefore the signal-to-noise ratio at this lower output level would be smaller.

Sometimes the signal-to-noise ratio is stated not on the basis of the output level of a component, but rather on the basis of the amount of input signal. For example, the manufacturer of a control amplifier may state that when the signal from a magnetic cartridge is reproduced, the signal-to-noise ratio is 60 db for an input of 10 millivolts (10 mv, .010 volt). If the 10-mv input results in 1 volt output, a signal-to-noise ratio of 60 db would mean, in voltage terms (see Table 2-3), that the audio voltage is one thousand times greater than the noise voltage; conversely, the noise is one-thousandth the signal magnitude, or .001 volt.

DYNAMIC RANGE

Because of the inter-relationship between noise and dynamic range, a discussion of the latter is appropriate here. Dynamic range refers to the relative magnitudes of the loudest and softest sounds appearing in the program material on disc, tape, or radio. For a live symphony orchestra, the dynamic range is typically about 60 db; that is, the loudest sounds contain about 1,000,000 times more power than the quietest; depending

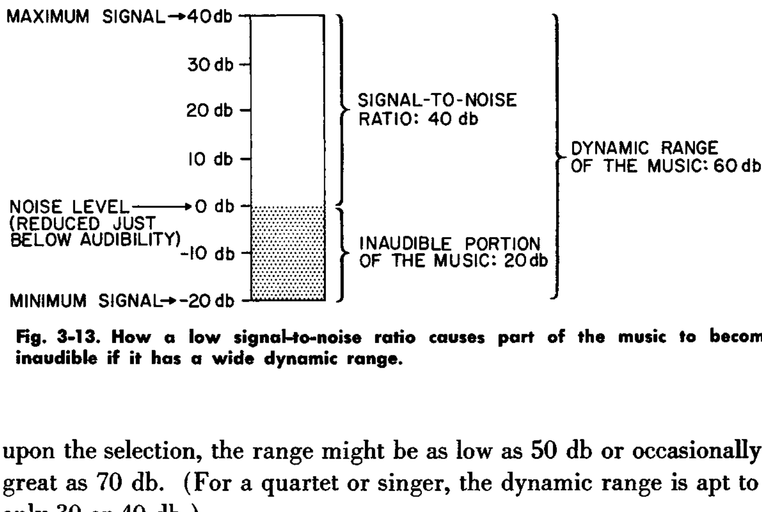


Fig. 3-13. How a low signal-to-noise ratio causes part of the music to become inaudible if it has a wide dynamic range.

upon the selection, the range might be as low as 50 db or occasionally as great as 70 db. (For a quartet or singer, the dynamic range is apt to be only 30 or 40 db.)

The presence of noise makes it difficult or impossible to hear desired sounds that are softer or about as loud as the noise. Although the loudest passages of a musical selection may be well above the noise level of an audio system, it can happen that the softest passages, because of a wide dynamic range, are of the same order as the noise. This results in *masking*; the noise drowns out the desired audio signal, which becomes lost to our ears. On the other hand, if the dynamic range of a musical selection is limited, say 30 db, then not only the loudest passages but also the softest can be kept well above the noise level in virtually any respectable audio system. Thus a low noise level (a high signal-to-noise ratio) is imperative to permit full reproduction of a wide dynamic range. To illustrate, assume that on the loudest passages the signal-to-noise ratio is

60 db. If the dynamic range is 50 db, the lowest passages will still be 10 db (60 minus 50) above the noise level and clearly discernible. By operating the system at a level that causes noise to fall just below our hearing limit, we can hear all the music without any noise. If, however, the signal-to-noise ratio were a poor one, say 40 db, reducing the noise below audibility would simultaneously reduce many of the soft sounds below our ability to hear them. Part of the music would simply disappear, as shown in Fig. 3-13.

4. the original sound and the playback system ■

To understand what constitutes high fidelity, it is not enough to scrutinize the audio equipment used to play music in the home. It is also necessary to take into account the other basic elements of the total picture: the original sound, the program sources that bring the music into the home (radio, phonograph record, prerecorded tape), and the listener. (See Fig. 4-1.) This chapter will deal with the original sound and the audio equipment; discussion of the other two factors will take place in the following chapter.

THE ORIGINAL SOUND

The fundamental objective of high fidelity is to imitate the original sound in all respects—ideally, to the extent where original and imitation are indistinguishable. We seek to hear in the home that which we might hear in a large concert hall, in a small recital hall, at a parade ground, in a night club, etc. Of course we do not wish to hear an orchestra, for example, as it might sound if all the players were crammed into the living room. Rather, we wish to feel transported to the original performance. The goal, in short, is "natural sound."

High-fidelity reproduction is that which, allowing for the state of the audio art, approaches this goal. Obviously, therefore, an appreciation of high fidelity depends upon first-hand acquaintance with natural sound. The individual who frequently attends symphony concerts, recitals, jazz performances, or wherever else his musical preferences take him, is in a position to know whether audio reproduction deserves to be called high-fidelity. A fresh memory of natural sound enables him to distinguish

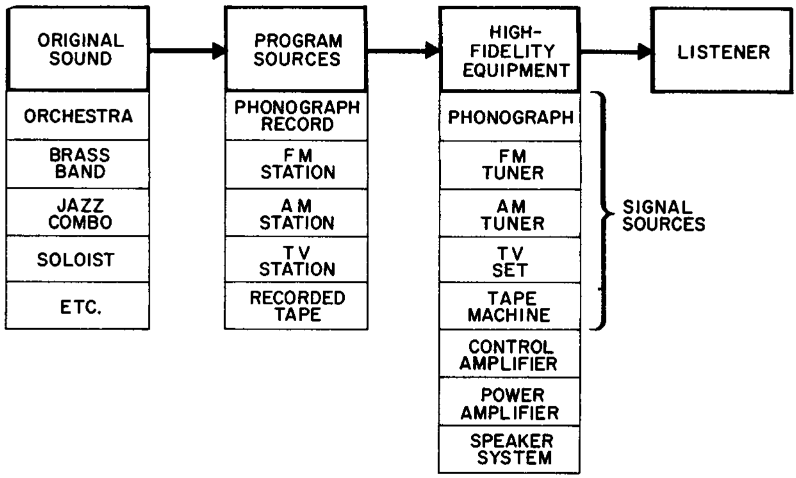


Fig. 4-1. Basic elements of a high-fidelity system.

between the spurious and the authentic in reproduced sound. In selecting equipment and operating it, this memory can serve him well toward achieving high fidelity.

Keeping in touch with natural sound is not equally easy for everyone, yet the obstacles can be surmounted, at least in part. For example, we may not be able to attend the performance of a first-rate orchestra or a big-time band as often as we might wish, but we may find it possible to attend a concert by a high-school orchestra, to listen to some dance music played by a local band, and a marching band at a football game. Although these performances may not be up to professional standards, the *sound* is authentic and we have the opportunity to refresh our concept of what reproduced music should sound like. This does not say that natural sound is the most pleasing sound under all circumstances. An individual may find that he can enhance his pleasure by altering the characteristics of the music in some manner, for example by means of the tone controls. This is a perfectly valid approach to musical listening because the purpose of high fidelity is to give pleasure. If, by the turn of a knob, the listener can enhance those characteristics of the program material that he likes best, then he should go to it.

THE EQUIPMENT

It would be nice (but not practical) to define high-fidelity equipment as that which contains no imperfections—as having perfect frequency response, no distortion of any kind, and no noise. In the present state of the audio art and in the foreseeable future, however, it is necessary to settle for a point part way to perfection. A practical approach in describing high fidelity performance in audio equipment is to set limits on the amount of imperfection that can be tolerated. In some cases, depending upon the audio component and the performance characteristic we are talking about, imperfection can readily be reduced to a level at which the ear can no longer hear it. In other cases, the minimum amount of imperfection that it is practical to achieve is a detectable one. (We have to put up with it because present technology can do no better, at least not at prices that place audio equipment within the reach of the average person.) It is necessary, therefore, to employ a sense of proportion and to set up a secondary group of standards that still come within the province of high fidelity.

We must not, of course, lose sight of the fact that the goal is perfection. Standards acceptable yesterday are no longer so today, and the same fate inevitably awaits present standards. A significant improvement in one component tends to show up hitherto unnoticed defects in other components. To illustrate, it was once considered that power amplifiers were about as good as they would ever need to be. Continued improvements in phono cartridges and speaker systems, however, revealed that further improvement in the power amplifier was worthwhile. All these facts make it impossible to draw a hard and fast line between what is high-fidelity equipment and what is not. For this reason, the quantitative indications in the following discussion of the tolerable amount of imperfection should be viewed as approximations.

The following listing of high-fidelity standards will try to keep three things in mind: (1) what is desirable; (2) what can be achieved by present-day top-quality equipment; (3) what is suitable performance on the part of equipment not quite of the best, yet deserving the label "high fidelity."

TABLE 4-1

APPROXIMATE FREQUENCY RANGES OF VARIOUS MUSICAL INSTRUMENTS

Instrument	*Musical Range (cycles)*	*Noise Range (cycles)*
Bass Drum	50-1500	1500-5500
Bassoon	60-7000	7000-13,000
Bass Tuba	40-4000	4000-7000
Bass Viol	40-5000	5000-9000
Cello	65-8000	8000-15,000
Clarinet	150-10,000	10,000-15,000
Cymbals	300-14,000	———
English Horn	175-7,000	7000-9,000
Flute	250-9000	9000-15,000
French Horn	90-5500	5500-9000
Oboe	250-15,000	———
Piano	27-6000	6000-15,000
Piccolo	500-10,000	10,000-15,000
Snare Drum	80-15,000	———
Soprano Saxophone	200-12,000	12,000-15,000
Trombone	80-8000	8000-10,000
Trumpet	150-9000	9000-10,000
Tympani	40-2500	2500-5000
Viola	150-7000	7000-10,000
Violin	200-9000	9000-15,000

EQUIPMENT FREQUENCY RESPONSE

In the late 1940's and early 1950's, when high fidelity was first becoming a household word, the term was very closely associated with frequency range, which was stressed almost to the exclusion of the other factors affecting performance. Initially, in speaking of frequency range, reference was chiefly to the ability of an audio system to go higher than the 8,000 cycles or so that had previously been the limit for home audio equipment. Attention then shifted to the low end, and the accent was upon reproduction of frequencies below 50 cycles. Later, attention was drawn to the fact that high-fidelity frequency response means more than the range covered; uniformity of response is important as well.

Since the extremes of human hearing are about 20 and 20,000 cycles, it might seem that a flat response between these limits should be stipulated as a high-fidelity standard. High-fidelity systems are, however, primarily concerned with reproduction of music, and one must take into account that below 30 cycles and above 15,000 cycles there is extremely little in the way of musical sound.

Table 4-1 indicates the approximate frequency ranges of various musical instruments. All told, a range of 30 to 15,000 cycles appears to be eminently satisfactory for the purpose of reproducing *music* and may be considered a prime standard so far as frequency range is concerned. (The sound quality of key-jingling or handclapping may suffer if the range does not extend to about 17,000 or 18,000 cycles.) However,

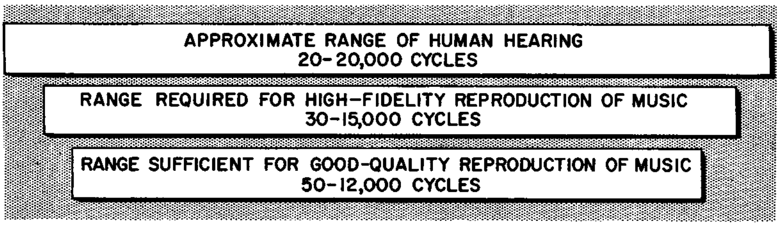


Fig. 4-2. The range of human hearing compared to acceptable frequency ranges for reproduction of music.

studies have indicated that response can even be limited to 11,000 or 12,000 cycles at the high end and to 50 or 60 cycles at the low end before constriction becomes noticeable. Hence one may stipulate a range of 50 to 12,000 cycles as a secondary standard consistent with high fidelity. In view of the difficulty that certain components have in getting below 50 cycles and/or above 12,000 cycles, this secondary standard is a realistic one. Figure 4-2 recapitulates the frequency ranges that we have been discussing.

Thus far we have been concerned with the frequency range of the entire high-fidelity system. In dealing with individual components, the requirements must be raised to allow for the fact that individually small and insignificant departures from required response can add up to a total departure that is significantly large. Component response does not, of course, stop abruptly, but instead tapers off more or less gradually. If a component has adequate response at, say, 40 cycles, response does not vanish at 39 cycles. Figure 4-3A shows an example of how response might drop off at the low and high ends. Response is down 2 db at 30 and 15,000 cycles relative to response at 1,000 cycles, and continues to fall off beyond these extremes. This decline at 30 and 15,000 cycles would not ordinarily be apparent on music, and therefore could be considered insignificant. If a number of components had similar drops, however, these individual deficiencies would add up to an appreciable sum for the entire system as shown in Fig. 4-3B. To provide a margin of safety, so that the sum of deficiencies in the 30- to 15,000-cycle (or 50- to 12,000-cycle) range shall not be too great, it is desirable to stipulate a range of 20 to 20,000 cycles for individual components, if it can be attained.

This objective can be realized more easily in some components than in others. The control amplifier and power amplifier present no great prob-

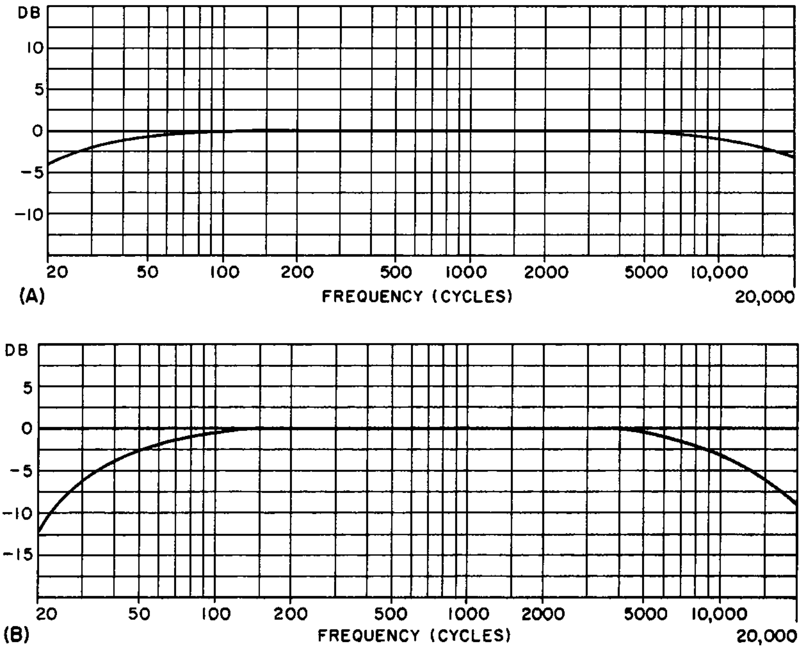


Fig. 4-3. (A) Response curve of a component showing a loss of 3 db at 30 and 15,000 cycles. (B) The frequency response of a high-fidelity system using three components whose response curve is like (A).

lem. Often these components are specified as having response far beyond 20,000 cycles, sometimes out to 100,000 and 200,000 cycles. This in itself does not contribute to high-fidelity reproduction, but is a byproduct of obtaining adequate response and other desirable characteristics within the audio range. (One may use the analogy of an automobile designed to operate at 120 miles an hour in order to provide complete satisfaction at 60 miles per hour.)

Response of 20 to 20,000 cycles can also be achieved (perhaps with a little more difficulty) in the FM tuner and the phonograph pickup. Real difficulties arise in the tape recorder and the speaker system, where 30 to

TABLE 4-2

SUMMARY OF PRACTICAL FREQUENCY RANGE REQUIREMENTS FOR INDIVIDUAL AUDIO COMPONENTS (CYCLES)

FM tuner	20-20,000
Phono pickup	20-20,000
Tape machine:	
Primary standard	30-15,000
Secondary standard	50-12,000
Control amplifier	20-20,000
Power amplifier	20-20,000
Speaker system:	
Primary standard	30-15,000
Secondary standard	50-12,000

15,000 cycles may be said to apply as a primary standard, and no particular stigma should be attached to them if they only meet the secondary requirement of 50 to 12,000 cycles. In the speaker system, the greatest difficulty occurs at the low frequencies; in the tape recorder, the high end is the greater problem. It should be pointed out that this applies to recorders operating at 7.5 ips (inches per second), the speed that has been virtually standard for home high-fidelity use; and it applies even more to 3.75 ips. Machines operating at 15 ips (a speed used almost entirely in semiprofessional and professional recorders) can easily reach 15,000 and even 20,000 cycles. Table 4-2 summarizes the practical frequency range requirements for the individual components.

UNIFORMITY OF RESPONSE

Uniformity of response is of equal or greater importance than range. The statement that a component covers a certain range does not tell us how flat (smooth) its response is within this range. There may be dips and peaks in the response that color the music so that it no longer sounds like the original. Figure 4-4A shows a frequency response characteristic that is sufficiently irregular to color the music and to be annoying. (A characteristic of this sort is sometimes found in an inexpensive or improperly adjusted tape recorder.)

Uniformity of frequency response need not be perfect because the ear will not detect small deviations from flat response. Instead, high fidelity requires that deviations from flat response be kept within certain tolerances. As a primary standard, it is desirable that a component stay within 1 db of flat response, if possible. In other words, at any frequency in the audio range, response should be no more than 1 db higher or lower than at 1,000 cycles. This requirement may be expressed, in the case of a component that can fully cover the 20- to 20,000-cycle range, as "20-

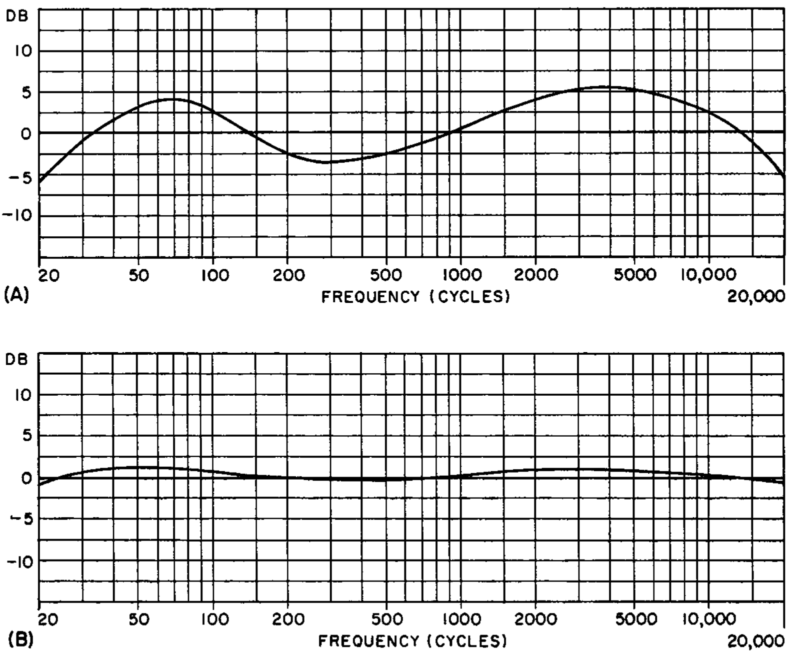


Fig. 4-4. Frequency charts of (A) a component that could color the reproduced sound and (B) a high-quality component with "20-20,000 cycles ±1 db" response.

20,000 cycles, plus or minus 1 db." An example of such frequency response appears in Fig. 4-4B.

The requirement of response flat within 1 db is easily met by a power amplifier deserving to be classified as high-fidelity, and can be met with almost as great ease by the control amplifier and the FM tuner. In the case of the phono cartridge and the tape recorder (allowing for the more limited range of the latter), a secondary standard of ±2 db variation is appropriate in view of the technical problems encountered in these components. (Deviations within ±2 db produce virtually no audible coloration of music and are therefore consistent with high-fidelity performance.) The fact that the deviations for each component may add up to a total that is appreciable makes it necessary to invoke the ±1 db require-

TABLE 4-3

SUMMARY OF PRACTICAL REQUIREMENTS WITH RESPECT TO UNIFORMITY OF RESPONSE FOR INDIVIDUAL COMPONENTS

FM tuner	±1 db	Power Amplifier	±1 db
Phono pickup	±2 db	Speaker system:	
Tape machine	±2 db	Primary standard	±3 db
Control amplifier	±1 db	Secondary standard	±5 db

ment where it is technically most feasible: in the control amplifier, power amplifier, and FM tuner.

The problem of achieving uniform response is most difficult in the case of the speaker system. Not so long ago, many of the best speaker systems were considered high-fidelity even though their acoustic output at some audio frequencies differed as much as 10 db or more from that at 1,000 cycles. In other words, their response was no better than ±10 db over the stipulated range (40 to 15,000 cycles for speakers). Such deviation from flat response means that the maximum difference between various frequencies could be 20 db or more.

The response characteristic of speaker systems has been considerably smoothened, so that today's best units remain within ±3 db of flat above 40 cycles. A deviation of not more than ±3 db above 40 cycles may be considered a current prime standard for speaker systems. A secondary standard that gives slight additional coloration would be ±5 db. Table 4-3 summarizes these practical requirements for uniformity of frequency response.

EQUIPMENT DISTORTION

At one time, IM (intermodulation) distortion of 5% or even higher was considered tolerable for average listening levels. However, as improvement (including increase in frequency range) in one type of component served to show up defects in other components, it became apparent that the acceptable limit of IM distortion was not more than 1 or 2%. The recent trend has been to consider still lower IM, if achievable, a requisite of high-fidelity equipment. In power amplifiers, for example, competent listeners have found that those units which sound best at moderate levels also turn out to have the lowest measured IM, even when the comparisons involved IM well below 1%.

Where feasible, IM distortion should not exceed 0.1% at moderate levels and should not be greater than 1% at the maximum levels ordinarily encountered. This is quite possible in the power amplifier and in the control amplifier. It is also attainable, but with somewhat more difficulty, in the FM tuner. Such low IM distortion is a good deal more difficult, or even impossible, to obtain in other components, but this does not seem to obviate the need for 0.1% IM where it is possible to keep it that small. For one thing, distortion (like imperfections in frequency response) can be additive, so that individually unnoticeable amounts can add up to a perceptible total. Furthermore, it has been noted that reduction of the distortion in a component that already has low distortion often results in improved sound, despite the fact that another component may be producing several times as much distortion. (This appears to be the exceptional case where strengthening a single link makes the chain stronger.) In the case of phono cartridges and tape recorders, and speaker systems, IM distortion not exceeding 1% at average levels and not more than 5 to 10% at maximum levels may be considered a suitable requirement.

