Tom Gilchrist

HI-FI STEREO
HANDBOOK

by William F. Boyce

Hi-Fi
Stereo
Handbook

by William F. Boyce

Preface

Since the first edition of this book in 1956, constant change and improvements in sound communication, storage, reproduction techniques, and the development of low-cost, high-quality stereo compacts have made hi-fi stereo an economic reality for everyone. These remarkable developments are highlighted by such effects as precision dynamic coupling of pickup devices to much-improved mediums, transistorization, miniaturization, combination of separate chassis into single integral units, acoustic suspension, and compact speakers—all giving more quality for less cost than was heretofore available.

It is more than a century since geniuses of music creation brought forth the need for mediums, techniques, and equipment to capture and store their great works. The world will forever lament that it was not able to record and store the work of these highly disciplined and talented artists of yesterday. At least from now on, we can achieve this for our own age and for future generations.

Modern concepts and standards of high fidelity, or hi-fi, provide for the ultimate in the endeavor toward reproduction of sounds exactly as they were originally created. True fidelity, as you will learn in the first chapter, is only occasionally achieved, but modern recording techniques provide results with maximum control of source, vocal, instrumental, and other elements arranged with emphasis, presence, and balance to create a final mix dramatically superior to live programs. The art of recording, storing, and reproducing sound electronically has progressed to the point where you, with eyes closed, can almost imagine yourself sitting fifth row center at Carnegie Hall.

But once your appetite for high fidelity has been whetted, what then? What equipment should you buy? . . . How much will it cost? . . . And these considerations are only the beginning.

This book was prepared as a reference and guide for all persons interested in high-quality sound reproduction. In addition to supplying information which will help you in planning, selecting, and installing appropriate systems, it also describes the various system components, plus what they do and how they operate. Thus, no matter whether the subject is new to you or whether you're a technician or an experienced hi-fi enthusiast, *Hi-Fi Stereo Handbook* has been written for you. I trust that you will find it informative and highly useful.

WILLIAM F. BOYCE

Contents

CHAPTER 6

CHAPTER 7

CHAPTER 8

Fidelity, Sound, and Distortion

It is universally agreed among the experts on high fidelity that there is as yet no exact scientific operational definition for a high-fidelity system. Standards and specified measurements of performance of a system have not been possible to establish because of limitations of the human ear and because of variations in human taste, room acoustics, system distortions, noise, and comparative volume levels.

WHAT *IS* HIGH FIDELITY?

A commonly accepted concept of high-fidelity sound is that it is reproduced sound with a high degree of similarity to that of original or live sound that has traveled from a source and has undergone several conversions through a system or several systems. High fidelity is felt to be achieved when the sound that is reproduced has negligible distortion from the original, when it has little extraneous noise, and when the volume levels and room acoustical effects are pleasing to hear. This reproduced sound might even be more pleasing to the listener at the output of the system than the original live sound would have been if heard at its source.

A reproduction of sound is something like a photograph. The picture cannot carry the original scene to the viewer in every detail. Some features of the picture may be de-emphasized, whereas other features may be emphasized intentionally, or distortion may be introduced for purely aesthetic reasons. Distortions of this sort can greatly improve the illusion that the photographer is trying to create. In the same way, the picture can be spoiled by undesirable distortions and effects, such as poor focus, poor film, and improper lighting.

Like photography, modern high-fidelity techniques encompass controls for modification of the original (live) sound to compensate for certain defects and make provision to actually improve the effects according to an individual listener's tastes. Undesirable distortions, differences in comparative sound levels, and injection of extraneous noise are also held to a minimum so that the pleasing qualities of the original sound will not be reduced.

In addition, modern concepts of high fidelity take into consideration the listener, his ear mechanism, and his nervous response, plus his listening experience and training.

Psychophysical reactions and imagination contribute to the realism of high-fidelity reproduction. The word "presence" is used to describe the degree of realism of the reproduced sound. This term suggests that the reproduction is so real that the listener can feel the presence of the source that is causing the live sound, even though that source is many miles away or even extinct. Furthermore, psychologists have shown that the trained human mind will fill in missing sounds that should appear in a musical rendition, even though these sounds are not present in the reproduction.

The application of the term "high fidelity," then, is largely a personal matter. Everyone can be a hi-fi expert—at least as far as his own tastes in equipment and quality of reproduction are concerned.

SOUND

The word "sound" is used in different ways. In the psychophysical sense, "sound" means to the listener the sensation of hearing audible vibrations conveyed from any medium (such as air, usually) through the ear to the brain. As used in physics, however, "sound" means the external cause of the sensation. In hi-fi, we are concerned with both meanings.

Sources of sound are bodies in vibration. Vibrations of a low-note, bass-viol string can actually be seen. The sounds caused by such a source have only a few vibrations per second. They are therefore called *low-frequency sounds*, *low notes*, or *lows*. On the other hand, the tinkles of a glass or of a musical instrument such as a triangle have comparatively many vibrations per second and are said to vibrate at *high frequencies*. Such sounds are known as *high notes* or *highs* in the range of human hearing sensations.

Audible sounds to the average person range between 20 and 20,000 hertz (cycles per second). The average human hearing system has certain characteristics of receiving, converting, and interpreting sound that are important considerations in the production of high-fidelity impressions.

Loud sounds are heard with good fidelity over a comparatively wide frequency spectrum. In other words, highs, lows, and in-between-frequency sounds are heard in proper relation to each other when all these sounds are loud; however, when the volume is reduced, the ear tends to attenuate (or cut down) the highs and lows but leaves the in-betweens proportionately louder. This is nicely demonstrated by the curves shown in Fig. 1-1.

The normal human hearing system has directional characteristics, receiving all sounds best from the forward position (source in front of observer) and with the ability to distinguish the direction of the source

or of a reflection. The ear can distinguish several sounds of different frequencies at the same time to a high degree. It is possible for the ear to distinguish between sounds with frequency differences as low as 3 cycles (vibrations) per second and volume differences of 1 decibel. The *decibel* is the standard unit of measurement of the loudness of sound.
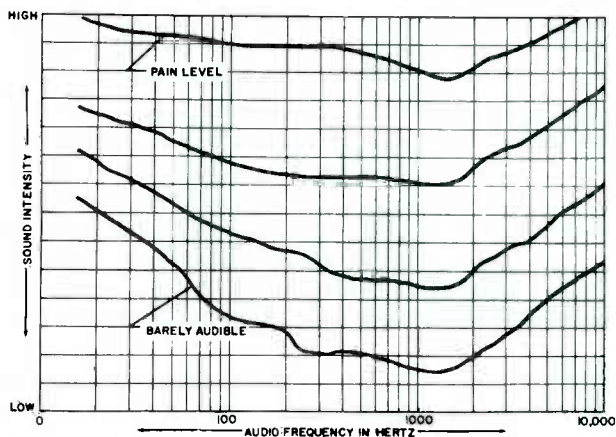

Fig. 1-1. Sound intensity required at different frequencies to produce uniform response in the ear's sensory system.

The ear can to some degree detect relative phase changes of sound. This means that a small change, increase or decrease, in the frequency or pitch of one note in relation to the frequencies of other notes simultaneously played can be detected.

Requirements for the components of a high-fidelity system that can please many individual ears of various listening tastes are quite exacting because of the fineness of the mechanism of the human hearing system.

DISTORTION OF SOUNDS IN HIGH-FIDELITY SYSTEMS

The best high-fidelity systems are substantially less than perfect. The ways in which the audio-frequency output sound differs from an input or a desired ideal output sound are classified as *distortion*. A complete high-fidelity system may be divided into functional sections, as illustrated in Fig. 1-2. Distortion may be created in any one or more of these sections. If more than one section is causing distortion, the final output sound may reflect the sum of the distortions from all distorting sections; however, in other cases, a section may be purposely designed to introduce distortion of a type which compensates for inherent distortion in another section. For example, bass and treble boost circuits can be used to offset (at least partially) the falling-off of response of a speaker at the highest and lowest frequency portions of the response range.

The high-fidelity system is somewhat like a chain, which is likely to be limited in overall performance by its weakest section; but the chain analogy breaks down in the foregoing case, in which the compensation introduced by one section may be used to compensate for distortion in another.

The speaker system is the weakest link in the high-fidelity equipment chain. Its inherent limitations arise primarily from the fact that two conversions of energy must take place: (1) the conversion of electrical energy from the amplifier output stage to mechanical energy in the motion of the diaphragm or cone to acoustic energy (sound) suitable to the listener's ear. Such energy conversion is known as *transduction*, and the devices which effect it are known as *transducers*. Input devices such as phono pickups and microphones are also transducers and have many of the same weaknesses as speakers, though to a lesser degree, because of the relatively low power levels at which they operate. Input devices provide transduction between sound input (or physical motion of a phono-pickup needle) and electrical output, just the reverse of the action in speakers.

The amplifier portion of the system can also contribute distortion if not properly designed or properly used, or not in proper working order. The voltage amplifier stages are basically the least troublesome. Being of the resistance-coupled type, they usually respond well over the required frequency range with very little distortion; however, the power amplifier stage and the output transformer which couples it to the speaker system are ordinarily important contributors to the overall distortion in the system.

Let us consider what we expect from an ideal system and how such a system would perform. The specifications of an ideal system are not
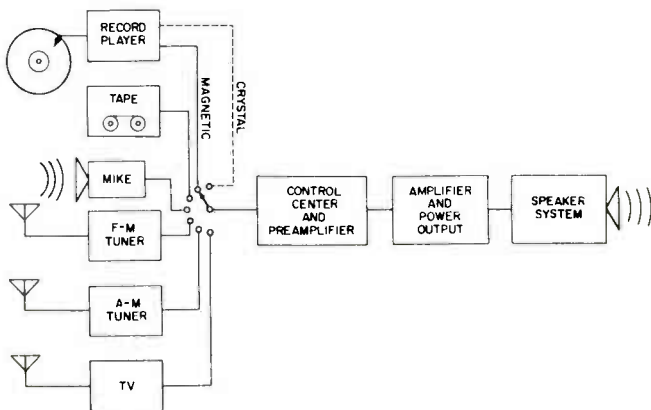

Fig. 1-2. Block diagram of a high-fidelity system.

easy to state, because, even in the actual attendance of a listener at a concert, the location of his seat, the arrangement of the orchestra, and the acoustics of the hall can greatly influence just exactly how the music sounds to the listener. Most of the tastes and reactions of the listener are conditioned by experience and are too complex to be classified in any complete manner.

To keep our discussion concrete and practical, therefore, we must concentrate on those electrical and physical features which distinguish a given system from other systems and which, in the most direct way, provide the information needed by the prospective purchaser of such a system.

The most generally accepted concept of perfection in a high-fidelity system is that which envisions reproduction sounding to the listener exactly as though he were present at the location of the original source of the music at the time this music was being recorded or transmitted. Seldom, if ever, will a system approach such a condition, but this is the earnest objective of the high-fidelity enthusiast.

Imperfections are generally classified according to their effects on performance. These effects are as follows:

1. Frequency distortion.
2. Amplitude distortion.
3. Spatial distortion.
4. Phase distortion.
5. Transient distortion.

Frequency Distortion

Frequency distortion is the variation of sound output intensity with frequency, for constant input intensity. This ordinarily has the effect of limiting the range of sound frequencies which can be usefully reproduced, or at least of reducing the relative amplitude of certain frequency components so much that the sound loses its naturalness. Frequency distortion may arise electrically in the amplifier, in the transformer which couples it to the speaker system, and in the speaker voice coil. Frequency distortion may arise mechanically in the diaphragm or cone, in its mounting and orientation, and acoustically in the transfer from the diaphragm to the space into which the sound is radiated. Input and storage devices such as tapes, records, phono pickups, tuners, and microphones can also introduce frequency distortion.

Amplitude Distortion

Amplitude distortion is the failure of the instantaneous amplitude (intensity) of the sound output of any or all frequencies to be directly proportional to the instantaneous amplitude of the electrical signal input. An ideal system would have an output-versus-input amplitude relation

which is plotted as a straight line and is thus *linear*. For example, in a linear system, doubling the voltage (or current) of the electrical input would double the intensity of the sound output. Tripling the electrical input would triple the output. Any variation (an increase or decrease) of the input will cause a corresponding proportional change in the output. Since in this ideal system the output is always directly proportional to the input, no amplitude distortion is introduced, and the waveform of the sound-pressure (intensity) output is an exact replica of the electrical voltage or current input waveform.

Practically, however, some amplitude distortion is introduced at some point or points in every system, so that the input-output relation cannot be plotted as a straight line; the relation is thus nonlinear to some degree. For this reason, this type of distortion is often referred to as *nonlinear distortion.*

The effects of linearity on a sine-wave input are shown in Fig. 1-3. The characteristic of the linear system is illustrated in Fig. 1-3A. With the straight-line characteristic, the ratio between input voltages before and after any change is the same as the ratio between resulting sound intensities. For example, the variation *a-b-c* on one side of zero is the same as variation *c-d-e* on the other side, for both input and output. On the other hand, this is not true for the nonlinear system illustrated in Fig. 1-3B. There, the curvature of the characteristic is such that portion *a-b-c* of the input signal produces a much smaller variation of output sound intensity than does portion *c-d-e*. The output waveform is therefore distorted. Its nonsinusoidal characteristic indicates that harmonic distortion (a form of amplitude distortion) has been introduced.

Nonlinearity in the output section of audio systems can result from poor or limited design in the output transformer, from changes in the radiation efficiency of the diaphragm or cones due to flexing with amplitude, from a change of effective magnetic-flux density with a change in the motion of the voice coil or diaphragm, and from nonlinearity of the air. (The ratio between compressed volume and compressing force is not constant with variation of level.) Poorly designed input devices, amplifiers, and control sections can introduce nonlinearity as the sound passes through the system.

There are two main types of amplitude distortion: harmonic distortion and intermodulation distortion.

Harmonic Distortion—This results from the fact that passage of a signal through a nonlinear system generates frequency components not present in the original signal and having frequencies which are integral multiples (1, 2, 3, 4, etc. times) of the frequency of the signal from which they are generated. For example, a nonlinear system to which a pure sine-wave electrical signal of 400 Hz is applied would generate and radiate sound energy at such frequencies as 800, 1200, and 1600 Hz in addition to that at 400 Hz. Fig. 1-3C illustrates harmonic distortion due

(A) Linear system.
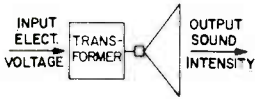
(B) Nonlinear system.
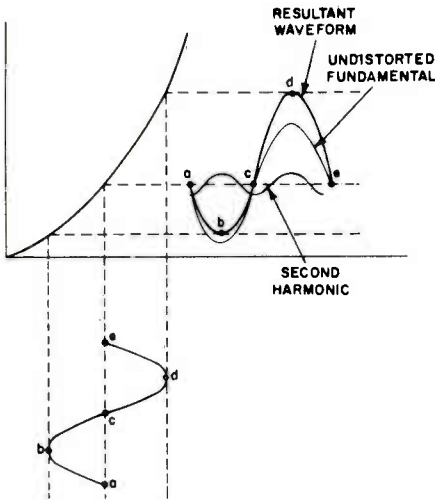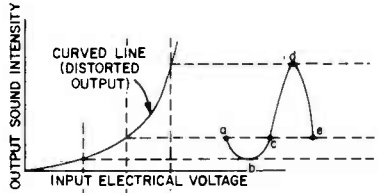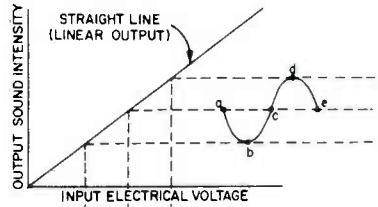
(C) Harmonic distortion.

(D) Intermodulation distortion.

Fig. 1-3. Effects of linear and nonlinear systems on sine-wave inputs.

to a nonlinear system. Notice that the output waveform, which is "flattened" somewhat on the negative alternation and "peaked" somewhat on the positive altern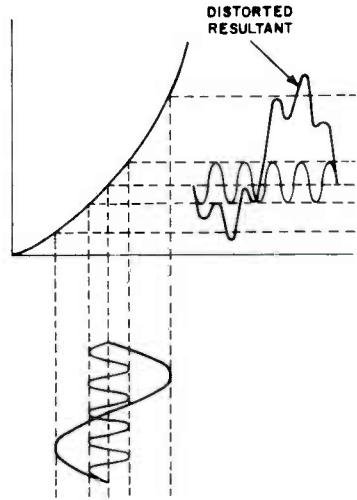ation, is a combination of the undistorted fundamental and its second harmonic. If both alternations had been "flattened," the output waveform would have been composed of the fundamental and the third harmonic. Similarly, more complex output waveforms contain higher-order harmonics.

Intermodulation Distortion—When two pure sine-wave signals of different frequencies are applied to a good speaker system, they should have no effect on one another and should appear separate and distinct as sound output components. In a nonlinear system, however, the two signals heterodyne in the same way as the oscillator and incoming signals in the mixer of a superheterodyne receiver; they produce new undesired frequency components with frequencies equal respectively to the sum and difference of the frequencies of the original sine-wave signals. The harmonics arising from harmonic distortion are also obtained, along with frequency components of the sums and differences of these harmonics. This mixing process is similar to that used in modulation; hence its designation is *intermodulation distortion*.

Introduction of intermodulation distortion by nonlinearity is illustrated in Fig. 1-3D. In this figure two input signals are applied to a nonlinear system. At this point, the two signals intermodulate and produce a waveform with both harmonic and intermodulation distortion. Although the original frequency components are still present, the output signal also contains new distortion components with frequencies which are respectively equal to the sums and differences of the frequencies of the applied input signals. Harmonic-distortion components are also present. The same factors in audio-system design and construction which cause nonlinearity and produce harmonic distortion also cause intermodulation distortion.

Spatial Distortion

Spatial distortion is distortion which manifests itself to the listener as a real or apparent wrong location of the source of sound. It arises either from a narrow directivity characteristic of the speaker or from the failure of the system to simulate the true spatial distribution of the sources of sounds being reproduced. The directivity characteristic is a feature of the speaker system alone; on the other hand, spatial location of sound sources to stimulate the original can be obtained in the speaker system only when the original sound is transmitted or recorded with this in mind. Binaural and stereophonic techniques are examples of the latter.

Phase Distortion

Phase distortion is distortion resulting when the different frequency components are reproduced in improper time relation to each other. The

causes of phase distortion are generally the same as the causes of frequency distortion, and substantial frequency distortion is practically always accompanied by phase distortion.

Transient Distortion

Transient distortion is failure of a system to follow exactly the sudden large changes in sound level. If the speaker system is not properly designed, pulses of sound energy tend to shock the system into oscillation at its natural frequency. The flywheel effect of these oscillating circuits causes the oscillation to continue after the true pulse which excited it has ceased. This effect is often referred to as *hangover*.

PERFORMANCE GOALS

To ensure our selection of the best type of high-fidelity system for our needs and desires, we should be thoroughly familiar with the ultimate in performance. In other words, if we have performance goals to aim at, we will best know what to look for in practical but less-than-perfect equipment. Keeping in mind the types of imperfections reviewed in the foregoing and the kinds of distortion they can introduce, we may now summarize the features of a theoretically ideal system. Such an ideal system would do the following:

1. Interpret, amplify, compensate, and reproduce sound components of any and all frequencies in the audible range with good efficiency.
2. Add negligible frequency components not in the original sound.
3. Distribute the sound in such a way that its sources would appear to be located nearly the same as they were in the original and so that the quality of the sound would be independent of the location of the listener with respect to the speaker system.
4. Allow negligible unnatural delay of some frequency components relative to others.
5. Reproduce, without resonance effects or hangover, the sudden large changes in sound volume level.

Monophonic and Stereophonic Sound

The word "monophonic" is derived from two Greek roots: *mono*, meaning "one," and *phone*, meaning "sound." Thus, the combined roots mean "one sound," or as we use it in hi-fi parlance, "one-source sound" (one-amplifier channel, one-speaker system).

The word "stereophonic" is also derived from two Greek roots: *stereos*, meaning "solid," and *phone*, meaning "sound." Thus "stereophonic" denotes "solid" or three-dimensional sound—sound coming from different sources, at different locations, with different volume levels (two or more separate channels).

Other terms used in describing monophonic and stereophonic sound are "monaural" and "binaural." Since *mono* means "one" and *auris* means "ear," the root of "monaural" is "one-ear." The term "monaural" was used for some time to describe what we now know as monophonic. However, due to the limited scope of the word "monaural," the word "monophonic," meaning one channel from start to finish, has superseded it.

The other term often associated with hi-fi systems is "binaural." This word is derived from two roots: *bi*, meaning "two," and *auris*, meaning "ear." Thus, "binaural" might be literally translated as "two-ear sound."

The association of binaural sound with stereo sound arises from the fact that the binaural system was an early method of obtaining stereo effects. The basic idea behind binaural techniques is the fact that we, as human beings, have a sensation of direction in the sound we hear because our two ears work separately. The sound sensation to the brain from one ear is kept separate from the sensation from the other ear, and both are transmitted to the brain through separate auditory nerves. The brain compares the two auditory signals received by the ears and, from the differences between them, determines the direction from which the sound came. The theory behind binaural systems is that if the two sound signals, which would be heard by a pair of human ears at the source, are transmitted, reproduced in the same relationship, and applied to the corresponding ears at a remote location, all the directional effects of direct live listening will be preserved. Binaural systems are discussed later in this chapter.

MONOPHONIC SOUND

In a monophonic sound system, the sound usually emanates from only one location when being reproduced. Dual speakers, large horns, and the location of the speaker system in a corner of a room, can be used to spread the sound so that it is difficult to place the sound source at one point. However, there is no stereo effect until two separate channels utilizing several microphones, amplifiers, and speaker systems are used.

While monophonic sound may be very pleasing to listen to, stereophonic sound has advantages for the music lover that cannot be equaled by monophonic systems.

STEREOPHONIC SOUND

Modern stereophonic sound, with its directivity and depth properties, adds the third dimension to the sound. It makes clear distinction between foreground, middle, and background, as well as between right, middle, and left sound sources. Stereo has thereby been able to produce a greater amount of clarity and instrumental sound color than monophonic sound. The reception of complex sound sources is also made possible. The directional effects and small time delays from echoing and reverberation of the elemental sounds of music are separately channeled from source to ear, thus providing high-quality simulation of live music.

History of Stereo

The idea of stereophonic sound is not new. Ever since electrical sound systems were first devised in the form of telephone circuits, engineers realized the spatial characteristics of the reproduced sound are important. It is known that as far back as 1881 experiments with binaural sound were being made. In that year, performances from the Paris Opera were transmitted via a pair of telephone lines to the Paris Exposition. Each telephone line constituted a channel, with the two channels corresponding to the two ears of the listener.

In the early 1920's, shortly after standard a-m broadcasting began in the United States, experiments were made with dual radio broadcasts. However, the fact that two receivers were scarce prevented interest from becoming great enough to maintain these broadcasts.

On April 27, 1933, engineers of the Bell Telephone Laboratories transmitted the music of the Philadelphia Symphony from the Academy of Music in Philadelphia, to Constitution Hall in Washington, D. C. A three-channel system was used in this experiment.

Perhaps one of the most spectacular steps forward in the field of stereo sound reproduction was the introduction in 1940 of Walt Disney's *Fantasia*. This was a movie for which the accompanying sound approached true stereo characteristics. The sound was recorded using a large number of microphones, each feeding a separate recording chan-

nel. In the the theater in which the picture was shown, each of the many speakers was placed in the same relative position as a microphone in the recording setup. Speakers were mounted in positions all around the perimeter of the theater, even in the back. To the audience, the sound could come from any direction, including the sides and the back. This production was a huge success, due in large degree to the realism and unusual nature of the sound. Today, practically all movies use wide-screen projection along with some form of stereo sound, but none approach the number of sound channels used in *Fantasia*.

All the foregoing progress was confined to the commercial theater or communications business. For a long time, people in the sound business have realized that, until a simple and convenient method of recording stereo sound on discs was perfected, stereo could not be introduced into the majority of homes. In 1931, A. D. Blumlein obtained a British patent on a system for cutting and reproducing two-channel recording discs. Later, in 1936, Bell Laboratories engineers A. C. Keller and I. S. Rafuse obtained United States patents for two-channel disc recording. However, these early ideas did not blossom into commercial reality because materials, methods, and techniques had not advanced to the point at which production and distribution were feasible.

In 1952, another pioneer, Emory Cook, developed binaural disc records using two normal pickups spaced about two inches apart. The outer portion of the record surface was used for recording one channel, and the inner surface for the other channel. Because two pickups were needed, there were tracking problems with this arrangement and it never gained wide acceptance. It wasn't until single-groove systems were developed that stereo discs became practical.

Single-groove stereo records became a reality to the public in 1956, when the London (British Decca) Company developed its system. The next year the Westrex stereo-disc system was introduced in the United States. (The Westrex system has been standardized in this country, and it is the subject of detailed discussions in later chapters.) Since the advent of stereo discs in large quantities other components of stereo systems have followed rapidly. Today there is available to the public a wide variety of stereo tuners, amplifiers, speaker systems, and other accessories, all of which are discussed in the following chapters.

STEREOPHONIC SYSTEMS

In Chapter 3 it is shown that speakers have directivity which varies with frequency. This means that the frequencies heard from a speaker vary with the position of the listener with respect to the speaker. To put it more simply: If you're in front of the speaker, you hear highs, and if you're off to the side, you don't. This is a kind of spatial distortion. But there are several other kinds of spatial distortion which engineers strive to overcome with stereo systems.

If the foregoing type of distortion is eliminated, and all the frequency components of the sound are radiated equally in all directions, all of the reproduced sound will still emanate from the point at which the speaker is located. The fact that the instruments or voices which originally produced the sound were widely separated spatially means that the single-point speaker source is not realistic. In other words, there is distortion; one might quite accurately call this "apparent-source direction distortion."

For example, if we attend a concert and don't sit too far back from the stage (or orchestra pit), we will be clearly aware that the piano is, say, to our left, the violins are the right, and the drums possibly in the middle portion of the stage. Of course, our eyes tell us these things, but, when the orchestra begins to play, our ears will also tell us. As explained more fully later, the human auditory system is keenly directional, with the slight difference between the sound components which enter the two ears indicating to the brain the direction of the source. If, in our example, we don't hear the piano, the violins, the drums, and the other instruments as though they are coming from their respective directions, we are not hearing an exact simulation of the source material. In this case, we cannot experience the realism of high fidelity. True stereophonic sound overcomes this lack and restores to the listener a sense of the direction of the original sources of the instruments creating the sound.

Recording Techniques

Realism is preserved in stereophonic sound by picking up and reproducing sounds at different depths (distances) as well as from different directions in proper or exaggerated relationships. Delays to produce an effect of one sound being deeper (further away) in relation to another may be introduced artificially within the recording equipment. This method is known as *delay stereophony*.

An old trick in the recording business is to add an artificial second channel having the same material as the first, but with a time delay introduced in it. This provides an artificial stereophonically enhanced sound. The most effective time delay is believed to be between 8 and 12 milliseconds.

Variations in intensity of the same sounds reproduced at properly spaced positions within a room will provide a stereophonic effect if reproduced at properly spaced positions within the room. This is easily demonstrated by the ping-pong effect, where the sound of a ping-pong ball bouncing back and forth is reproduced.

Recording music from an orchestra is a refined process of organization, placement, separation, emphasis, combination, equalization, reverberation, limiting, rolling off, takes, inserts, overdubs, and editing. Usually the orchestra is arranged in a circular manner around the conductor, spaced in a manner suited to recording—which is different from arrange-

ments made for performing for an audience. There may be an isolation booth for a singer who hears (and keeps time with) the orchestra through a low-level speaker or headphones. Each musical instrument or groups of the same instrument may have one or more microphones at strategic locations to take best advantage of the output of each instrument or instrument group. The object is to separate sounds and to reduce "leakage," which is any sound picked up by a microphone that is not intended to be picked up. Separation is essential to achieve clarity, or emphasis, and balance control for the production of a good record. Low-volume instruments such as flutes must be separated from louder instruments, or their presence will be obscured.

For these reasons the musicians are sometimes separated by space or by sound-reflecting walls. Microphones are placed closer to softer musical instruments to increase their relative outputs. On occasion an instrument such as a bass viol is placed behind a V-shaped wall to prevent its high-level sound waves from spreading around the room. Dozens of microphones may be used with artful placement to develop the desired separation, presence, or intimacy. Hard music, such as rock-and-roll, requires closer "miking" and more microphones. For violins, several microphones may be placed over a group and the heights varied to change the effect during the program.

The output of all this is combined in the recording room on tape. The output level of each microphone is adjustable on the control engineer's panel so that the separate sounds can be narrowed, broadened, increased, decreased, and parts of the audio spectrum emphasized or expanded, de-emphasized or rolled off, according to the art of creating a beautiful and dynamic combination of the input elements of sound. It can be seen that the engineer must be a talented artist, as well as a good technical man, to be able to produce a creative "mix." There is a striking difference between the sounds in the orchestra room and the sound output of the final recording.

The control room is essentially concerned with mixing, emphasis, reverberation, and limiting. Since each mike has its own control, the mix is accomplished in the control room, not in the orchestra room nor by the conductor. *Emphasis* or *de-emphasis* is the increase or decrease of selected portions of the a-f spectrum used to enhance an instrument's sound or delete its lesser outputs by means of electronic equipment. High frequencies are boosted to bring out the overtones of violins; this makes them "brighter." The mid-range frequencies are often boosted when recording guitars, drums, and percussion instruments to create the hard sounds of rock-and-roll records.

Reverberation is applied by the electronic equipment to sustain musical sounds. This has the same effect as enlarging the room containing the orchestra or the output of the specific instrument so treated. It is said that this adds excitement but it is often used to "cover up" poor musical

performance. Reverberation of sounds can also be created in specially designed, small echo chambers or with electromechanical devices.

Limiting is provided by automatic electronic equipment designed to instantaneously reduce volume peaks that exceed the capabilities of the electronic systems used. This protects the recording against damage and distortion due to overloading.

The *take* is the actual recording of the live music, mixed, emphasized, balanced, etc., amplified, and recorded on tape.

The *insert* is a partial take of the program of that part that is desired to be improved. This is spliced in place on the original master tape. Good inserts are hard to achieve because it is difficult to repeat the same texture of performance from any group at a separate time from the first or "master" play. Tempo, mix, balance, and emphasis must come together to sound the same.

The *overdub* is a technique which is used to apply the vocal performance onto the master tape after the orchestra is recorded. The master tape of the orchestra performance is played back while the vocalist accompanies this playback. The orchestra's playback is wired directly from the recorder to another tape deck to produce a copy of the orchestration while the new vocal performance is added to the vocal track of the copy. This type of operation is used to improve an original poor vocalization and/or to reduce the cost of recording. However, there must be available a master tape of the orchestra without vocal. Two tapes are usually made: one with the vocal and one without it. In order to achieve this, an isolation booth is required for the vocalist. Overdubs are also used to create special effects, such as doubling the number of strings in an orchestra.

Editing is the elimination of lesser parts of the performance or reduction of length to fit standard record or tape-cartridge sizes. The best parts are pieced together to make a finer performance, or a selection may be shorted, or "clams" and studio noises eliminated. Following the editing process, master records or master processing tapes are made from the edited master. The master records and tapes are used to reproduce records, tapes and tape cartridges at a small fraction of the cost of making the original master.

Channels

In the course of our discussions of stereo, we often use the word "channel." As applied to this subject, "channel" means a separate and distinct path for an electrical or acoustic signal. For example, an ordinary monophonic audio-frequency amplifier is part of a single channel, because it can carry only one signal at a time. If we attempt to use the amplifier for more than one signal at the same time, the two signals will interfere with each other and cannot be separated at the output. In non-stereo systems, only one channel is used. One microphone picks up the

sound, which is amplified in one amplifier of conventional design, and fed to one speaker system. In a two-channel system, two or more microphones are used; the sound signal from each microphone is passed through a separate amplifier system, so that one sound signal has no effect on the other. As we shall see later, three amplifiers are not always necessary for three channels, because there are techniques of providing three electrical channels in two amplifiers. The effect is the same as though three complete amplifiers were used. One of three channels can be simulated in a two-channel amplifier by additive or subtractive methods to gain the practical effect of a third channel.

Multichannel Systems

As was said, stereo sound is "solid sound," that is, sound which, even though it is artificially reproduced, seems to the listener to come from the same directions that it would if he were present at the source. One theoretical way to reproduce stereo sound would be to mount speakers all around the listener. Then we would design the system so that the speaker or speakers in the respective directions are activated at the right times with the right portions of the sound program.

Let us imagine that one listener is listening to a live performance of an orchestra in a concert hall, and a second listener is in a similar position in a similar room in a remote location to which the sound is being transmitted by a stereo system. If the piano is playing directly in front of the concert-hall listener, then the stereo system should reproduce the piano primarily from a speaker directly in front of the remotely located listener. If another instrument is located to the left side of the local listener, the listener in the remote location should hear it from the speaker or speakers to his left, and so on.

This kind of arrangement is employed in elaborate systems such as were used for *Fantasia*, and, to a lesser degree, is being used in movie theaters today. The method is depicted in the diagram of Fig. 2-1. Theoretically, for "perfect" stereo effect, we would need an infinite number of speakers, so that there will be no "holes" in our reproduction. However, as we shall see, this is no problem because the same effect as additional speakers can be obtained by the "sharing" between adjacent units. As can be seen from this illustration, such an elaborate system is very expensive, and there are many electrical problems that are not immediately apparent.

Binaural System

The foregoing method operates on the principle of providing actual sound from all the directions involved. Obviously, this method is far too complex to be practical for home use. Other methods, including the stereo methods now in general use, provide a sense of direction and depth even though as few as two channels are used.
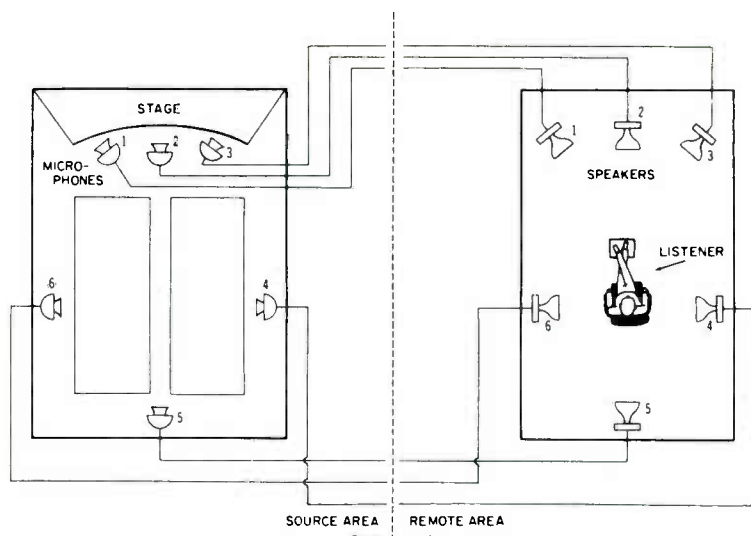
Fig. 2-1. An elaborate stereo system.

The binaural system illustrated in Fig. 2-2 is an example of a two-channel system. Rigidly correct binaural systems use an actual "dummy" head at the source location as shown. A microphone is mounted at each ear of the dummy. Thus each microphone should "hear" exactly what each ear would hear if the dummy were a human being. The whole head is used, instead of just placing microphones at the two ear locations to simulate the effect of the human head on the sound waves. Each microphone is connected to a separate amplifier and then transmitted to the listener's location. There, each channel terminates in an earphone on the same side of the head as the source microphone for that channel. If we assume that the microphones, amplifiers, transmission systems, and earphones have a high degree of fidelity, each ear of the listener should receive the same sound as the corresponding ear of the dummy at the source. The listener's auditory system puts the two sounds together to provide indication of the directions and depth from which the sound components are reaching the dummy at the source.
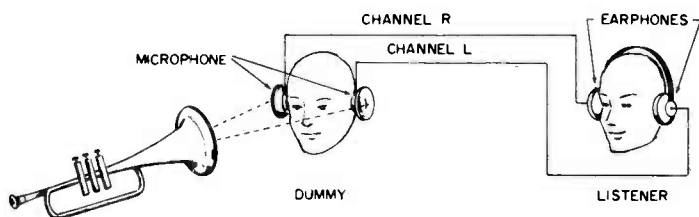
Fig. 2-2. Simple representation of a binaural system.

23

Two-Channel Stereo

Binaural systems give a good degree of realism, but most people don't want the inconvenience of wearing headphones. That is why the stereo system of today uses speakers. However, the two channels of a stereo system operate on the principle that if the two speakers are located the same distance apart as two pickup microphones at the source the outputs of the speakers will combine in such a way that the listener is given the sensation of sound direction and depth comparable to that of a listener at the source. A simple diagram of this kind of stereo system is shown in Fig. 2-3. The system is designed to have each speaker reproduce the sound which is present at its corresponding position at the source.
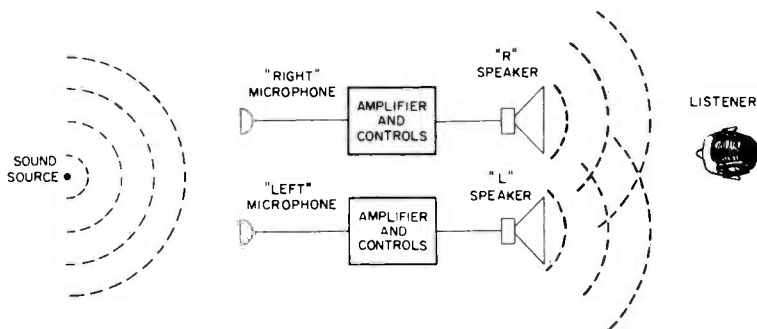
Fig. 2-3. Simple representation of the stereophonic principle.

Speaker and Microphone Placement—The obvious advantage of stereo reproduction is that, added to other high-fidelity characteristics, it gives the listener greater enjoyment through the sensation of the relative direction, depth, and intensity of the various parts of the program. It affords a greater satisfaction because it gives an added dimension of realism.

However, there is difficulty in locating the listener in the same relative position as the "assumed listener" at the source. The listener (at the source) is considered to be sitting along the center line of the hall, and about equidistant from the two microphones. But suppose the remote listener's setup is such that he can't place himself at the same distance from the speakers as the assumed listener is from the microphone at the source. It has been found that small differences have little negative effect, but if the listener is sitting directly in front of one of the speakers, he hears practically nothing from the other speaker, and his sensation of sound direction is spoiled. Therefore, proper positioning of equipment and the audience is an important consideration.

Another important consideration is how far apart the microphones should be from each other when recording, and how far the speakers

24

should be from each other when reproducing the recording. It is logical, and generally accepted, that the speakers should have the same spacing as the microphones. To understand better the problem of deciding what this spacing should be, let us consider the extreme cases. If the microphones and speakers were as close as they could be to each other, they would appear as one, and we would have a monophonic system. But let's assume that they're just a few inches apart. Then if the listener stands twenty feet away they will still effectively appear as one source. If he moves closer, until his distance from the speakers is comparable with the distance between them, the stereophonic effect is reestablished.

Now suppose that we spread the microphones and speakers very far apart. In the extreme, they would be so distant that they would not pick up sound or reproduce it so it could be heard. Closer together, but still widely spaced, the reproduced sound would be heard from two separate and distinct sources. This effect is known as the "hole in the middle," because nothing seems to be coming from the area between the two speakers, whereas the center of the orchestra would be at this point at the source.

To overcome the "hole-in-the-middle" effect, stereo may employ any number of channels. An arrangement may provide for pickup of an orchestra with six or even twelve microphones proportionally spaced around the orchestra. A corresponding number of amplifiers are used to reproduce the program. The amplifiers feed the speakers which are placed at the points where microphones were located in the original recording. Certain motion-picture extravaganzas have used these techniques.

The complete stereo system just described produces a very realistic stereophonic effect, imparting breadth, depth, and even height to the sound output. In addition, many detailed directional effects are possible. However, three speakers with two channels can provide nearly the same effect. Improved techniques of recording—microphone placement, channel intensity control, and speaker placement—have made it possible to approach the quality of three-channel stereo with two channels (and a third speaker). This is accomplished by combining certain portions of the right- and left-channel signals and feeding the resulting signal to the center speaker.

To provide signals for the third speaker, three microphones, properly spaced from left to right, may be employed at the sound source. The signal from the middle microphone is mixed with the signals from the left and right microphones, and the resultant signals are fed to the left and right channels respectively. Thus, the left and right channels contain a certain proportion of the effects of the center microphone output.

The signals at the output of the preamplifiers may be divided so that the majority of the output from either channel is fed to its respective left or right speaker, but a small portion (less than a third) of the output of

each channel is fed to the center speaker. Therefore the center speaker operates at a lower input level than do the other two.

Only a small difference in the intensity of the sound from the two outer speakers is sufficient to give the desired directional effect. The subdued level of the center speaker output allows the signal from the left channel to appear at the left and the signal from the right channel to appear at the right, and the directional effect is maintained. However, the stereo effect is produced over a greater area by the third speaker. The listener can sit nearer or farther from the speakers, or move more to the right or left without losing the stereo effect. The illusion of a "curtain of sound" spreading across the three speakers is developed.

It can be seen from the foregoing, that adding the third speaker involves more than parallel connections. Technical instructions for setting up three-speaker stereo are given later in this book.

COMPATIBILITY

It is likely that monophonic equipment will be used to some degree for some time to come. It is therefore important that stereo systems be compatible with monophonic systems. By "compatible" we mean that one should be able to play a monophonic record or a broadcast tuner on a stereo system and be able to play a stereo record or tuner on a monophonic system. If any part of the system—the record, the tuner, or the amplifier—is limited to monophonic performance, the result is monophonic reproduction. However, we say a system is "compatible" if monophonic and stereophonic reproduction is practical and normal in the same system. Design for compatibility has been mainly one involving records, phono pickups, and broadcasting techniques; it is relatively easy to combine or separate the two channels of the remainder of a stereo system to produce monophonic or stereo output.

The Westrex stereo recording method, now standard, is designed for near-perfect compatibility in record reproduction, as will be shown in Chapter 3. Unfortunately, in practice, true compatibility does not yet exist because of mechanical limitations of the older monophonic pickups. As a result, stereo records should never be played with monophonic pickups. However, monophonic records can be played with stereo pickups, and often the reproduction is more pleasing than when the record is played on a monophonic system. Of course, there is no stereo effect. Details about compatibility and difficulties in attempting to play stereo records with monophonic pickups are discussed in Chapter 3.

3

Stereo Techniques

The desirability of stereo reproduction, and what had to be done to realize it, has been known for many years. However, it was only recently that definite techniques were developed and applied to hi-fi equipment. In this chapter, these techniques are discussed.

Stereophonic sound equipment utilizes all the circuitry contained in monophonic equipment. In addition, the circuits and devices necessary to add a second channel and provisions for two or three different outputs must be included in stereophonic equipment.

In the past decade the reproduction obtained from monophonic, single-channel hi-fi equipment has reached a high state of development. The quality of single-source sound, with the best systems, is such that the average human is unable to detect any distortion.

BASIC COMPONENTS OF STEREO SYSTEMS

The basic components of a monophonic system are shown in Fig. 1-2. This arrangement may be adapted to two-channel stereo by adding another channel of amplification and an additional speaker system, which will permit stereo reproduction of simulcasts. Then, by adding a stereo phono pickup, tape deck, or multiplex adapter, the other input devices can be converted for stereo operation.

Improvements in the form of adapters are available to provide one control for balancing the outputs of the separate channels, and other conveniences are available for patching up the old single-channel system to produce stereo. However, this kind of arrangement is not competitive in convenience of operation, flexibility, simplicity, or other refinements when compared with equipment designed especially for stereo operation.

STEREO RECORDING

To us, as hi-fi listeners, the manner in which the recorded or broadcast programs which we enjoy are picked up at the source is only of

27

academic interest. With discs and tuners we concentrate on reproducing the music available to us, and the studio techniques over which we have no control do not interest us. However, with the availability of multitrack tape recorders for the home, there is growing interest in amateur recording, and the addition of stereo makes it much more interesting. Also, a knowledge of microphone placement problems for pickup helps us in deciding speaker placement for our reproducing system.

The most important general type of stereo pickup technique is called *time-intensity pickup*. This means that the signals in the two stereo channels will vary in both intensity and time according to the difference in direction and in distance of the sound source from the two microphones.

As explained in Chapter 2, the stereo effect can be obtained by picking up the sound with two microphones spaced a certain distance apart and located in front of the sound source. But how far should the microphones be separated? If they're too close together, the differences in time and intensity of the two signals are negligible and the stereo effect is lost. Experiments have been made with microphones as much as 30 feet apart. At such distances, the reproduced sound seems to the listener to divide into two sources, especially if he is close to the speakers.

(A) *Speaker placement.*   (B) *Microphone placement.*
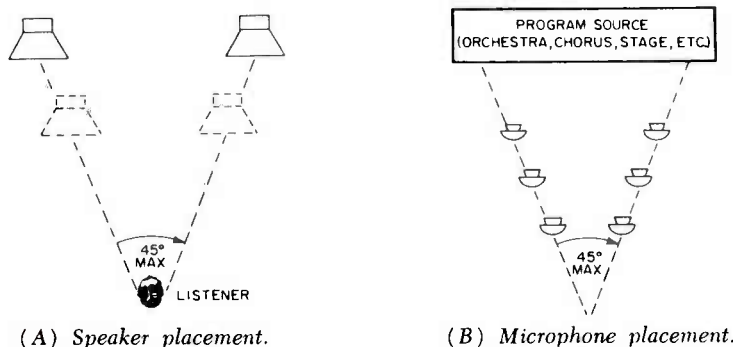
Fig. 3-1. Triangles indicating suitable locations for speakers during reproduction and microphones during pickup.

Experiments with the relative spacings involved have led to the conclusion that there is a definite optimum relation between the microphones' separation and their distance from the source. A similar relation exists between the listener's speakers and their distance from him. This relation can be illustrated by the triangle formed by the two speakers and the listener as shown in Fig. 3-1A. In practice the angle at the listener should not be less than 30 degrees nor exceed 45 degrees. This relation keeps the distance of the listener from the speakers approximately the same as the distance between the speakers, and ensures sufficient time-intensity variation to provide stereo effect.

It is generally agreed that in the ideal two-channel system, the listener's speakers should be the same distance apart as the studio microphones. However, if we stop and think of the size of a full symphony orchestra, and how far the microphones need to be spaced to have a good part of the orchestra between them, we realize that the same spacing at both ends of the circuit is not always practical. In practice, compromises are made. A good rule to follow is to spread the microphones as far apart as you can and still pick up an appreciable amount of sound from the middle portion of the source. Usually the pattern will be such that they can be placed along the sides of a triangle like that shown in Fig. 3-1B.
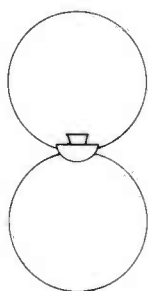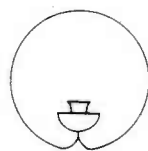
(A) *Figure 8.*   (B) *Cardioid.*

Fig. 3-2. Typical directional-microphone pickup patterns.

Another method, referred to as the *intensity-difference system*, is sometimes used in stereo recording. In this system, the microphones are not spaced; however, they have a directional characteristic as shown in Fig. 3-2. The stereo effect is obtained by proper orientation of the microphones. One method is to mount the microphones at 90 degrees to each other and 45 degrees to the center of the source, as shown in Fig. 3-3A. Another method is to use one microphone aimed directly at the source for the main signal, and a second microphone to pick up side signals. This is illustrated in Fig. 3-3B.

A directional microphone especially designed for stereo pickup is pictured in Fig. 3-4A. This microphone contains two directional elements as shown in the cutaway side view of Fig. 3-4B. Each element has a cardioid pickup pattern, and the two elements are aimed 90 degrees apart. Thus, each element is aimed 45 degrees from the center of the sound source, producing the response given in Fig. 3-4C.

DISC RECORDING

It was not until practical methods of stereo disc recording were developed that standardized stereo could reach our homes. Although other good methods have been considered, the Westrex method of stereo re-

29

CENTER OF SOURCE
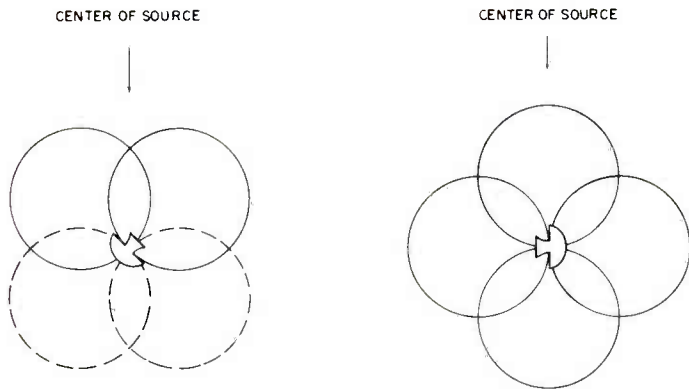
CENTER OF SOURCE

(A) *Two microphones mounted at 45°*
from center of source.