Because the factors that cause harmonic distortion also cause IM distortion, the evaluation of a component in terms of IM implicitly takes into account its harmonic distortion. In addition, measurement of IM corresponds more closely to the reaction of the ear. However, in some instances, particularly in the case of the speaker, distortion varies a good deal with frequency. In order to evaluate performance from one end of the audio range to the other, it may be desirable to measure harmonic distortion for selected frequencies distributed throughout the range. A given percentage of harmonic distortion generally indicates that IM distortion is at least three or four times as great. For this reason, the acceptable limit of harmonic distortion is quite low, under 1% at any level in all components except the speaker and tape machines. In speakers, it should remain under 1% at average levels except below 50 cycles, where even the best speakers produce 3 to 5% harmonic distortion at reasonably loud levels. Although this much distortion is not desirable, in the current state of the art, it must be considered in keeping with high-fidelity performance. Harmonic distortion of tape machines should not exceed 2 to 3%.

It is not yet possible to give numerical values to permissible transient and phase distortion, which means that the listening test becomes all the more important in evaluating a high-fidelity system. Thus if the sound is muddy or boomy, it may be due to transient distortion at low frequencies (although it could also be due to room characteristics). If the sound has a ringing, rattly, or tinny quality, this may be due to transient distortion at the high end. Fuzziness or lack of clarity or a seeming "out of step"

TABLE 4-4

SUMMARY OF PRACTICAL REQUIREMENTS WITH RESPECT TO DISTORTION OF INDIVIDUAL COMPONENTS

	ELECTRICAL DISTORTION	
	Intermodulation	*Harmonic*
FM tuner	0.1%-1%	Under 1%
Phonograph	1%-5%	Under 1%
Tape recorder	1%-10%	Under 2-3%
Control amplifier	0.1%-1%	Under 1%
Power amplifier	0.1%-1%	Under 1%
Speaker system		Under 3-5%

	MOTION DISTORTION			
	Wow and Flutter		*Speed Error*	
	Primary Standard	*Secondary Standard*	*Primary Standard*	*Secondary Standard*
Phonograph	0.1%	0.25%	0.2%	1%
Tape recorder	0.1%	0.25%	0.2%	1%

relationship between the high frequency and low frequency components of a sound (such as a drum beat) may be due to phase distortion.

There are numerical guides to high fidelity for wow and flutter distortion. Some persons can detect amounts of wow and flutter as low as .001% and even less at single frequencies, particularly the higher ones. For the mixed sounds that we normally encounter, however, wow and flutter can be appreciably higher before causing annoyance. Although a limit of .01% wow and flutter would be desirable, because it is sufficiently low to escape detection by virtually any ear, it would be most difficult to achieve except at inordinately high cost. For practical purposes, keeping wow and flutter below 0.1% will suffice for most persons. As a secondary standard, one may stipulate a limit of about 0.2 or 0.25%.

Another type of speed error should also be considered. This refers to inaccurate speed. If a turntable rotates at a higher speed than it should, the pitch of all sounds is raised. (The opposite effects are obtained when a record or a tape is played below rated speed.) A certain amount of speed inaccuracy is tolerable. This amount varies a good deal among individuals—from about 0.2% for persons with very sensitive ears, to as high as 10% for other individuals. For highest fidelity, the maximum speed error should be 0.2 to 0.3% in either direction (higher or lower than rated speed). Errors up to 1% may be considered as satisfying a secondary high-fidelity standard, because they will escape detection by most ears. Table 4-4 summarizes the practical requirements with respect to IM, harmonic, and other forms of distortion.

EQUIPMENT NOISE

There is no such thing as complete absence of noise. To one degree or another, noise is always with us, both in electrical form in the audio equipment and in our everyday listening environment. The problem in high fidelity is not to reduce noise to zero, which is virtually impossible, but to bring it to a subaudible level or at least sufficiently low so that it will not interfere with the music.

The problem of keeping noise inaudible ties in very closely with the dynamic range of the original material, as was shown in Chap. 3. If the signal-to-noise ratio is greater than the dynamic range of the material being reproduced, it is possible to reduce the gain (volume) of the audio until the noise falls below audibility, without losing any of the desired signal. The maximum dynamic range encountered in live performance of a full orchestra is about 70 db. The range for soloists or small musical groups would be on the order of 30, 40 or 50 db.

It would seem that in view of an approximate 70 db maximum dynamic range of program material, the signal-to-noise ratio of an audio system should be at least as great, so that the softest as well as loudest sounds could exceed the noise. This ratio can be achieved by the control unit and power amplifier, particularly the latter. (When the signal source is a magnetic cartridge, one can only expect a 55 db ratio for the control amplifier.) It is appropriate to stipulate such a signal-to-noise ratio as being a high-fidelity requisite for these components. Even if something less than a 70-db ratio were sufficient (as we shall very soon see is the case), this high a ratio can eliminate the control unit and power amplifier as sources of noise.

Seldom if ever do the program sources—phonograph disc, tape, or FM broadcast—have a dynamic range greater than 55 db. Because the very softest sounds would take the signal down into the noise level and because the loudest sounds would raise problems of excessive distortion (the higher the level, the greater the distortion), dynamic range is compressed either during performance or by electronic means *(limiting)* during signal processing. The loudest sounds are reproduced below their original magnitude relative to sounds of a moderate level, and the softest sounds

TABLE 4-5

SUMMARY OF PRACTICAL REQUIREMENTS WITH RESPECT TO SIGNAL-TO-NOISE RATIO FOR INDIVIDUAL COMPONENTS

FM tuner	55 db
Tape machine	50-55 db
Control amplifier:	
High-level sources*	70-75 db
Magnetic phono source	50-55 db
Power amplifier	70-75 db

*FM tuner, tape machine, TV, etc.

are reproduced at higher than their original relative strength. For this reason, as a secondary standard, one may view a 55-db signal-to-noise ratio as compatible with high fidelity. In the present state of the art, a signal-to-noise ratio of 55-db is about the best that can be expected of tape recorders and FM tuners at prices that a significant number of audiophiles can afford. It is especially difficult for a tape recorder to attain this high a signal-to-noise ratio, and (in the case of this component) a ratio of even 50 db may be considered within high-fidelity limits.

Signal-to-noise ratios above 55 db can be attained in moderate-price FM tuners and in tape recorders, but only at the cost of a sacrifice in frequency response and distortion, so that the equipment is no longer of high-fidelity caliber in these respects. (This conflict among performance requirements is treated in a later chapter.) Table 4-5 summarizes practical requirements with respect to signal-to-noise ratio.

EQUIPMENT RELIABILITY

Two audio components of the same type, for example, two power amplifiers, may have virtually the same performance characteristics with respect to frequency response, distortion, or noise, yet the price of one may be double the other. Assuming that in each case the manufacturer is giving proportionate value, what accounts for the price difference? It is generally safe to say that the answer lies in the degree of reliability of the equipment. Although reliability is not something that we hear, it can make a decided difference in *what* we hear. Therefore it is fitting that reliability of performance be considered one of the attributes of high-fidelity equipment. It includes (1) initial performance of the equipment; (2) durability of the equipment; (3) stability of the equipment.

1. *Initial Performance.* When a manufacturer advertises the specifications (performance characteristics) of his product, they apply to the *average* of what he produces. The audiophile, however, buys just one unit, not the average, and he is vitally interested in having that particular component operate correctly. An excellent average is of no good to him if his own unit fails to operate properly. The assurance that each unit will perform satisfactorily depends in part upon the design and engineering behind the product. Good engineering seeks to do away with critical factors, so that slight departures from design or from preferred operating conditions will not result in considerable degradation of performance.

In most audio components, quality of performance also depends upon alignment. That is, the performance of these components can be adjusted by a technician (not the user) to affect frequency response or noise or distortion or other characteristics. Even though design may be inherently good, the full potential is not fully realized unless the manufacturer puts proper care and time into alignment.

Finally, performance of the component depends upon quality control, which involves adequate testing both of incoming supplies and of the finished product. As with alignment, testing of the finished product can be fairly superficial or it can be extensive, involving more time and higher cost. Extensive testing assures one that the component is not merely performing, but that it measures up to high-fidelity standards.

2. *Durability.* A component's performance may measure up to high-fidelity requisites at the outset, yet deteriorate in a relatively short time.

Fig. 4-5. A high-quality power amplifier.

Marantz

The quality of parts used is important, but use of high-quality parts is not enough. It is also necessary to provide safety factors, so that parts are not worked too hard and therefore subject either to destruction or altered performance. For example, a part may be exposed to 350 volts, and one manufacturer may use a part rated to withstand 400 volts, whereas another uses a part rated at 600 volts. The second manufacturer is providing a greater safety margin. Parts layout is another factor related to durability. Heat is detrimental to parts life and performance, and provision has to be made for adequate ventilation. Figure 4-5 is a photo of a power amplifier that uses heavy-duty long-life components and construction.

3. *Stability*. Performance may be degraded by parts or entire audio components that fail to operate in a consistently satisfactory manner. For example, due to design or to the parts used, a tape recorder may provide a different frequency response characteristic from time to time. An FM tuner may fail to hold a station for an appreciable length of time, even though it is retuned three, four, or more times. In these and other ways, performance that is adequate at one moment may not be adequate at another.

5. program sources and the listener ■

Having considered the original sound and the reproducing equipment as two of the ingredients of high fidelity, it remains in this chapter to discuss the other two basic elements, namely the program sources and the listener.

THE PROGRAM SOURCES

All too often the audiophile blames his high-fidelity system for inadequate performance (poor imitation of natural sound) when the fault really lies in the program source: the FM (or AM or TV) station, the record disc, or the prerecorded tape. For various reasons, usually technical problems, but too often indifference, carelessness, or a mistaken concept of high fidelity, the material provided by these sources is not adequate in terms of frequency response (range and uniformity), distortion, and noise, as well as lacking in certain other important characteristics.

Lest there be any misunderstanding, it had best be stated early in this discussion that today's program sources are not so derelict that the material they provide is of insufficient caliber to warrant the purchase of high-fidelity reproducing equipment. On the contrary, a fair amount of what is available on radio, on disc, and on tape ranges from good to breathtaking in its quality, largely compensating for the lapses from high fidelity at other times. However, the lapses, at least on the part of some broadcast and recording studios, have been too frequent and too great to be ignored.

Being aware of this situation, and assuming that he maintains a fresh recollection of natural sound (by attending a sufficient number of live performances), the audiophile can be selective when purchasing recordings, giving close attention to the quality of sound as well as to the musical merits of the performance. He may find that some manufacturers of disc or tape recordings seem more than others to turn out a high-fidelity product, and he may thus come to prefer certain brands. However, he should be alert to the fact that because of personnel and other changes, recording companies' quality of product may vary from time to time, and so he should not follow his bias blindly. As with recordings, the audiophile may soon come to know which FM stations—assuming he has a choice of several—come closest to observing high-fidelity standards.

SIGNAL-TO-NOISE RATIO

Although this book is basically concerned with home audio reproducing equipment, an appreciation of high fidelity also calls for at least a brief look at the problems that confront the recording and broadcast studios. Some of these problems are not so much due to the equipment used by them as to deficiencies in the audiophile's equipment, for which the studios are trying to compensate. A major problem is keeping distortion down to undetectable, or at least tolerable, levels. Small amounts of distortion result in the listener subconsciously experiencing a sense of fatigue rather than detecting a change in the character of the sound. Moderate and large amounts of distortion audibly alter the sound, imparting a rough or even a broken texture. It is characteristic of the electronic and mechanical equipment used in connection with audio, whether in the home or in the studio, that distortion tends to increase as the level of the sound is increased. (Figure 5-1 shows how IM distortion goes up as more signal is put on a tape.) In recording, it is necessary to cope with the fact that the media used—disc and tape—contain a certain amount of inherent noise. In the case of the phonograph record, minute irregularities in the groove as well as dust and dirt that gather there (Fig. 5-2) produce random electrical signals that emerge as noise from the speaker system. To override the noise, it is desirable to record at high levels, but

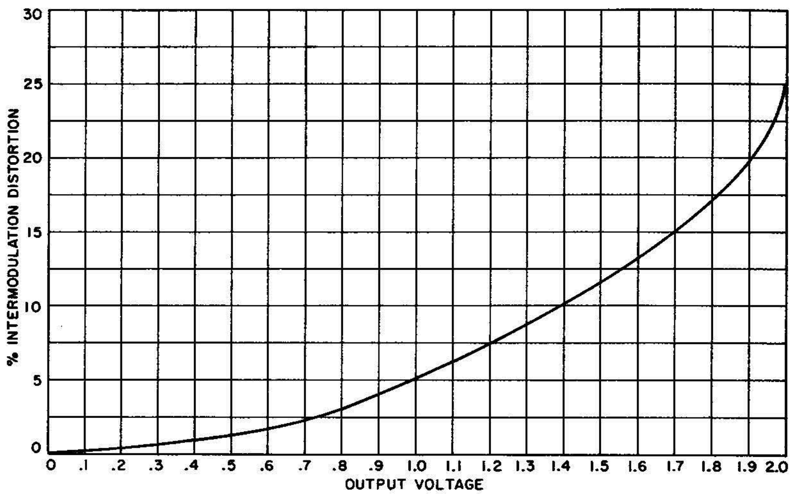


Fig. 5-1. The increase in intermodulation distortion with increasing signal recorded on tape.

this breeds relatively high distortion. Similarly, tape has an inherent noise that produces a distinctive kind of hiss, making it necessary to record at high levels. Noise is always present in electronic equipment, originating in the tubes, transistors, resistors, etc. Thus, the electronic equipment employed in broadcasting and processing records and tapes also contributes noise.

Taking all these factors into account, it is very difficult for program sources to achieve a signal-to-noise ratio much beyond 60 db. That is,

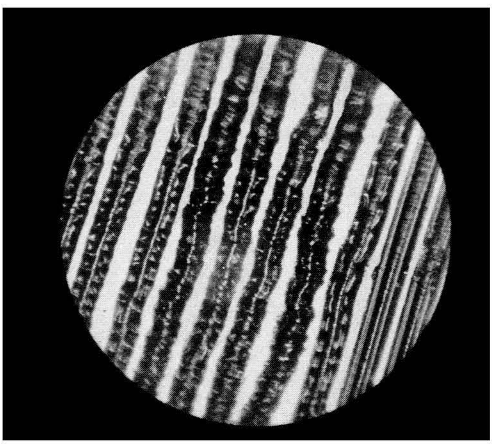


Fig. 5-2. Photomicrograph of a record groove.

Cook

the loudest audio signal on a record, tape, or FM broadcast might be 60 db higher than the noise level, but not more. In fact, this high a signal-to-noise ratio probably exists only in FM broadcasts, with a top ratio of 55 db being more characteristic for the other sources. These figures assume that recording is at a level that keeps distortion to a reasonable minimum. If caution is cast aside, one can record or broadcast a louder signal and increase the signal-to-noise ratio at an expense of inordinate distortion; this happens.

In deciding how much signal to record or broadcast, the studio engineers are concerned not only with their own equipment and its limitations, but also with that of the listener. For example, many tape recorders—those in the lower price classes, particularly—produce relatively large amounts of noise. Even though a commercial prerecorded tape contains very little noise, the machine adds a substantial quantity, so that the resultant signal-to-noise ratio is poor. To overcome this limitation in the playback equipment, tape recording studios sometimes put a relatively large amount of signal on the tape, causing an increase in distortion. It is the expectation of the studio that the improvement in signal-to-noise ratio will add more listening pleasure than is subtracted by the increase in distortion. The audiophile who possesses a truly high-fidelity tape machine that produces very little noise in playback, however, suffers a loss because of this high distortion. The same kind of thing happens in disc recording and FM broadcasting.

As observed in the previous chapter, recording and broadcast studios do not transmit the full dynamic range of such program material as symphonies and large choral works because this range would exceed the practical signal-to-noise ratio. The resulting effect is adverse in two ways. First, reduction of the dynamic range—often it is limited to 50 db as compared with the original range of 70 db—deprives the music of some of that shading which is characteristic of the original sound. Second, reduction of the dynamic range, accomplished by electronic limiting devices, in itself produces a certain amount of distortion. The severity of this distortion depends upon the amount of limiting and the design of the limiting equipment.

FREQUENCY BALANCE

A major flaw that appears in program material concerns frequency balance—keeping the same relative proportion in the strength of low, middle, and high frequencies as occurs in natural sound. In the process of recording a live performance, it is legitimately necessary for the

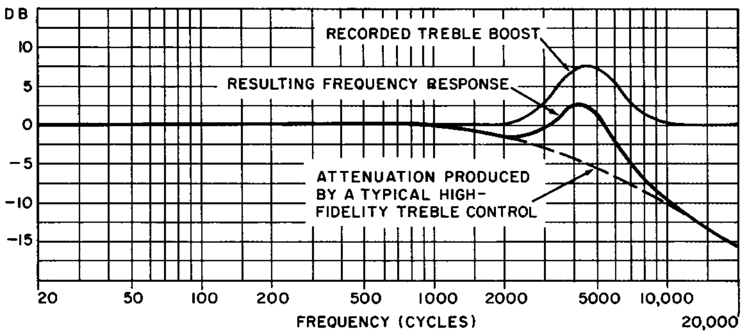


Fig. 5-3. How a typical treble control fails to correct for a treble peak in recording.

recording engineers to emphasize or deemphasize various parts of the audio spectrum in order to compensate for characteristics of the microphone used, location of the microphones, and acoustic characteristics of the performance site. The purpose of these adjustments is to make the final result, when heard by the listener, bear a close resemblance to natural sound. In making these adjustments in frequency response, the recording engineers must trust their ears and their monitoring equipment (the audio playback systems used at the recording studio to guide the engineers in frequency, volume, and balance adjustments). Unfortunately, the recording engineer may compensate not only for microphone characteristics and acoustic conditions but also for imperfections in his own hearing, for imperfections in his monitoring equipment and environment, and for his personal predilections in musical balance. Add to this the still extant tendency in some quarters to identify high fidelity with exaggerated bass and exaggerated treble, and one can understand why frequency balance is not always what it should be on recorded or broadcast material.

The audiophile can undo a good deal of the harm by using the tone controls. For example, if the program material has exaggerated highs, he can adjust the treble control so that the highs are reduced in level. Too often, however, adequate correction for frequency imbalance cannot be obtained by means of the reproducing equipment ordinarily available. The recording or broadcast engineer has access to special equipment that shapes the audio spectrum quite differently than does the audiophile's equipment. The engineer can emphasize or deemphasize a very narrow band of frequencies, if he desires, whereas the audiophile's tone controls affect a wide band of frequencies. To illustrate, the engineer may decide to impart an easily perceived kind of brilliance, manifest on even the poorest reproducing systems, by boosting frequencies between 3,000 and 6,000 cycles. The audiophile, on the other hand, cannot deemphasize these frequencies without severely cutting response above 6,000 cycles as is shown in Fig. 5-3.

It bears repetition that this sort of thing does not occur all the time or often enough to deprive a high fidelity installation of value. It happens, though, with sufficient frequency that the audiophile must be alert to the possibility that it is not his equipment which is at fault when reproduction differs with his mind's recollection of natural sound. On the other hand, one must allow for the fact that perfection is an unattainable goal, so that not every disc or tape or FM program can live up to the highest prevailing standards.

REVERBERATION

In addition to the problems of keeping noise and distortion low, dynamic range wide, and frequency response in proper balance, the program sources must consider at least one other important factor in striving

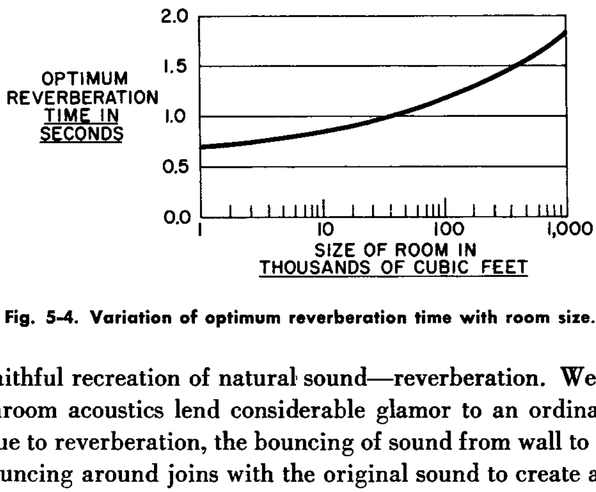


Fig. 5-4. Variation of optimum reverberation time with room size.

for the faithful recreation of natural sound—reverberation. We all know that bathroom acoustics lend considerable glamor to an ordinary voice. This is due to reverberation, the bouncing of sound from wall to wall. The sound bouncing around joins with the original sound to create a composite effect that covers up many defects that would be heard if the original sound were sharply defined. Similarly, reverberation in a concert hall can enhance the music, providing a cloak that hides stridency. Too much reverberation, however, produces a confusion of sound. Thus, certain concert halls are preferred over others in part because their reverberation is closest to what the ear finds pleasing.

Clearly, it is necessary to take reverberation into account when a recording is made. This becomes complex because the reverberation that the ear finds ideal in the concert hall is not ideal in the living room. The smaller the listening area, the shorter is the optimum reverberation time, as shown in Fig. 5-4. In a typical living room, the reverberation time—defined as the period required for the sound to diminish 60 db (to one millionth of its original value)—is ideally about 0.7 second; in a concert hall, the optimum is about twice as long. The desirable approach is for a recording company to supply not all the needed reverberation, but just an amount which, when added to that of the listening room, will give the most pleasant effect. But this is difficult to do because differences in size, shape, material, and furnishings cause rooms of about the same size to have different reverberation characteristics.

The net result is that a recording cannot be all things to all listeners. Different companies supply different amounts of reverberation in light of

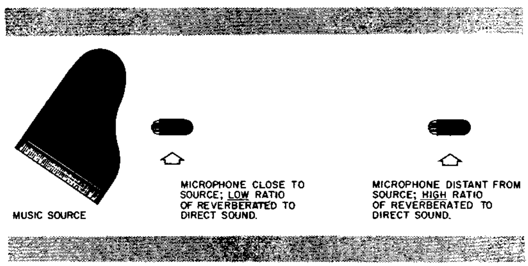


Fig. 5-5. How microphone placement controls reverberation time in recording.

what they deem best for the majority of listeners. They can control reverberation by selection of the performance site and microphone placement (Fig. 5-5), and they can produce it by mechanical and electronic means. The audiophile may well find that the amount of reverberation typically supplied by one recording company is more suited to his listening environment or tastes than that of another company.