(B) *One microphone for main signal,*
another for side signals.

Fig. 3-3. Placement of two directional microphones for stereo pickup.

cording is the one adopted by the recording industry. We shall, there-
fore, confine our discussion here to that method.

Requirements

Let's review briefly the general requirements of a stereo disc recording
method. They are as follows:

1. Two completely independent left and right signals must be re-
   corded separately in such a way that there is a minimum of inter-
   ference or mixing between them, and so that, in playback, they
   can be separately recovered in the same form as that in which they
   were recorded.
2. The system must be compatible with monophonic recording and
   playback systems. A stereo playback system should play mono-
   phonic records without loss of fidelity; likewise, a monophonic sys-
   tem should play stereo records and produce as good a monophonic
   output as when it is playing monophonic records.

When Edison first developed phonograph recording, he used vertical
motions of the recording stylus to record the sound vibrations. This came
to be known as the "hill–and–dale" method of recording. However, in
such a system, the "hills" in the record groove had to lift the playback
arm and cartridge. Since early arms and cartridges were relatively heavy
this motion caused excessive wear. Because of this, the record industry
adopted the lateral recording method, in which the needle goes from side
to side in accordance with the sound vibrations. All commercial mono-
phonic records for home use are laterally recorded, and monophonic
pickups are designed accordingly.

(A) Photo.
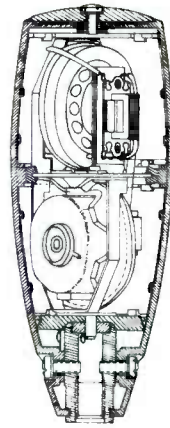
(B) Construction.

(C) Response pattern.
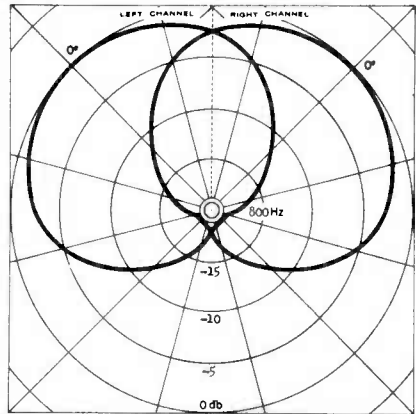
Courtesy North American Philips Co., Inc.

Fig. 3-4. A directional microphone designed for stereo pickup.

The Westrex System

In the Westrex stereo recording system, both stereo and mono, lateral and vertical motions are employed. Modern pickup and arm design is such that vertical motions are usable under the proper conditions. The basic principle of the Westrex system is illustrated in Fig. 3-5. The sound signal for each stereo channel is recorded along a direction at 45 degrees to the horizontal. In other words, each channel is recorded on one side of the record groove in such a way that its undulations will move a playback stylus back and forth along one of the 45-degree directions. The two directions are at right angles (90 degrees) to each other, and motions along either one alone do not affect motion along the other. Thus,

31

the motions of one channel do not interfere with the motions of the other, and two separate output signals, each corresponding to the motions for its respective channel, are obtained from the stereo cartridge.

A symbolic representation of how the axial motions in the two 45-degree directions can generate two separate signals is given in Fig. 3-6. Imagine the assembly of the stylus and the two bar magnets free to move in any direction but not to rotate. Then, if the stylus is pushed to the left, it goes into a position like that of Fig. 3-6B. The magnet at the left moves directly into the coil, thus generating a current in it. The magnet at the right merely moves across the diameter of the coil perpendicular to its axis. There is no axial motion in or out of the coil, so no current is generated in the right coil.
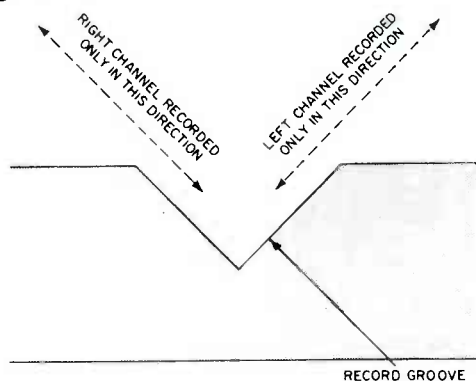
Fig. 3-5. How sound vibrations are cut at 90° to each other and 45° from the horizontal in the Westrex system.

Now suppose that the stylus-magnet assembly is pushed to the right, as shown in Fig. 3-6C. Conditions are now the reverse of those in Fig. 3-6B. A current is generated in the right coil but none is generated in the left coil.

Thus, vibrations recorded in the groove so that they push the stylus at 45 degrees toward the left produce pickup output current only in the left coil. Likewise, vibrations recorded in the groove so that they push the stylus at 45 degrees toward the right produce pickup output current only in the right coil.

It is difficult to imagine any case in which both left (L) and right (R) signals would not be present. Because of this, you may wonder how the playback needle can move in both 45-degree directions at once. Naturally, it cannot do this. Instead it moves in a direction and with amplitude dictated by the vector resultant of the two forces along the 45-degree paths. This means the needle can move as a whole in any resultant direction, including side to side and straight up and down. Five combinations of relative L and R signals are analyzed in Fig. 3-7. Only one signal
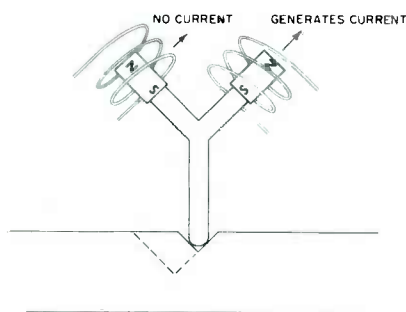
32
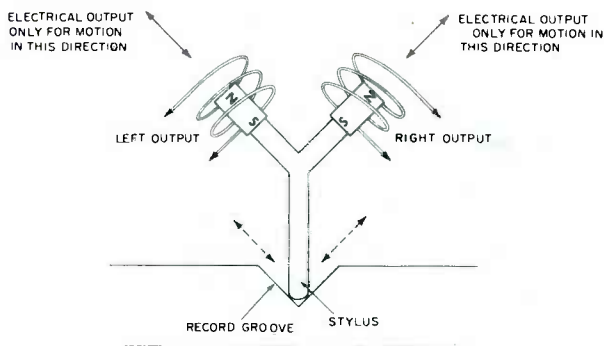
(A) Rest position.
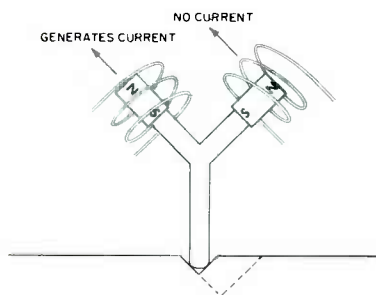
(B) Movement to left.

(C) Movement to right.

Fig. 3-6. How stylus motion in either of the 45° directions causes pickup in one coil but not in the other.

is present in Figs. 3-7A and B. Accordingly, in each of these cases the needle moves along the direction of the single channel present. Thus if the needle's motion is along one of the 45-degree sides of the groove, it

"senses" that it is receiving signal from only one channel. Fig. 3-7C illustrates the special case of equal signals from L and R. In this case, the resultant motion is always vertical, and the record "looks" to the needle like a "hill-and-dale" recording. The vertical motion indicates that both signals are equal in amplitude and the amount of motion indicates their amplitude. Fig. 3-7D indicates the motions when the L signal is much stronger than the R signal. The larger the L signal with respect to the R signal, the further the resultant is to the left of vertical. The angle of resultant motion "tells" the pickup the ratio between the two amplitudes, and the amplitude of the resultant indicates their amplitudes. The

(A) L signal but no R
signal.

(B) R signal but no L
signal.

(C) Equal L and R
signals.

T = RESULTANT MOTION OF NEEDLE

(D) Strong L signal,
weak R signal.

(E) Strong R signal,
weak L signal.
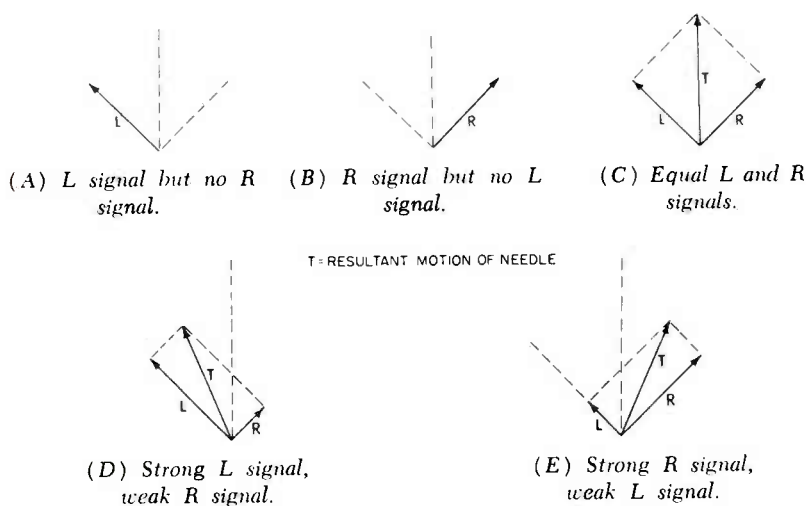
Fig. 3-7. How motions along the two 45° directions in a stereo record groove add up to a single resultant motion.

situation depicted in Fig. 3-7E is the same except that the R signal is stronger than the L signal. The examples in Fig. 3-7 show that for any combination of resultant needle angle and amplitude of motion, there is a distinct amplitude for both the L signal and the R signal.

Since the amplitude of the L and R signals are usually approximately equal at any instant, the conclusion to be drawn from Fig. 3-7 and the preceding discussion is that the groove modulation in stereo records is predominantly vertical. However, vertical groove modulation tends to be more distorted and produce more record wear than lateral modulation. To change the Westrex recording system to one in which groove modulation is predominantly lateral, the phase of one of the signals driving the stereo record cutter is reversed. This changes the resultant needle force from a predominantly vertical motion to one which is predominantly lateral. This method is illustrated in Fig. 3-8. Note that Figs.

34

3-8A, B, and C are the same as Figs. 3-7C, D, and E, respectively, except that the R vector is reversed. The Record Industry Association of America (RIAA), which accepted the Westrex system, has stipulated that "equal in-phase signals in the two channels shall result in lateral modulation of the groove." This requirement is satisfied by the phase reversal of one of the signals to the cutter as shown in Fig. 3-8; but a compensating reversal must take place at the pickup or later in the system for playback. However, this is a simple matter, because it involves only reversing the connections of one of the pairs of leads from the pickup.

(A) Equal L and R signals. (B) Strong L signal, weak R signal. (C) Strong R signal, weak L signal.
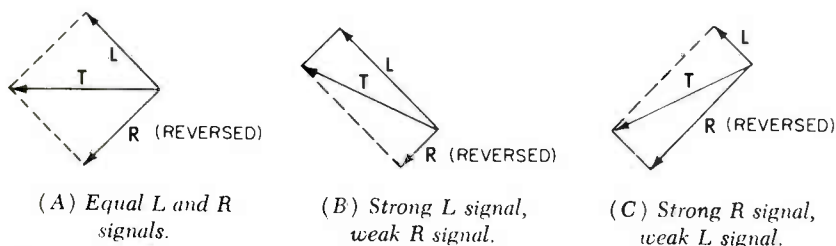
Fig. 3-8. Vector addition of motions of needle in stereo record groove after electrical inversion of one channel.

Special Factors in Stereo Disc Systems

Although the phase reversal of one of the signals to the recording head makes the modulation of a stereo record predominently lateral, the fact still remains that vertical motions play a vital part in recording and playback. This fact brings with it some special problems in stereo pickup design.

The stereo pickup must have high vertical *compliance;* that is, it must allow for easy motion of the needle in the vertical direction. If it does not, the needle exerts undue pressure on the record, causing rapid wear. A good stereo pickup is designed to have the necessary vertical compliance, but monophonic pickups are not. Thus, although as far as motions required of the pickup needle are concerned there is compatibility, use of a monophonic pickup on a stereo record will destroy it after relatively few playings. Therefore, never play stereo records with a monophonic pickup.

Another important factor is the *pinch effect* illustrated in Fig. 3-9. It arises from the fact that, in its lateral motions, the cutting stylus' flat front edge is not at right angles to the direction of motion of the record it is cutting. For example, at A in Fig. 3-9, there is no signal and the stylus is laterally motionless; therefore, the flat front of the stylus is at right angles to the direction of the groove, which at this point has its greatest width. (W). However, at B modulation has been applied; the

35

stylus still has the motion it had at A, but now it also has the lateral motion imparted by the signal applied to the pickup cartridge. The flat front of the stylus, with its cutting edges, now plows sideways through the record materials, so the width of the groove here is W′, which is considerably less than W, the width at A. Unless the tip of the playback needle is extremely fine, the narrowing at B tends to lift it out of the V-shaped groove. This lifting motion is the same as might be experienced with vertical groove modulation. It does not affect a monophonic pickup (except to make the record wear faster), but with the vertical-sensitive stereo pickup it introduces an unwanted signal and distortion.

To attempt to overcome the effects of pinching, the tip radius of stereo pickup needles has been reduced from the over 1-mil (0.001 inch) monophonic microgroove value to 0.5–0.7 mil. However, this reduction in tip diameter introduces another disadvantage—the tip pressure on the record is increased. Pressure is force per unit area, so even if the force of the needle on the record has not increased, a decrease in its area causes a corresponding increase of pressure. Pressure is a main factor in wear on records, so the smaller needle tip results in greater wear, if the other factors remain equal. To keep wear down, stereo pickups are operated at very low needle force. Four grams is usually considered a reasonable maximum.

The same types of pickup devices are used for stereo as for monophonic reproduction. They include moving-coil and moving-magnet magnetic, piezoelectric, and ceramic cartridges. The suspensions used for the needle and its associated mechanical vibrating assembly are very critical and many clever arrangements have been worked out by the engineers.

The principle of how moving magnets are used to generate the two stereo outputs was illustrated in Fig. 3-6. Although that diagram does not correspond to any manufactured cartridge, the manufactured ones
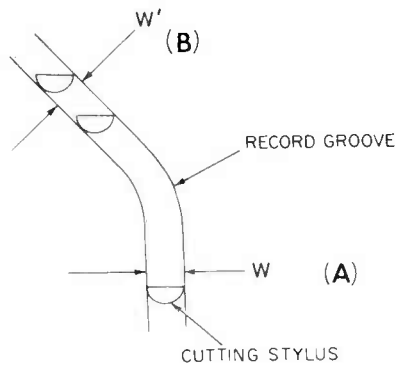
Fig. 3-9. The cause of the pinch effect.

do have a small magnet (or two) moving near a core having a pair of coils. The motion of the magnet(s) generates voltages in the fixed coils. The pole pieces and moving magnets are oriented so that motion of the needle in one 45-degree direction cuts magnetic lines of force for one coil but moves the lines of force parallel to the other coil so that no voltage is generated in it. For movement in the other 45-degree direction, the situation is reversed and voltage is generated in the other coil.

As implied by the name, in the moving-coil magnetic-type (dynamic) pickup, the coils move and the magnet stands still. A special linkage mechanism with jeweled bearings imparts rotational motion to either or both of the coils as the needle is moved by the record undulations. Each coil responds to needle motion in one of the 45-degree directions. The coils operate in a magnetic field, so an audio voltage is generated in them when they turn.

Ceramic elements can also be used in a stereo pickup, as illustrated in Fig. 3-10. As the ceramic crystals are bent, a voltage is generated. As the needle moves to the left, one ceramic element is bent (Fig. 3-10B);

(A) No signal.

(B) Signal moves needle up and to left.

(C) Signal moves needle up and to right.
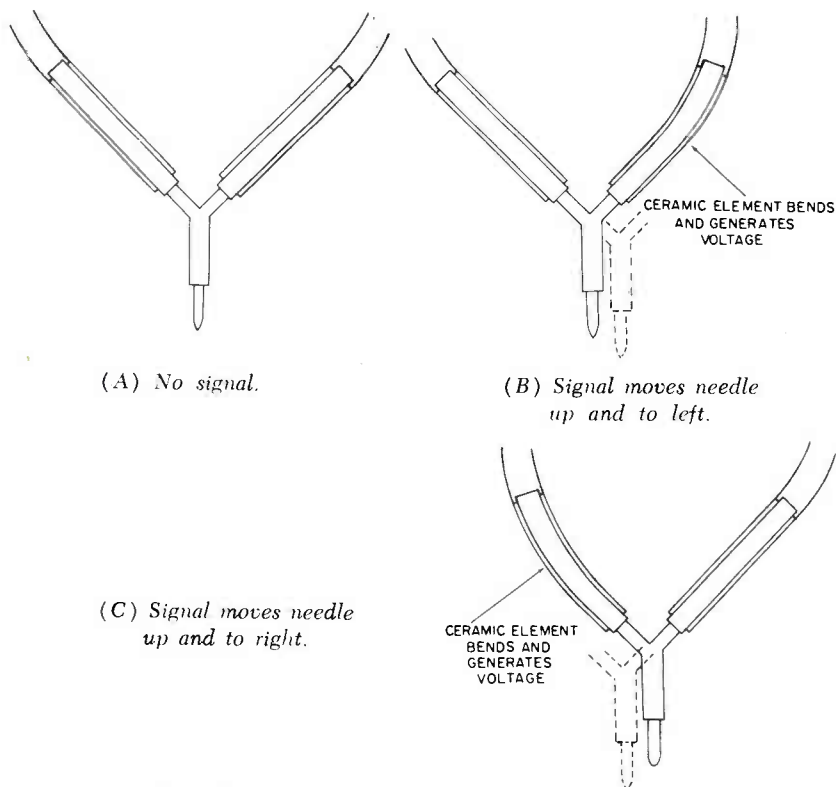
Fig. 3-10. Stereo playback using ceramic elements.

similarly, movement to the right bends the other element (Fig. 3-10C). Thus, two isolated stereo outputs are provided.

Tracking Problems

Trackability distortion is produced when a phono stylus/cartridge transducer does not track or "trace" the grooves of a record in the same manner as originally cut by the cutting stylus. This action causes all kinds of distortions to be produced and parts of the original sound are not reproduced. As techniques of record and tape making improve, wider frequency ranges and greater transients are accurately recorded in the program material mediums. These improvements in program material mediums create a requirement all the way through the hi-fi system for improved pickup, amplification, and reproducing techniques, and for the equipment to keep pace. However, the essence of a quality system is the ability of its pickup device to provide precise dynamic coupling to the medium and accurately transduce the program material as it was recorded.

A number of factors must be just right to ensure proper *tracking*, that is, to ensure that the needle stays in the groove and does not cause excessive wear. Good tracking is achieved when the needle follows both sides of the groove with equal pressure on each side at all times. Tracking is much more important in stereo playback than in monophonic playback because the needle will wear one side of the groove more than the other if it is forced against the side of the groove. In stereo, wear on one side of the groove reduces the amplitude of one stereo channel with respect to the other stereo channel; therefore, if the wear is appreciable, a stereo record would become useless much sooner than a monophonic record under the same conditions.

This wear on the sides of the grooves is another reason why the needle force of a stereo pickup must be less than that of a monophonic pickup. But, reduction of needle force means more difficulty in maintaining tracking because the needle will sometimes have a tendency to "skate" across the record. Fortunately, the reduction of the needle-tip radius to 0.5 to 0.7 mil to overcome pinch effect as explained before also improves tracking.

It can be seen from the preceding discussion that the pickup arm for stereo cannot be too carefully adjusted. It must have an absolute minimum of resistance to lateral motion so that it tracks smoothly when balanced for the recommended needle force.

Pickup arm and needle adjustment is easier with a "single-play" turntable than with a record changer because of the elimination of complex change mechanism. With a changer, the needle must rest heavily enough in the grooves to allow the pickup arm to trigger the change mechanism. This takes a lot more force than necessary to move the arm across the record. In addition, because the records are stacked on the turntable

during operation of a changer, the needle has a different angle for each record. Thus distortion is introduced in the reproduction and record wear is much greater than with single-play mechanisms. This does not mean that changers cannot give top-quality reproduction, but, everything else being equal, the single-play setup is simpler to adjust and operate.

Turntables

At first thought it would seem that the requirements for turntables and turntable drives would be the same for stereo as for monophonic systems. However, this is not so. Stereo turntable and drive requirements are more exacting because of the greater inherent sensitivity of stereo pickups to vertical vibration and the susceptibility (in the case of magnetic pickups) to hum pickup from the motor.

Rumble

Rumble is the effect produced in the sound output by low-frequency signals generated by vibrations in the motor and drive systems. These signal components usually have frequencies of from 30 to 60 Hz. Therefore, if the system as a whole does not have extended low-frequency response, rumble is not such a great problem. Thus, if your speaker system cuts off at about 100 Hz, you can stop worrying about the fine points of rumble production. However, any high-fidelity system worthy of the name reproduces signal components down to 50 Hz or below, and turntable rumble is an important factor.

A stereo pickup is more sensitive to rumble than a monophonic pickup because it is sensitive to vertical vibrations, and the vertical vibrations are usually two or three times as strong as the horizontal vibrations in a turntable drive assembly. Therefore, special measures must be taken for stereo-system turntables to minimize vibration and its effects. Otherwise, severe rumble is present in the output signal of the pickup. Although there is no standard, rumble is usually measured with respect to a fairly strong 1000-Hz signal obtained from a standard test record. Low-level passages of music may be as much as 40 db below the test record output, so rumble should be at least 45 db down. At −60 db it is usually completely inaudible, so this is a desirable objective.

In record players designed for stereo, rumble should be minimized by damping in the drive system and by the use of motors which deliver power as smoothly as possible.

Hum Pickup

The reason for the greater susceptibility of some stereo pickups to hum is the fact that there are two channels instead of one. The hum currents in the two coils of a magnetic-type stereo pickup combine in the output during operation. When a monophonic record is being played, the hum currents can be made to cancel by connecting the coils in parallel.

The source of most hum pickup is the turntable drive motor. Induction and synchronous motors have coils carrying alternating current from the power line, and thus they radiate hum. In general, the higher the motor power, the more hum is radiated. It would seem that the motor power should be made as low as possible. However, the lower the motor power, the more difficult it is to get good speed regulation and the more likelihood of wow being introduced. It is general practice to make the turntable relatively heavy to provide the inertia for good speed regulation. But more motor power is required to drive the heavier turntable; hence more hum is produced. Record changers require more drive power than single-play systems, so changers tend to be more subject to hum pickup.

TAPE RECORDING

Stereo recording and playback from tape may soon be as common as stereo records, because of technical improvements, price reduction, and greater convenience in use. Many audiophiles believe that tape offers the greatest opportunity for the ultimate in high-fidelity stereo reproduction.

Although stereo tapes were available long before stereo records came into general use, certain basic problems have slowed them from enjoying a wide distribution. The first of these problems is price. At present, stereo tapes cost more than comparable discs and they must be played on a machine of relatively high quality, which also costs more than record players providing comparable reproduction. The second basic problem, now overcome, was the inconvenience of handling. A disc can easily be slipped onto a turntable, the pickup placed on it, and music obtained with little delay. When a tape is to be played, ordinary rolls must be carefully keyed into position, and the tape threaded through the guides and past the heads of the machine to the pickup reel. Magazine-type tapes are now available which eliminate such time-consuming operations, but they are still expensive.

In spite of the price disadvantage, tape does have many advantages. It is practically immune to wear and deterioration of quality with playing. A tape can be played thousands of times without noticeable degradation, providing reasonable care is used in its storage. The transfer from storage on the tape to an electrical signal in the amplifier is accomplished without the necessity of mechanical parts vibrating at sound frequencies as they do in phonograph pickups. Thus wear and resonance effects are minimized.

The difference between a stereo tape system and a monophonic tape system is that the stereo system simultaneously uses two tracks, recorded on the tape as shown in Fig. 3-11. The recording and playback heads each have two units, one for each track. The tracks and the gaps in the head are separated by a guard band so that there is no interaction between the two signals.
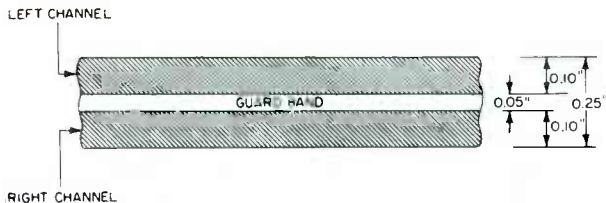
Fig. 3-11. Dimensions of tracks and guard band for two-track stereo tape.