THE LISTENER

A discussion of the ingredients of high fidelity can no more leave the listener out of the picture than one can properly draw a seascape without the sea. Sound is not merely a series of waveforms, but involves a listener and its effect upon him. Whether sound is high-fidelity depends not only upon the nature of the waveform but also upon the reaction of the listener. Sound that delights one person may be deemed by another as an onslaught upon his ears.

To deserve being called high-fidelity, audio equipment and program sources must come up to certain standards. If listening to reproduced music is to constitute an experience in high fidelity, the listener's attitudes should also conform to certain requirements. If these attitudes do not already exist, they can be cultivated: (1) an inclination to overlook the faults, to avoid being hypercritical; (2) an appreciation of what variety can do to augment listening pleasure.

Overlooking the Faults: Most of us have the faculty of concentrating on what we really want to hear. We can concentrate upon a single voice amid the babble of a party, or we can overlook the hiss, hum, rattle, and whistle that frequently accompany the music emanating from the 4-inch speaker of an inexpensive table radio. Not expecting perfection from a small radio, we are not disturbed by its absence. Where high-fidelity equipment is concerned, however, the audiophile, having spent much money and effort, expects the best and is therefore extremely conscious about imperfections. Instead of focusing upon the music, his senses are overly alert to the faults. If he wishes to truly enjoy the pleasures of reproduced music, the listener should discipline himself to overlook the faults—within reason—because perfection is unattainable. This is a question of proportion and common sense. When the imperfections in reproduced music are reduced approximately to the extent permitted by the current state of the art, they should be forgiven and forgotten. The performance of one's high-fidelity system should be met with an attitude of forbearance, not with one of high challenge. On the other hand, age and use do cause the performance of the system to deteriorate gradually, and this of course must not be tolerated too long. Therefore, as one does with any fine device such as a camera or car, the components of a high fidelity system should be checked periodically. But between checkups and repairs, the listener should relax and allow himself to enjoy the music.

Appreciating Variety. The user can vary the use of his high-fidelity system in a number of ways so as to prevent the system from becoming a source of weariness instead of pleasure: he can vary the length of the listening period from time to time instead of always letting the system drone on for endless hours. He can let the system stay silent several nights a week (or longer), so that it will again have a fresh appealing quality when he returns to it. He can do his listening at various volume levels. (Not all types of program material are most pleasing at a high level; on the other hand, one may find new meaning in a subdued composition by listening to it played loudly.) Whether the audiophile likes classical or popular fare best, it may do his ear good to listen sometimes to what the other man likes. At worst, when he returns to his preferred music, the contrast will heighten his enjoyment. Some listeners devote themselves exclusively to records or tape, and others exclusively to FM radio. The audiophile may well find that his total pleasure is substantially increased by taking advantage of more than one of the available sources of good music. A number of FM listeners tend to anchor to one particular station. By the mere turn of the dial, they can possibly obtain a different quality of sound, either desirable in itself or serving by contrast to enhance the pleasure derived from their favorite station. Finally, the audiophile may find that he can increase his enjoyment by varying certain components. For example, one phono cartridge may sound better on certain types of program material, while a second may sound better on other types. Or, if he has a secondary speaker system in another room, he may find that serious listening to this speaker (not just at background level while reading) may reveal facets of the music undisclosed by his principal speaker system.

SUMMARY

We have seen that there are four basic constituents of the listening experience we call high fidelity: the original sound, the program sources that convey this sound into the audiophile's home, his audio-reproducing equipment, and the listener himself. For the program source and the reproducing equipment to be considered high-fidelity, the following requisites must be met: virtually complete coverage of the musical sounds audible to the human ear; smooth reproduction of these sounds without undue accentuation or loss of particular frequencies; very low distortion; and very little noise. Furthermore, the reproduction should possess a dynamic range that comes close to being as wide as that of the original sound. The reproduced sound should contain reverberation approximating that of the original, but with allowance for the fact that the most pleasing amount is apt to be less in the home than in the large hall. The playback level, for natural sound, should be close to that of the original performance. This does not mean that the speaker should be capable of producing a volume of sound equal to that of an orchestra or large band or a group of 100 voices. Rather, the listener should hear about the same amount of sound as though he were in the seat of his preference in the concert hall. (Because of the listener's proximity to the speaker and the reduced size of the listening quarters, this means that a relatively small amount of sound has to be generated by the speaker system.)

6. matching high-fidelity components ■

A high-fidelity system consists of a fairly considerable array of components, as brought out in Chap. 1 and recapitulated here in Table 6-1. It is natural to expect that in putting these components together to form a home music system there would be some problems of properly matching one to another. And so there are, at least if the maximum potentialities of the components are to be realized.

The problem of matching has several frames of reference. One concerns quality of components; another has to do with frequency response; and, in addition, there is the question of the level of signal fed from one component to the next. (We shall devote the most attention to this because it involves the all-important factors of low distortion and high signal-to-noise ratio.)

COMBINATION VERSUS SEPARATE COMPONENTS

It might seem that one could eliminate the entire problem of matching components if one simply purchased a packaged unit already assembled by the manufacturer. However, this is only partly true; moreover, what

TABLE 6-1
COMPONENTS IN HIGH-FIDELITY SYSTEMS

Signal Sources:	*Control Amplifier*	*Speaker System:*
FM tuner	*Electronic Crossover**	Woofer
AM tuner	*Power Amplifier*	Mid-range
Phonograph:		Tweeter:
Turntable		Low treble
Tone Arm		High treble
Cartridge		Dividing network
Tape machine		Enclosure
TV set		

*Not used if there is a dividing network.

one gains in the way of minimizing the problem of matching may be sacrificed in terms of quality.

Matching is not only a matter of assembling the components correctly (something that the manufacturer of a packaged system undertakes to do), but it is also a matter of using them correctly, which is the listener's task. Moreover, manufacturers do not always perform the best possible job of matching, in part because they cannot foresee the individual problems that arise among various audiophiles.

By knowing how individual components can be matched, the audiophile is in a position to get the best out of his home music system, whether it consists of many components, of a few integrated units or of a complete packaged unit. Perhaps he cannot always solve a problem by the mere turn of a control, but understanding the problem and knowing how the solution can be attained, he may be able to make an adjustment in the equipment or to have a technician make this change for him. To the extent that he understands the situation, the chances are increased that the adjustment will be made correctly and at reasonable cost.

For the ultimate in quality, recourse is usually necessary to separate components. Since no one manufacturer makes the best of each type of component, the audiophile seeking the best possible system or the best system for a given budget will often discover that he can reach his objective by purchasing components one by one. Furthermore, there is the question of flexibility or versatility. Although many audiophiles have simple demands, others have fairly complex ones in terms of the functions and features they desire in their high-fidelity system. It is nearly impossible for combination components or packaged systems to meet all the possible wants and combinations of wants.

Ease of servicing must be considered in deciding between combination components and individual components with their greater problems of matching. If a combination component (such as a tuner-control amplifier-power amplifier on one chasssis) develops trouble, the entire unit must go out for service, and one may be without use of the phonograph for days or weeks even though the trouble lies only in the tuner. In addition, it is easier to find a fault in a single component than in a combination unit, so that servicing of a combination unit may be more costly.

Moreover, when the system consists of individual components, the audiophile himself can often determine in which component the trouble lies. For example, if his phonograph operates satisfactorily but the tuner does not, he knows the trouble lies not in the control amplifier, power

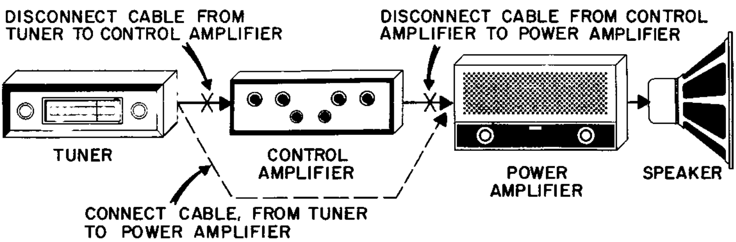


Fig. 6-1. Determining whether the control amplifier is malfunctioning by bypassing it.

amplifier, or speaker, but in the tuner. If neither tuner nor phonograph operates properly, he can bypass the control amplifier and connect the cable from the tuner directly to the power amplifier, as shown in Fig. 6-1. If this produces satisfactory sound (although of course the bass, treble, loudness, etc., facilities are not available), the finger of suspicion points to the control amplifier.

MATCHING IN TERMS OF QUALITY

That "a chain is no stronger than its weakest link" is partly true in high fidelity. We may think of a high-fidelity system (proceeding from signal source to control amplifier to power amplifier and finally to speaker system) as an audio chain. If any of the basic elements or its subdivisions are appreciably lower in quality than the rest of the system, the difference will quite likely be apparent in the quality of the sound.

TABLE 6-2
APPROXIMATE ALLOCATIONS OF A HIGH-FIDELITY BUDGET

$200-$300:	
Record changer and cartridge	30%
Combination control and power amplifier	35%
Speaker system	35%
$300-$500:	
Phonograph (changer or manual)	20%
Combination control and power amplifier	25%
FM or AM-FM tuner	25%
Speaker system	30%
$500 and Up ($800 typical):	
Manual turntable, arm, and cartridge	15%
Control amplifier	15%
Power amplifier	15%
AM-FM tuner	20%
Speaker system	35%

Since most audiophiles cannot spend as freely for high fidelity equipment as they might wish, their problem is to allocate the available budget as judiciously as possible among the four basic elements. A mistake all too commonly made is to spend a disproportionately large part of the budget upon components prior to the speaker system and then to buy a speaker system which qualifies barely if at all as high fidelity. The expenditure on the speaker should range from about one-third to half of the total sum, yet it often is one-fifth or less. For example, an audiophile may spend $200 for a control amplifier and power amplifier, $150 for an AM-FM tuner, $125 for a manual phonograph (turntable, arm, and cartridge purchased separately), and $50 for a speaker—less than one-fifth of the total expenditure. A wiser apportionment would be about $150 for the speaker system, with compromises in several of the other components to keep the total at $525 as before. (A good record changer with cartridge might be purchased for $75, saving $50, a less expensive tuner for $125 would save another $25, and the final $25 could be made up in the control amplifier or power amplifier or in both.)

Table 6-2 suggests approximate allocations of expenditure for three total amounts that the audiophile might spend. The table is based upon the FM tuner and phonograph as the most common signal sources. If a tape recorder and/or a TV set are to be part of the system, their cost must be added to the totals shown in the table. A tape machine compatible with high-fidelity standards and designed for home use will cost between $200 and $500. Semiprofessional and professional machines will range well upward of $500.

Although the chain can be no stronger than its weakest link, in audio, the weakening of other links can make the chain weaker still. Because one or more of the audio components contain limitations not yet overcome by the state of the art, an audiophile may think it fruitless to seek perfection in the other components. (This attitude stems in particular from the inadequacies of the speaker system.) Nevertheless, as listening can reveal, this does not mean that the purchase of fine components prior to the speaker is a waste of money. The tests that exist for measuring distortion are only indexes to quality, but the final judge must be the ear. Tests by ear indicate that, despite the limitations of the speaker, the high-grade phono cartridge, tuner, amplifier, and control amplifier produce superior sound.

MATCHING FREQUENCY RESPONSE

As indicated in Chap. 4, frequency response covering the full audio range can be obtained from all audio components but with more difficulty and therefore at more expense in the case of some components than others. The speaker system and the tape machine are particularly subject to constriction of range; low-priced phonograph cartridges also tend to be somewhat limited. When seeking high-fidelity reproduction, one must first ask whether moderate or top quality is sought. If top quality is the goal, then one must be prepared to make the necessary expenditures on those units in which it is most difficult to obtain full frequency response. Obviously, no matter what the frequency range of the control amplifier and power amplifier, they cannot reproduce all the sound if some other component in the chain, such as the speaker, has a limited range.

MATCHING SPEAKERS

Within the speaker system, the problem of matching frequency response can be a vital one, particularly if one purchases the individual components rather than an integral system (speakers and enclosure designed and sold as a unit by the manufacturer). Assume that the system comprises a woofer for the range below 800 cycles and a tweeter for the range above 800 cycles. It is unlikely that the two speakers would have the same efficiency, that is, produce the same volume of sound for the same amount of electrical power fed into them. Typically, the tweeter is more efficient.

Therefore its level should be reduced by means of a device called an attenuator. (A typical attenuator, the *L-pad,* is discussed in Chap. 9.) When the attenuator is put in, the problem is to equate the level of the two speakers. This can be done by a technician using instruments, or by the owner by ear. If the audiophile has a phonograph record that he trusts, one with good balance between the low and high frequencies when played on another system of proven quality, he can adjust the attenuator when playing this record on his own system.

A more scientific approach, though not necessarily one that produces better results, is diagrammed in Fig. 6-2. A frequency test record available in many high-fidelity stores, is played through the system, but

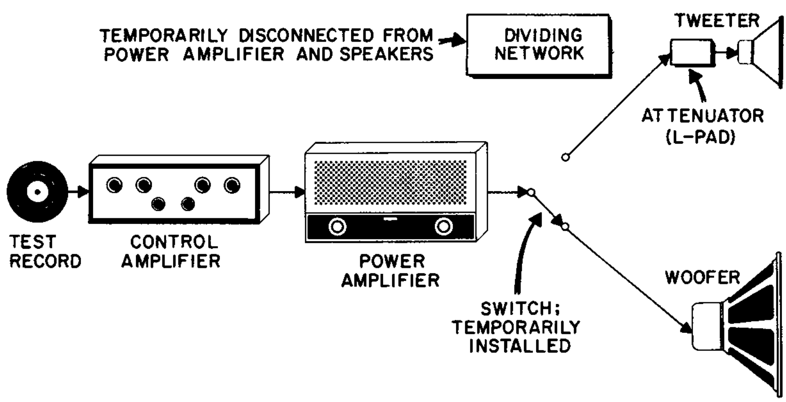


Fig. 6-2. Method of balancing two speakers.

the crossover network that divides the music between the woofer and the tweeter is temporarily removed. A switch is employed to feed the signal from the power amplifier either to the woofer (switch down) or to the tweeter (switch up). Asssume, as before, that the crossover frequency is 800 cycles; that is, 800 cycles is the dividing frequency between the two speakers. One plays a tone on the test record (all the tones are identified) as close as possible to the crossover frequency. The switch is alternately flipped up and down, meanwhile adjusting the tweeter attenuator, until both speakers alternately seem to produce the same amount of sound.

The position of the attenuator should be noted and considered a *tentative* point of balance. Speakers do not have perfectly smooth frequency response characteristics. Thus the woofer and tweeter may seem to be balanced at one frequency, but not at another nearby (Fig. 6-3).

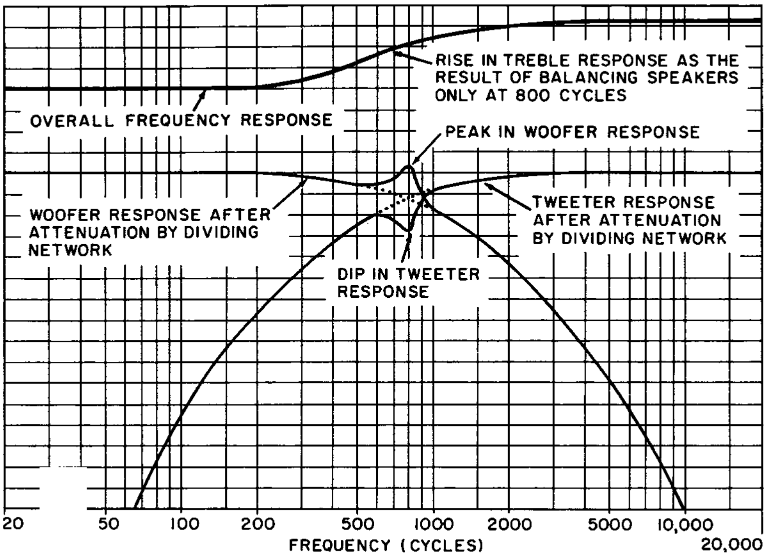


Fig. 6-3. How the treble range might become emphasized by balancing two speakers at one frequency.

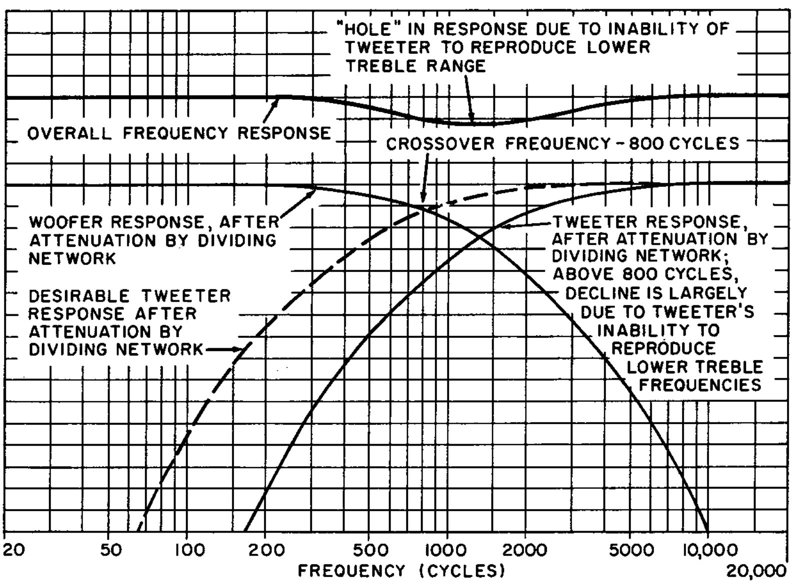


Fig. 6-4. How a hole in frequency response may result from insufficient tweeter response.

It may be seen that at the crossover frequency of 800 cycles the woofer has a peak and the tweeter has a dip. Thus balancing at that one frequency would result in considerable emphasis of the treble range compared with the bass.

The audiophile assembling his own speaker system may find that one of the speakers has inadequate frequency range, as shown in Fig. 6-4. It is assumed that the crossover frequency is 800 cycles, with the dividing network causing frequencies above this to be significantly attenuated in the woofer. Although the dividing network does not cause appreciable attenuation above 800 cycles in the tweeter, the tweeter may have an inherent decline in response below, say, 2000 cycles instead of below 800 cycles or less. (This is the situation shown in the diagram.) The net result is the response characteristic shown by the upper line. Response is then said to have a "hole" in the 800-cycle region. The dashed line in Fig. 6-4 shows what the frequency response of the tweeter should be for flat response throughout the audio range.

THE EFFECT OF THE ENCLOSURE AND THE LISTENING ROOM

In order to obtain the best possible response at bass frequencies, the speaker must be carefully matched to its enclosure; this is a complex subject. The most that can be said here is that before building or buying an enclosure for the speaker of his choice—assuming that this course is preferred to purchasing an integral speaker system—the audiophile should obtain as much information as he can from the speaker manufacturer as to the recommended enclosure in terms of size, construction, internal bracing, internal lining with sound absorbent material, etc. If possible, before the speaker is bought, it should be heard in an enclosure similar to that which is to be purchased or built.

Although a particular speaker and enclosure may sound well together in a sound salon, it is possible that they may not sound well in the audiophile's home. (The reverse is also possible.) The speaker system must be matched to the room in which it is to be used. The skilled audio salesman, given a complete description of the listening room in terms of size, shape, and furnishings, can suggest speaker systems, by type and by brand, that are apt to provide good results. Some speaker systems have a more brilliant sound than others, and these may be most effective in a room that is heavily carpeted and draped, with plenty of overstuffed furniture. A speaker system that perhaps sounds dull in the store may come to life in a hard-surfaced room that has a minimum of furniture.

The principle of matching the speaker system to the room has an analogy in the choice of phono cartridge. Although the control amplifier, power amplifier, tuner, and tape machine of high quality are largely devoid of coloration, this is not as true of the phonograph cartridge. Accordingly, a skilled audio salesman may recommend a phono cartridge with a slightly dimmed high end to offset the coloration of a speaker system with some exaggeration in the treble. Similarly, a brilliant cartridge may go well with a somewhat "bassy" speaker.

MATCHING SPEAKERS FOR STEREO

If the audiophile plans on a stereo installation and wishes the best possible results, each speaker system should be of the same kind, although this may be more costly and space consuming than adding a smaller and less expensive speaker system to the one already in use for monaural listening.

Figure 6-5 shows what can happen if poorly matched speakers are employed. Speaker B is assumed to have relatively flat response, whereas

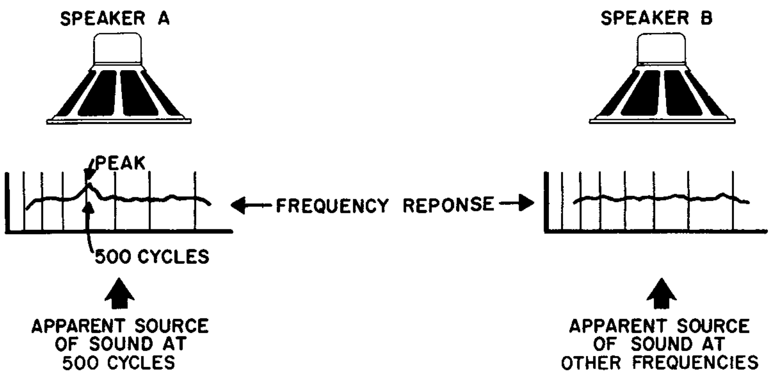


Fig. 6-5. Apparent shift in the position of a sound source due to unmatched speakers in a stereo system.

speaker A has a peak in the region of 500 cycles. Suppose that a violin is playing a note with a dominant frequency of about 500 cycles, and that the sound of the violin is supposed to issue principally from the speaker B. (The instrument appears to be on the right-hand side of the orchestra.) The peak of speaker B will cause it to produce much more of the sound of the violin at 500 cycles than it should. Therefore the violin may appear to be on the left instead of the right. But if the violin plays another frequency, at which speaker B does not exhibit a peak, the violin will seem to shift back to the left side. A dip in the response of one of the speakers will produce a similar effect. In either case, the apparent wandering of the sound source destroys much of the realism of which stereo is capable.

MATCHING SIGNAL LEVELS

One of the identifying and delightful characteristics of a truly high-fidelity music system is that the music issues from a background of dead silence, given a high-quality record, tape, or FM program. When there is a pause in the program material, nothing should be heard from the speaker at a distance greater than a foot or two. Another all-important characteristic is low distortion, which gives a seeming transparency (often described as cleanness) to sound that is smooth in texture. It is in large part for these two qualities that the audiophile spends his money.

Despite careful selection of well-recommended and expensive components, the audiophile often winds up with a system producing an objection-

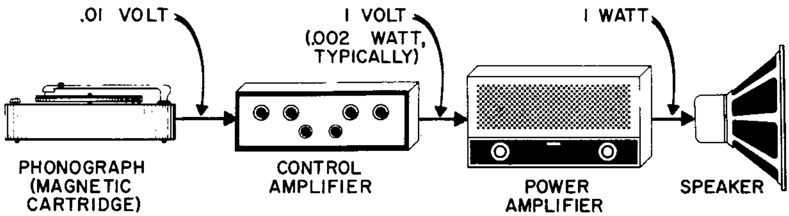


Fig. 6-6. Signal levels in the audio chain.

able amount of noise (we use this term to include hum as well) or an objectionable amount of distortion or a combination of both. Let us assume, as will be true in the majority of cases, that there is nothing wrong with the components as such, and that they live up to the manufacturers' technical ratings. The fault will then lie in two things: (1) the purchase of components ill-suited to each other or (2) improper use of the components in conjunction with each other. In both cases, the source of the difficulty is the amount of signal presented by each component to the next. To illustrate how the audio signal varies in level as it proceeds through the audio chain, Fig. 6-6 follows the course of a signal voltage obtained from a magnetic phono cartridge.

Too much audio signal going into a component can cause distortion *(overloading)*; too little signal results in excessive noise. Each component produces some noise which is present at its output. If the incoming audio signal is weak, the outgoing audio signal is proportionately weak. Therefore the ratio between the outgoing signal voltage and the outgoing noise voltage is low. The problem of matching signal levels is to find the optimum intermediate point between too much distortion and too much noise. When high-caliber components are used, this intermediate point is not critical, although it tends to become so when components of lesser quality are employed. In the following discussion we shall examine the relationship between each basic element of the audio chain and the following element with respect to the effect of signal level upon noise and distortion.

MATCHING THE POWER AMPLIFIER AND SPEAKER SYSTEM

Whereas power amplifiers of 10 or 20 watts were generally considered ample for home use in the earlier days of high fidelity, there has been an

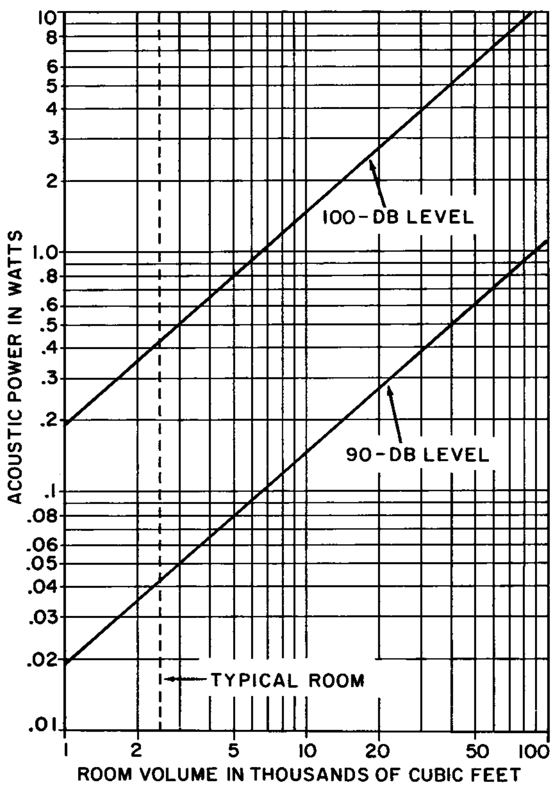


Fig. 6-7. Acoustic power required to produce sound levels of 90 and 100 db in rooms of various sizes.

increasing trend toward units capable of producing 50 watts, 100 watts, or even more, although many music systems do not actually put more than 1 to 5 watts into use. The principal reason for these relatively tremendous wattages has been to insure that the power amplifier will not develop noticeable distortion in driving the speaker system to the desired volume. By designing an amplifier for, say, 60 watts, excellent performance is assured at operating levels below 10 watts. On the other hand, it is possible to design an amplifier of relatively low power, say 10 or 12 watts, that will perform just as well as the high-power amplifier at a few watts.

In purchasing a power amplifier, particularly if cost is also a consideration, account should be taken of the power requirements of the speaker and of the acoustic demands of the listener. Figure 6-7 shows the acoustic power required for rooms of various size in order to produce a volume level of 100 db and 90 db (above the threshold of hearing, as was explained in Chap. 2). 100 db is about the maximum tolerable level of sound in the home and is very close to the maximum level of sound that one would encounter at a seat well up front in a symphony hall.

For purposes of discussion, let us consider the acoustic requirements of a room that is typically furnished and rather good-sized, say 20 $\times$ 15 $\times$ 8½ feet, comprising about 2,500 cubic feet. As can be seen in Fig. 6-7, such a room requires about 0.4 watt to produce a level of 100 db, which most persons would find intolerable in the home. A level of 90 db is probably closer to comfort for the majority.

Speakers do not convert all the electric power supplied them into acoustic power, but are inefficient to varying degrees. A speaker with an efficiency of 8% (the other 92% going into heat instead of sound) can be said to be about average. An efficiency of 20% is high, and a speaker with 1% efficiency would be classed as a low-efficiency unit. Based on these three typical efficiencies, Fig. 6-8 shows the electrical power required to produce 100 db of sound in rooms of varying size. It may be seen that for a typical good-sized room of 2,500 cubic feet, the high-efficiency speaker would require an electrical power of 2 watts; the moderate-efficiency speaker, 5 watts; the low-efficiency speaker, 40 watts.

In order to have low distortion at all sound levels, the audiophile purchasing a speaker system known to have low efficiency (although as a rule, it has high quality) does well to match it with a power amplifier rated at 40 watts or more. The individual with a speaker of average to high efficiency may be acquiring extra noise along with extra watts if he buys a power amplifier with a capacity that he does not require. Every amplifier produces some noise. The more power that an amplifier is

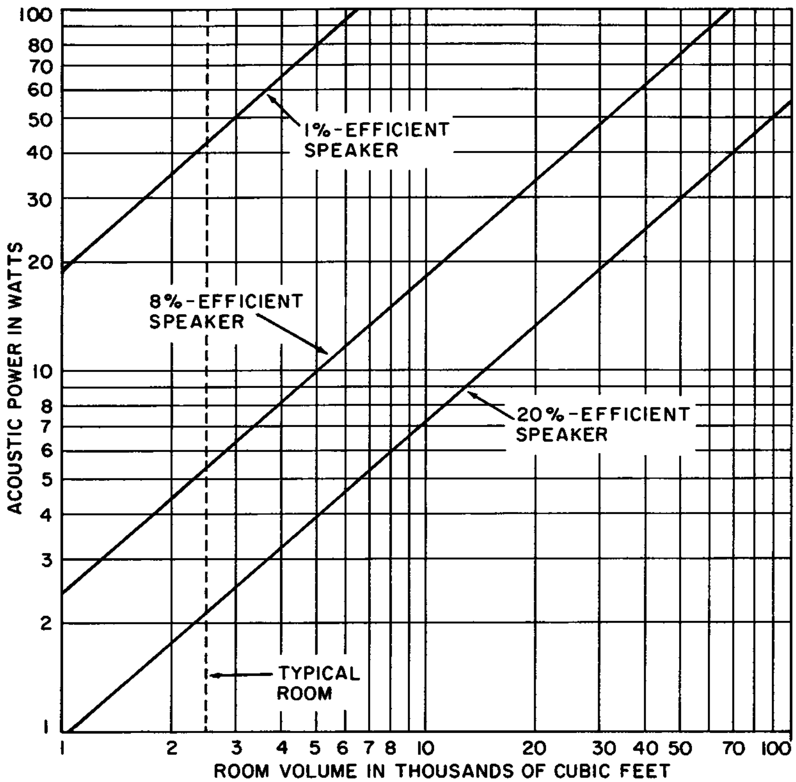


Fig. 6-8. Electrical power required to produce 100 db of sound in rooms of various sizes with speakers of three different efficiencies.

capable of developing, the more it amplifies its internal noise. As a general principle, more noise is fed to the speaker by a high-power amplifier than by a low-power amplifier, assuming the two units are otherwise equal (in terms of design, quality of parts, etc.).

Can the additional noise of a high-power amplifier be heard? The answer depends upon the quality of the amplifier and upon the efficiency of the speaker. In the case of a moderate quality power amplifier, producing a fair amount of noise, this noise may be inaudible with a low-efficiency speaker, which attentuates the noise along with the audio signal. If a more efficient speaker is used, however, the noise may become audible. Its constant and insistent presence can be quite annoying, particularly when listening at low sound levels in a very quiet room. If the audiophile were to use a power amplifier of the same quality but less power, the noise level might be just enough smaller to become inaudible, even with an efficient speaker.

On the other hand, the audiophile able and willing to purchase expensive, top-quality components can find power amplifiers that have not only a large capacity but also a signal-to-noise ratio that is so high that their noise in inaudible even through very efficient speakers.

In using an amplifier with a good deal more power than is required, one must consider the possible danger of damaging the speakers by accidental application of too much power. High-fidelity speakers are customarily designed to withstand from 10 to 25 watts of power. An accidental blast of 60 watts from a high-power amplifier can damage or destroy a speaker. A blast of this kind may occur if the volume control is improperly set, if a lead breaks inside the amplifier, if a switch malfunctions, or if the phono arm is dropped onto a record. Because of this danger, at least one of the finest power amplifiers on the market provides a switch that cuts its maximum power output by half, for use with the more efficient speaker systems. It may be noted that a number of reviews of this amplifier have commented on the absence of a noticeable difference in quality of sound when the available power output is cut in half. (Another solution to this problem is to use a fuse in the speaker cable.)

All told, the well-advised audiophile purchases a power amplifier that is suited to the speaker system in terms of producing sufficient power at acceptably low distortion, keeping noise at an inaudible level, and minimizing the danger of damaging the speaker.

MATCHING THE CONTROL AMPLIFIER AND POWER AMPLIFIER

The volume control on the control amplifier covers an arc of about 300 degrees; it may be said to rotate from about 7 o'clock at minimum volume to about 5 o'clock at maximum volume, as shown in Fig. 6-9. The

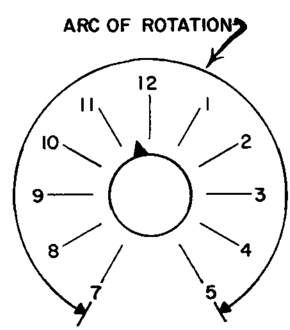


Fig. 6-9. Identification of volume-control position.

level of sound that one would hear at a live performance should be found at about the 2-o'clock or 3-o'clock position of the control, leaving the remainder as a reserve of power for occasions when one wants the music even bigger than life, or when the signal source is unusually low in level.

Often, however, one finds the situation where a tremendous volume of sound is produced when the control is hardly "cracked open." Thus, in some systems, turning the control to about 10 or 11 o'clock produces all the sound that one can bear. This is a case of improper matching between the control amplifier and the power amplifier. It is bad because turning the volume control all the way up can do injury to the speaker and possibly to the listener's nerves or ears. But the trouble may go deeper than this. It may be symptomatic of a situation in which noise or distortion or both are unnecessarily excessive.

Possibly one may hear appreciable noise from the speaker, caused by the control amplifier rather than the power amplifier, (as can easily be ascertained if the noise vanishes when the control amplifier is temporarily taken out of the system). In this event, the signal from the control amplifier should be cut down to match the needs of the power amplifier. Typically, a power amplifier requires about 1 volt to drive it to full power. Considering that a speaker of average or high efficiency ordinarily uses only a fraction of the power amplifier's capacity, the maximum signal required to drive the power amplifier may be well under 0.5 volt. Yet the control amplifier, with its volume control full on, may be delivering 3 to 15 volts to the power amplifier. Along with the extra voltage, the control amplifier is supplying extra noise.

Most power amplifiers incorporate an input-level set to cut down the signal from the control amplifier. Figure 6-10 is the electrical represen-

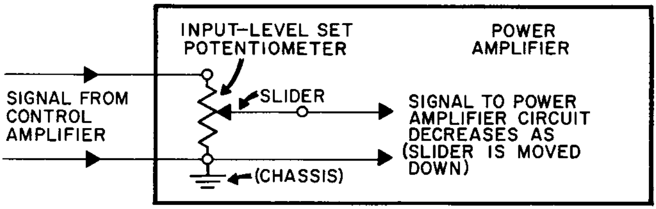


Fig. 6-10. The input-level set in a power amplifier.

tation of this simple device, often called a potentiometer. The signal voltage coming into the power amplifier is applied across a resistor, represented by the wavy line. A moving arm, called the slider, can be placed at any point on the resistor and thus take off any desired fraction of the incoming signal voltage. If the control amplifier presents too much signal to the power amplifier, the slider arm is moved down until the power amplifier (and, in turn, the speaker) produce the limited amount of power that is desired. Thus the noise as well as the audio signal from

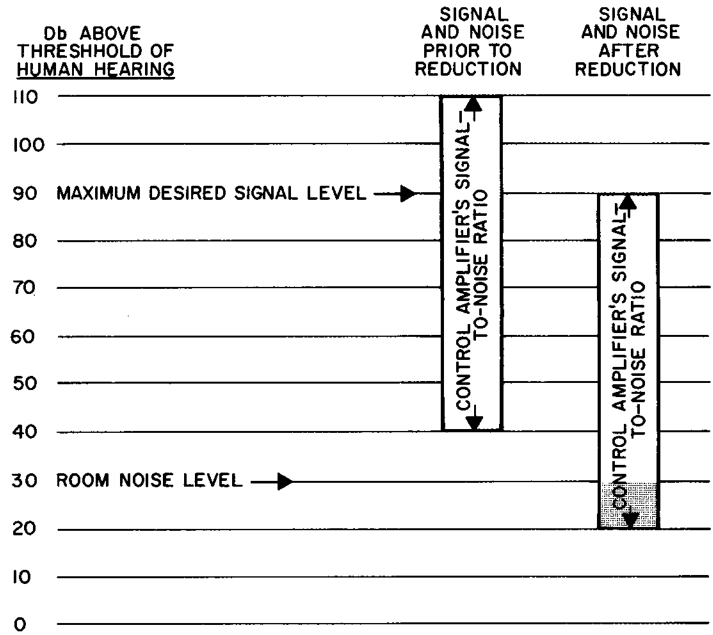


Fig. 6-11. Reduction of signal and noise output from the control amplifier by means of an input-level set on the power amplifier.

the control amplifier is reduced as shown in Fig. 6-11. The maximum audio signal is brought down to a level that suits the ear, and at the same time the noise from the control amplifier is reduced until it is masked by the ambient noise of the room.

Whether the power amplifier's input-level set may be turned down far enough to eliminate all the noise produced by the control amplifier depends upon the characteristics of the control unit. If the unit is of but moderate quality, it may be unable to furnish enough high-quality signal to drive the power amplifier when the input-level set is turned down enough to eliminate all the noise.

The magnitude and quality of the signal produced by the control amplifier depend upon the amplifier's sensitivity and upon the distortion that it produces. Sensitivity is the maximum ratio (volume control all the way up) between the signal voltage coming out of the control unit and the signal voltage going in. Thus if the incoming voltage from the FM tuner is 1 volt, a control amplifier with a sensitivity of 10 can turn out a maximum of 10 volts. If the sensitivity of the control amplifier is low and if the signal from the control unit is substantially reduced by the power amplifier's input-level set, the remaining signal may not be strong enough to drive the power amplifier and speaker to the desired volume.

Even though a control amplifier has sufficient sensitivity to permit its outgoing signal to be substantially reduced, the remaining signal may not be usable because it contains excessive distortion. To illustrate, assume that a control amplifier can produce 5 volts from the signal fed to it by the FM tuner. Assume that after the input-level set of the power amplifier has been turned down there is a 0.5-volt signal left, which is just enough to drive the power amplifier and speaker. If, however, the control amplifier produces 5% IM at an output level of 5 volts, one would have to settle for lower signal voltage from the control amplifier. The input-level set of the power amplifier would have to be turned up, resulting in more audible noise at the output.

MATCHING SIGNAL SOURCES AND THE CONTROL AMPLIFIER

There is a limit to the amount of audio voltage that a control amplifier can accept from the signal source. Beyond this limit, the control amplifier is overloaded and produces distortion. Some control amplifiers can accept as much as 1 volt without producing more than 0.1% distortion, whereas others may be limited to a 0.1-volt input for the same amount of distortion. Accordingly, means must often be employed for reducing the amount of signal fed to the control amplifier.

A number of control amplifiers permit attenuation of the incoming signal by means of input-level sets similar to that already shown in Fig. 6-10. Sometimes an individual level set is provided for each signal source as shown in Fig. 6-12. Other control amplifiers, however, do not contain individual level sets. Instead, they tend to follow the configuration of Fig. 6-13, where the signal goes directly into the volume control after being selected by the function switch. The volume control in effect also serves as an input-level set.

This technique may result, at least for some signal sources, in an excessive level of sound below the 2-o'clock position of the volume control. The cure for this is to reduce the signal level at the source itself. In the case of an FM or AM tuner, this can easily be done because the tuner generally contains a volume control of its own. This is also true

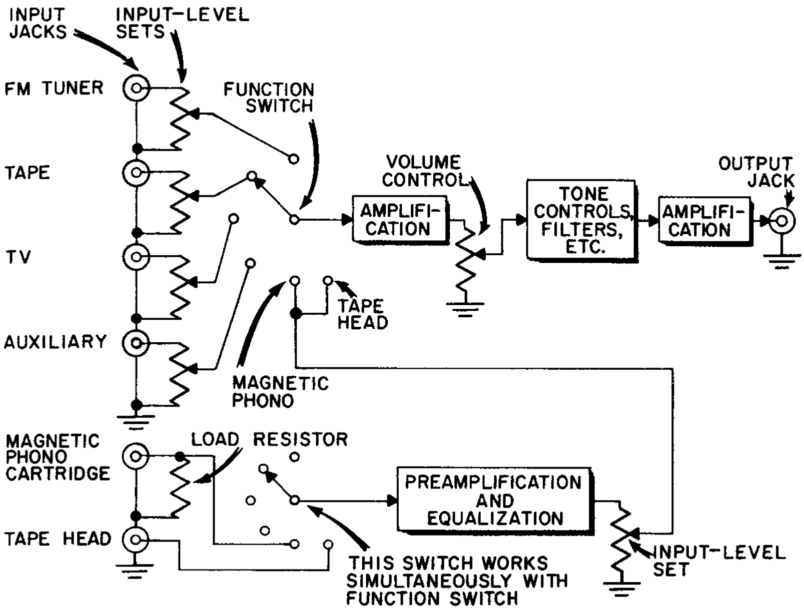


Fig. 6-12. Block diagram of a typical control amplifier equipped with input-level sets.

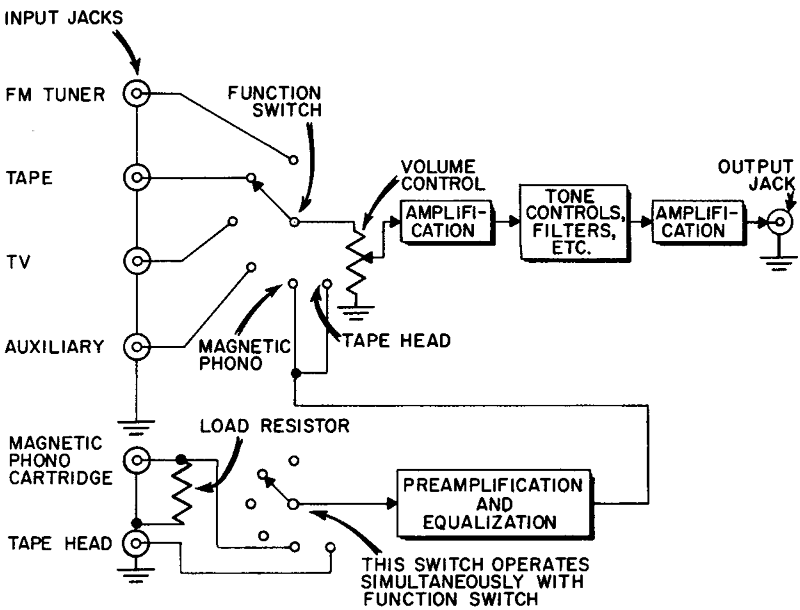


Fig. 6-13. Block diagram of a typical control amplifier in which the volume control acts as an input-level set.

of a tape machine or TV. However, there is no volume control associated with a phonograph pickup. To meet the problem in the case of a magnetic cartridge, many control amplifiers provide two inputs, one designated by a term such as "high magnetic" and the other as "low magnetic."

Often there is no special input for piezoelectric cartridges, a number of which are capable of high-fidelity performance. This makes it necessary to feed this type of cartridge into one of the inputs designated for a high-level source. Such an input, for example, might be marked "auxiliary." The signal from the piezoelectric cartridge is a rather large one, often over 1 volt. Therefore in the case of a control amplifier such as the one shown in Fig. 6-13, high volume would be attained at a low setting of the volume control. To cut down the signal from the piezoelectric cartridge, a simple change can be made in the control amplifier, a task ordinarily for the audio technician. (Of course, before getting into the problem of making a change in a control amplifier, the audiophile might reconsider his choice of pickup or control amplifier.)

MATCHING THE COMPONENTS BY EAR

Although it is preferable to have a competent technician perform the task of matching the components to each other, so that the various signal voltages and the level of distortion can be measured, the audiophile can generally do a satisfactory job of matching by ear, once he understands the problems involved. The better the components that he owns—the lower their noise and distortion and the more voltage the control amplifier can turn out at low distortion—the easier his task. The following is a suggested procedure.

1. With the volume control of the control amplifier full on, the input-level set of the power amplifier should be adjusted until no noise is audible a few feet from the speaker. (This assumes that power amplifier noise is insignificant.)

2. With the volume control of the control amplifier at 2 o'clock or 3 o'clock and with the function switch turned, say, to the FM tuner as a signal source, the input-level set of the control amplifier should be adjusted until maximum desirable volume is obtained. If the control amplifier has no input-level set, the volume control of the tuner should be appropriately adjusted.

3. With other signal sources fed into the control amplifier, the input-level set for each source (or, in the absence of input-level sets, the volume control of a TV set or tape machine) should be adjusted as in

step 2. If the control amplifier has no input-level set for the phono cartridge, the audiophile may have a technician make the necessary change to reduce the incoming signal from it; or he may rest satisfied in having loud volume occur substantially below 2 o'clock on the volume control.

If the components used are appreciably below top quality, it is possible that this procedure may lead to excessive distortion as the price of quiet operation. To ascertain whether this procedure produces too much distortion, the audiophile can play (at loud level) a phonograph disc known to be of high quality.