The arrangement in Fig. 3-11 allows two tracks to be simultaneously recorded or played back. Four-track tape permits additional playing time for the same length of tape. Using four-track tape, the tape can be played in one direction first and then turned over and played in the other direction the same as monophonic tape. The four tracks are recorded on the tape as illustrated in Fig. 3-12. The guard bands do not need to be as large as for the two-track tape in Fig. 3-11 because in Fig. 3-12 only alternate tracks are used during tape travel in a given direction.

All stereo tape machines now use "in-line" heads, where the gaps for the two channels are exactly centered along the same vertical line. However, there were some machines made for staggered heads; in these the tape passed over first one head, then the other. Tape recorded for this arrangement is obviously not playable on the in-line–head machine. It was at first thought that staggering was necessary to prevent crosstalk between the channels, but the desired isolation is now obtained by proper spacing between the in-line gaps.

One of the advantages of tape as a stereo medium is the fact that isolation between channels is inherently much better than for phonograph pickups. Even with the small spacing between the tracks of a two-track system 40 db of separation is normally obtained. The separation is obtained much easier in the four-track system because head gaps have more physical separation.

STEREO BROADCASTING AND RECEPTION

We have already considered two sources of stereo high-fidelity music for the home listener: disc recordings and tape recordings. A third source is the signal of a radio broadcast station. Instead of a record
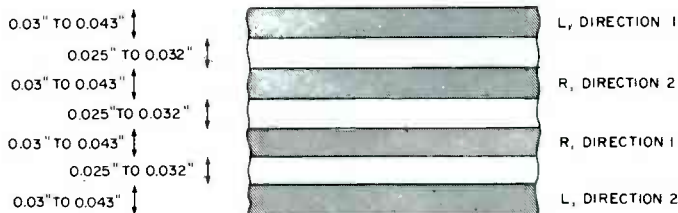
Fig. 3-12. Dimensions of tracks and guard bands for four-track stereo tape.

41

player or tape machine, a tuner is used. Tuners for stereo are shown in Chapter 4. They are similar to single-channel tuners except for the technique of multiplexing, which will be described in this chapter.

Two-Station Stereo Broadcasts

The first method used for stereo broadcasting and reception is illustrated in Fig. 3-13. Two complete transmitters were used, one sending out the left (L) stereo signal, the other the right (R) stereo signal of the same program. All combinations of a-m broadcast, f-m broadcast, and television sound-channel transmitters have been tried. Most popular were the "f-m/a-m" (one f-m station and one a-m station) and the "f-m/f-m" (two f-m stations) methods.

The two-transmitter method of transmitting stereo was convenient because it utilized existing transmitting and receiving equipment with no circuit changes. However, it did have the following disadvantages:
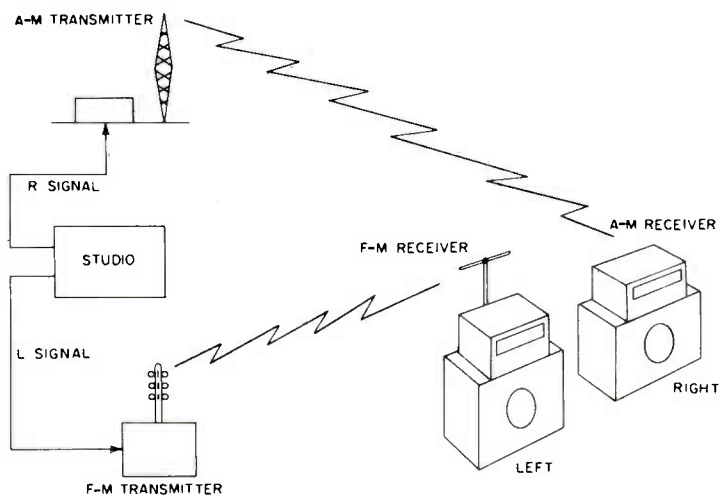
Fig. 3-13. Two-transmitter method for radio transmission of stereo signals.

1. It wasted frequency spectrum space. Two station channels had to be used for each program.
2. Differences in propagation characteristics of the waves radiated by the two transmitters led to variation in signal amplitude and quality within the separate channels. This was particularly true of f-m/a-m combinations, where the carrier frequencies are so widely separated.
3. In some cases, especially with f-m/f-m, the duplication of complex receiving equipment represented an excessive expense. Ninety percent of the programs were nonstereo, requiring only one re-

ceiver; but to receive stereo, two receivers were needed. Also, a number of a-m/f-m tuners could not receive both bands simultaneously, and so they had to be supplemented with additional receiving equipment.

Stereo Multiplex

The disadvantages of the two-station method made it imperative that a method of transmitting both stereo signals on one carrier be devised. The result is the method called *stereo multiplex*. "Multiplex" means "a method or arrangement for sending two or more messages simultaneously on one carrier or circuit." This is exactly what stereo multiplex does— it transmits both channels on the same f-m carrier.

The standard f-m broadcast system is based on an audio frequency response of approximately 0 Hz to 15 kHz. There is nothing about the basic modulating system to prevent extension of this audio-frequency range to 75 kHz (further extension would require more bandwidth than is currently allotted). Therefore, if at some point above the limit of human hearing (but still within the 75-kHz bandwidth) another signal is added, it will be amplified and detected by the regular receiver circuits. However, being above the range of hearing (supersonic), it will not interfere with the regular audio signal. Thus, the carrier is modulated by the regular audio signal plus the supersonic signal.

This supersonic modulation component is known as the *subcarrier*. It, in turn, is modulated by another audio signal. As long as the bandwidth of the subcarrier is not allowed to extend downward into the range of the regular carrier, no interference will occur. This complete modulated subcarrier signal is part of the composite signal which modulates the main carrier. One of the stereo signals is transmitted as modulation of the main carrier, and the other signal as modulation of the subcarrier.

In receiving, two demodulating circuits are needed. First the main carrier is demodulated, to get the first stereo signal and the modulated subcarrier. Then the modulated subcarrier is separated and demodulated, to get the second stereo signal.

Compatibility—There will always be some listeners who don't care about being equipped for stereo reception. Also, for portable use, and where cost is to be minimized, a conventional single-channel receiver is called for. For this reason it is important that the stereo system be compatible. By "compatible" we mean that while stereo multiplex transmission is in progress a listener with a conventional receiver (one not equipped for stereo) should be able to receive the transmission as a full monophonic signal.

If one channel of the multiplex system is used for the L signal and the other for the R signal, compatibility is not provided. The owner of the conventional receiver hears only the modulation of the main carrier, which in this case would be just the L signal; the subcarrier frequency

43

is above audibility, so neither the subcarrier nor its modulation is used. For a truly compatible system, the nonstereo listener should hear the combination of both L and R signals.

To meet this compatibility requirement, the main carrier can be modulated with a full monophonic signal (L plus R), and the subcarrier channel can be modulated by the difference between the two signals (L − R). The L + R modulation signal provides the nonstereo listener with his full monophonic reception; his receiver does not respond to and thus ignores the subcarrier.

For stereo reception, the L and R signals are recovered by demodulating the subcarrier signal and adding or subtracting the difference signal from the main carrier (L + R) signals.

Electrical addition of the signals can be accomplished simply by applying both signals across a common impedance. Subtraction can be accomplished by first inverting one of the signals (changing its phase by 180 degrees), then adding it to the other signal.

STEREO *VERSUS* HIGH FIDELITY?

One point should be emphasized: Stereo reproduction is part of high fidelity, not an "additional feature." The impression is sometimes wrongly given that "hi-fi" and stereo are two separate stages of development of audio reproduction. This is a fallacy, unfortunately sometimes encouraged by a statement that some equipment features "high fidelity and stereo."

Stereo is a "fine point" in high fidelity; that is, if you do not have all the good qualities of high-fidelity monophonic systems to start with, other forms of distortion will prevent appreciation of the benefits of stereo effect. If the reproduction of your system is clean and clear of harmonic intermodulation and transient distortion, it is likely that improvements in spatial sound effects through stereo will be appreciated.

4

Program Source Equipment

Any hi-fi installation will require one or more pieces of equipment to provide program material. Those commonly used may be described as follows:

1. *Record Players:* There are two popular types. The most popular is the automatic record changer; for optimum results, the preferred type is the professional but plain turntable, manually fed. Both units are available in models that will play records of all sizes and speeds.

2. *Tuners:* These are a-m and f-m radio receivers especially designed to noiselessly receive, amplify, and rectify signals of wide bandwidth to provide a hi-fi audio signal from lower broadcast-band frequencies and vhf f-m frequencies.

3. *Tape:* Playback equipment can provide several hours of recorded program material as does the record player. Its primary advantage is that one may easily record, store, and play back one's own recorded material of music, entertainment, or information for any purpose.

4. *Television:* The audio components of a television signal may be picked up by a special television audio tuner, and either may be picked off the television receiver or may be fed through a system to provide hi-fi audio to accompany the television picture program.

5. *Microphones:* These are electromechanical transducers designed to convert sound waves into electrical program material pulses used occasionally in hi-fi to record, entertain, or announce, or for other special purposes.

RECORD PLAYERS

All record players have a motor, turntable, pickup arm (tone arm), pickup cartridge, and needle. The professional high-quality reproduction systems generally have a manually fed turntable. Record changers with automatic devices to change or feed records at the end of play of a former record, providing continuous operation for long periods, are more popular. We will cover the component parts of both the manually fed and automatic types, starting with the needle.

45

To minimize wear and secure the maximum useful life from a recording, a properly shaped needle must be used. A properly formed reproducing needle will also minimize background noise.

Fig. 4-1 illustrates a group of playback needles seated in the grooves of a recording. The needle at (1) is of a theoretically ideal shape. At (2) and (3) the needles are too sharp and will gouge the bottom of the groove. The one at (4) is too blunt and will cause excessive wear on the walls of the groove, resulting in their eventual breakdown. At (5) is a needle of satisfactory shape. However, the groove that the playback equipment must track is cut by a stylus that is shaped like a diamond. The difference in shape between the cutting stylus and playback needle causes trackability distortion and pinch distortion.
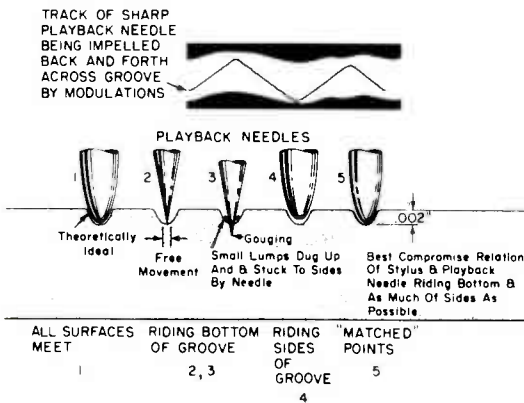
Fig. 4-1. Properly and improperly shaped playback needles.

When a cutting stylus cuts a sine-wave groove, it moves back and forth in the same plane and the points of contact in the groove are in a line parallel to the radial from the center of the record being cut. However, when a round needle follows this groove, the only time the points of contact are in a line parallel to the radial from the center of the record is at the peaks of the sine-wave excursion. (See Fig. 4-2A.) At the midpoint or steepest-slope point of the sine-wave groove, the points of contact of the cutting stylus are still parallel to the radial, but the points of contact of a round stylus are inclined nearly 45 degrees to the radial passing through the center of the record. The resulting tracking distortion is comparatively large.

This condition is mostly overcome by the use of an elliptically shaped needle. The needle is mounted in a position so that the long axis of the ellipse is nearly parallel to a radial passing through the center of the

record. Thus, when the elliptically shaped needle tracks the groove, its points of contact nearly coincide with the original points of cutting-stylus contact at any given position. (See Fig. 4-2B.) The cutting stylus cuts at a 0-degree operating angle with the radius of the groove circle.

(A) Cutting stylus, round needle, and elliptical needle have same points of contact at waveform peak in record groove.
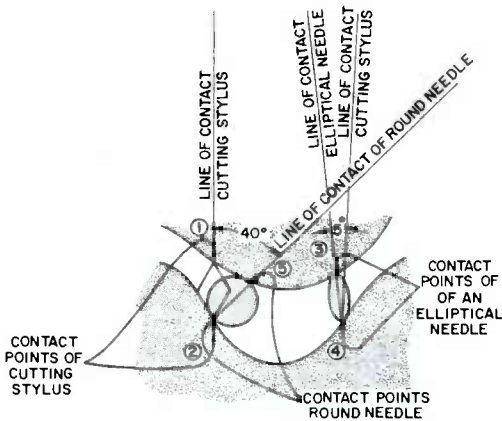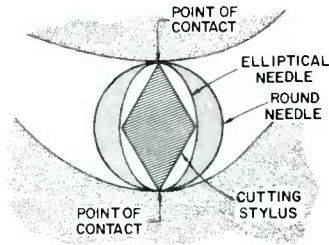
(B) Cutting stylus and elliptical needle have similar contact conditions, but round needle is in contact at a 40° angle.
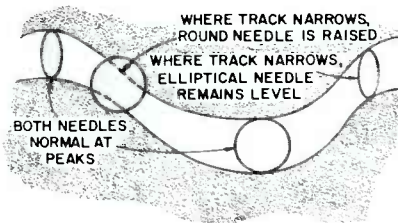
(C) Pinch-effect distortion of round and elliptical needles.

Fig. 4-2. Needle shape and performance.

(See points 1 and 2 of Fig. 4-2B.) The round needle may follow, making contact on points (2 and 5 of Fig. 4-2B) of a line making a 40-degree angle with the line of contact of the cutting stylus, whereas the el-

47

liptical needle will follow with only a 5-degree difference angle (points 3 and 4 in Fig. 4-2B) with a proportionate increase in fidelity or reduction of tracking distortion.

The elliptical needle has another advantage in that it also corrects for pinch-effect distortion caused by the changing width of the groove as the angle of groove sides and line of excursion of the cutting stylus change. As the groove width narrows, the round needle rises because it is pinched upward. Since the elliptical needle rides almost the same points of contact as the constant-width cutting stylus, it maintains a fairly constant vertical position (Fig. 4-2C).

Regular records usually contain enough abrasive material to quickly wear a metal needle to the proper shape. Instantaneous or master recordings do not contain abrasive material, and because of this it is particularly important that a needle of the correct shape be used with these types of recordings.

OSMIUM   SAPPHIRE   DIAMOND
10 hours   50 hours   400 hours

Fig. 4-3. Needle wearing qualities.

Reproducing needles are made of a number of materials. Plain, hardened steel and osmium needles are good for only a limited number of plays and therefore are seldom used in hi-fi playback. Those made of sapphire are the most common. Sapphire needles have a longer life than steel needles. A properly used sapphire needle will give hundreds of plays. Diamonds are also used in reproducing needles and are considered to be the best. Diamond needles will give satisfactory performance for thousands of playings, and when properly shaped and polished they are superior to all other types. Their only disadvantage is that they are very expensive and delicate and may be fractured by a slight impact. A chipped diamond or sapphire needle will quickly ruin any type of recording. The comparative life of diamond needles against other materials varies according to the manufacturer and the user. A life expectancy for a diamond needle many times longer than that of competitive materials is not unusual. Wearing qualities of different needles are shown in Fig. 4-3. Prices on diamond needles are dropping to reasonable levels, making them best buys for top hi-fi playback.

The needle-tip size for monophonic (single-channel) records may be over 0.001 inch (1 mil), while needle tips for stereo records are either 0.5 or 0.7 mil.

The construction and the electrical characteristics of the various available pickups vary greatly. They can be classified into five distinct groups. These are: crystal, ceramic, magnetic, dynamic, and capacitance.

Units of varying quality can be obtained in each group. The tastes and desires of the user and the use to which the pickup is to be put may govern its choice. With systems designed to reproduce the voice for reference purposes only, an inexpensive pickup with a comparatively narrow frequency range is suitable. Where the highest quality of reproduction is required, a special high damped pickup having a frequency characteristic flat to beyond 15,000 Hz is necessary.

Crystal and Ceramic Pickups—Crystal and ceramic pickups are the lowest-cost units mentioned in the foregoing. They are simple in design and construction, have fair frequency response characteristics, and hi-fi types have low distortion content.

Some crystalline substances possess the ability to produce a charge under certain conditions. When they are stressed mechanically, a charge is produced on their surfaces. If a voltage is applied to the surfaces of a crystal with piezoelectric properties, a mechanical deformation of the crystal will take place.

The *piezoelectric crystal* acts as a generator and converts mechanical motion into an electrical charge. Crystal microphones and phonograph pickups can be thought of as piezoelectric generators. A crystal is also similar to a motor. When a potential is applied to a crystal, it moves. It converts electrical energy into mechanical motion. Crystal headphones and record cutters are piezoelectric motors.

A piezoelectric crystal as used in microphones and pickups is a formation of crystalline Rochelle salt. The Rochelle salt crystal possesses the piezoelectric property to a high degree. It is approximately 100 times as active as a regular quartz crystal.

Rochelle salt crystals are formed in large bars. These bars are cut into slabs or plates for use in the manufacture of crystal elements. The two commonly used crystal plates are usually referred to as *expander* and *shear plates*, as shown in Fig. 4-4. The crystal is either a shear or expander plate, depending on the way it is cut from the bar.

A crystal plate is said to have three axes. The latter are the electrical (AA), the mechanical (BB), and the optical (CC) axes. An expander plate is cut at a 45-degree angle to the optical and mechanical axes of a crystal bar. A shear plate is cut with its edges parallel to the mechanical and optical axes of the crystal bar.

When a potential is applied to the two large faces of each plate, mechanical motion is developed at an angle of 45 degrees from that of the mechanical and optical axes. (When a force is applied, an electrical potential will be developed at the faces.) Therefore, the expander plate

will increase its length and, at the same time, decrease its width. If the polarity of the faces of the crystal is changed, the crystal will decrease its length and increase its width.

The same action takes place when a potential is applied to a shear plate, except that expansions and contractions occur along the diagonals of the plate instead of parallel to the edges, as in the case of the expander plate. When mechanical pressure is applied, a potential voltage is produced.

EXPANDER PLATE USED IN BENDER ELEMENTS

SHEAR PLATE USED IN TWISTER ELEMENTS

AA - ELECTRICAL AXIS   BB - MECHANICAL AXIS   CC - OPTICAL AXIS
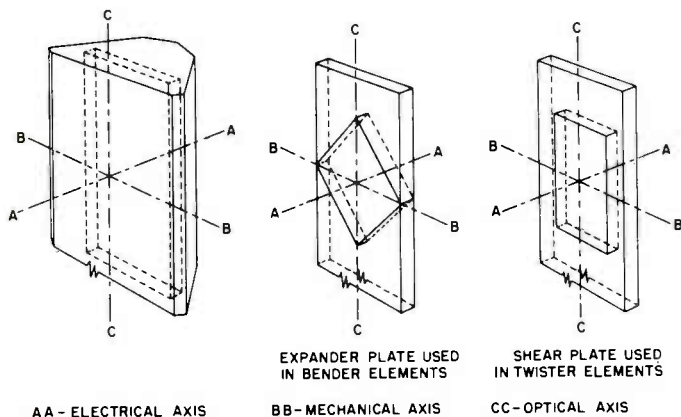
Fig. 4-4. Types of crystal plates.

In order to form a crystal element for use in a crystal cartridge or other device, a number of crystal plates are cemented together. This makes possible more effective utilization of the properties of the crystal.

An element which consists of a number of expander plates cemented together is referred to as a *bender* element, while an element formed from a number of shear plates is known as a *twister* element. The names "bender" and "twister" refer to the action which takes place when an electrical potential is applied to the element.

The multiplate crystal has a number of important advantages over a crystal employing a single plate. It greatly decreases the undesirable effects of saturation and hysteresis and reduces the effects of temperature on the impedance and sensitivity of the unit. Fig. 4-5 shows the construction of a bender and a twister crystal element. The faces of each crystal plate are milled smooth, and foil or graphite electrodes are applied. Leads are connected to the electrodes, and after they have been properly oriented, the plates are bonded together with cement.

The completed crystal element is coated with a special moistureproof material to protect it against deterioration under very dry or damp conditions. The crystal element is mounted in a nonconducting case and held at one end by a metal clamp. The other end of the crystal is free,

permitting it to move torsionally. A bearing and chuck are mounted on the free end of the crystal. The bearing usually consists of rubber or a similar synthetic material. The chuck usually consists of a light metal such as aluminum. To restrain the crystal from vibrating at more than one mode, it is customary to restrain it slightly along its axes of motion. This is accomplished by cementing a strip of damping material along the length of the element. This strip also gives some damping effect in the other modes and helps to reduce the amplitude of the resonant peak of the crystal. Because the crystal is very stiff, its resonant frequency is normally not in the frequency range to be recorded.

(A) Bender element.   (B) Twister element.
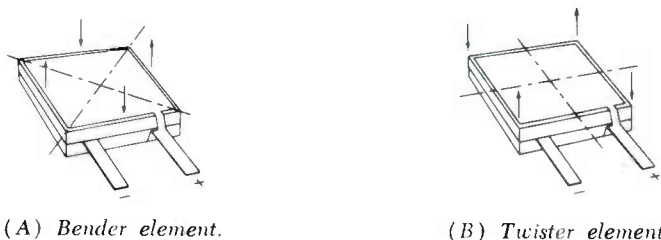
Fig. 4-5. Construction of crystal elements.

As a circuit component, a crystal cartridge acts in the same way as does a capacitor and can be considered as such. Extremes of temperature drastically affect the operation of a crystal. The maximum sensitivity of a crystal is usually at about 75 degrees. As the temperature rises above or falls below 75 degrees, the sensitivity of the crystal falls off slowly. Cartridges using barium titanate have much improved temperature characteristics.

At temperatures in the neighborhood of 130 degrees, Rochelle salt crystals may permanently lose their piezoelectric properties. At a rule, temperatures slightly below this, that is, from 110 degrees to 120 degrees will not injure a crystal.

Ceramic pickups have overcome this difficulty and are now commonly used in place of crystals. A ceramic pickup has the advantage that it is quite stiff, and, because of this, variations in mechanical load do not greatly affect the performance. Ceramic pickups have operational disadvantages in that they have some distortions and resonances which are almost always present. Also most have roll-off of the high frequencies. Types such as the Columbia cartridge shown in Fig. 4-6 have overcome these faults to a major degree.

Magnetic Pickups—The magnetic pickup is a current-operated device. The construction of magnetic pickups varies greatly. Essentially they consist of a coil and magnet and another magnet to which the needle is attached. The movable magnet to which the needle is affixed is damped.

This is accomplished in a number of ways, depending on the construction of the particular cartridge.

The coil is connected directly to the input of the preamplifier. Current through it varies with the change in the density of the flux about the fixed magnet. This variation in flux is produced by variation in the force exerted on the movable magnet, for each change in position. Since the reproducing needle is connected to this magnet, it produces output in proportion to the movement.
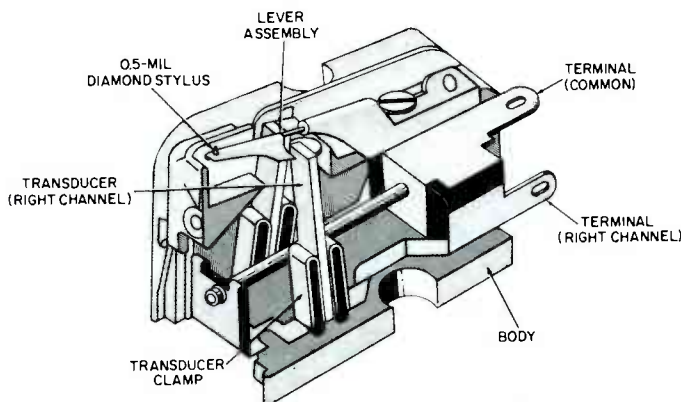
Fig. 4-6. Construction of the Columbia *Professional 55* ceramic stereo cartridge.

The frequency range of a magnetic pickup is greater than that of the crystal type. The finest magnetic pickups have a frequency range of from 50 to over 15,000 Hz. The distortion content of a fine magnetic pickup may be as low as 0.1 percent at 400 Hz, and 1 percent to 4 percent at high frequencies. A typical hi-fi reluctance-type magnetic cartridge is shown in Fig. 4-7.

Dynamic Cartridges—The dynamic cartridge, as the name implies, is of the moving-coil type. The unit consists of a movable coil, to which the needle is mounted, and a permanent magnet. The coil is connected to a low-impedance input of a preamplifier. When the needle moves the coil, causing a magnetic field to be set up around it, this field interacts with the field of the permanent magnet; and as a result, an output voltage is produced. The results obtained with a dynamic pickup are as good or better than that obtained with magnetic pickups. A dynamic cartridge is illustrated in Fig. 4-8. The response of this cartridge extends beyond 20,000 Hz, and it gives almost as much output as a good reluctance pickup.