It is possible that after adjusting the power-amplifier level set, the signal from the control amplifier may be too small to drive the power amplifier, apart from the question of distortion. In this case it is necessary to readjust the input-level set of the power amplifier and bear with a certain amount of noise from the control amplifier.

ELECTRICAL MATCHING

There are problems of matching components to each other so that they operate properly in terms of electrical performance. Of course, this is a technical matter and not appropriate for detailed discussion here. However, enough should be said on the subject to alert the audiophile to the fact that such problems exist and can stand between him and high-fidelity performance. Three important instances will be mentioned here.

One case in point is proper electrical matching between a phonograph cartridge and the control amplifier. The frequency response of a piezoelectric cartridge (and usually that of a magnetic one, as well) is dependent upon the resistance at the input of the control amplifier, (the *load resistance*). The required load resistance varies according to type and brand of cartridge. Poor frequency response of a cartridge—too much or too little bass; too much or too little treble—can often be traced to incorrect load resistance. In connecting components to each other by means of cables, there is always the danger of high frequencies being attenuated. Properly designed equipment minimizes this danger. More will be said on this in Chap. 8.

In connecting a microphone to a tape recorder, a power amplifier to a speaker, and in other instances, the problem arises of obtaining the most efficient transfer of signal voltage or signal power. This problem will also be dealt with in Chap. 8.

7. loudness compensation ■

One of the principal attributes of high-fidelity sound is flat frequency response, but we cannot merely be concerned with flat response in the audio system. We wish to *hear* a balance among frequencies that is as close as possible to the original sound; we desire response that is flat *to the ear.*

Unfortunately, it is a characteristic of human hearing that when the volume of sound is reduced appreciably below the original level, the bass seems to be attenuated more than the middle and high notes. By preference or necessity, most of us listen to reproduced music at levels substantially lower than the live performance. We seldom attempt to bring into our homes the actual sound level of a symphony orchestra, a brass band, a 100-voice choir, or a jazz combo. However, as we reduce the volume to fit the circumstances, we find the bass frequencies seem to be attenuated or even completely lost. (To a smaller degree, there is also some apparent attenuation of high frequencies, as will be discussed later.)

The apparent loss of bass at reduced volume is widely known as the Fletcher-Munson effect. It can be seriously detrimental to the pleasure derived from reproduced music, because this range supplies body and warmth to music. We are not referring only to the very lowest notes (those below 70 or 80 cycles), but rather to the frequencies up to about 400 cycles, which are produced in some degree by virtually all the instruments. (Refer back to Table 4-1.)

When these frequencies are reduced in apparent intensity relative to the rest of the audio range, the music as a whole tends to sound "thin," and is likely to become irritating. When the bass is in apparent balance, it preserves both the texture of the sound as a whole and the individual tone quality of each instrument.

The Fletcher-Munson effect can be overcome in two ways. One method is simply to turn up the volume until it is about as loud as the original sound, thereby restoring balance of the bass with the rest. Many persons, such as those living in apartment houses or those whose inherent preference is for music played quietly, do not find this solution practical. The other method is to counteract the seeming loss of bass at reduced volume by adding bass emphasis in the control amplifier. This is known as *loudness compensation.*

In the following discussion, we shall have a close look at the Fletcher-Munson effect and the means of providing compensating bass boost. Through a clear understanding of the "loudness problem," the audiophile is in a position to obtain the correct amount of loudness compensation for maximum listening enjoyment.

THE FLETCHER-MUNSON CURVES

In the early 1930's, an extensive program of research into the characteristics of human hearing was conducted by Fletcher and Munson of the Bell Telephone Laboratories. Perhaps the most publicized of their findings concerns the variation in sensitivity of the ear at various frequencies. To measure relative loudness, 1000 cycles was used as a standard of reference, and the amount of acoustic power required to produce a sound of the same loudness at other frequencies was determined. Figure 7-1 shows the results of the investigation.

The horizontal scale represents audio frequencies, and the vertical scale represents various levels of acoustic power, measured in db (decibels) above the threshold of hearing at 1000 cycles. Let us consider the curve marked 60 db. Following the curve to various frequencies (to its intersection with some of the vertical lines), we see that where it intersects the line representing 200 cycles, it has risen to a level of 65 db. This means that 200 cycles must be reproduced at an

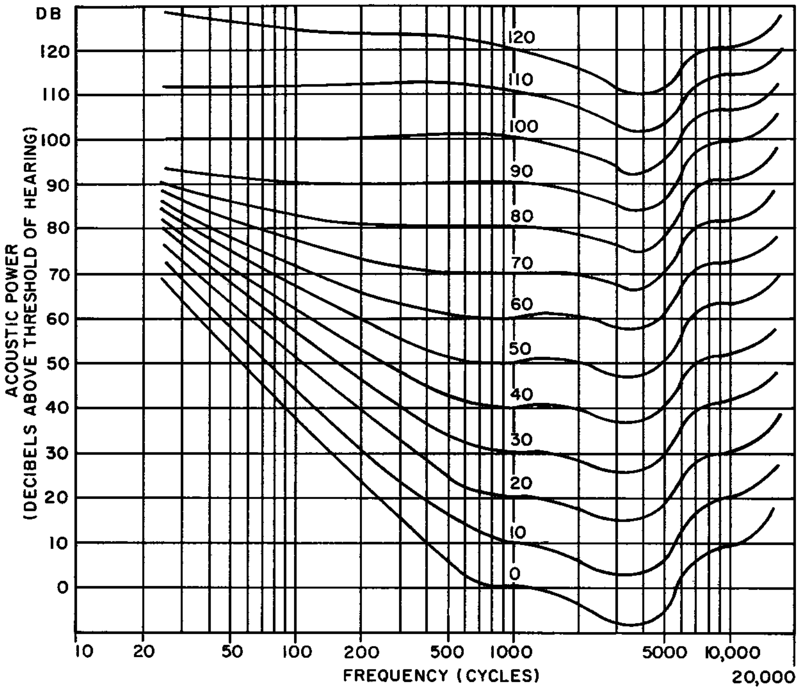


Fig. 7-1. The Fletcher-Munson curves.

acoustic power of 65 db in order to sound just as loud as 1000 cycles reproduced at 60 db (1,000,000 times the minimum audible level). In other words, the acoustic power at 200 cycles must be 5 db more (about three times as great) in order to sound as loud as the 1000-cycle reference tone. At 100 cycles, the curve has risen to 71 db. That is, the 100-cycle tone must have 11 db (about 13 times) more power than a 60-db 1000-cycle tone to sound as loud. At 50 cycles, a tone must contain 18 db (63 times) more power, than a 60-db 1000-cycle tone in order to sound as loud.

Turning to the 100 db curve, we can readily see that if 1000 cycles is produced at a level of 100 db (10,000,000,000 times more power than the minimum audible amount), all the frequencies below 1000 cycles will sound just as loud as 1000 cycles if they are produced at the same power. Clearly, when the acoustic level of 1000 cycles is less than 100 db, more power has to be generated at lower frequencies if all of them are to sound equally loud. The greater the reduction in volume at 1000 cycles, the more the low frequencies have to be boosted relative to 1000 cycles for equal loudness.

A more direct way of viewing the Fletcher-Munson effect is by means of Fig. 7-2, which is a transposition of the curves shown in Fig. 7-1. It shows the apparent decline of bass frequencies to the ear as volume is reduced. At a seat well up front at a symphony concert, the maximum

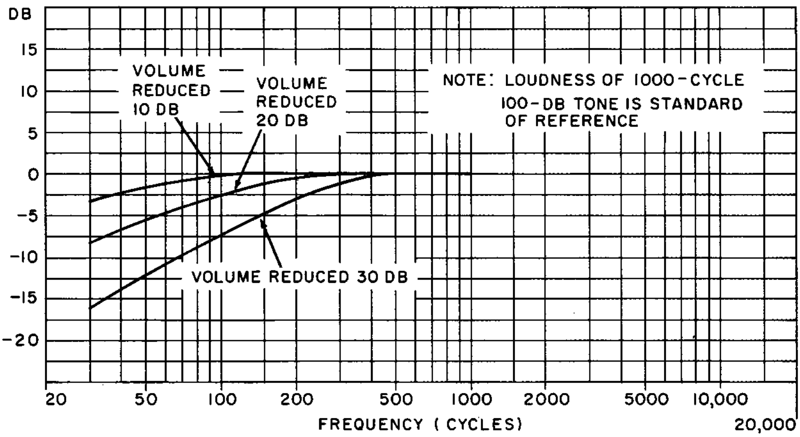


Fig. 7-2. The decline in apparent loudness in the bass as volume is reduced.

sound level heard by the ear is about 100 db. In the home, this level is apt to be reduced to varying degrees, although probably not by much more than 30 db (to 1/1000th of the original power). To represent practical situations, it is assumed that there is attenuation of 10, 20, or 30 db below the sound level heard at the original performance.

Looking at the curve marked —20 db, for example, we see the effect upon the ear of dropping volume level 20 db below the level that would have been heard at the original performance. The curve shows virtually no apparent loss in bass response until one gets below frequencies of 300 cycles. At 100 cycles, there is an apparent loss of nearly 3 db. There is a loss of 6 db at 50 cycles and of 8.5 db at 30 cycles. In the case of a 30-db reduction from original volume, the apparent bass losses become much greater—over 3 db at 200 cycles, 7.5 db at 100 cycles, 125 db at 50 cycles, and 16.5 db at 30 cycles. This effect also takes place with music that never reaches a maximum level of 100 db at the original performance, although the amount of apparent loss in the bass may not be quite the same as indicated in Fig. 7-2. The curves shown represent average bass losses experienced by a group of persons participating in the Fletcher-Munson experiments, and they are not necessarily the losses that would be experienced by a given individual.

THE EFFECT UPON TREBLE FREQUENCIES

Figure 7-1 shows that varying amounts of power must be generated at high frequencies as well as at low ones for each frequency to sound as loud to the ear as 1000 cycles. Accordingly, loudness compensation is also indicated for the treble range, particularly above 5000 cycles. However, the amount of treble compensation called for is much smaller than the bass compensation required.

To illustrate this point, consider the curve marked 100 db. At 15,000 cycles, about 12 db more acoustic power is necessary than at 1000 cycles in order for both frequencies to sound equally loud. In the case of the curve marked 70 db, 15,000 cycles must be reproduced at about 18 db more power than 1000 cycles in order for both frequencies to sound equally loud. Thus the requirements for equal loudness increase only 6 db (from 12 to 18 db), although the volume level at 1000 cycles is dropped 30 db. For the same drop in volume level, the bass boost at 30 cycles should be 16.5 db. Figure 7-4 translates the Fletcher-

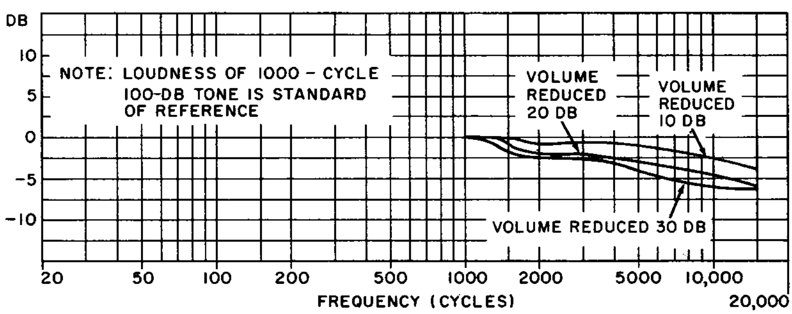


Fig. 7-3. The decline in apparent loudness in the treble as volume is reduced.

Munson curves into the apparent decline in treble response as volume is reduced from 100 to 70 db, and shows the relatively small amount of treble boost that is needed.

As a result, many high fidelity control amplifiers provide automatic loudness compensation only for bass frequencies. To the extent that treble compensation is required but not automatically supplied, a slight turn of the treble control is usually sufficient. In view of this fact, the following discussion deals only with problems of bass boost; however, what is said there may be taken to apply in principle to treble boost as well.

ACHIEVING LOUDNESS COMPENSATION

Bass boost to compensate for the Fletcher-Munson effect can be obtained in two ways: (1) by turning up the bass control; (2) through the loudness control, which is used in place of the volume control and

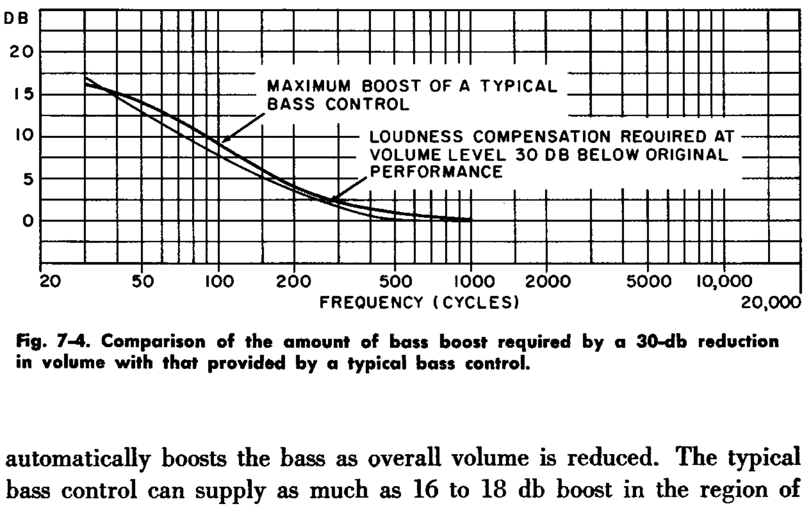


Fig. 7-4. Comparison of the amount of bass boost required by a 30-db reduction in volume with that provided by a typical bass control.

automatically boosts the bass as overall volume is reduced. The typical bass control can supply as much as 16 to 18 db boost in the region of 30 to 50 cycles. Thus the 16.5-db compensating bass boost required for a 30-db drop in volume is within the range of the typical bass control as shown in Fig. 7-4.

If volume is decreased more than 30 db, however, the usual bass control is no longer adequate to the task of loudness compensation. On the other hand, volume decreases greater than 30 db are infrequent. When they do occur, many of the softer passages are masked by the noise of the listening room. It then becomes pointless to try to balance the bass notes with higher notes that cannot be heard. Figure 7-5 illustrates this point, assuming that the original sound has a maximum level of 100 db, an average level of 70 db, and a minimum level of 50 db. It is further assumed that the room is fairly quiet, having a noise level of only 25 db. If volume is reduced by 35 db, this brings the maximum level down

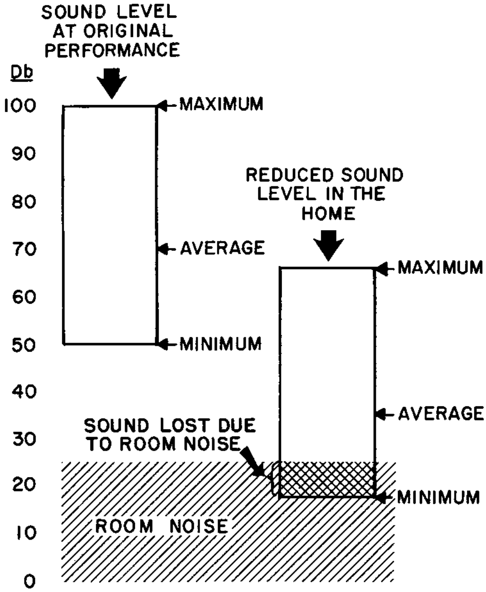


Fig. 7-5. How a reduction in the volume level causes part of the music to become inaudible.

to 65 db, the average level down to 35 db, and the minimum level down to 15 db. Although the loudest and moderate sounds remain well above room noise and are therefore quite audible, the softest sounds are obscured by the 25 db of room noise.

THE LOUDNESS CONTROL

Like the volume control, the loudness control is used to increase or decrease the total sound level. But unlike the volume control, which affects all audio frequencies equally, the loudness control is a discriminatory device. It automatically boosts the bass *relative to the other frequencies* when volume is lowered. The greater the decrease in volume, the greater the relative bass boost. Figure 7-6 shows the effect of a typical loudness control. Using 1000 cycles as a standard of reference, volume reductions of 10, 20, and 30 db from the original level are represented. Note how the amount of bass boost increases as the volume is lowered. One may look upon the loudness control as a differential device that changes the level of the bass notes at a slower rate than that of the other frequencies.

If the loudness control is operating properly, it should present no bass boost when the overall volume corresponds to that of the original performance. Accordingly, an adjustment is required so that when the loudness control is turned to maximum or near-maximum position, the reproduced sound level will approximate the original one. In a typical

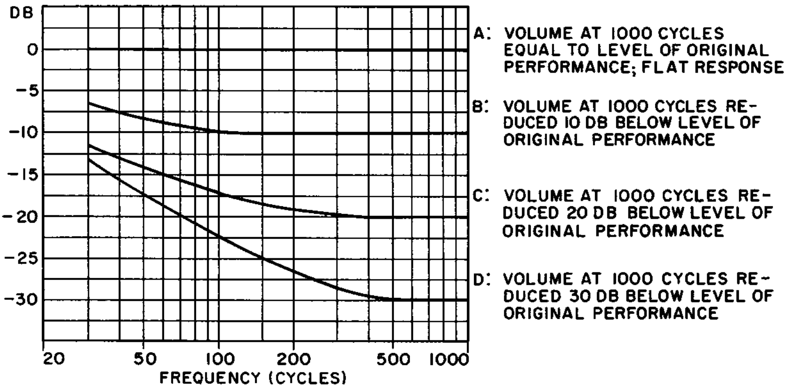


Fig. 7-6. Bass response of a typical loudness control at various volume settings.

control amplifier, this adjustment is performed by means of the volume control. First the listener turns the loudness control all or most of the way up. Then he sets the volume control so that the music sounds about as loud to his ears as it would have had he been present at the original performance. Then he backs down on the loudness control to the desired listening level. As he gradually reduces level by means of the loudness control, it automatically introduces an increasing degree of bass boost, as already explained, to compensate for the Fletcher-Munson effect.

Although the loudness control can go a long way toward restoring balance at low listening levels, it must be realized that in two respects its action may be less satisfactory than at first appears. First, the degree of bass compensation it supplies generally conforms to the Fletcher-Munson curves, which are based upon the *average* hearing characteristics of a number of persons rather than upon the *specific* hearing characteristics of the individual listening to the music. Second, the loudness control is not a completely automatic device. In order for it to operate properly, as already pointed out, the volume control must be set so that maximum position of the loudness control corresponds to the sound level at the original performance. Unfortunately, not all signal sources are of the same level. If one happens to set the volume control on the basis of a relatively weak signal source, the loudness control does not operate correctly when a strong signal source comes in, and it is necessary to reset the volume control.

Because of these facts, some audiophiles have found that they derive greater pleasure from reproduced music by deactivating the loudness control (turning it all the way up, so that it produces no bass emphasis), and they obtain bass boost through the bass control instead. The overall sound level is then governed by the volume control. At low listening levels they accentuate the bass, and the treble as well, in accordance with their personal tastes. On the other hand, many audiophiles find that the loudness control in their music system is a very satisfactory means of coping with the Fletcher-Munson effect. The only safe conclusion to be drawn, therefore, is that the audiophile owes it to himself to try both methods of bass compensation. A short period of experimentation should enable him to conclude which method sounds best to *his* ears.

OTHER TYPES OF LOUDNESS CONTROL

Although the loudness control described in the preceding section is the most common type, there are some variations that deserve mention here. In one, there are no separate loudness and volume controls. Instead, the volume control is converted into a loudness control by means of a switch. In one position of the switch, the control affects all frequencies equally as sound level is changed. In the other switch position, the bass frequencies are boosted relative to the middle and high notes when the control is turned appreciably below maximum position. In order to enable the loudness control to operate properly, the volume should be adjusted by the input-level sets of the control amplifier or the volume controls of the signal sources.

Another variation is sometimes called a loudness contour switch. It has two or more positions that provide various amounts of bass boost. The bass compensation is each position of the switch may or may not vary as the volume control is turned up or down.

8. equalization ■

The functions of a high-fidelity system are in large part concerned with shaping the frequency characteristic of the sound. There are bass and treble controls to accentuate or deemphasize the low and high frequencies; sharp-cutoff filters to attenuate the extremely low frequencies or extremely high frequencies, so as to suppress rumble or noise; presence controls to emphasize the midrange, imparting a closeup characteristic to the music; and loudness controls to overcome the Fletcher-Munson effect. Despite all these possible departures from flat frequency response, it is nevertheless important that at easily recognized settings of these controls one should be able to obtain flat response so far as the electrical signal is concerned. Flat response is a reference point to which the listener can always return when different program sources, different acoustic conditions, or other factors require a different shaping of the response of the audio system.

Although flat response is a basic requirement, the audiophile will soon discover that high-fidelity program sources do not furnish a signal with a flat frequency characteristic. Either bass or treble or both are exaggerated or deemphasized; it is a vital task of high-fidelity equipment

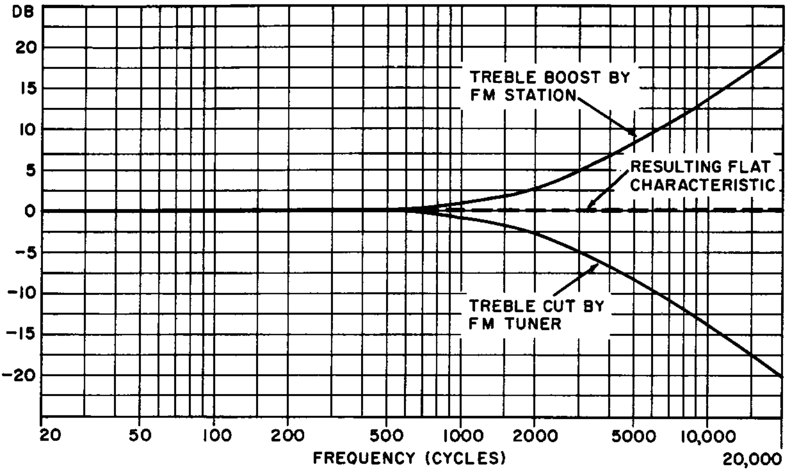


Fig. 8-1. FCC FM equalization.

to offset accurately (equalize) these departures on the part of the program sources.

Why, the audiophile may well ask, do the high-fidelity program sources fail to present signals with a flat characteristic? The answer lies in the fact that flat response is not the only requisite of high-fidelity performance. Low noise and low distortion are equally important. By proper shaping of the frequency response of the program source and reshaping in the playback system, it is possible to maintain flat response and at the same time reduce noise and distortion. It can well happen, however, that improper equalization will result in frequency response that deviates considerably from flat. This results in sound that displays excessive shrillness, tubbiness, or other forms of imbalance. For this reason, basic understanding of equalization can be of aid to the audiophile in getting the most from a high-fidelity system.

FM EQUALIZATION

The FCC (Federal Communications Commission), which regulates radio communications in the United States, requires that commercial FM stations (including TV) provide the specific amount of treble boost shown by the upper curve in Fig. 8-1. In order to yield flat frequency response, the FM tuner must deemphasize the treble frequencies, as shown. If a tuner fails to provide correct treble attenuation, the highs will either be exaggerated or underemphasized, as shown in Fig. 8-2.

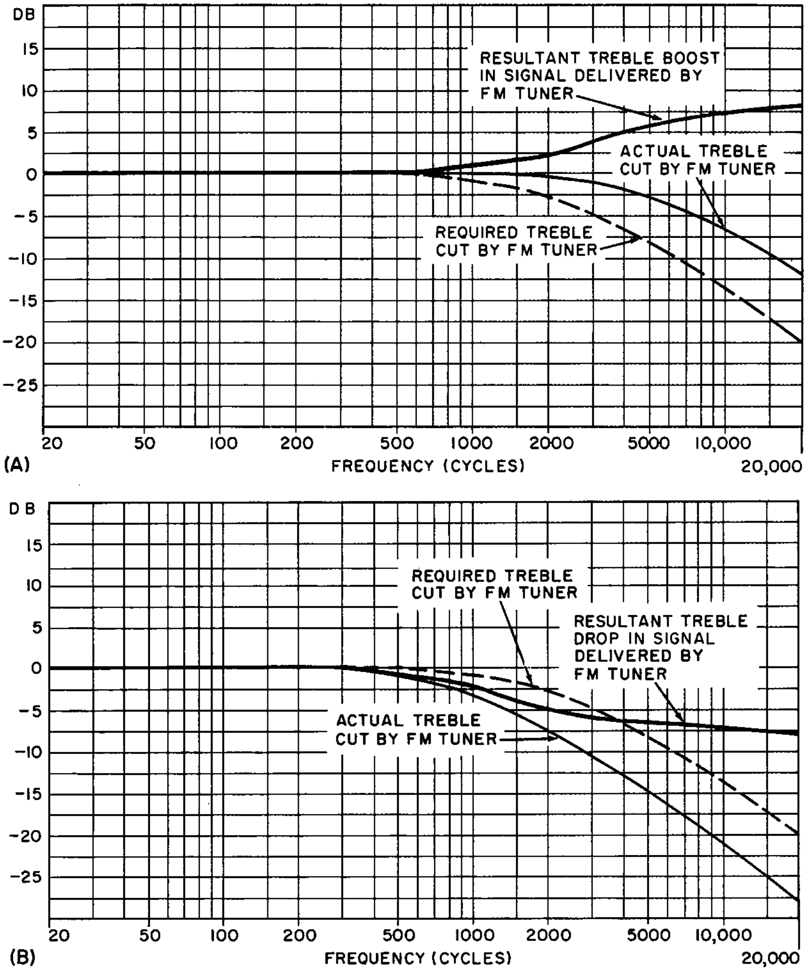


Fig. 8-2. Treble emphasis (A) resulting from insufficient treble cut, and treble loss (B) resulting from excessive treble cut in an FM tuner.

Part A shows what happens if the tuner supplies too little treble cut, which results in exaggerated treble. (This defect was at one time deliberately built into some FM tuners to produce a brilliant sound that uninformed listeners identified with high fidelity.) Occasionally (usually by accident rather than design), too much treble cut is built into the tuner, resulting in a decline in treble response as shown in Fig. 8-2B.

The audiophile who feels that the sound of his FM tuner is too brilliant or too dull can have it checked by a competent audio technician

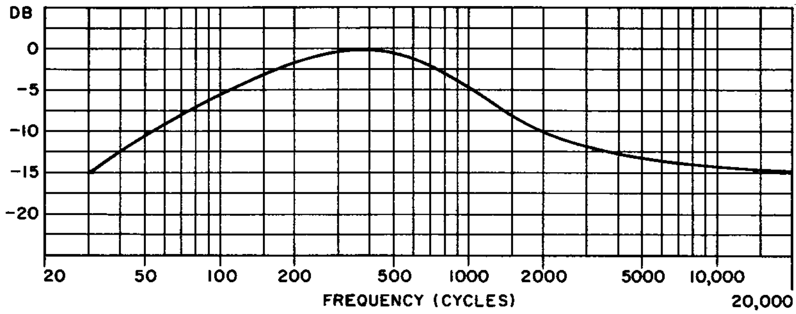


Fig. 8-3. Variation with frequency of the maximum audio energy in music.

in a matter of moments. If the tuner is at fault rather than some other part of the system, the trouble can very easily and quickly be cured by the technician at minimal cost. It is quite possible that a tuner with uneven response serves to balance an opposite frequency characteristic of the speaker being used. The deliberate choice of such a tuner may be the result of a suggestion by the audio salesman interested in helping his customer to assemble an audio system with smooth overall sound. However, the customer should be aware of this, because (in the event he replaces the speaker system) the sound may be less smooth than with a tuner having flat response.

The basic reason for requiring treble boost on the part of the FM (and TV) station is to reduce noise produced in broadcasting and in the tuner. Noise is created in any electronic apparatus, and is equally distributed among all frequencies. Because there are more frequencies between 800 and 20,000 cycles than between 20 and 800 cycles, most of the noise exists in the treble range. Treble boost at the FM station increases the ratio between the audio signal and noise at the transmitter, and the treble deemphasis in the tuner not only restores flat frequency response, but also cuts down the noise.

Any electronic equipment can accept only a certain amount of signal before it begins to produce an unacceptable amount of distortion. For this reason, the amount of treble emphasis that can be employed at the FM transmitter is limited. However, there is a saving factor—acoustic power tends to be relatively small at high frequencies.

Investigations have revealed that music sources and the human voice generally produce a decreasing amount of power as frequency goes up. Figure 8-3 shows the approximate distribution of audio energy produced by a symphony orchestra playing a typical composition. The curve applies to the maximum power produced at each frequency because it is the maximum rather than average power that causes the most distortion. It can be seen that the most audio power is produced in the neighborhood of 400 cycles. If we use 400 cycles as a standard of reference, calling it 0 db, we find that the audio energy is about 13 db down at 10,000 cycles and about 15 db down at 15,000 cycles. This decline tends to offset the treble boost in FM equalization, and the audio power actually transmitted in the treble range is thus much more nearly constant than indicated by the preemphasis curve of Fig. 8-1. For this reason, the treble boost does not introduce excessive distortion.

PHONOGRAPH EQUALIZATION

Virtually all phonograph records made today in the United States, and the majority of those made abroad, have their frequency response shaped according to the RIAA (Record Industry Association of America) curve. This characteristic is shown as Fig. 8-4. The high frequen-

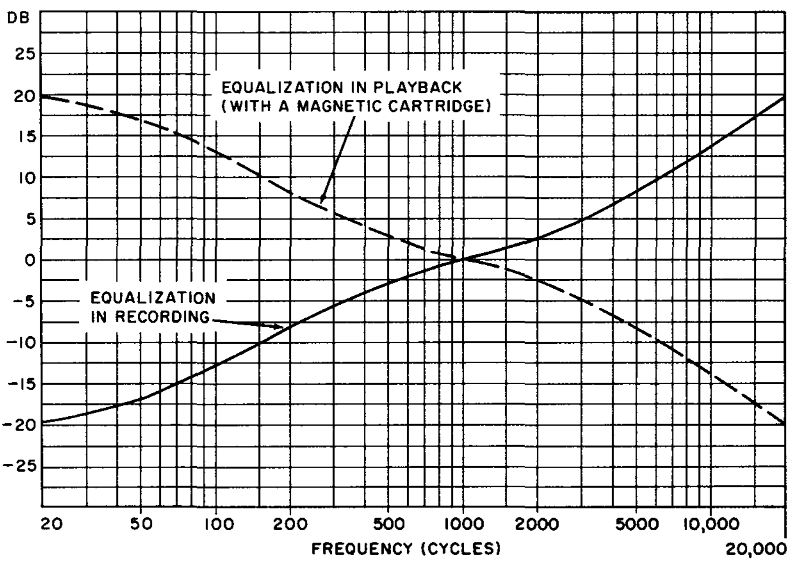


Fig. 8-4. RIAA phonograph record equalization.

cies are emphasized in recording, and the low frequencies are deemphasized. In playing back a phonograph record, equalization of the opposite kind must be provided in the control amplifier in order to provide flat response. This playback equalization, consisting of treble cut and bass boost, is also shown in Fig. 8-4. (At the moment we are assuming that a magnetic cartridge is used for playback; later we shall discuss the equalization required for other pickups.)

Some of the older discs, particularly those made prior to 1955, have a somewhat different frequency characteristic. Such records require different equalization in playback, and most control amplifiers therefore provide one or more phono equalization characteristics in addition to RIAA. If only RIAA equalization is provided, one can easily compensate for the difference between the RIAA and the other curves by small adjustments of the bass and treble controls, using one's ears to determine when the tonal balance is satisfactory.

TREBLE BOOST IN RECORDING

The reason for using treble boost in recording a disc is the same as in FM broadcasting; it serves to reduce noise. In the case of a record, most of the noise is encountered in playback because minute irregu-

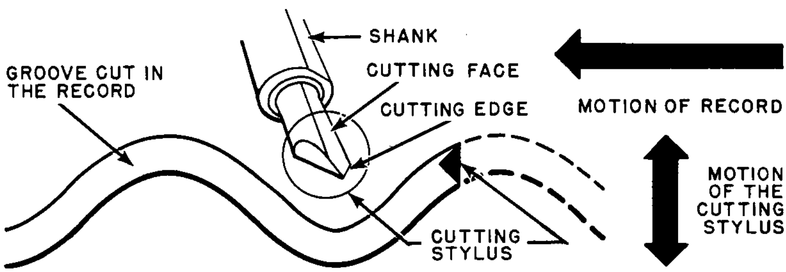


Fig. 8-5. The groove pattern produced by a cutting stylus.

larities and bits of dust in the record grooves are translated into noise signals by the phono cartiridge. If the audio signal picked up by the cartridge is subjected to treble cut to achieve flat response, the noise is reduced. As with FM, there are limits to the amount of treble emphasis that can be put on a record. To make clear why this is so, and at the same time pave the way for further discussion, an explanation of what takes place in disc recording is in order.

When an audio signal is recorded, the stylus that cuts the groove swings back and forth (also up and down in the case of stereo discs), producing a pattern such as shown in Fig. 8-5. The greater the audio signal, the greater speed with which the cutting stylus moves from side to side. If the audio signal is constant, the cutter velocity is constant.

This means that (for an audio signal of constant strength) the cutter moves less far at higher frequencies because it must swing back and forth more times per second. The groove amplitude—the amount that the cutter swings—thus grows smaller as frequency rises. This is shown in Fig. 8-6A. If we record a frequency of 2000 cycles instead

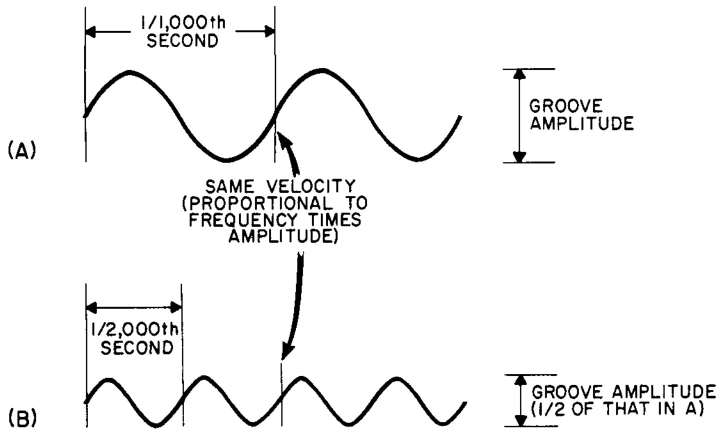


Fig. 8-6. Groove amplitudes at 1000 and 2000 cycles when cut with a constant-velocity characteristic.

of 1000 cycles, the greater number of cycles per second means that the amplitude of each cycle must be proportionately smaller in order to maintain the same velocity. In Part B it can be seen that the recorded amplitude is one-half as great at 2000 cycles as at 1000 cycles. Figure 8-7 shows how the amplitude of the recorded signal declines steadily for signals of equal strength as one goes from the lowest to the highest audio frequency.

Boosting the treble frequencies increases the groove amplitude. Assuming that the treble boost doubles the amplitude of a 2000-cycle signal (which would normally have half the amplitude of 1000 cycles), its amplitude will become the same as that of 1000 cycles, as shown in Fig. 8-8. This makes the groove undulations sharper at the top and bottom, which makes it more difficult for the phonograph stylus used in playback to follow the groove. This, in turn, results in a type of distortion known as *tracing distortion.* If the groove undulations are extremely sharp, tracing distortion can become very severe. For this reason, there are limits to the extent to which treble boost can be used in disc recording. As with FM, treble boost can be employed to the degree to which there is

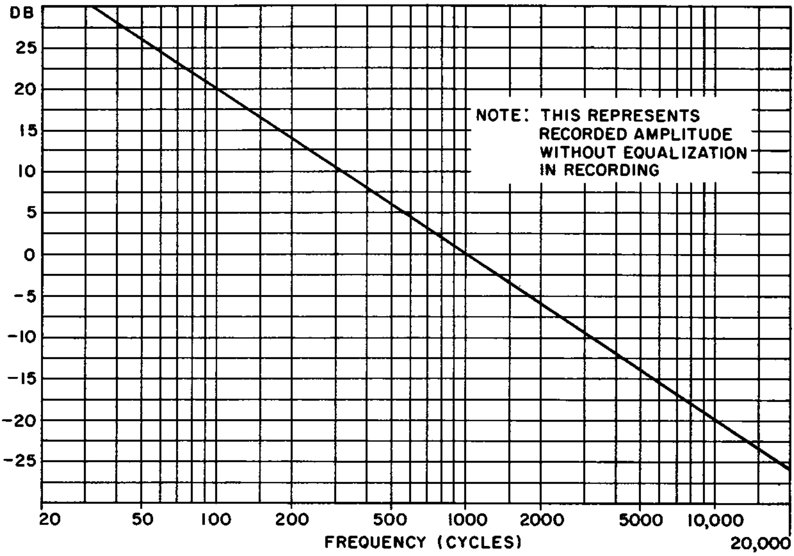


Fig. 8-7. Variation in groove amplitude with frequency for signals of equal magnitude cut with a constant-velocity characteristic.

a decline in audio energy at high frequencies. (The reader should note that the RIAA treble boost—Fig. 8-4—is exactly the same as the treble boost used by FM stations, shown in Fig. 8-1.)

BASS ATTENUATION IN RECORDING

In addition to treble boost, Fig. 8-4 shows that bass cut is employed in recording a phonograph disc. As shown in Fig. 8-7, groove amplitude increases with decreasing frequency, assuming that the same amount of signal is recorded at all frequencies. Below 500 cycles, the groove amplitude would tend to become so great as to produce excessive distortion, because both the cutting stylus and the stylus of the playback cartridge would have considerable difficulty in making such wide excursions. In addition, there would be danger of one groove overlapping an adjacent one (overcutting), and excessive amplitude would reduce the number of grooves that could be put on a record, decreasing playing time per record.

To some extent, the problem of excessive groove amplitude at low frequencies is solved automatically, because most sources of sound produce a decreasing amount of power at bass frequencies. Nonetheless, the record manufacturer must provide the additional deemphasis of the bass frequencies shown in Fig. 8-4. This amount of bass cut is limited in order to take into account the problems of the playback cartridge. The cartridges employed in most high-fidelity installations are magnetic types, which produce a relatively weak signal. In amplifying this weak signal and at the same time adding bass boost to compensate for the deemphasis used in recording, there is considerable chance of amplifying hum (60 and 120 cycles) and rumble (frequencies below 40 cycles created by irregular motion of the phonograph turntable) to an audible level.

EQUALIZATION FOR AMPLITUDE PICKUPS

The preceding discussion has taken into account only the magnetic type of pickup, which responds to the *velocity* of the signal recorded on the disc. However, there are other pickups that turn out a signal proportional to the *amplitude* of the recorded signal. These are the capacitive pickup and the piezoelectric cartridges employing either a Rochelle Salts crystal or a ceramic element for converting the motion of the stylus into an electrical signal.

We have seen how groove amplitude decreases with rising frequency before application of RIAA recording equalization. After applying RIAA treble boost and bass cut, the recorded amplitude varies with fre-

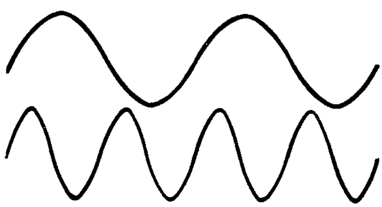


Fig. 8-8. Comparison of the grooves for a 1000-cycle signal and a 2000-cycle signal cut at the same amplitude.

quency in the manner shown by Fig. 8-9; again one should be reminded of the assumption that signals of equal strength at all frequencies are recorded on the disc. The resulting amplitude characteristic emphasizes the bass and deemphasizes the treble. In order to obtain flat frequency response, it is necessary to achieve a corresponding amount of bass cut and treble boost in playback.

The required treble boost is generally supplied by the piezoelectric cartridge itself, which is designed to have a rising treble characteristic that offsets the treble cut shown in Fig. 8-9. The bass cut, however, must be provided by the control amplifier. This is accomplished by putting the signal from the piezoelectric cartridge across a load resistor of proper value, usually in the range of 1 million to 3 million ohms,

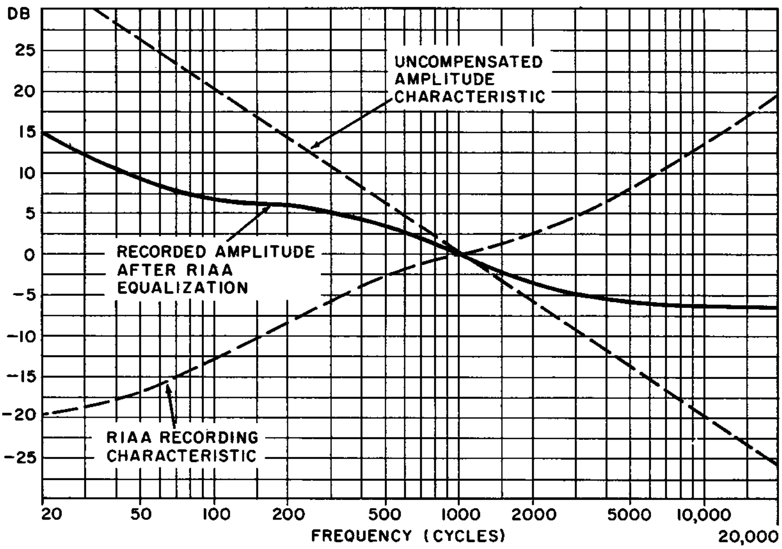


Fig. 8-9. How the uncompensated amplitude characteristic and the RIAA recording characteristic combine to yield the RIAA amplitude characteristic.

depending upon the specific cartridge. (The reason why the load resistor produces bass cut is outside the scope of this discussion.)

The piezoelectric pickup has a relatively high output, often in excess of 1 volt. Therefore it is intended to be connected to one of the high-level input jacks of the control amplifier (those intended for such sources as the tuner, tape machine, and TV). These jacks are generally provided with a 500,000-ohm load resistance. This resistance only rarely yields the proper bass cut for piezoelectric cartridges; in most cases, a load resistor of only 500,000 ohms will cause too much attenuation of low frequencies, resulting in a deficiency of bass.

To prevent this bass loss, some control amplifiers provide an input specifically intended for a piezoelectric pickup that has sufficiently high load resistance. If the audiophile finds this not to be true of his control amplifier and he wishes to use a piezoelectric cartridge (some of which have excellent quality at relatively low prices), the cartridge can easily, quickly, and inexpensively be adapted to effect the proper amount of bass cut in conjunction with the usual 500,000-ohm load resistor. It is merely necessary to wire a capacitor of suitable value—generally in the range of .001 to .002 microfarad—directly across the cartridge. An audio technician can do this in a few minutes or the audiophile capable of deft soldering can do it himself. In either case, one should find out from the cartridge manufacturer the proper value of capacitor to wire across the pickup so that it will produce RIAA playback equalization in conjunction with a 500,000-ohm load resistor.

A substantial number of control amplifiers handle the problem of equalizing the amplitude pickup in another manner. By means of a simple electrical network (consisting of resistors and capacitors), they *in effect* convert the piezoelectric cartridge from an amplitude-responsive device to a velocity-responsive one, just like a magnetic pickup. In this case, the signal from the piezoelectric cartridge is treated in the same manner as one from a magnetic pickup.

TAPE-RECORDER EQUALIZATION

Equalization requirements for tape recorders vary with the speed at which the machine operates. However, because the 7.5-ips (inches-per-second) speed has been virtually standard for high fidelity in the home and for nearly all commercially recorded tapes, we can limit the following discussion to equalization at 7.5 ips.

Figure 8-10A shows the equalization required when playing a commercially recorded tape. The solid line is the equalization characteristic called the NARTB (National Association of Radio and Television Broadcasters[1]) curve. Although the NARTB curve is an official standard only for tapes operating at 15 ips, it has been put into virtually universal use in the United States for 7.5 ips as well. The dotted line showing a small amount of treble boost is not required by NARTB, but depends upon the head used for playback. The head contains a minute gap that contacts the tape. The narrower the gap, the better the high-frequency response of the head. High-quality heads typically have gaps of .00015 inch to .00025 inch. If the gap is sufficiently narrow, there is very little loss of high frequencies up to about 15,000 cycles at a

[1]Now the NAB (National Association of Broadcasters).