Stereo Cartridges—In stereophonic application, one of the most critical components is the pickup cartridge. (See Fig. 4-9.) The basic prin-

ciples of stereo cartridge operation were covered in Chapter 3. Now we shall consider those characteristics important in selecting a cartridge and some examples of cartridges.

The weight of a cartridge and pickup arm assembly is transmitted to the record as needle force. The maximum allowable needle force without excessive record wear is related to the size of the needle tip. This is because wear is dependent on pressure, which is force per unit area; thus the smaller the needle tip, the less the needle force must be to prevent wear. If needle-tip size were made too small, the necessary reduction

Courtesy North American Philips Co., Inc.
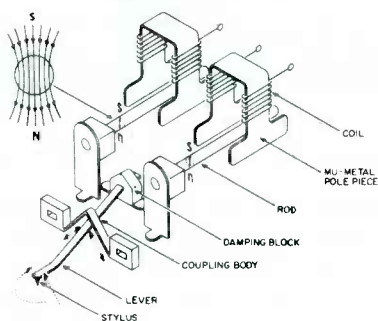
(A) *Pickup.* (B) *Construction.*

Fig. 4-7. A moving-magnet type of stereo pickup.

in needle force would result in loss of tracking; that is, the needle would skate over the record instead of staying in the groove. Thus the minimum needle-tip size is kept to 0.5 mil, and the range of needle force is roughly from ½ to 6 grams.

As is the case with all high-fidelity system elements, we are interested in the frequency response of a cartridge. Response is usually stated in terms of frequency range and the deviation, in db, of the response over that range. We are naturally interested in the widest possible frequency range, but it is doubtful if response below 20 Hz makes much difference. However, the high limit of the range should extend to near 20,000 Hz or beyond to take full advantage of the better recordings.

The output voltage of a cartridge is important, not only because it is used to calculate how much amplification is needed, but also because the higher the output, the more chance there is of having a good signal-to-noise ratio. However, sometimes it is desirable to sacrifice some signal strength to ensure minimum distortion, best frequency response, and minimum record wear. Output voltage does not establish criteria unless the level at which the needle is driven is also specified. Standard records are used to provide the drive for output voltage tests. Some cartridge

53
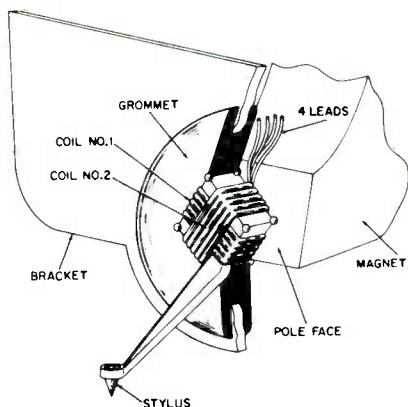
(A) *Pickup.*   (B) *Construction.*

Fig. 4-8. The Grado moving-coil type (dynamic) of stereo pickup.

manufacturers state output for a 5-cm/sec needle velocity, and others for a 10-cm/sec needle velocity. Naturally, the output should be higher for the greater velocity. In Fig. 4-10 the needle velocity is shown in relation to needle force and frequency. For total outputs below 10 millivolts, a separate preamplifier may be necessary, depending on the gain of the system used.

Channel separation is the indication of how well the left signal is kept out of the right channel, and the right signal out of the left channel. Good separation is necessary for good stereo effect, since the difference between the two signals is what produces the spatial effect. If a cartridge were not carefully designed with separations in mind, one channel would affect the other and the outputs of the two channels would tend to become the same. Tests have shown that a minimum of 15-db separation should be maintained.

To operate properly, a cartridge must be connected to the proper amplifier input impedance. Generally speaking, ceramic and crystal pickups must work into a relatively high resistance (15 kilohms to several megohms) compared with magnetic cartridges (5 to 100 kilohms). Each cartridge manufacturer specifies the proper load characteristics for his models.

Compliance is a measure of how easy it is to move the needle in the directions it must be driven during playing. It is ordinarily measured in millionths of a centimeter per dyne (10^{-6} cm/dyne), i.e., the distance, in millionths of a centimeter, that the needle can be pushed by a force of 1 dyne in that direction. For stereo cartridges it is important that they not only have a high compliance, but that this high compliance apply in all directions of motion. This is why many stereo cartridge manufacturers specify both lateral and vertical compliance. It will be shown
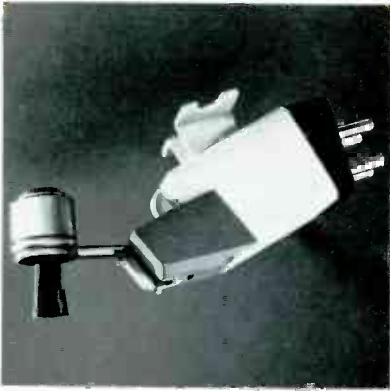
54

(A) *Pickering* Dustamatic.

(B) *Ortofon SPE/T*.

(C) *Shure V-15 Type II*.

(D) *Empire 888D*.
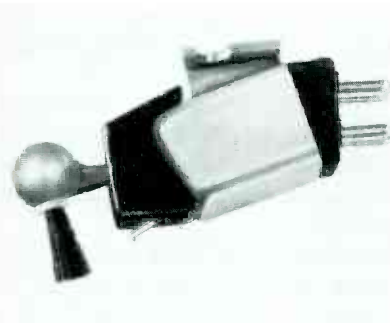
(E) *Stanton* Longhair.

(F) *Dusting action of brush*.

Fig. 4-9. Typical stereo cartridge.
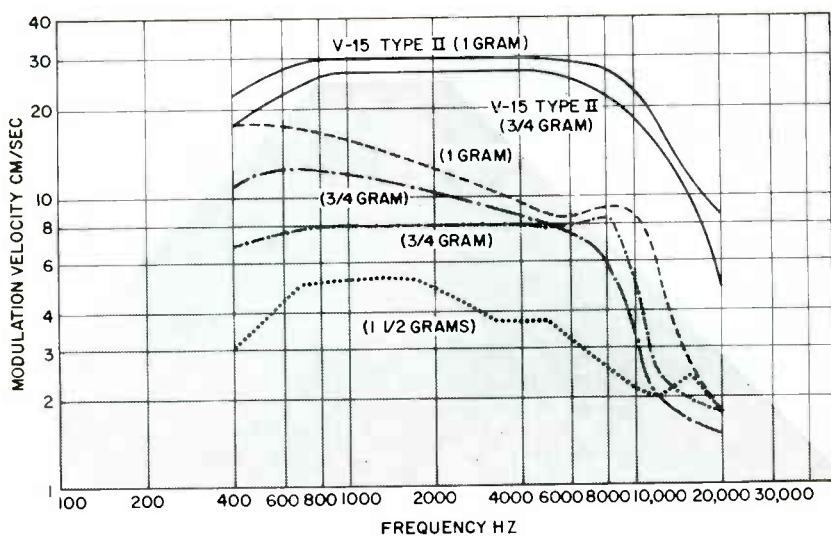
Fig. 4-10. Performance and needle force of Shure high-performance cartridges compared with earlier models.

that it is the low vertical compliance of most monophonic pickup cartridges that makes it necessary that stereo records never be played with monophonic pickups.

Another characteristic that is sometimes specified by cartridge manufacturers is channel balance. It is given as the deviation from balance, that is, the difference in db between the outputs of the two channels for the same drive. Actually this characteristic is not very critical because just about all stereo systems include a channel-balance adjustment to compensate for reasonable differences in output. However, a relatively large unbalance, such as more than 3 db, is an indication of a defective cartridge. The optimum arrangement is a system that is naturally balanced throughout and balance deviation of 1 db or less should be considered good.

Application—The pickup cartridge should be mounted in a suitably designed arm. The needle pressure should be left at the lowest value consistent with the design of the cartridge used. For optimum results, the arm should have an offset head to minimize tracking error, and side-to-side motion should be as free as possible.

Vertical sensitivity is an important characteristic of any cartridge. This sensitivity is mostly related to the vertical movement sensitivity of the cartridge and stylus. Low vertical sensitivity in a properly designed monophonic unit will reduce hum and other table noises without affecting the pickup quality of the lateral groove modulation from the record. However, as explained previously, vertical sensitivity is important for

56

proper reproduction of stereo records. Hum may be picked up by induction in magnetic- and dynamic-type pickups. Hum sensitivity is another important consideration in pickup selection.

In general, reluctance-type magnetic pickups are most widely used. Dynamic pickups reproduce extremely well, but they are less sensitive. Ceramic units are best for medium-quality work requiring high output from the pickup unit. These units will drive a basic amplifier directly without need for a preamplifier. In comparison with ceramic units, the response of the better reluctance types is more nearly flat, and they have the least distortions and resonances. While hi-fi crystal pickups are entirely suitable for most reproduction, the best crystal units usually have a number of small resonances and distortions. Capacitance units are suited to special applications requiring extremely wide frequency response. However, much of the operating quality of any cartridge depends on the pickup arm and the needle pressure.

The Pickup Arm (Tone Arm)

The pickup arm or tone arm is the carriage for the cartridge and needle. Its design and principle of operation are very important to high-fidelity reproduction. To permit the needle to respond without distortion to record groove deflections (tracking ability) the tone arm should have free movement in all directions. The force of weight of the arm applied vertically to the needle should be adjustable to provide for the needs of various kinds of cartridges. Optimum pressure will generally be found between ½ gram and 6 grams, with an average setting of 1 recommended for most cases. Several companies make a device available for measuring this weight near the point of application. The weight must be within the range required for a certain needle-cartridge combination. It must be
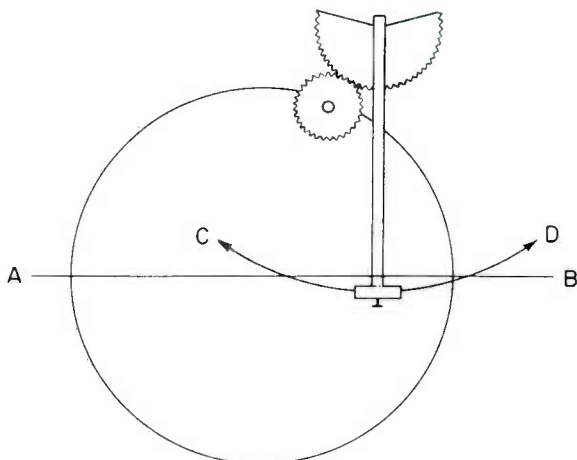
Fig. 4-11. Tracking error of tone arm.
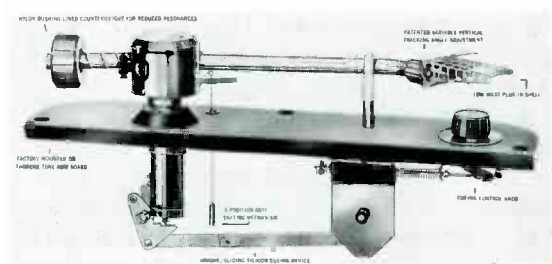
(A) *Thorens TP14.*

(B) *Rek-o-Kut* Micropoise.
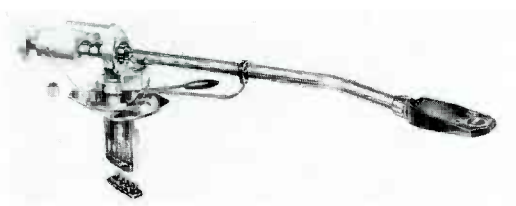
(C) *Shure SME Series 2.*
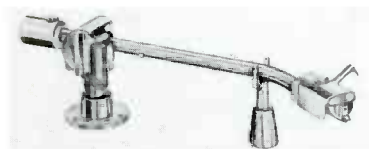
(D) *Empire 980.*

Fig. 4-12. Typical pickup arms designed for stereo.

heavy enough to keep it in the groove to follow the modulation, and light enough to allow free movement and low wear on the needle and the record. The pressure on the needle therefore should be adjustable and the adjustment made preferably with a calibrated scale.

The pickup arm should track as nearly as possible to the original line of cut; otherwise, a tracking error will develop and cause distortion. See Fig. 4-11. The record is cut along line *AB*, and the usual pickup arm operates in an arc, as shown by line *CD*. Placement of the mounting for the pickup arm can vary the tracking to a large degree. When the play-back is made with the arc of contact of needle to record too far off, the needle follows the modulation with a modified movement because the round needle tip is moving in a different direction and amount per mod-ulating element from the direction and movement of the original cutting stylus. This causes tracking distortion in the signal generated by the cartridge.

The arm is best designed to ride on precision aligned bearings for the greatest reduction of resistance to movement and for the lowest inertia. Some units use ball bearings.

Pickup arms that provide for different kinds of cartridges enable one to try and use different cartridges.

For stereo application the pickup arm (tone arm) and turntable and drive arrangement must meet special requirements which are more rigid than those sufficient for monophonic reproduction. Pickup arms designed particularly for stereo are illustrated in Fig. 4-12. Let's consider some of the features to be considered when one is selecting a pickup arm for stereo.

One of the characteristics most important in stereo pickup arms is the tracking error just described. The angular difference between the direc-tion of motion of the groove the needle is in and a line between the pickup arm lateral pivot and the needle will generate unbalance in the two outputs, as well as tracking distortion. In other words, instead of the needle pulling "in line" with the pickup, at some parts of the motion across the record it pulls in a different direction, producing proportional unbalance. The lateral component of force caused by tracking error in a stereo record groove produces more wear on one side of the record groove than on the other. Hence, one stereo component signal will be reduced in relation to the other. Also, stereo cartridges operate at much lower needle force and cannot tolerate as much lateral tracking error force without jumping out of the groove.

Tracking error is minimized in pickup arms by orienting the pivot point and needle point locations with respect to the record and making the angular offset of the pickup with respect to the longitudinal axis of the arm optimum, as shown in Fig. 4-11. By proper adjustment of both of these variables, arm manufacturers have been able to reduce tracking error to a fraction of a degree in some models.

A good stereo pickup arm should have a needle-force adjustment finer than that adequate for monophonic pickup arms. Stereo models require more precision, because of the lower forces and smaller tolerances to which they must be adjusted. The normal arrangement is for adjustment first for perfect balance (zero needle force), then adjustment of another control to the exact number of grams of force desired.

Sometimes pickup arm manufacturers state the *tracking force*. This is the force that must be exerted laterally against the needle to move it in across the surface of the disc. In arms for stereo, the tracking force seldom exceeds 2 grams, and in many cases it is below 1 gram. The tracking force of an arm must be less than that of the cartridge used with it, or the cartridge rating would be exceeded as the needle moved inward on the record.

Turntables

The construction of the turntable used has a great deal to do with the quality of the reproduction fidelity. Wabble, wow, and other turntable defects show up unfavorably when a record is played back.

Three standard turntable speeds are used. They are 78.26, 45, and 33⅓ revolutions per minute.

Fixed speed is a very important factor in playback, since variations of as little as 1 percent are detectable. Speed variations are caused by changes in the power source of the turntable and motor.

"Wabble" results when the turntable, spindle, and bearing assembly are not carefully aligned to be concentric. When wabble occurs, the surface of the record moves closer to and farther from the pickup head as the turntable revolves. This results in changes in the strain on the needle, modifying the modulation as it is picked up. In extreme cases, the stylus may completely leave the surface of the record at one or more points.

The motor used to drive the turntable is usually powered by alternating current and, as is universally the case, vibration at the power frequency is set up within the motor. If this vibration is transmitted to the turntable and pickup, it will result in hum modulation of the pickup. This is known as *rumble*. In good turntable assemblies, the motor is insulated from the turntable sufficiently to eliminate rumble. Rubber mats lying on top of the table and under the record will further reduce this effect.

To avoid the defects described, a carefully designed and constructed motor and drive system is necessary. Motors can be designed to give fairly constant speed, but it is not possible to design and build a motor inherently of constant speed without some degree of undesirable fluctuation. The fluctuation will cause distortion. Fluctuation should be reduced by the turntable drive system. This may be achieved by use of resilient frictional drivers or application of governors or other frictional drags to provide regulation of the speed. The inertia of a heavy turn-

table will also contribute to regulating the speed and reduction of the effect of motor fluctuation.

Frictional drives and drags must be carefully designed and applied so that the frictional force applied is constant, or the table speed will vary through each 360-degree revolution. This causes the effect known as "wow."

Fig. 4-13 shows a number of driving methods commonly employed in record players and changers.

The gear-chain or direct motor drive shown at the top requires a very powerful motor free from vibration. If properly designed, this type of drive is very satisfactory. Because good units are very expensive, this drive system is rarely encountered.

The next arrangement shown in Fig. 4-13 shows the direct rim-drive method. This method provides a single speed. The turntable is driven by a rubber wheel attached to the motor shaft. The rubber wheel also serves to isolate the motor from the turntable, reducing the transmission of motor variations. Modern arrangements use one or more rubber idlers, further reducing transmission of motor fluctuations and vibrations.

One of the defects of this system arises from the deformation of the rubber wheels which results if the wheel is left in contact with the turntable rim when the equipment is not in use. The wheel at its point in contact with the turntable rim is flattened. In the better units, provisions are available for removing the pulley from contact with the turnable when the equipment is not in use.

In the next sketch shown in Fig. 4-13, a multiple-speed rubber-wheel drive system is shown. Two rubber wheels are provided, one giving a 78-rpm turntable speed, and the other a 33⅓-rpm turntable speed. The desired speed can usually be chosen by changing the position of a lever connected to the drive mechanisms.

The single- and dual-speed rubber-wheel drives are often constructed so that the rubber wheel mounted on the motor shaft drives an idler which, in turn, drives the turntable. This greatly simplifies the design of facilities for removing the wheel from contact with the turntable, and it permits better control of the pressures between the rubber wheel and the turntable rim. The possibility of slippage, which is a common fault of a direct rubber-wheel drive, is also greatly reduced. This system is also applied to three-speed players.

DIRECT MOTOR DRIVE

TWO-SPEED RUBBER IDLER-WHEEL DRIVE

RIM DRIVE
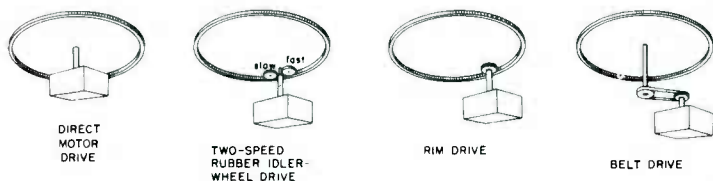
BELT DRIVE

Fig. 4-13. Turntable drive methods.

The remaining sketch of Fig. 4-13 shows a belt-drive arrangement often encountered in playback equipment. A rubberized composition belt is connected between pulleys on the motor shaft and turntable spindle. This system is very good with respect to low transmission of vibration.

Another drive method uses a conical driving member that is continuously variable. This provides for adjustment to any of the three popular speeds of 78, 45, and 33⅓ rpm and in addition allows for setting an exact speed around any one of the three standards to get a precise pitch from a particular record according to individual desire.

(A) A typical turntable, by Rek-o-Kut.   (B) Marantz SLT-12 turntable.

Fig. 4-14. High-fidelity turntables.

Record Changers Versus Turntables

Turntables have design and application advantages in that they are simple and have limited function compared with an automatic record changer. It is generally conceded that the highest-quality reproduction can be obtained from a good turntable. Record changers, on the other hand, can provide convenience with excellent quality. However, the additional mechanisms, devices, and controls necessary to automatically change the record may cause problems in maintenance of constant speed, rumble, and other defects. Each has its place—the turntable for flawless reproduction, and the record changer for hours of excellent-quality continuous program without attention. Fig. 4-14 shows two hi-fi turntables and Fig. 4-15 shows two typical record changers.

The cartridge, the pickup arm, and turntable requirements for stereo operation are so interrelated that many audiophiles prefer to buy the whole combination as a unit. Some typical combinations are illustrated in Fig. 4-16. As explained in Chapter 3, variation of needle height, greater tracking force, and stronger rumble effects make record changers somewhat inferior to manual players for stereo. Changers that are used for stereo are very similar to those employed for monophonic reproduction. However, they contain refinements to reduce rumble, tracking force, etc.

(A) *Garrard 50 Mk II, an example of a moderately priced automatic turntable.*

(B) *Garrard 60 Mk II, with adjustable-pressure pickup arm and cue control.*

Fig. 4-15. Typical record changers.

TUNERS

Turners are available to receive f-m or a-m signals separately or in one combined unit. Some tuners provide for the preamplifier-equalizer functions, as has been described. These functions are provided for because in some installations no other program material source other than the tuner is required. In other cases a separate preamplifier control is not as desirable as having all controls on the same panel with the tuning controls. The performance of such a tuner preamplification-compensation system should not be expected to be as good as a top-quality separate control preamplifier, but the design of this arrangement usually provides as much quality as is usable for f-m or a-m reception or other

(A) *A precision automatic turntable: the Garrard LAB 80 Mk II.*
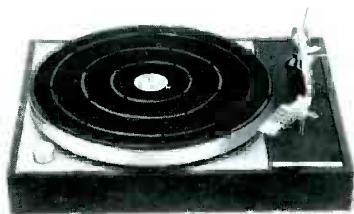
(B) *Thorens turntable with arm and cartridge.*

Fig. 4-16. Combination turntable, pickup arm, and cartridge assemblies for stereo.

63

average pickup. Amplitude-modulation reception is usually limited to a 10,000-Hz limit of frequency response or less because a-m broadcast stations are separated only by this amount and reception of two nearly equal signals only 10 kHz apart will produce a strong 10-kHz beat note
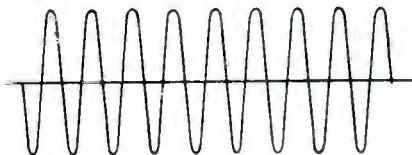
Fig. 4-17. An unmodulated radio-frequency wave.

which is unpleasant to sensitive ears if not filtered out. Frequency-modulation tuners can provide a greater range of reproduction and high fidelity over the entire audible range, if all elements of the systems involved are designed, arranged, and operated to achieve maximum performance.

A-M Tuners

Amplitude-modulation tuners, covering the broadcast band from 500 to 1500 kHz, are usually of the superheterodyne type, having an r-f preamplifier to reduce image interference, a first detector–oscillator to convert the signal frequency to a lower level, where there are better r-f amplification conditions, a second detector to rectify or pick off the sound signal from the carrier signal, and an audio amplifier and output arrangement to bring the signal to sufficient level to drive a basic audio amplifier or control unit of hi-fi type. In addition, there may be the usual control features, such as automatic volume control, noise limiting, and others.

Amplitude Modulation—Amplitude modulation is defined as the process of changing the amplitude of an r-f carrier in accordance with the intelligence to be transmitted. When there is no modulation, the radio-frequency carrier portion of an amplitude-modulated wave is of constant frequency and constant amplitude, as shown in Fig. 4-17. An audio modulating frequency is superimposed on this carrier in a manner that causes the amplitude of the carrier signal to vary as illustrated in Fig. 4-18, leaving the carrier frequency unchanged. The pattern shown in Fig. 4-18 is commonly referred to as a *modulation envelope.*

Sidebands—An amplitude-modulated wave is composed of a number of frequencies: the radio frequency of the carrier wave, the modulating audio frequency or frequencies, and combinations of these frequencies. These combination frequencies are called the *sideband frequencies* and are the result of mixing the radio frequency and the modulating frequencies. Whenever any two frequencies are mixed together, two new frequencies are produced. One of these is the sum of the two frequencies,

and the other is the difference between the two original frequencies. Thus, for a modulating frequency of 5000 Hz and a carrier frequency of 1000 kHz, sideband frequencies of 995 kHz and 1005 kHz are produced. If the modulating frequency is increased to 10,000 Hz, sidebands will be produced at 900 kHz and 1010 kHz.

Fig. 4-18. A modulated radio-frequency wave.
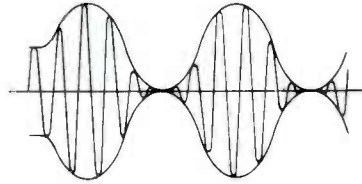
It is these sideband frequencies that carry the intelligence in an amplitude-modulated wave. When an r-f carrier is modulated by many audio frequencies, such as occur in speech or music, the side frequencies consist of a band of frequencies above and below the carrier frequency. The width of this band is determined by the highest modulating frequency. For this reason, hi-fi a-m signals must have an available bandwidth equal to the highest frequency to be reproduced. Since a-m transmitting stations are only 10 kHz apart from each other they are necessarily limited to 10 kHz bandwidth. Therefore the highest audio frequency possible to be transmitted by a-m radio is 5000 Hz.