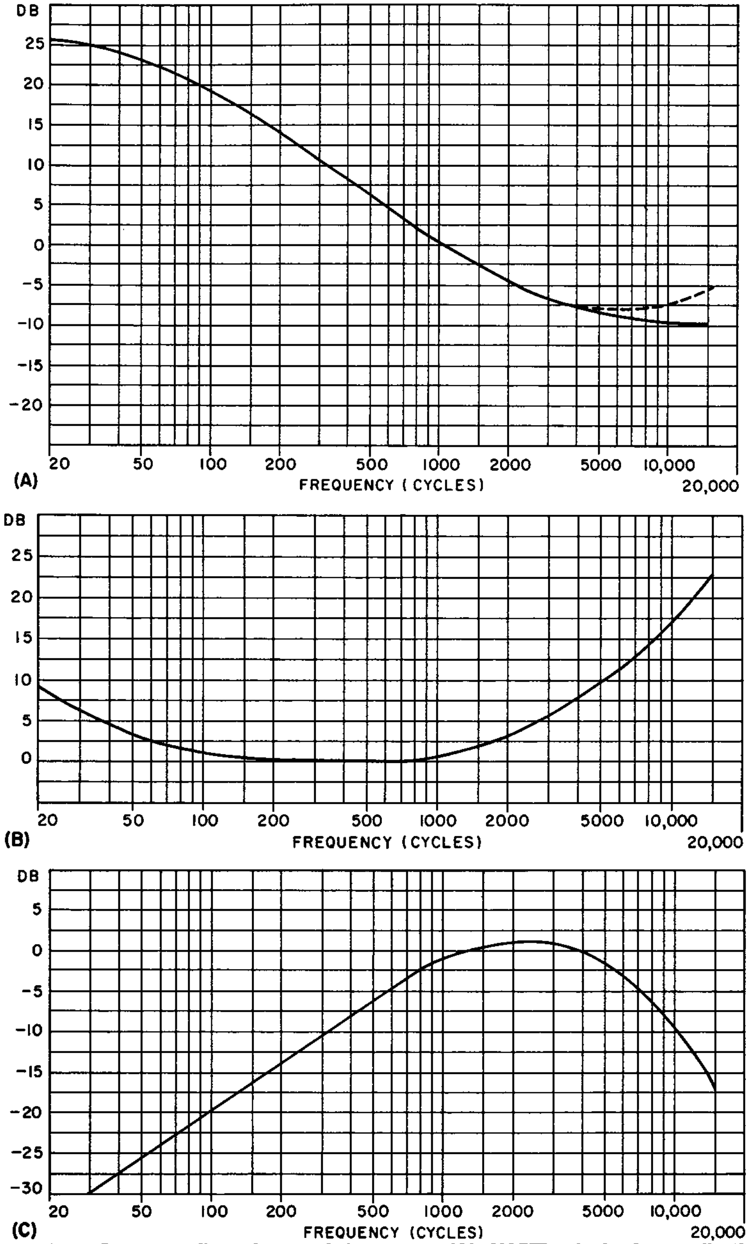


Fig. 8-10. Tape-recording characteristic curves: (A) NARTB playback equalization (dotted line shows optional treble boost that depends on the playback head used), (B) NARTB-type equalization on a tape (C) typical uncompensated response of a tape recorder operating at 7.5 ips.

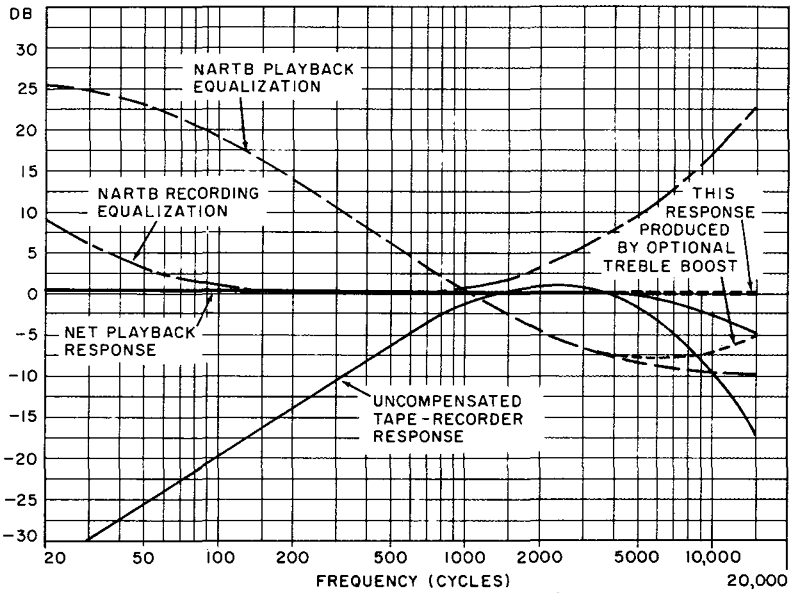


Fig. 8-11. How the curves of Fig. 8-10 combine to yield flat response.

speed of 7.5 ips. Other heads may show somewhat greater loss. The dotted curve in Fig. 8-10A shows the treble boost that might be required to compensate for the high-frequency losses of a head with a gap of .00025 inch.

In recording a tape, a very substantial amount of treble boost and a slight amount of bass boost are employed, as shown in Fig. 8-10B, which represents recording equalization used by a typical high-quality tape machine conforming to NARTB equalization. The reason for the great quantity of treble boost is to compensate for losses that occur in the recording process due to various magnetic phenomena associated with the tape. These losses become increasingly severe as frequency goes up.

The total picture of tape equalization can be made clear with the aid of Fig. 8-10C, which shows the frequency response of a tape recorder if unequalized audio signals of the same magnitude at all frequencies are fed into the machine and the resulting tape is then played back without equalization. When NARTB equalization is followed in recording and playback, the net result, theoretically, is the flat frequency response shown in Fig. 8-11. In practice, response is permitted to drop a bit at each extreme of the audio range; in a good-quality machine, the drop is typically 2 to 4 db at 30 and at 15,000 cycles.

Some tape recorders do not follow the NARTB equalization pattern (providing most of the treble boost in recording and most of the bass boost in playback). Instead, half the bass boost and half the treble boost are provided in recording and the other half of both in playback. (This may be called *half-and-half equalization.*) By using exactly the same equalization in recording and playback, it is possible to sim-

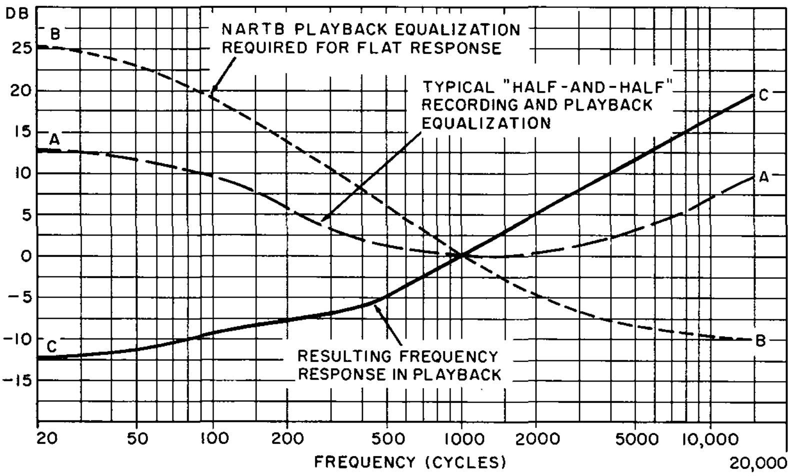


Fig. 8-12. The effect of playing a tape an NARTB-type characteristic on a tape recorder that has half-and-half equalization.

plify the tape amplifier and lower its cost. Figure 8-12 shows the effect of this type of equalization on the playback frequency response of a tape prerecorded with NARTB equalization; note the severe bass cut and treble boost.

There are additional disadvantages to half-and-half equalization. Less treble boost in recording means a smaller amount of signal (treble frequencies) recorded on the tape; hence there is a lower signal-to-noise ratio. By using treble boost in playback, tape hiss and noise generated in the tape amplifier are accentuated. This further decreases the signal-to-noise ratio. In recording, the tape overloads (distorts) more easily at low frequencies. Since half-and-half equalization provides a substantial amount of bass boost in recording, this tends to distort bass notes.

On the other hand, half-and-half equalization does have a pronounced advantage in reduction of hum. The great amount of bass boost called for in playback by NARTB equalization boosts hum as well as low-frequency audio signals. Therefore, extensive precautions must be taken in shielding the playback head and components in the tape amplifier against hum pickup. In half-and-half equalization much less bass boost is used in playback, so that hum is amplified to a considerably smaller extent.

9. impedance ■

Impedance is an important word in high fidelity. If the audiophile has a basic understanding of this frequently encountered term, he is in a better position to select and operate his equipment properly. In this chapter, the meaning of impedance will be explained, and we shall see how it concerns the audiophile in several respects.

If the reader has by chance taken a glimpse at the text and figures ahead, he may have the erroneous impression that he is about to be immersed in mathematics. But, as he will soon discover, there is nothing in this chapter beyond simple multiplication, division, and addition. In exchange for following a few easy calculations, he will acquire a basic understanding of electrical behavior that should not only be satisfying to his intellectual curiosity, but will enable him to understand certain problems that arise in audio, and to avoid or cope with these problems.

THE MEANING OF IMPEDANCE

Electrical circuits contain three fundamental quantities: current, voltage, and impedance. Current refers to the quantity of electrons that flow

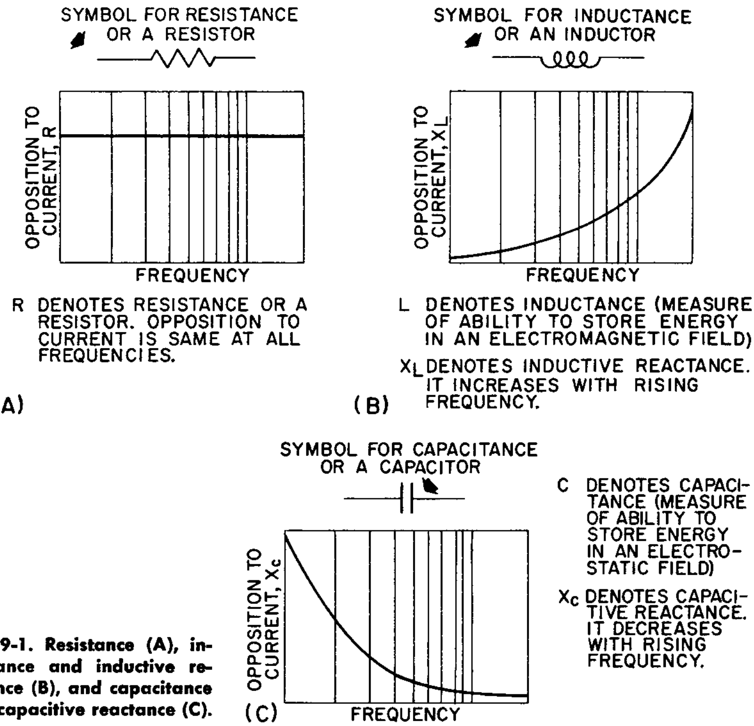


Fig. 9-1. Resistance (A), inductance and inductive reactance (B), and capacitance and capacitive reactance (C).

in a second through an electrical conductor, and is measured in amperes. Voltage is an "electrical pressure" that causes current to flow, and is measured in volts. *Impedance is the opposition to flow of electrical current, and is measured in ohms.*[1]

The flow of water provides a helpful analogy for understanding the flow of electric current. Pressure (corresponding to voltage) is required to make water (corresponding to current) flow through a pipe, which presents a resistance (corresponding to impedance) to the flow. In electricity, the resistance to flow depends upon a variety of factors, chiefly the nature of the material through which the current must pass. For example, current travels more easily through copper than through iron. To push a given amount of current through iron requires more voltage than through copper. Or, if the same voltage is applied, less current goes through iron than through copper.

[1]Fractions of a volt or ampere are often used as a unit of measurement. A millivolt is one thousandth of a volt, and a microvolt is one millionth of a volt. A milliampere and a microampere denote one thousandth and one millionth of an ampere, respectively. In the case of impedance, a frequently used unit is the megohm, one million ohms.

To some readers, *resistance* is perhaps a more familiar term than impedance for expressing opposition to current flow. We use the term impedance because we are not dealing only with direct current, that is, with electrons moving steadily in one direction. We are also dealing with alternating current, where electrons move in a given direction, then in the opposite one, again in the first direction, and so on, repeating this pattern a number of times per second. In the case of alternating current, there are three kinds of opposition to current flow: resistance (Fig. 9-1A), which is constant at all frequencies; inductive reactance (Fig. 9-1B), a property of a coil that opposes high frequencies more than low frequencies, and offers no opposition to direct current; and capacitive reactance (Fig. 9-1C), a property of a capacitor (two plates of conductive material separated by a nonconductive material) that opposes low frequencies more than high frequencies. (A capacitor does not permit any flow of direct current.) Impedance, denoted by the symbol Z is the general term for the total opposition to alternating current, i.e., any combination of resistance, inductive reactance, and capacitive reactance.

OHM'S LAW

In this section we shall examine the relationship of voltage, current, and impedance. If an electrical circuit presents a small impedance to current flow, a low voltage is sufficient to force a desired amount of current through it. Conversely, a large impedance would necessitate a high voltage

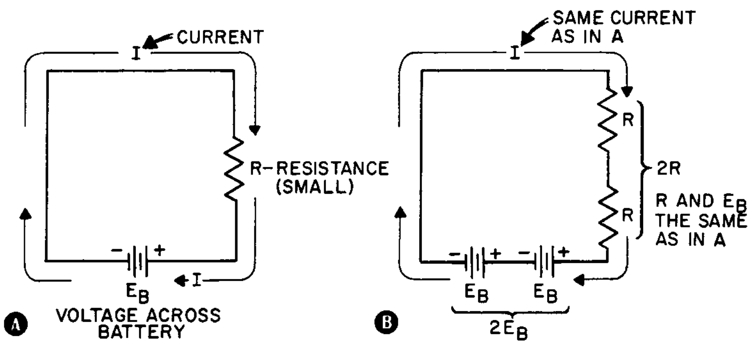


Fig. 9-2. Voltage varies directly with impedance.

to make the same current flow. In short, for a given amount of current flowing through a circuit, voltage varies directly with impedance. This is shown in Fig. 9-2. In part A, a small voltage is represented as driving a certain amount of current through an impedance, in this case a resistance. In Fig. 9-2B, the resistance has been doubled. Accordingly, the voltage must also be doubled (two batteries in series) in order to drive the same amount of electric current through the circuit as before.

If a given voltage is applied to an impedance, the current that flows will be large if the impedance is small, and the current will be small if the impedance is large (Fig. 9-3). To sum up, if current is constant,

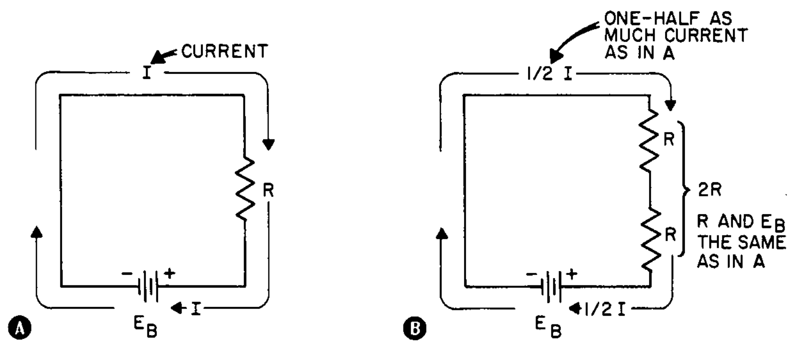


Fig. 9-3. Current varies inversely with impedance.

impedance and voltage increase or decrease together. If voltage remains the same, current goes up as impedance goes down, and current goes down as impedance goes up. (Impedance varies directly with voltage and inversely with current.) This entire relationship is expressed by Ohm's Law:

$$Z = E/I$$

where Z is impedance, E is voltage, and I is current.

This simple formula is most important and most useful, enabling us to deduce from any two of the basic quantities the value of the third one. For example, given the voltage and current in a circuit, we can find Z by means of the expression $Z = E/I$. The formula can also be rearranged so that we can find either current (I) or voltage (E); thus:

$$I = \frac{E}{Z} \quad \text{and} \quad E = I \times Z$$

WHY HIGH OR LOW IMPEDANCE?

In the case of water flow, power is the result of applying a certain amount of pressure to a certain number of gallons per second; in short, power is pressure times rate of flow. In the case of electricity, power is voltage multiplied by current. This statement can be summed up by the formula

$$P = E \times I$$

where P is power (in watts), E is voltage (in volts), and I is current (in amperes).

This formula reveals that a given amount of electrical power can take an infinite number of forms: a small voltage and a large current, a large voltage and a small current, a moderate voltage and a moderate current, etc. For example, if 10 watts of power are required, it could be achieved by a current of 10 amperes and a voltage of 1 volt, by 1 ampere and 10 volts; by 5 amperes and 2 volts, by 0.1 ampere and 100 volts, etc.

The form in which power is developed depends upon the task of the electrical circuit. Sometimes voltage is the most important requirement (as when supplying the audio signal to the power amplifier), and sometimes it is current (as when supplying the audio signal from the power amplifier to the speaker).

Impedance is important because the form in which electrical power is obtained depends upon it. A high impedance is associated with a high

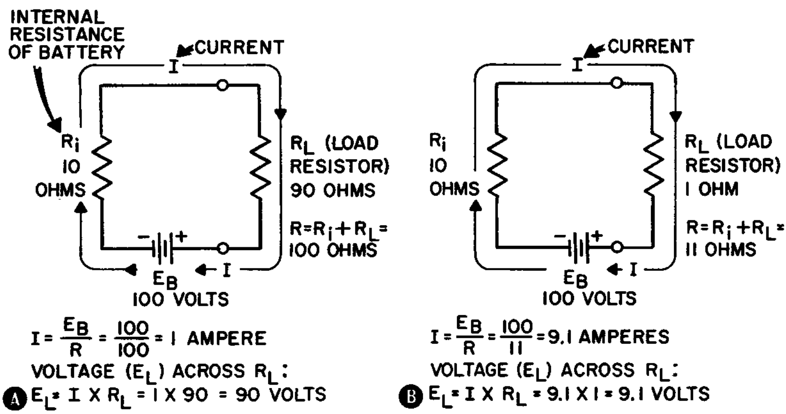


Fig. 9-4. The same quantity of electric power may exist in the form of high voltage and small current (A) or low voltage and large current (B).

voltage and a small current. A low impedance enables us to have power in the form of low voltage and large current. This is shown in Fig. 9-4. (For simplicity, the figures deal with direct instead of alternating current, showing the impedance as a resistance.) Figure 9-4A shows a battery, which is sending current through R_L (the load resistor). It also shows a resistance R_i, which is the resistance to current flow *within* the battery. (If there were no such internal resistance, all the life would go out of a battery the moment we touched its two terminals together.) The internal resistance is assumed to be 10 ohms, the load resistance 90 ohms, and the battery voltage 100 volts. Thus the total resistance of the circuit R, is 100 ohms.

Now let us put Ohm's Law to work to discover the current through R_L and the voltage across it. As shown, we can calculate from the various forms of Ohm's Law that current flowing through the circuit is 1 ampere and that the result of 1 ampere flowing through 90 ohms is 90 volts across R_L. Thus, this relatively high load impedance (in terms of our illustration) dissipates power in the form of high voltage (90 volts) and low current (1 ampere).

In Fig. 9-4B, the battery voltage and internal resistance of the battery are assumed to be the same as before, but the load resistor is only 1 ohm, yielding a total circuit resistance of 11 ohms. The calculations show that 9.1 amperes flow through R_L and that the voltage is 9.1 volts. Thus, by using a small load impedance, the load dissipates power in the form of low voltage and high current.

MAINTAINING SIGNAL VOLTAGE

By now the reader may be saying to himself that impedance is not his concern, but that it is for the manufacturer of an audio component to worry about impedance and its relationship to proper operation of the component. However, this is not altogether true. When it comes time to connect one component to another (for example, a control amplifier to a power amplifier, a speaker to a power amplifier, or a microphone to a tape recorder), the audiophile enters the picture, and impedance may well become *his* problem.

When the audio signal passes from one component to another, attention must be paid to the impedance of each of them and to the effect that the impedance of one component has upon the operation of the other. It is quite possible that unless impedances are in proper relationship, performance will be significantly impaired. Reduction of signal voltage is one of the ways in which performance may be caused to deteriorate. (The worst form of this deterioration is where some frequencies are attenuated more than others, with high frequencies being most apt to suffer.)

To illustrate, assume that we are connecting the output of a control amplifier to the input of a power amplifier, as in Fig. 9-5. At the output of the control amplifier, there exists a certain signal voltage across a

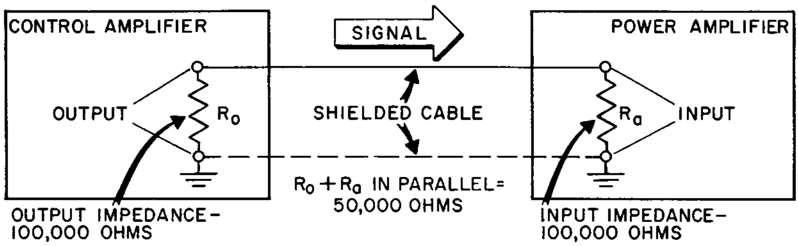


Fig. 9-5. How the effective output impedance of one component is affected by the input impedance of the following component.

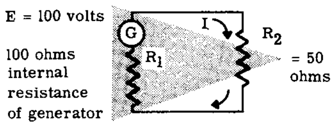
certain impedance (the control-amplifier output impedance, R_o). If R_o is reduced, the output voltage across it would drop, as we have learned from Fig. 9-3.

By connecting the power amplifier, via a cable, to the control amplifier, we place the input impedance of the power amplifier (R_a) across the output impedance of the control unit. Thus R_a and R_o are in parallel. Two impedances in parallel have a smaller total impedance than either one alone. In the case shown, the parallel value of two 100,000-ohm impedances is 50,000 ohms.
about 1 db.) In Fig. 9-5, however, R_a is the same as R_o, and the output impedance of the preceding component, the output voltage of the latter will not be significantly affected. (It will not go down more than 10%, or about 1 db.) In Fig. 9-5, however, R_a is the same as R_o and the output voltage drops 50%. This occurs because the output impedance has dropped from 100,000 to 50,000 ohms.

In practice, control amplifiers generally have output impedances of less than 50,000 ohms, often as little as 500 ohms. Therefore a power amplifier with an input impedance of 500,000 ohms, which is typical, is usually satisfactory. On the other hand, as we shall see in a subsequent section, the connecting cable between the two amplifiers has capacitance; at high frequencies, the impedance (capacitive reactance) across the cable may be appreciably less than 500,000 ohms, especially in the case of a long cable. The reduced impedance at high frequencies may cause a significant drop in high-frequency signal voltages.