A-M Superheterodyne Receivers

A *superheterodyne receiver* is one in which the desired signal is mixed with a locally generated signal to produce an intermediate-frequency signal. This intermediate-frequency signal is then amplified and detected to produce the audio frequency. Fig. 4-19 is a simplified block diagram of a typical superheterodyne receiver.

The r-f amplifier stage receives the weak signal intercepted by the antenna, amplifies it, and passes it on to the mixer. In the mixer stage, the received signal is heterodyned with the output of the local oscillator. The output of the mixer stage is an intermediate-frequency (i-f) signal which has the same modulation characteristics as the received signal.
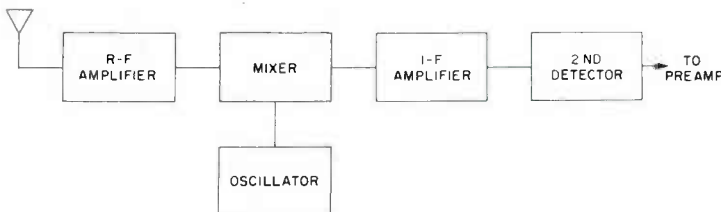
Fig. 4-19. Block diagram of an a-m superheterodyne receiver.

65

The i-f signal then passes through a number of amplifiers, referred to as intermediate-frequency amplifiers, whose output is applied to the second detector. This stage removes the i-f component from the signal, leaving the undistorted audio signal, which is then amplified and applied to the speaker.

Frequency Conversion—The converter stage consists of the mixer and local oscillator. The purpose of the frequency converter is to produce an intermediate-frequency signal having the same modulation characteristics
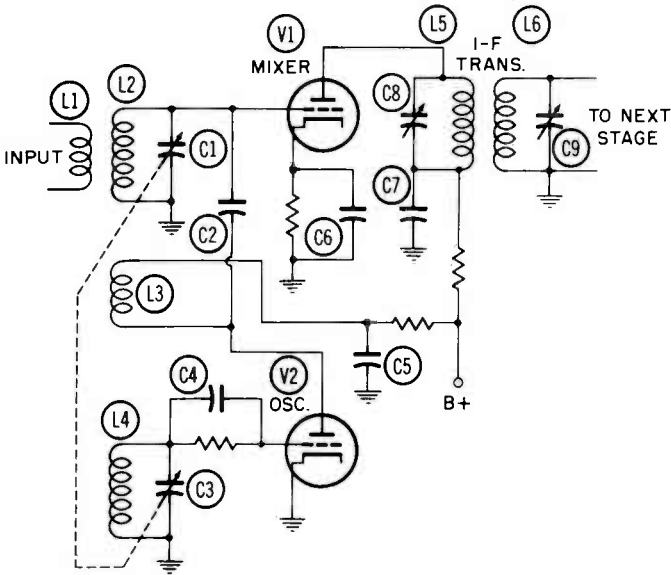
Fig. 4-20. A converter using separate mixer and oscillator tubes.

as the received signal. This is accomplished by generating an unmodulated r-f signal in the receiver and heterodyning it with the received signal. By this method, a third signal is generated, whose frequency is equal to the difference between the locally generated and incoming signal frequencies.

Two circuits are required to generate the i-f signal, an oscillator and a mixer. Tubes of special design have been developed so that both functions can be accomplished by one tube. Many receivers, however, employ separate mixer and oscillator tubes. A typical converter circuit using separate mixer and oscillator tubes is shown in Fig. 4-20. The r-f input is coupled to the mixer grid-tuner circuit, L2 and C1, by means of coupling coil L1. This circuit (L2-C1) is tuned to the frequency of the incoming signal and is applied to the mixer grid. V2 is connected in an Armstrong oscillator circuit. The oscillator operates at a frequency equal

to the incoming signal frequency plus the intermediate frequency. Output from the oscillator is coupled to the mixer grid through capacitor C2. The signal on the plate of the mixer tube is thus the result of both the incoming signal and the oscillator signal. Signals at the oscillator frequency, the received signal frequency, the difference frequency, and several others appear in the mixed output. The circuit of L5 and C8 is tuned to the difference frequency, and this signal builds up to a high amplitude while other signals are largely eliminated.

Capacitors C1 and C3 are ganged so that when the mixer grid circuit is tuned to the frequency of an incoming signal, the oscillator is tuned so that its frequency remains equal to the incoming signal plus the intermediate frequency. The most common intermediate frequency is 455 kHz. If the received signal is at a frequency of 1000 kHz, the oscillator frequency must be 1455 kHz to produce an intermediate frequency of 455 kHz. If the mixer is tuned to a new signal, to say 2000 kHz, the oscillator must be changed to 2455 kHz.

Oscillator Signal Injection—In the converter described previously, a capacitor is used to inject the oscillator signal into the grid circuit of the mixer. This arrangement is called *capacitive injection.* Two other methods of injecting the oscillator signal into the mixer circuit are shown in Fig. 4-21. In Fig. 4-21A *inductive injection* is used. The oscillator grid coil (L4) is inductively coupled to the mixer cathode circuit by means of coupling coil L3. In Fig. 4-21B *electronic injection* is used. A pentagrid mixer tube is used in this circuit, providing a separate grid for the oscillator signal. The incoming signal applied to the control grid and oscillator signal applied to a second control grid both act upon the electron stream through the tube to produce the intermediate frequency in the plate circuit. The injection method illustrated in Fig. 4-21A is superior to that of Fig. 4-20 in that it reduces interaction between the mixer and oscillator circuits. The circuit of Fig. 4-21 is superior to both of the others in this respect.

I-F Amplifiers—The i-f amplifiers provide most of the voltage amplification of the signal of a superheterodyne receiver. One or two and sometimes three i-f amplifier stages are used. A typical i-f amplifier circuit is shown in Fig. 4-22. The input and output circuits are inductively coupled by means of i-f transformers T1 and T2. The primaries and secondaries of the transformers are tuned. Since the incoming signal is always heterodyned to the same intermediate frequency, the four tuned circuits are operated at the same frequency at all times. This makes it possible to design and adjust the circuits to obtain maximum gain. The i-f transformers are mounted in small metal cans and are adjusted to the proper frequency by means of variable capacitors, as shown in the figure, or by means of movable powdered-iron cores. The capacitor-tuned types are often provided with fixed powdered-iron cores to increase gain and selectivity.
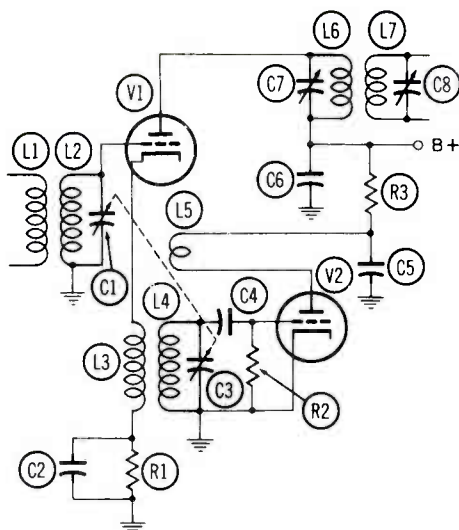
(A) *Inductive injection.*
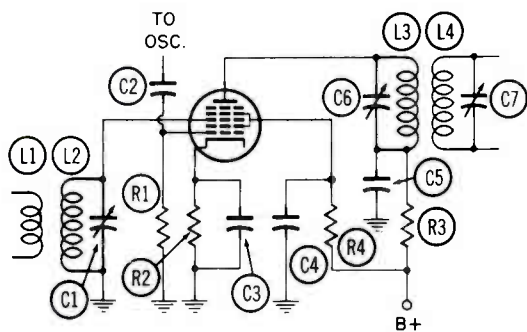
(B) *Electronic injection.*

Fig. 4-21. Two methods of injecting the oscillator signal into the mixer.

Because of the high gain of i-f amplifiers, coupling between input and output circuits must be kept to a minimum. This is accomplished by careful shielding and placement of parts and by providing suitable decoupling networks in plate, screen, and grid circuits. Decoupling networks usually consist of a resistor and capacitor connected as shown in Fig. 4-22. The plate decoupling network consists of R2 and C5, while that of R3 and C6 provides screen decoupling.

Hi-Fi Bandpass and Image Rejection—The two most important factors influencing the choice of an intermediate frequency are bandpass and image rejection. For several reasons it is possible to obtain greater bandpass as the intermediate frequency is raised. Therefore, when max-

imum bandpass is desired, the intermediate frequency is made as high as possible, consistent with other factors.

If the oscillator of a superheterodyne is tuned to 1455 kHz and the intermediate frequency is 455 kHz, signals at 1000 kHz (oscillator minus intermediate frequency) and 1910 kHz (oscillator plus intermediate frequency) may be received by tuning the mixer to the desired signal. This is possible because both frequencies when heterodyned with the oscillator signal will produce the same difference frequency. In practice, the mixer is tracked so that it is always tuned to either the oscillator frequency plus the intermediate frequency or the oscillator frequency minus the intermediate frequency. If the mixer frequency is equal to the intermediate frequency plus the oscillator frequency, then the intermediate frequency minus the oscillator frequency is referred to as the *image frequency*. If the mixer is tuned below the oscillator frequency, then the higher frequency is called the image frequency. Regardless of which frequency the mixer is tuned to, some signal energy will appear in the mixer output if a strong image-frequency signal is present. This difficulty occurs because the mixer circuit is not selective enough to reject the image signal. Suitable image rejection is obtained by choosing an intermediate frequency high enough to provide sufficient separation between the received-signal and image-signal frequencies. As the intermediate frequency is increased, the image frequency moves further away from the frequency to which the mixer is tuned and the image rejection increases. In the broadcast band and at somewhat higher frequencies, intermediate frequencies in the neighborhood of 455 kHz are satisfactory; while at higher frequencies, the intermediate frequency must be increased to obtain suitable image rejection. Generally, it is necessary to make a compromise and choose a frequency somewhere between that which gives optimum image rejection and that which gives the greatest selectivity.

R-F Amplifiers—An r-f amplifier is not absolutely necessary in a superheterodyne receiver; in fact, many receivers do not include such a
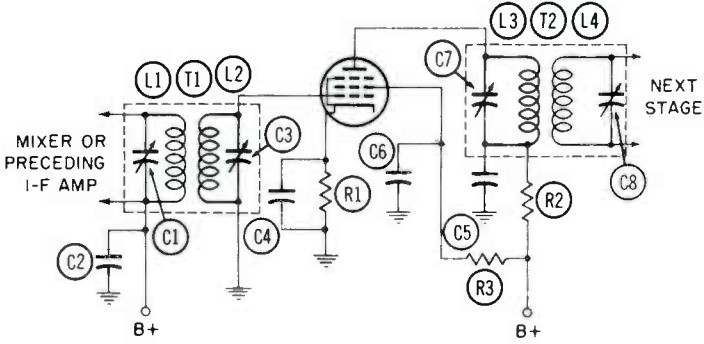
Fig. 4-22. A pentode i-f circuit.

69

stage. However, the incorporation of an r-f amplifier greatly improves the performance of a receiver. The purpose of an r-f amplifier is to improve the image rejection and the sensitivity of the receiver. As explained in the paragraph on i-f amplifiers, the mixer stage does not have sufficient selectivity to completely reject strong signals at the image frequency. The r-f stage increases the image rejection by amplifying the desired signal. The image signal is not amplified, and thus image interference is reduced. Some receivers use as many as three r-f stages to secure optimum image rejection in combination with an intermediate frequency low enough to permit high selectivity.

Considerable noise is generated in converter tubes. This noise is superimposed on the signal and appears in the output of the receiver. To be received, a signal must have an amplitude greater than the noise generated in the converter stage. An r-f amplifier increases the amplitude of the incoming signal before it reaches the converter stage. Since the converter noise remains constant, the additional signal amplification makes it possible to receive signals which would otherwise be lower than the converter noise level. Some noise is also generated by r-f amplifiers, and, when such stages are employed, the absolute sensitivity of the receiver is determined by this noise in the first r-f stage. Radio-frequency amplifiers, however, generate much less noise than converters. The ability of an r-f stage to improve the sensitivity of a receiver is particularly important at frequencies above 10 MHz. Below 10 MHz, man-made noise is too great to make very high sensitivity useful.

A typical r-f amplifier circuit is shown in Fig. 4-23. It consists of a pentode tube with a tuned-grid circuit and an impedance load. Pentodes are generally used because of their high gain and low interelectrode capacitance. Because of their high gain, r-f amplifiers must be carefully shielded and decoupled to prevent oscillation.

Second Detector—The second detector removes the i-f component from the signal, leaving the audio that was impressed on the carrier at the transmitter. The simplest and most common type of detector is the diode detector shown in Fig. 4-24. Grid-leak detectors overload too
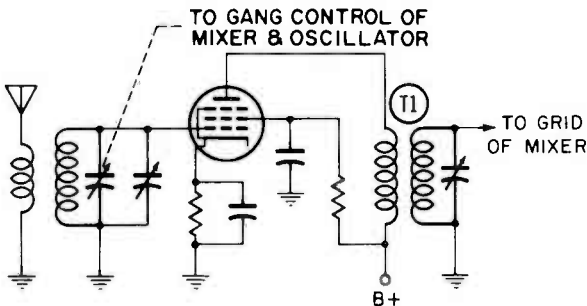
Fig. 4-23. A typical r-f amplifier circuit.

easily for use in superheterodyne receivers. The plate detector is sometimes used, but it is not as popular as the diode detector because it is more difficult to obtain avc voltage from the former.

The diode detector is used in many forms. It has the advantage of a cathode-follower output. Cathode-follower circuits have lower impedance, allowing use of longer lines to feed the amplifier unit. Cathode-follower circuits also have low-distortion and high-stability characteristics.
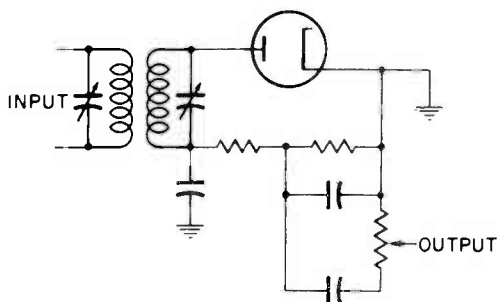
Fig. 4-24. A diode detector circuit.

Automatic Volume Control—The function of avc is to maintain constant output from a receiver when the amplitude of the incoming signal changes. This is accomplished by rectifying part of the received signal, at the output of the i-f amplifier and developing a voltage across a suitable resistor. The magnitude of the voltage is proportional to the amplitude of the incoming signal. By using remote-cutoff tubes in the r-f and i-f stages of the receiver, this voltage may be applied to their grids, as bias, to vary the gain of the receiver inversely as to signal strength.

A typical avc circuit is shown in Fig. 4-25. Tube V1 operates as a conventional diode detector. Tube V2 is the avc rectifier. Signal voltage is fed from the detector plate to the plate of V2 through coupling capacitor C6. The rectified signal current produces a voltage across diode load resistor R5. Since the current through R5 is from the diode plate to ground, the upper end of R5 is negative with respect to ground. This negative voltage is applied to the grid circuits of the r-f and i-f amplifiers through a filter and individual decoupling networks. An increase in the amplitude of the incoming signal increases the avc bias and reduces the gain of the receiver to maintain constant output. If the signal amplitude decreases, the avc bias decreases and the receiver gain is raised. Resistors R6 and R7 form a voltage divider operating from the receiver B+ supply. The voltage divider places a positive potential on the cathode of the avc diode. This potential delays the development of avc voltage until the signal reaches a predetermined minimum value. On weak signals, there is no avc bias and the receiver operates at full gain.

71
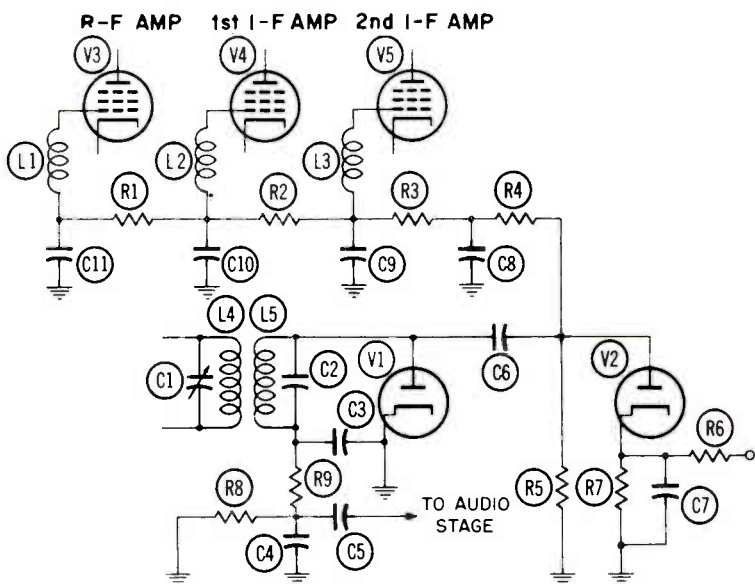
Fig. 4-25. An automatic volume control circuit.

Some receivers employ the circuit of Fig. 4-26. Here, the second detector, avc rectifier, and the first audio amplifier are combined in one tube. The upper diode is the signal detector. The lower diode, which acts as the avc rectifier, is coupled to the detector plate through capacitor C4. A rectified voltage is developed across R2 and applied to the r-f and i-f grid circuits through a filter consisting of R1 and C3. The cathode
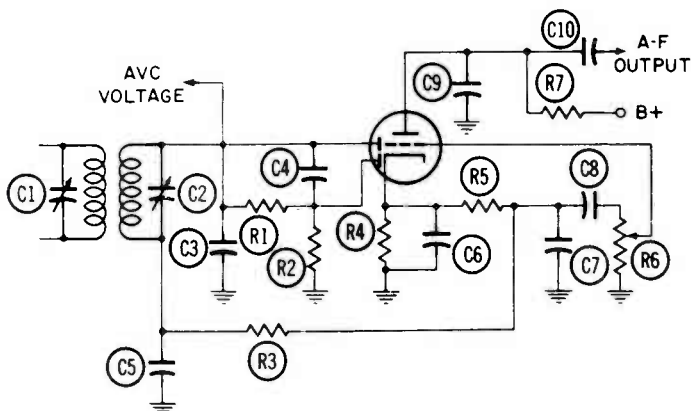
Fig. 4-26. A combined avc, second detector, and audio amplifier.

current of the audio amplifier section produces a bias voltage across R4 which delays avc action until the incoming signal is great enough to develop a voltage exceeding the bias.

The filters, R1 and C3 in Fig. 4-26, and R4 and C8 in Fig. 4-25, play an important part in the operation of these circuits. The filters remove audio-frequency variations from the avc voltage. Their time constants must be long enough to remove all audio fluctuations but not so long as to prevent the avc voltage from following rapid changes in input signal amplitude.

Tuning Indicators

The tuning eye, or "magic eye," is a cathode-ray type of tuning device that may be used to simplify the tuning of receivers and other equipment. It consists of a miniature cathode-ray tube and a triode amplifier in the same evacuated glass envelope. The usual circuit arrangement is shown in Fig. 4-27. A rodlike common cathode is mounted vertically
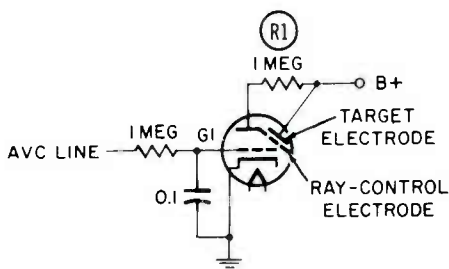
Fig. 4-27. A cathode-ray tuning indicator.

and surrounded by a funnel-shaped anode that tapers downward. The inner surface of this anode, or target electrode, is chemically treated so that it glows when struck by the electrons emitted by the heated cathode, producing the familiar ring of light visible at the end of the glass envelope. Between the cathode and target is a vertical wire known as the *ray-control electrode*. If this electrode is at the same voltage as the target, the glow will be a continuous ring. However, if the voltage is less positive than that of the target electrode, a field is set up that repels electrons flowing from the cathode to the target. Thus, the part of the target in line with the ray-control electrode will be dark; the extent of the dark area depends on the voltage difference between the target and the ray-control electrode. In a typical circuit, the control grid of the triode section is connected to the avc line in the receiver. Without a signal there is no avc voltage and no bias on the control grid (G1) of the triode section of the indicator. This results in a high anode current in the triode section, and therefore, a high voltage across the resistor R1 (usually about 1 megohm). Thus, the voltage at the ray-control electrode is con-

73

siderably less positive than that at the target, and a wide shadow angle is obtained. As the signal increases, the negative avc voltage increases and the control grid G1 becomes more negative. This results in less anode current in the triode section and a smaller voltage drop across resistor R1. The difference in the voltage at the ray-control electrode and the target electrode becomes less, and the shadow angle becomes smaller. Thus, as a station is tuned in, the dark portion of the ring becomes smaller.
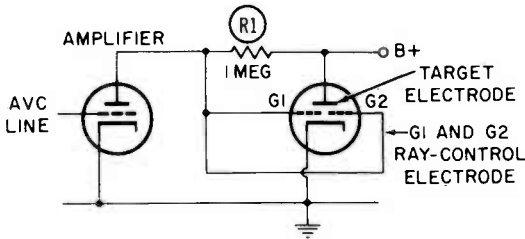
Fig. 4-28. A dual-shadow tuning indicator.

A dual magic-eye indicator is shown in Fig. 4-28. Its operation is similar to that of the single type except that it employs an external d-c amplifier. The cathode-ray section uses two ray-control electrodes, and two shadow angles are produced. These shadows will be symmetrically opposite if the ray-control electrodes are connected together. Two dissimilar patterns are obtained if the ray-control electrodes are connected to separate circuits.

Signal-Strength Meters

A plate-current signal-strength meter is shown in Fig. 4-29. A milliammeter is connected in the plate lead of several of the r-f or i-f tubes which have avc voltage applied to their grids. As the signal strength increases, the avc voltage becomes more negative and the plate current through the meter decreases. Resistor R is adjusted so that the milliammeter reads full-scale with no signal (highest plate current). This point is called "zero signal." Thus the meter indicator moves counterclockwise with increasing signal. In many commercial receivers, the meter is mounted in an inverted position, so that the pointer will move to the right with increasing signal strength.

A bridge-type signal-strength meter is shown in Fig. 4-30. Tube V1 is used to amplify the avc voltage. The current through R1, M, and R3 tends to cause the meter needle to move to the right, while the current through R2, M, and the tube tends to make the needle move to the left. At zero signal, these currents are made equal by adjusting the resistance

74
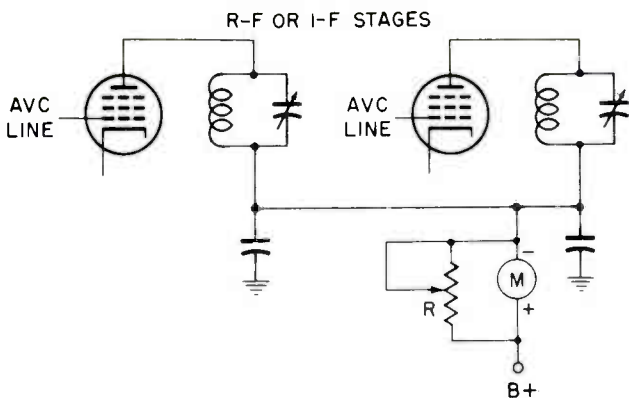
Fig. 4-29. A plate-current signal-strength meter.

of R1. The operation of this circuit is based on the fact that a change in grid bias will cause a variation in the d-c plate current of the tube. As the received signal amplitude increases, the avc voltage becomes more negative. The voltage is applied to the grid of V1 and the d-c plate current decreases. Thus, the meter needle moves to the right with increasing signal strength.
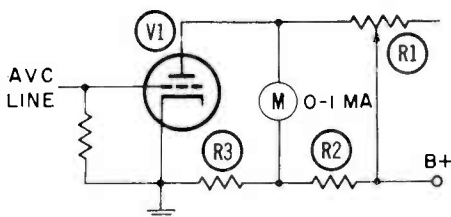
Fig. 4-30. A bridge-type signal-strength meter.

Automatic Frequency Control

Automatic frequency control circuits are used in many superheterodyne receivers to compensate for frequency drift. This drift may be due to such factors as small changes in the oscillator or carrier frequencies. It is compensated for by automatically adjusting the oscillator frequency.

An afc system consists of two basic parts: a frequency detector and a variable reactance circuit. Fig. 4-31 shows a typical circuit. The discriminator is of the Foster-Seeley type and is excited by the i-f signal from the final i-f amplifier stage. The discriminator output is a d-c voltage whose polarity depends on whether the intermediate frequency has deviated above or below its correct value, and whose magnitude is proportional to the amount of deviation. This d-c voltage is applied to the control grid of the reactance tube, which is connected across the tank

75
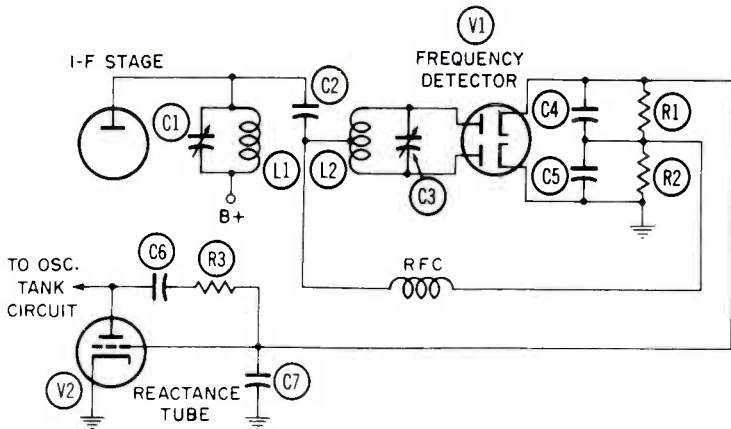
Fig. 4-31. An automatic frequency control circuit.

circuit of the local oscillator. The effective reactance this tube offers to the oscillator varies with the grid bias. A deviation in the intermediate frequency from its proper value causes a change in the d-c grid voltage of V2, which produces a change in the reactance presented to the local oscillator. This change in reactance is such that the oscillator is automatically adjusted to bring the intermediate frequency back to its correct value.

F-M Tuners

Frequency-modulation tuners operate in the very high frequency regions and consequently have operating requirements different from those of a-m tuners. The r-f circuits and components must be of optimum design to provide sufficient sensitivity to pick up f-m stations under all conditions, and the audio aspects of all circuitry must be of sufficient quality of design and construction to receive and pass audio frequencies to 15 or 20 kHz as may be broadcast by f-m stations.

Frequency Modulation

High-fidelity a-m reception is usually limited by the presence of high-level man-made noise and atmospheric disturbances which a-m receivers cannot reject without loss of fidelity. Also, because of the propagation characteristics of frequencies used for standard a-m broadcasting, out-of-area broadcast station signals can interfere with local reception (especially at night). At the frequencies used for f-m reception, distant stations cannot interfere, and the inherent noise-rejection characteristics of the f-m receiver minimize the noise problem.

In frequency modulation, the frequency of the r-f carrier is varied in accordance with the a-f or other signal to be transmitted. Amplitude and

76

frequency modulation are compared in Fig. 4-32. Fig. 4-32A shows an unmodulated carrier, Fig. 4-32B shows an amplitude-modulated carrier, and Fig. 4-32C shows a frequency-modulated carrier. In the a-m carrier the frequency remains constant and the amplitude varied during modulation, while in the f-m carrier the amplitude remains constant and the frequency varies during modulation.

F-M Receivers

Receivers for frequency modulation are of the superheterodyne type and are somewhat similar to ordinary amplitude-modulation superheterodynes. Block diagrams of the two most widely used f-m receivers and an a-m superheterodyne are shown in Fig. 4-33. All three receivers employ r-f amplifiers, mixer stages, oscillators, and i-f amplifiers. The most

(A) Unmodulated carrier.

(B) Amplitude-modulated carrier.

(C) Frequency-modulated carrier.
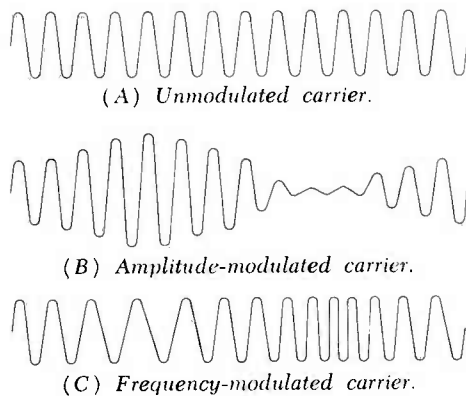
Fig. 4-32. Comparison of a-m and f-m carriers.

important difference between a-m and f-m receivers is in the detector circuit. A number of f-m detectors have been developed. The ratio detector used in the receiver in Fig. 4-33B removes the audio signal from the carrier and at the same time rejects amplitude impulses which may accompany it. The receiver Fig. 4-33C employs a discriminator detector to remove the audio signal from the carrier. This detector is sensitive to amplitude impulses and in order to eliminate them before detection, limiter stages must be provided. The limiter removes all amplitude fluctuations from the carrier before it is applied to the detector.

R-F Amplifier—Radio-frequency amplifiers are used to secure improved signal-to-noise ratio, higher gain and selectivity, and improved image rejection for frequency modulation, as in amplitude modulation as previously described. Improvement in signal-to-noise ratio is more important to secure in an f-m r-f amplifier than in an a-m r-f amplifier because considerably more noise is generated in converter stages and, as described, the addition of amplification before the converter increases

the signal amplitude without increasing the noise. The tubes used in vhf r-f amplifiers must have high mutual conductance, low interelectrode capacitance, and high input resistance. Components and circuits must be used that generate as little noise as possible. Because of the high frequencies at which these circuits operate, short leads, careful shielding, and high-quality insulation must be used.
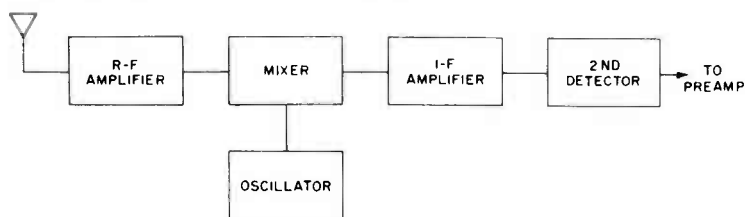
(A) Amplitude-modulation receiver.
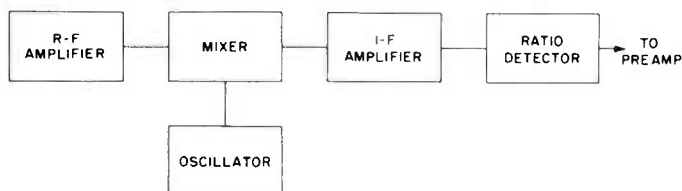
(B) Frequency-modulation receiver using ratio detector.
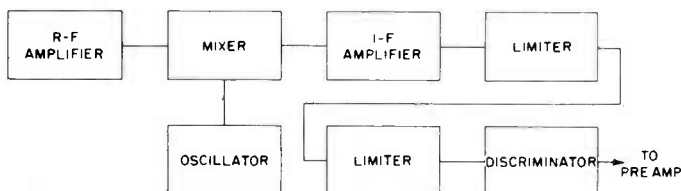
(C) Frequency-modulation receiver using a limiter and discriminator.

Fig. 4-33. Block diagrams of a-m and f-m superheterodynes.

A typical r-f amplifier circuit is shown in Fig. 4-34. The antenna transmission line is coupled to the input coil by means of a separate winding. This is required to match the high input impedance of the stage to the comparatively low impedance of the antenna transmission line. Most receivers are designed to match a 300-ohm line. To pass the complete f-m signal, the r-f amplifier must respond to a wide band of frequencies. This is accomplished by using low-Q coils. The low Q broadens the response curve of the amplifier.

Mixer-Oscillator—Frequency-modulation receivers generally use separate mixer and oscillator tubes to achieve greater efficiency although in some cases these functions are combined in one tube specifically designed for this application. The circuits employed are similar to those found in

78
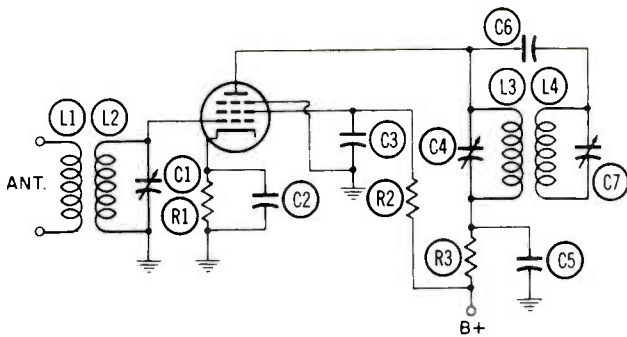
Fig. 4-34. A typical vhf r-f amplifier circuit.

a-m receivers, with modifications to make them more suitable for use at high frequencies. The difficulties encountered in using a combined mixer-oscillator stage stem from interaction between the mixer and oscillator, which becomes troublesome at high frequencies and results in oscillator pulling and instability. These difficulties are largely avoided by using separate tubes and loose oscillator-mixer coupling.

It is much more difficult to minimize oscillator drift at the frequencies used for f-m broadcasting than it is at a-m broadcasting frequencies. Heating, humidity, and B+ supply voltage variations (regulation) all contribute to oscillator drift. The effects of changing humidity are minimized by coating circuit components with moistureproofing materials and by permitting a certain amount of temperature rise in the area surrounding critical components. Heating causes drift because it expands parts of critical components, which results in increased capacitance. It is minimized by using insulation materials with low temperature coefficients and by shunting tuned circuits with negative–temperature-coefficient capacitors to counteract the increase in capacitance taking place in other components. The effects of poor regulation in B+ voltage supplies are minimized by careful decoupling of the various circuits in the receiver.

I-F Amplifiers—The i-f amplifiers used in f-m receivers employ conventional amplifier circuits. To a major extent, they determine the overall gain and selectivity of the receiver. Two stages of amplification are generally used in which three double-tuned transformers are employed. To secure the required broad-band response, the Q of the i-f transformer windings is made comparatively low. In addition, one or more of the transformers is often overcoupled to broaden its response. The i-f frequency used in most modern f-m broadcast receivers is over 10 MHz. This high intermediate frequency gives excellent image-frequency interference rejection.

Limiters—In receivers using discriminator-type detectors, some means must be provided to remove amplitude variations from the re-

ceived signal before it is applied to the detector stage. The function of the limiter is to remove noise pulses and to restore uniformity to the signal over the passband. A simplified limiter circuit is shown in Fig. 4-35A. A sharp-cutoff tube is used. The plate and screen voltages applied to the tube are much lower than those normally applied to an amplifier, and no fixed bias is provided. Bias is obtained by placing a capacitor and resistor in the grid return of the tube. Under these operating conditions, a grid signal of comparatively low amplitude will drive the tube to saturation on positive peaks, and to cutoff on negative peaks.

(A) Circuit.   (B) Limiting action.
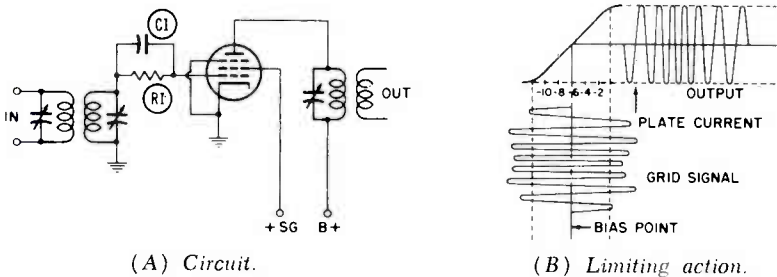
Fig. 4-35. A typical limiter circuit.

This action is illustrated in Fig. 4-35. When the input signal to the limiter has sufficient amplitude, all the negative and positive peaks are clipped and the signal at the plate of the limiter has a constant amplitude. If the signal is not of sufficient amplitude, only partial limiting will take place. Resistor R1 and capacitor C1 play an important part in the operation of the limiter. During the positive cycles of input voltage, the grid of the tube draws current, loading the input tuned circuit and providing a diode clipping action. During this period, current flows through resistor R1, and capacitor C1 is charged. During the negative portion of the input-signal cycle, C1 discharges through R1, developing a negative bias on the grid which is inversely proportional to the amplitude of the input signal. When the amplitude of the input signal increases, the negative bias on the tube becomes more negative. Thus the bias on the tube is automatically controlled by the amplitude of the input signal. The time constant of R1 and C1 is chosen so that it is long enough to maintain substantially constant grid bias during the negative portions of the i-f signal applied to the stage. However, the time constant is short enough to permit an increase in bias when sudden-amplitude impulses occur.

In hi-fi receivers, two limiter stages are recommended to secure proper limiter action.

Discriminators—The discriminator circuit of Fig. 4-36 illustrates one way in which the audio modulation may be removed from the frequency-modulated carrier. The i-f signal is coupled to the discriminator plates
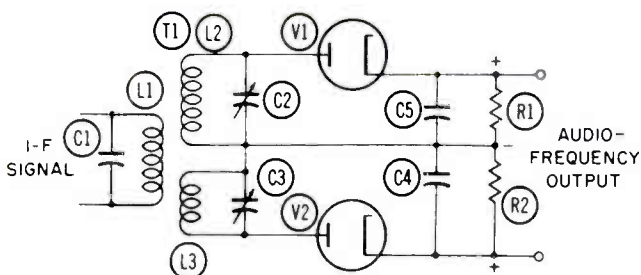
Fig. 4-36. A double-tuned discriminator circuit.

by means of transformer T1. The secondary of the transformer consists of two windings, L2 and L3, tuned by means of capacitors C2 and C3. Tuned circuits L2-C3 and L3-C3 are resonated at different frequencies, one above the frequency of the received signal and the other below it. The frequencies to which the resonant circuits are tuned are equal to the carrier center frequency plus the maximum r-f carrier deviation, and the carrier center frequency minus the maximum r-f carrier deviation. For f-m broadcast reception one tuned circuit is tuned to the carrier center frequency plus 75 kHz, and the other to the carrier center frequency minus 75 kHz.

Fig. 4-37 illustrates the response curves of the tuned circuits as indicated by the voltages at the plates of the diodes. When a frequency-modulated signal is applied to the input of the circuit, the instantaneous voltages on the diode plates vary as the signal swings each side of the center frequency. If L2 and C2 are resonant above the carrier frequency, the voltage on the plate of V1 will be higher than the voltage on the plate of V2 when the carrier swings higher in frequency than its center frequency. When the carrier swings lower in frequency than its center frequency, the voltage on the plate of V2 will be higher than the voltage on the plate of V1. When the carrier frequency is above its center frequency and the voltage on the plate of V1 is higher than the voltage on
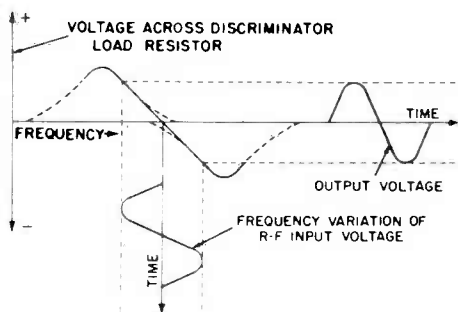
Fig. 4-37. Discriminator response curves.

81

the plate of V2, the current through V1 is greater than the current through V2; consequently, the voltage developed across R1 is greater than the voltage developed across R2. The resistors are connected so that the voltages developed across them are of opposite polarities. As a result, the voltage difference from the top of R1 to ground is equal to the voltage across R1 minus the voltage across R2. Under the conditions described, the voltage at the output is positive. When the carrier swings below its center frequency, the voltage developed across R2 is greater than that across R1, and the voltage at the output of the circuit is negative in polarity. Thus, as the carrier swings above and below its center frequency, it produces a voltage in the output to the discriminator which varies in amplitude and frequency in accordance with the modulation of the f-m carrier.
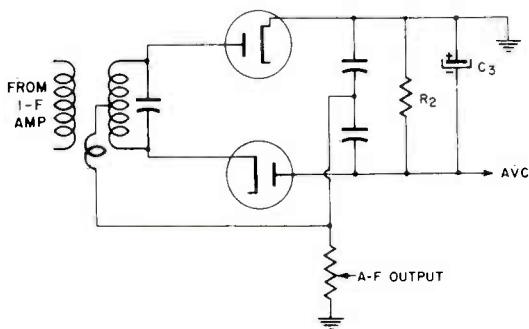
Fig. 4-38. A ratio detector circuit.

Ratio Detectors—Ratio detectors are used in place of limiter-discriminator circuits for applications requiring less fidelity under all conditions. Fig. 4-38 shows a ratio detector circuit. It is similar to the circuit just described, except that the diodes are connected in opposite phase and C3 is connected across load resistor R2, and there are no limiters required as shown in Fig. 4-33B. Outfits using this circuit perform very well under optimum conditions and are recommended for reasonable-cost installations.

Tuner Compensation

Pre-emphasis and de-emphasis are used in standard f-m broadcasting and reception to minimize noise at the receiver. The high-frequency portion of the audio modulating signal is purposely emphasized, that is, boosted to a relatively high level at the transmitting station than the remainder of the audio-frequency range. This is known as *pre-emphasis*. Then, at the receiver, the exact reverse of this is done, to restore the high audio-frequency components to their normal relative level. This is known as *de-emphasis*. The advantage is that in the de-emphasis proc-
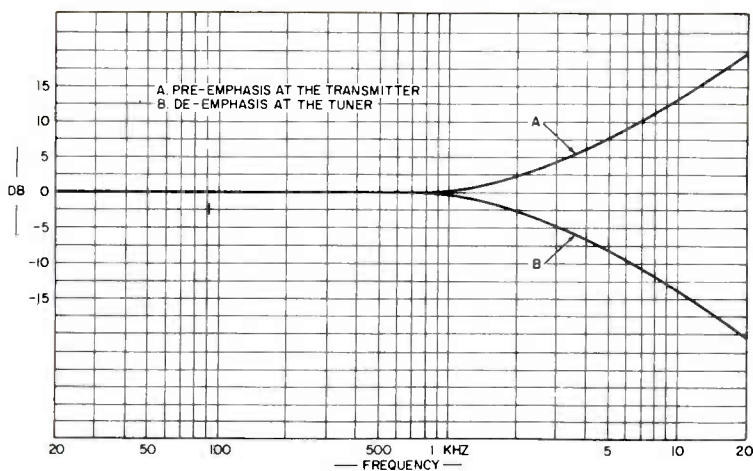
Fig. 4-39. Standard pre-emphasis and de-emphasis characteristics for f-m broadcast and reception.

ess, the reduction of relative response to high frequencies also reduces the response to noise. The standard pre-emphasis and de-emphasis characteristics are shown in Fig. 4-39.

Stereo Tuners

As explained in Chapter 3, tuners for stereo are similar to those for monophonic reception, but with differences as follows:

1. The original method of stereo broadcast and reception used separate a-m and f-m channels. Two separate tuners were used for stereo, one for amplitude modulation and one for frequency modulation. Both a-m and f-m facilities were available in hi-fi tuners before the advent of stereo and were employed as a means of carrying the two channels without new techniques or equipment. For stereo, the tuners were completely separate, so that both were played at the same time for a-m/fm two-station stereo reception.

2. Provisions are now made for reception of multiplex signals with f-m tuners. This consists of providing a separate "multiplex" output for feeding to a multiplex adapter, which is integrated in the receiver or may be bought separately.

The specifications important in the selection of tuners for monophonic reception as discussed up to this point apply also to tuners for stereo reception. In the use of stereo-type tuners, for two-station reception, the output of the a-m section was one stereo signal and the output of the f-m tuner was the second stereo signal. These outputs were simply connected to separate chanels in the stereo preamplifier. For multiplex

83

reception, the regular f-m tuner output and the multiplex (mx) f-m tuner output are both connected to the multiplex adapter; the adapter then delivers the left and right signals to the preamplifier.

Some examples of combination tuner units are illustrated in Fig. 4-40.

Combination tuners and dual preamplifiers allow such functions as volume, bass and treble boost, and stereo balance to be controlled on the same panel as the tuner dial. Only a two-channel amplifier and speaker system is needed with such an arrangement. Sometimes the combination process is carried one step further and the tuner, preamplifier, and amplifier are all combined. Such units include everything needed for reception of stereo broadcasts except the speaker system. Examples of these combinations are shown in Chapter 5.
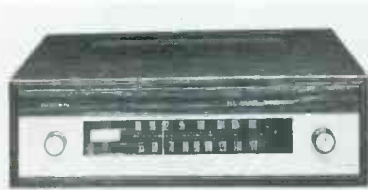
Courtesy Bogen Communication Div.,
Lear Siegler, Inc.

(A) Solid-state a-m–f-m/f-m tuner.
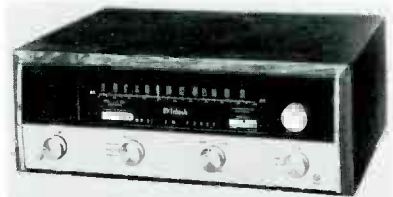
Courtesy McIntosh Laboratory, Inc.

(B) McIntosh f-m/f-m tuner.
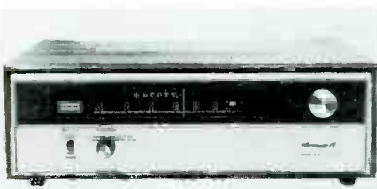
Courtesy H. H. Scott, Inc.

(C) Scott solid-state tuner.

Courtesy Kenwood Mfg. Co., Inc.

(D) Kenwood solid-state stereo.

Courtesy Heath Co.,
a subsidiary of Daystrom, Inc.

(E) Heathkit a-m–f-m/f-m tuner.

Courtesy Marantz Co., Inc.

(F) Marantz f-m tuner.

Fig. 4-40. Some f-m/f-m and a-m–f-m/f-m tuners and receivers.

Fig. 4-41. An adapter to convert early tuners for reception of f-m multiplex.

Stereo Multiplex Adapters

A typical multiplex adapter unit is illustrated in Fig. 4-41. This kind of unit can be connected to any tuner to provide f-m multiplex reception. Fig. 4-42 shows an adapter that mounts directly on the receiver chassis, making it an integral unit suitable for f-m and f-m multiplex outputs.

A block diagram of a typical f-m tuner with multiplex is shown in Fig. 4-43. From the antenna through the main detector and de-emphasis network it is exactly the same as a single-channel nonstereo tuner.

Fig. 4-42. A plug-in stereo multiplex adapter.

The high-pass filter, demodulator, and matrix portion of Fig. 4-43 is often referred to as the *stereo converter* or *stereo translator*. It can be connected to a nonstereo tuner to provide stereo output, if the received signal is transmitted as multiplexed stereo. The multiplex input of the converter is connected to the "multiplex" output of the tuner's detector, before the de-emphasis network. The shielded lead between the multiplex jacks on the tuner and the converter should be kept as short as possible, to avoid undue attenuation of the subcarrier.
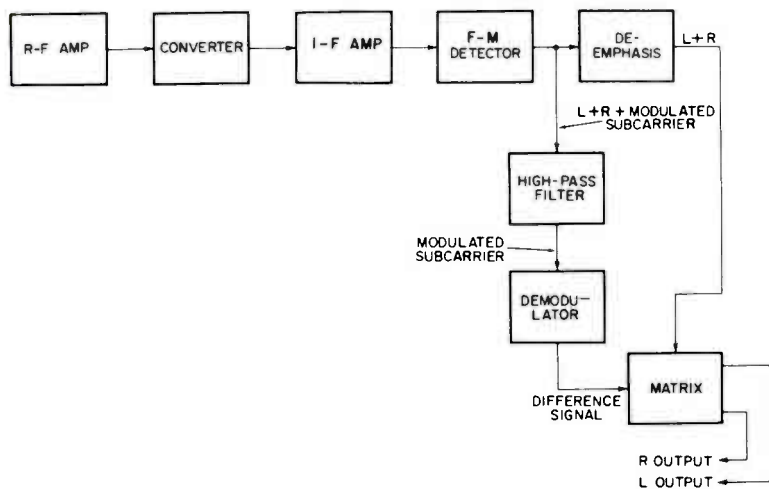
Fig. 4-43. Block diagram of a typical tuner equipped with a multiplex converter.

Stereo Multiplex Operation

As explained in Chapter 3, stereo multiplex is the system whereby the left- and right-channel signals of a stereo program are both sent out on one f-m broadcast channel. In the method standardized in the United States by the Federal Communications Commission, the following objectives are attained:

1. The left (L) and right (R) signals are sent, in separable form, within the 200-kHz f-m broadcast channel bandwidth.
2. Both signals are modulated onto the same carrier with negligible loss of signal-to-noise performance compared with that of monaural transmission. This is accomplished by means of the "interleaving" provided in the system.
3. The system is compatible in that the owner of an f-m receiver not equipped for stereo reception still receives an L + R signal equivalent to that received from nonstereo broadcasts.