MAXIMIZING POWER TRANSFER

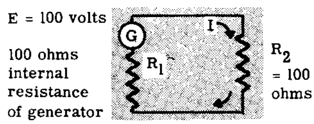
The more power that an amplifier produces, the more distortion is likely to be present. If power is transferred inefficiently from amplifier to speaker, the amplifier has to produce more power for a given volume of sound, which tends to increase distortion. Accordingly, it is highly desirable to maximize the efficiency of power transfer from the power amplifier to the speaker. For maximum transfer of *power* at minimum distortion, the output impedance of the power amplifier and the impedance of the speaker should be *equal*. We can demonstrate this important



$$I = \frac{E}{R_1 + R_2} = \frac{100}{150} = 0.66 \text{ amp}$$

$$P = I^2R$$

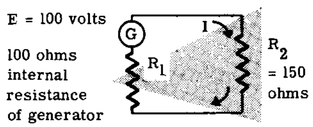
$$= 0.66^2 \text{x } 50 = 0.44\text{x}50 = 22 \text{ watts}$$



$$I = \frac{E}{R_1 + R_2} = \frac{100}{200} = 0.5 \text{ amp}$$

$$P = I^2R$$

$$= 0.5^2\text{x}100 = 0.25\text{x}100 = 25 \text{ watts}$$



$$I = \frac{E}{R_1 + R_2} = \frac{100}{250} = 0.4 \text{ amp}$$

$$P = I^2R$$

$$= 0.4^2 \text{x } 150 = 0.16\text{x}150 = 24 \text{ watts}$$

Fig. 9-6. To achieve maximum power transfer, the output and input impedances must be equal.

fact by following the calculations in Fig. 9-6, which again for ease of presentation are based upon direct current, although what we learn from them also holds true, in principle, for alternating current.

The illustrations show a 100-volt generator as the power source. It is assumed here that the internal resistance of the generator, R_1, is 100 ohms. Three load resistors of 50 ohms, 100 ohms, and 150 ohms are connected, one at a time across the generator, and the power dissipated in each of them is calculated. After finding the current in R_2, in each case the power is found by using the formula $P=I^2R$. (This formula is derived from $P=E \times I$ by substituting $I \times R$ for E; using it saves us the trouble of calculating the voltage across R_2 each time.) The results of these calculations show us that maximum power is dissipated in the load when the two resistances are equal. When a 2:1 impedance mismatch exists (R_1 is 100 ohms and R_2 is 50 ohms), the reduction in the power dissipated in R_2 (the power transferred) is 3/25 or 12%. This is a loss of only about 0.8 db.

The loudspeakers in common use have a low impedance. Since voltage varies directly with impedance and current varies inversely with it, this indicates that a speaker requires a relatively low voltage and a relatively large current to drive it. The tubes at the output of the power amplifier are designed to furnish enough power to drive the speaker. Unfortunately, this power is in the wrong form; the voltage is too high and the current too low for the needs of the speaker. In other words, the output impedance of the tubes is too high. For this reason, the usual power amplifier incor-

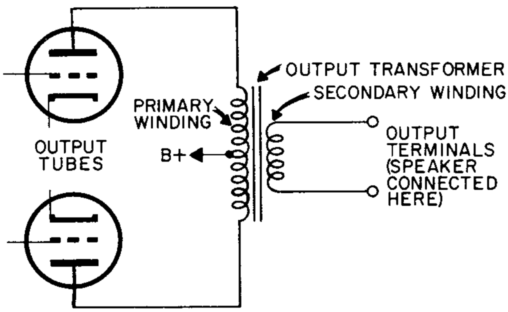


Fig. 9-7. Simplified output circuit of a typical power amplifier.

porates an output transformer following the output tubes, as shown in Fig. 9-7. The transformer has two windings (coils); the primary winding is connected to the tubes, and the secondary winding to the speaker. The transformer, by an electromagnetic process, converts the high voltage and small current available at the primary winding into a low voltage and a large current at the secondary winding. It is something like the common household bell transformer that steps down the 117-volt house current to about 6 volts.

The power fed into the transformer and the power coming out of it are virtually the same. (There is a very slight loss because the transformer is not quite 100% efficient.) Since the secondary produces a low voltage and a high current, we can see from Ohm's Law that its impedance is low. This reduced impedance matches the low impedance of the typical speaker, and permits maximum power transfer.

MATCHING THE SPEAKER TO THE AMPLIFIER

The manufacturer of a high-fidelity power amplifier designs it to have an output impedance (at the output-transformer secondary) that matches high-fidelity speakers in general use. A few amplifiers provide only one output connection, having an impedance of either 8 ohms or 16 ohms. Since most high-fidelity speakers have an 8-ohm or 16-ohm impedance, it is easy to find a speaker that matches the amplifier exactly. As indicated by Fig. 9-6, matching is not extremely critical, since reducing the load impedance by half reduces the power transfer by only 0.8 db. Thus a 1:2 mismatch between an 8-ohm amplifier output and a 16-ohm speaker or a 2:1 mismatch between a 16-ohm amplifier output and an 8-ohm speaker will not affect power transfer to an appreciable extent. Nonetheless, most

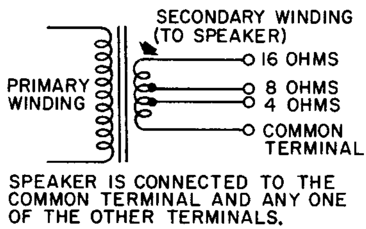


Fig. 9-8. Schematic representation of an output transformer that provides several output impedances.

power amplifiers provide several output connections (as shown in Fig. 9-8), each having a different impedance; typically, impedances of 4, 8, and 16 ohms are available, making it possible for the audiophile to match the speaker that he wishes to buy.

Many home music systems have speakers in several rooms, and by means of a switching arrangement, all the speakers are driven by one amplifier. It often happens that two or more of the speakers are connected to the amplifier simultaneously. This raises problems of impedance matching, because the impedance of several speakers in parallel is less than that of one speaker alone. If the speaker impedances are equal, their total impedance is the impedance of one speaker divided by the number of speakers. For example, assume that three 16-ohm speakers are connected at the same time to the amplifier, as in Fig. 9-9. Their total impedance is 16/3, or 5⅓ ohms. Accordingly, they should be connected to the amplifier output terminals with the nearest similar impedance. (They could be connected to the 4-ohm or 8-ohm terminals with about the same results.) As a general rule, however, if there has to be a mismatch, distortion is reduced if the speakers' impedance is higher than that of the power amplifier. Hence it would be better to connect the three 16-ohm speakers to the 8-ohm connection.

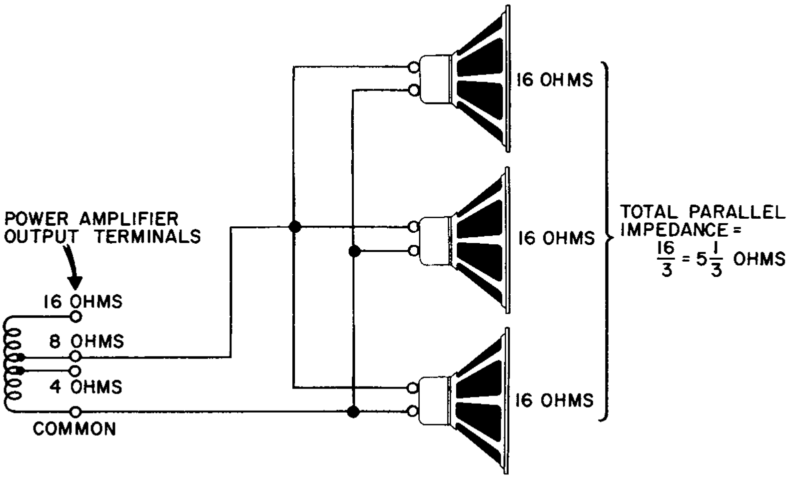


Fig. 9-9. Connecting three 16-ohm speakers to a power amplifier.

In some installations it will not be desired to operate all the speakers at once. Assume that two 16-ohm speakers are used, each in a different room. One or the other may be disconnected by a switch. Thus the speaker impedance presented to the amplifier will be either 8 or 16 ohms, depending on how many speakers are connected to the amplifier. No matter whether connection is made to the 8-ohm or 16-ohm terminals of the amplifier, a mismatch will result either when one speaker is used or when both are used. Another problem that arises when speakers are used in different rooms is how to control the volume of each of them separately.

Both problems—presenting an unvarying impedance to the power amplifier and separately controlling the volume of each speaker—may be solved by feeding each speaker through an L-pad, a form of variable resistance that controls speaker volume and at the same time, in conjunction with the impedance of the speaker, presents a constant impedance to the amplifier. The L-pad can be mounted on one of the panels of the speaker cabinet, and its value should be approximately equal to the impedance of the speaker. Figure 9-10A shows how an L-pad is constructed of two variable resistors with their moving arms ganged to each other, so that one resistance increases as the other resistance decreases.

Our knowledge of Ohm's Law enables us to understand how this useful device operates. As the volume is reduced (arms moved down), increasing resistance is placed in series with the speaker, and at the same time decreasing resistance is put in parallel with the speaker. The series resis-

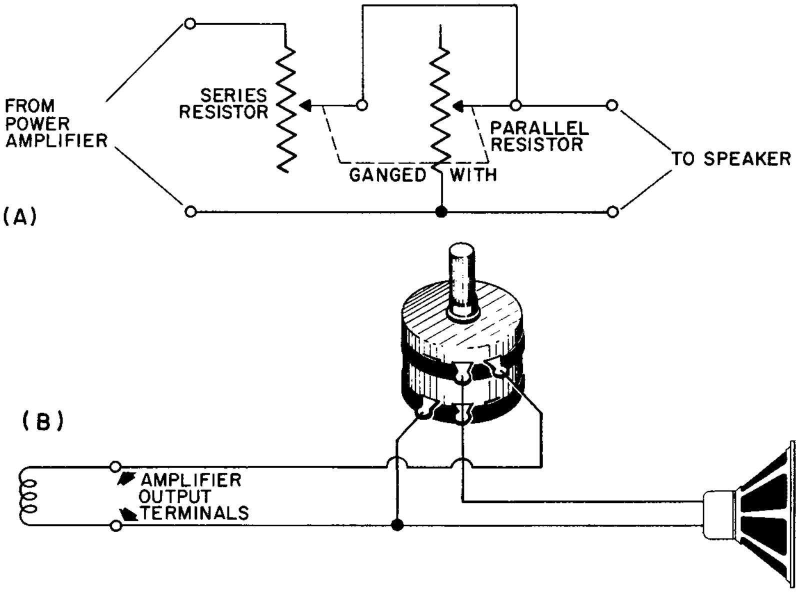


Fig. 9-10. The L-pad: circuit (A) and method of connection (B).

tance decreases the current through the speaker, while the decreasing parallel resistance bypasses more of the current that otherwise would flow through the speaker. Thus, with less current, there is less power to drive the speaker. The values of the series and parallel resistances are so chosen that, together with the speaker impedance, they present an unvarying impedance to the amplifier, no matter what position the arms are in. Figure 9-10B shows how the audiophile would insert an L-pad between a power amplifier and a speaker. The variable resistors are already ganged. (Both arms are moved by one shaft.)

Individual L-pads may be used with two, three, four, or more speakers, assuming that the amplifier is capable of supplying enough power to drive all of them simultaneously. Each L-pad and its associated speaker consume a constant amount of power at all times. If the volume is turned down by means of the L-pad, the L-pad consumes more power and the speaker less. The power requirements imposed upon the amplifier thus go up in proportion to the number of speakers used. For minimum distortion, speakers should be operated with their L-pads in as advanced a position as feasible, permitting the power amplifier to be operated at low level.

If a number of speakers are connected in parallel, their total impedance may be so low that they cannot be matched to the amplifier, but it is generally possible to rearrange them in such a manner as to improve the impedance match. For example, assume that four 8-ohm speakers are connected to the amplifier via L-pads. Their total impedance is 8/4, or 2 ohms. Assume that the only impedances available at the amplifier output are 8 and 16 ohms. Connecting the four speakers to the 8-ohm terminals would result in a severe mismatch, namely 8:2 (or 4:1).

However, the speakers can be rearranged as in Fig. 9-11 to obtain a correct match. Two of the speakers are connected *in series*, and the impedance of this pair is 16 ohms. (When speakers are in series, their

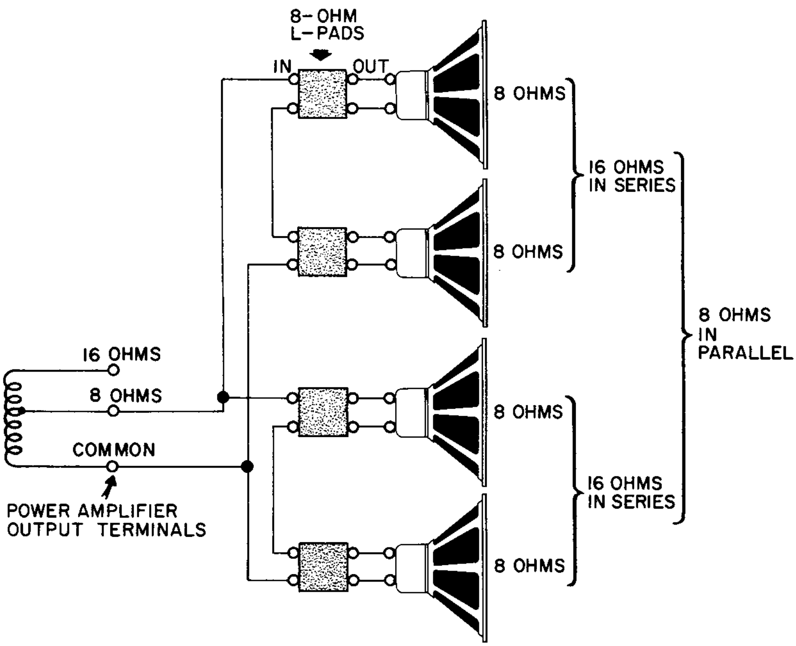


Fig. 9-11. Connecting four 8-ohm speakers to a power amplifier.

impedances are added.) The other two speakers are similarly connected in series and also have a total impedance of 16 ohms. Now the two *pairs* are connected in parallel with each other. A *parallel* connection of 16 ohms and 16 ohms yields 8 ohms. Thus the complete network of speakers and their respective L-pads is placed across the 8-ohm tap for correct matching.

It may happen that speakers have different impedances. Assume an 8-ohm speaker and a 16-ohm speaker are used in different rooms. If both speakers are placed across the same output terminals of the amplifier, more current and therefore more power will be drawn by the 8-ohm speaker. One way of meeting this situation is through the L-pads, which enable the volume of each speaker to be adjusted to the desired level.

Another procedure, applicable only where there is a variety of output terminals, is to connect each speaker to a different terminal, as in Fig. 9-12. Here the 8-ohm speaker is shown connected to the 4-ohm terminal and the 16-ohm speaker to the 8-ohm terminal. The 8-ohm speaker on the 4-ohm terminal draws the same power from the amplifier as the 16 ohm

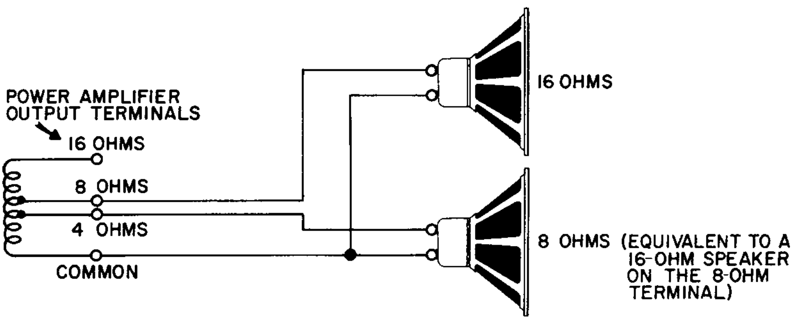


Fig. 9-12. Connecting speakers of different impedances to a power amplifier.

speaker on the 8-ohm terminal. Thus, *in effect,* there are two 16-ohm impedances across the 8-ohm terminal. And the value of two 16-ohm impedances in parallel is 8 ohms, so that a correct match is obtained.

In deciding what output terminal to use, the audiophile may have to calculate the parallel value of several speakers having different impedances. To illustrate, assume that an 8-ohm speaker and a 16-ohm speaker are used. Their total impedance is found by multiplying the two impedances and dividing by their sum.

$$\frac{8 \times 16}{8 + 16} = 5\tfrac{1}{3} \text{ ohms}$$

Placing a total impedance of 5⅓ ohms across either the 8-ohm or 4-ohm terminal results in sufficiently accurate matching. Suppose that we add another speaker. The same procedure is repeated, treating the impedance of the first two speakers as that of a single speaker. Thus we have:

$$\frac{5\tfrac{1}{3} \times 16}{5\tfrac{1}{3} + 16} = 4 \text{ ohms}$$

As additional speakers are installed, the new total impedance can be calculated by the same process each time. If the total speaker impedance becomes substantially less than half of the lowest impedance available at the output terminals, then it is necessary to arrange the speakers in some other fashion, as was illustrated in Fig. 9-11.

THE CATHODE FOLLOWER

We often are told that a cathode follower is desirable because it permits a long run of connecting cable between components. We have previously noted that the output impedance and therefore the output voltage of a component is not significantly reduced if the impedance at the input of the following component is at least 10 times as great. Connecting cable has impedance in the form of capacitive reactance, which permits high frequencies to travel *across* the cable (to be partially short-circuited). In other words, the cable has less impedance at high frequencies than at low ones. Therefore placing a cable across the output of, say, a control amplifier means that we are placing across it an impedance that decreases as frequency goes up. If the cable is long, it has low capacitive reactance at high frequencies, and in the treble range may have less than ten times the output impedance of the control amplifier. The result is significant attenuation of the high frequencies.

In order to prevent the impedance of the cable (and of other circuitry) from affecting control-amplifier output, the control amplifier can be designed to have a very low output impedance. One method of doing this is to use a *cathode follower* as the last stage. The cathode follower is a vacuum-tube circuit that does not provide voltage amplification, but characteristically has an output impedance of about 500 ohms. (A vacuum-tube stage serving as a voltage amplifier has an output impedance ranging from several thousand ohms to 100,000 ohms or more.)

MICROPHONE IMPEDANCES

The microphones most frequently used in high-fidelity applications are magnetic types that have impedances ranging from as low as 50 ohms to about as high as 50,000 ohms. (Those of a few hundred ohms or less are called low-impedance microphones, and those in the range of 10,000 ohms and upward are referred to as high-impedance microphones.) Low-impedance microphones have an appreciably smaller voltage output than high-impedance microphones. This is to be expected in view of the fact that voltage and impedance vary in the same direction. Thus the purchaser of a low-impedance microphone may well have an unpleasant surprise in store for him; the microphone often will not supply enough signal voltage to drive his tape recorder or amplifier to the desired level. It is then necessary to insert a step-up transformer between the microphone and the next component. The transformer converts the low voltage and *relatively* large current produced by the microphone into a *relatively* high voltage and small current. (In supplying audio signal to a tape recorder or control amplifier, only voltage is of concern.)

To avoid the expense of buying a step-up transformer and the effort of installing it (which often entails the problem of eliminating hum pickup), the audiophile may be well advised to purchase a high-impedance microphone, which provides adequate signal voltage in most circumstances. On the other hand, the advantage of a low-impedance microphone is that, as in the case of the cathode follower, its low impedance permits a long run of cable without the loss of high frequencies.

Crystal microphones are often used, particularly where it is not necessary to meet highest standards. These microphones have a very high impedance (particularly at low frequencies), often in the range of several million ohms. Consequently, to accomodate such microphones and prevent loss of *low* frequencies, the input impedance of the tape recorder or other component to which the microphone is connected must be quite high. For some crystal microphones, an input impedance of 5 million ohms or more is required.

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preface

Primers dealing with high fidelity are inclined to concentrate upon the equipment that makes up a home music system. For this reason, they are chiefly occupied with describing the functions, features, construction, and operation of each element of such a system. However, this is only one approach to explaining high fidelity. The fact remains that equipment is but a means to an end. An amplifier, for example, is not desired for itself, like a diamond. We are interested in what an amplifier does for us and how well it does it. *Quality of performance* is the basis for distinguishing between high-fidelity and other types of sound reproduction. A collection of audio components, separately or together labeled "high fidelity", is not what we really mean by the term. Our true interest is in the quality of reproduced sound that these components make possible.

Accordingly, only one chapter is devoted to the description of the components of a high-fidelity system. In the main, this book is concerned with spelling out the attributes of high fidelity performance—that which distinguishes a high-fidelity system from a garden-variety radio-phonograph—and with clarifying certain major problems that come up in striving for the best possible reproduction of sound.

It is very awkward to attempt to explore a subject such as sound reproduction without using at least a few basic technical terms. For this reason, Chaps. 2 and 3 are devoted to explaining the meaning of the decibel, frequency response, distortion, and signal-to-noise ratio—four all-important terms that appear everywhere in discussions of high fidelity, and serve to characterize what is (and what is not) high fidelity. Mastery of these terms should be of aid not only in following the subsequent discussions, but also in pursuing the subject outside this book—when conversing with an audio salesman, with an exhibitor at an audio show, with a serviceman, or with anyone else having competence in the field of audio.

In Chaps. 4 and 5, the four basic elements in audio reproduction—the original sound source, the program source (broadcast station, phonograph disc, or recorded tape), the audio equipment, and the listener—are discussed in terms of their limitations and their contributions. The next three chapters are concerned with vital problem areas of high fidelity: Chap. 6 with proper matching and connecting of high-fidelity components; Chap. 7 with obtaining satisfactory tonal balance when listening at levels much below the original sound; and Chap. 8 with "equalization," which concerns maintaining the low, middle, and high notes in proper relationship to each other, regardless of the listening level.

Finally, Chap. 9 takes up the meaning and relevance of "impedance," a term frequently encountered in high fidelity discussions but largely a mystery to the majority of audiophiles because of its technical nature. A fundamental understanding of this term can render the audiophile good service by enabling him to purchase, match, and operate components properly.

This book is intended for the reader completely unversed in the technical lore of electronics and audio. The only requirement is an interest in high fidelity. Because it is a basic book, it does not propose to answer all possible questions about the audio art. However, it is hoped that the book will supply the reader with a fundamental understanding, valuable not only in itself, but also of practical value, assisting him in the purchase of an audio system or in improving or adding to what he already has.

In several portions of this book, basic materials have been drawn from articles written by the author for *Audio, Audiocraft, Hi-Fi Music at home,* and *Radio & TV News.* He wishes to thank the publishers of these magazines for permission to use these materials, which have been completely rewritten and reworked for purposes of this book. Thanks are also due to the various manufacturers who were kind enough to supply photographs of high-fidelity components.

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