Sum and Difference Signals—Although the L + R signal is all the listener to a monophonic program needs, the proper reception and reproduction of stereo programs requires the use of an additional (L − R) signal. This signal is produced at the transmitter by inverting (phase changing by 180 degrees) the R signal (making it negative, or −R) and adding it in its negative form to the L signal. The resulting L − R signal, as well as the L + R signal, is transmitted on the f-m broadcast carrier. The two signals are received and separated at the receiver. Then, in the stereo receiver, they are added together to form the L signal alone, and the L − R signal is subtracted from the L + R signal to form the R signal alone. This is shown mathematically as follows:

$$L + R + (L − R) = L + L + R − R = 2L$$
$$L + R − (L − R) = L − L + R + R = 2R$$

Stereo Multiplex Modulation Signal—The L + R and L − R signals are both sent as part of the modulation signal which frequency modulates the main (radio-frequency) carrier. Before we proceed with details of the modulating signal, it will minimize confusion to emphasize a very important fact: As far as the main carrier of the f-m signal is concerned, there is only one composite modulating signal, all of whose components modulate the carrier. The composition of this modulating signal includes the subcarrier, L + R signal, L − R signal, and certain other components within this overall modulating signal.

The spectrum of the modulating signal is shown in Fig. 4-44. Its bandwidth is from 50 Hz to 75 kHz; the 75-kHz deviation of the carrier is the total range and limit of signal-component frequencies allowed within the modulating signal.

The lowest portion of the modulating-signal spectrum extends from 50 Hz to 15 kHz, and this range includes the components of the L + R
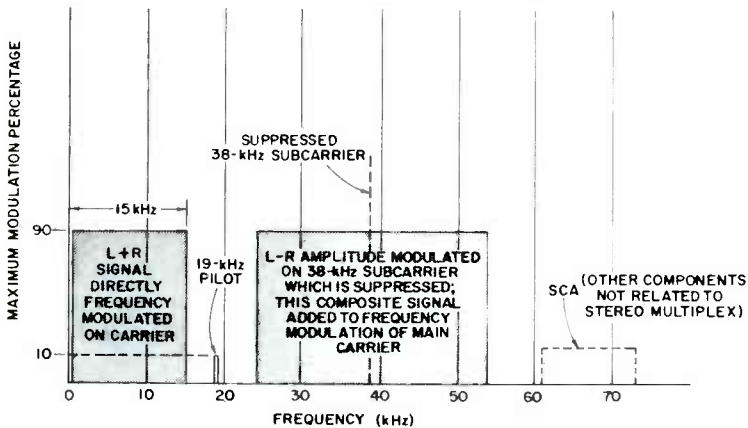
Fig. 4-44. Frequency spectrum of frequency-modulation signal on main carrier.

signal, or the signal that corresponds to or may be used as a normal monophonic modulation signal. In other words, if we remove all the signal components of frequencies above 15 kHz, we have a signal similar in effect to those signals used before stereo multiplex was developed.

Next in the spectrum is a 19-kHz pilot signal. However, let us ignore this for the moment and proceed to the $L - R$ portion of the signal, extending from 23 kHz to 53 kHz except for a 100-Hz quiet spot in the middle. This region consists of the sidebands resulting when the $L - R$ signal modulates the 38-kHz subcarrier. At the station, after the modulation has taken place, the 38-kHz subcarrier is suppressed and only the sidebands are used. Since the carrier component *was* at 38 kHz, the $L - R$ signal sidebands extend to 38 kHz $\pm$ 15 kHz (23,000 Hz to 37,950 Hz, and from 38,050 Hz to 53,000 Hz). These sidebands (the $L - R$ signal) are added to the modulating signal. In relation to the main carrier, all frequencies in this composite modulation of the main carrier are above audibility, so they do not interfere with transmission and reception of the main-carrier $L + R$ signal for the listener using a monophonic receiver.

The $L - R$ sidebands as an independent signal are not intelligible (until decoded for the stereo listener) in the form in which they appear in the modulation signal. They must be: (1) recombined with a 38-kHz carrier, (2) demodulated from that carrier, and (3) decoded.

To process the signal at the receiver, the composite frequency-modulation signal depicted by Fig. 4-44 must be demodulated from the main carrier and the $L + R$ and the $L - R$ sidebands separated from the other parts of the signal by filters.

Demodulating the $L - R$ Signal—The carrier with which the sidebands are combined at the receiver must have the desired 38-kHz frequency within very close limits and remain synchronized to prevent distortion when these sidebands are demodulated. The desired synchronization is provided by the 19-kHz pilot carrier added to the modulation signal. This steady 19-kHz unmodulated signal is demodulated from the main carrier at the receiver and passed through a frequency doubler, whose output at 38 kHz is combined with the $L - R$ sidebands to produce the $L - R$ signal. The output from the doubler may be used directly for combining with the sidebands, or, as in some designs, this output can be used to synchronize a 38-kHz oscillator whose output is in turn used to combine with the sidebands.

When the 38-kHz subcarrier has been combined with the $L - R$ signal sidebands, the result is an amplitude-modulated signal. It can be shown that the top and bottom envelopes of the modulated-signal waveform have the waveshapes of the L and R signals respectively. A detector that operates half-wave on only the positive peaks then produces only the L signal; in the same way, a negative-peak detector will "pick off" the R signal. Some stereo receivers (or adapters) use this method of

demodulation. In others, the signal is demodulated from the 38-kHz carrier by an ordinary a-m detector to obtain the L − R signal. The L − R signal is then matrixed with the L + R signal (as explained earlier) to obtain the separate L and R signals. In a third method of detection, the local 38-kHz subcarrier is used as a switching signal, connecting the output of the detector to the L output channel during the time of positive peaks and connecting the output of the detector to the R output channel during the time of negative peaks.

Stereo Adapter Circuits—The block and schematic diagram of a typical stereo adapter is shown in Fig. 4-45. The adapter obtains its signal input from the "stereo" output of an f-m tuner; this output is simply the "audio" output of the tuner, tapped off at a point in the circuit before the de-emphasis network. This output from the tuner is applied to the first stage in the adapter, which is a broad-band amplifier. This amplifier amplifies all the frequency components in the tuner output, which include the L + R signal (already in audio form), the 19-kHz pilot signal, and the L − R sidebands. Also included is, in some cases, another signal, called the *Subsidiary Communications Authorization* (SCA) *signal*, which, as indicated in Fig. 4-44, is centered around 67 kHz. This signal, which is used for commercial "storecasting" service, has nothing to do with home f-m reception, and it is filtered out in the adapter circuits—so we shall ignore it.

In the circuit illustrated, the broad-band amplifier is a tube. Its signal output is applied to three points: (1) a low-pass filter (inductor and capacitors) which removes frequency components above 15 kHz, leaving just the L + R signal, (2) a bandpass filter and trap, which passes only the L − R sidebands, and (3) a 19-kHz keyed oscillator doubler which doubles the frequency of the 19-kHz pilot carrier. The 19-kHz oscillator locks in frequency and phase to the amplified pilot signal, and provides a 38-kHz signal for the L − R detector.

The detector uses two solid-state diodes connected in opposite polarities. The 38-kHz and L − R sideband signals are both applied to the detector, where they mix with it and then undergo demodulation. Since the diode polarities are opposite to each other, one puts out a plus L − R signal and the other a minus L − R signal. These then are mixed with the L + R signal in a matrix (resistors and capacitor network in the right-hand corner of the diagram) and produce the desired left and right output signals.

Stereo Multiplex Reception Indicator Lights

It is difficult to determine what stations are broadcasting stereo multiplex by listening to each station on the dial. Therefore, stereo indicators —usually lights or tuning eyes—have been arranged on panels of most recent stereo receivers. These lights turn on when a stereo program is tuned in.
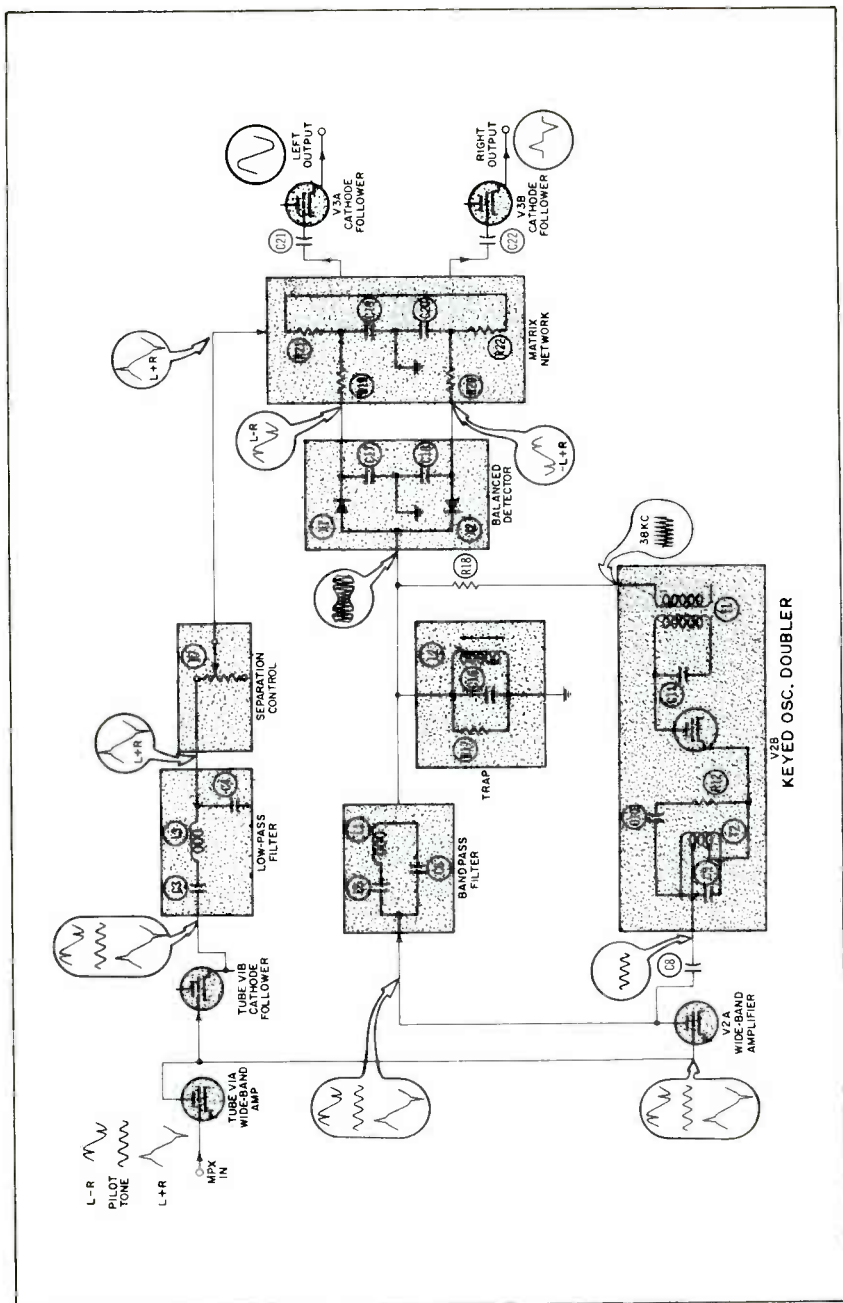
Fig. 4-45. Block diagram of Heath AC-11 adapter.

These lights are operated by a circuit triggered by presence of the 19-kHz pilot signal that is part of every stereo multiplex broadcast signal (see Fig. 4-44). Block diagrams and circuit examples are shown in Figs. 4-46, 4-47, and 4-48. Fig. 4-46A shows the block diagram for the receiver that amplifies and doubles the original 19-kHz pilot modulation component of the main carrier. In this type of circuit a portion of the voltage produced at the output of the doubler amplifier is stepped up through a transformer to over 70 volts minimum and applied to a neon indicating light. Since the pilot signal is only present on stereo multiplex broadcast, the light only lights up to indicate that stereo reception is possible.

In Fig. 4-46B the oscillator is always on, so the signal must be picked off before the pilot-oscillator amplifier circuits. Since this output is not sufficient to provide enough power to operate a signal device, an amplifier is used to build up voltage in the same manner as that in Fig. 4-46A. The circuit in Fig. 4-46C shows the circuit detail of a stereo multiplex indicator circuit. Fig. 4-47 shows a Scott multiplex circuit with stereo indicator output (terminal 10, on jack).

Automatic F-M/F-M Stereo Switching

Some higher-priced f-m multiplex receivers have automatic switching circuits which operate to change the receiving mode when f-m broadcast material changes from f-m mono to f-m stereo, and vice versa. Fig. 4-48 shows a method of accomplishing this objective, and, at the same time, it also provides for a stereo beacon light. This circuit is dependent on the 19-kHz pilot signal, and it operates with the agc circuits of the receiver as shown.

Tuner Sensitivity

It is a primary requirement that the tuner be able to bring in the desired stations clearly and without noise. The ability of a receiver to do this is related to its sensitivity. A sensitivity of 10 microvolts on f-m reception or 75 microvolts on a-m reception will give good results for receivers operating within 15 miles of the broadcast station. For more distance, look for proportionately more sensitivity. If you live over 50 miles from the desired station, you should obtain the best tuner available with (probably) less than 3 microvolts sensitivity on f-m reception and 10 microvolts on a-m reception. At greater distances a-m stations will probably be the more reliable source of reception, and your choice of equipment should be so governed.

TAPE RECORDERS

Tape recorders consist of tape-transport devices, motors, erase-bias head, recording head, playback head, amplifiers, and controls. Most lower-priced models combine the recording and the playback operations

91
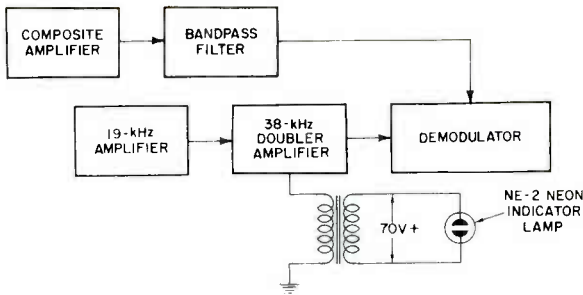
(A) *Stereo multiplex indicator light (no oscillator circuit).*
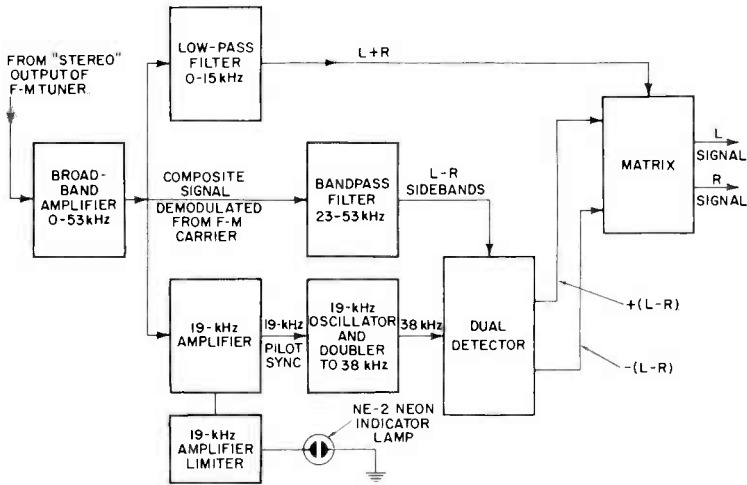
(B) *Stereo multiplex with indicator light takeoff before oscillator.*
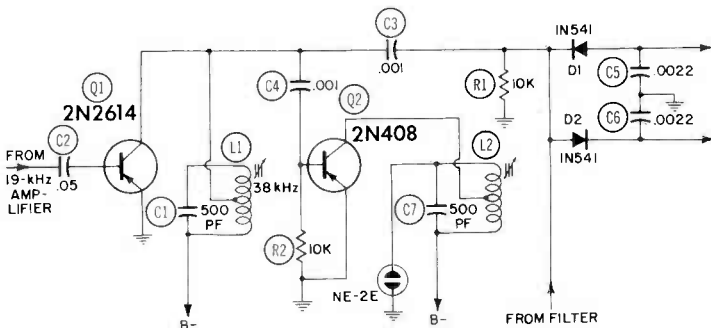
(C) *Stereo indicator circuit in Knight Model KN-265.*

Fig. 4-46. Stereo indicator circuits.
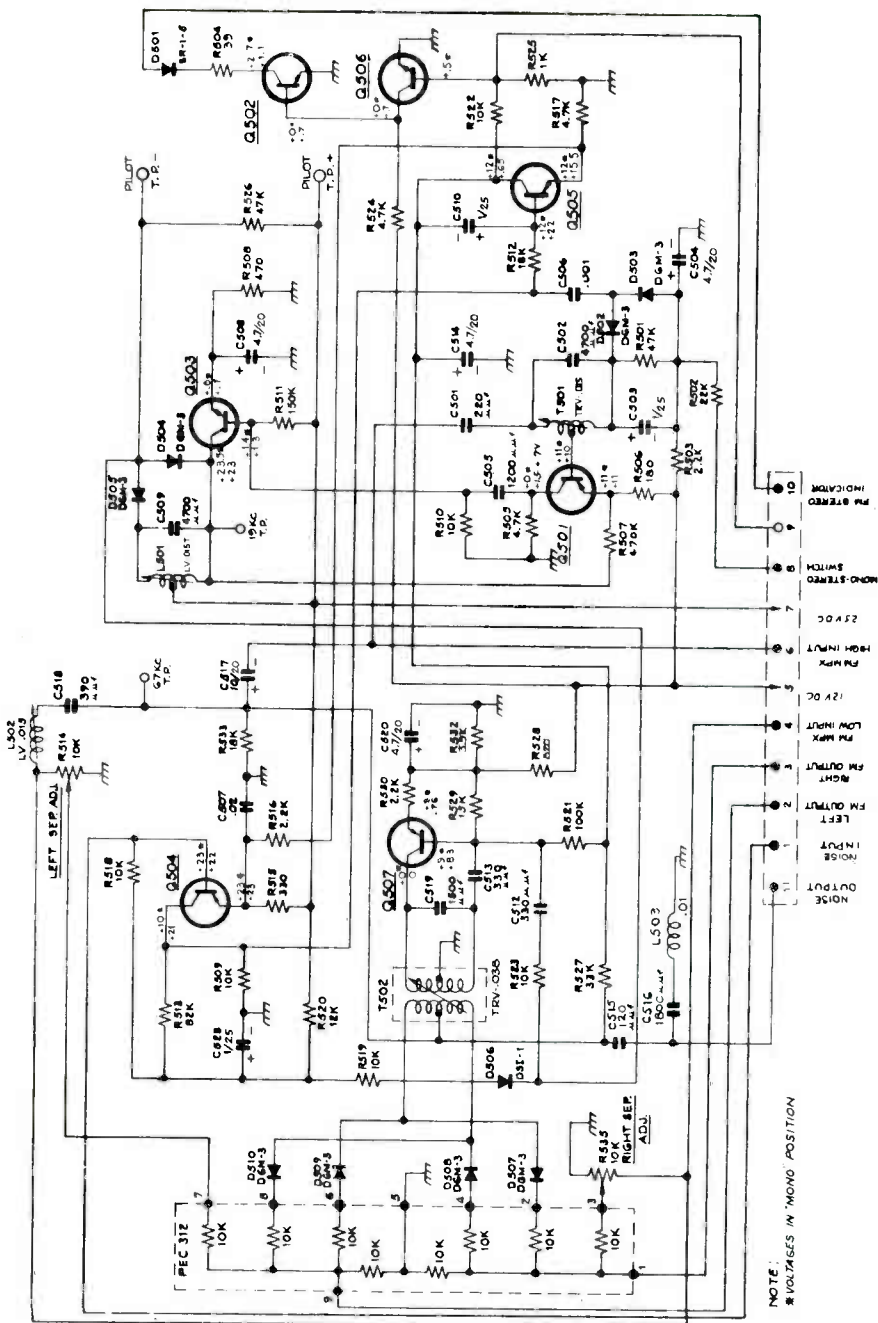
Fig. 4-47. Scott transistor multiplex circuit with indicator output.

93
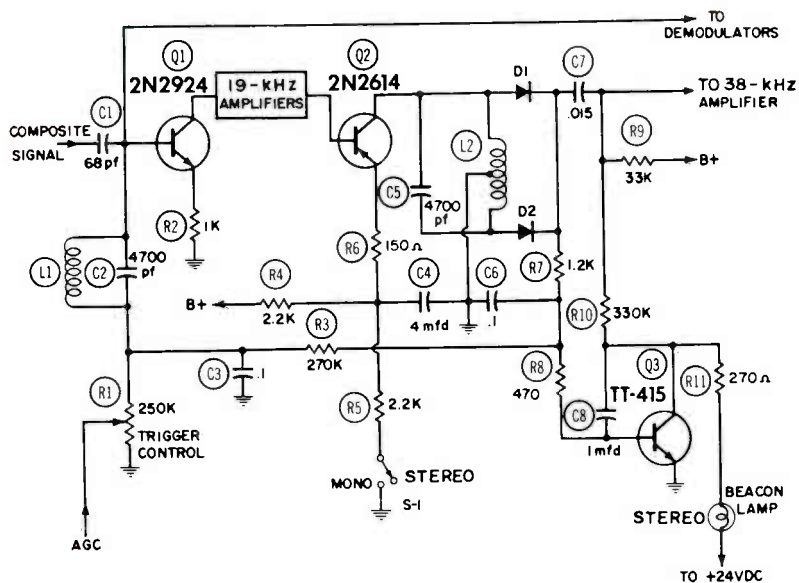
Fig. 4-48. Fisher Model 1249 beacon and automatic stereo circuit.

in one head, requiring one less unit. The tape is generally transported from a reel through the erase, recording, and playback heads, drawn by a capstan and pressure roller driven by a constant-speed motor.

The tape is finally wound on a second reel, usually operated by a separate motor (see Fig. 4-49). As the tape is pulled through the re-
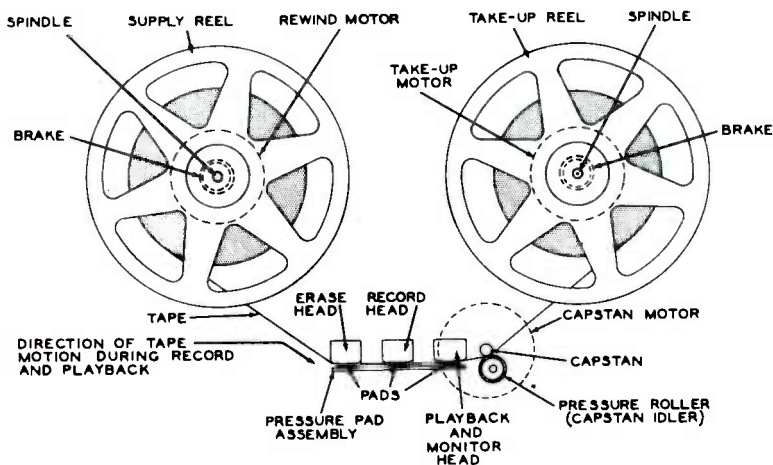
Fig. 4-49. Major parts of a tape-transport mechanism.

cording or playback head, variations in flux density in the gap surrounding the tape are produced both in recording and playback. When recording, the signal to be recorded is fed to the recording head, causing a changing flux field which magnetizes the particles in the tape. On playback, the moving magnetized particles in the tape cut the path of the field of the pole pieces in the playback head and induce a signal voltage proportional to the residual magnetism of the particles at any point, thereby inducing a reproduction of the recorded signal.
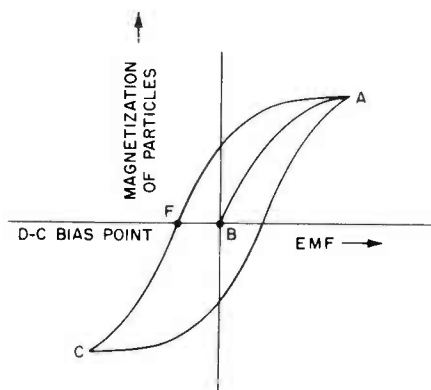
Fig. 4-50. Tape magnetization response.

The problem of producing a linear recording on magnetic material is similar to that of producing linear output from a class-A amplifier. The magnetization (hysteresis) curve shown in Fig. 4-50 has a changing slope with only a portion straight enough to use for hi-fi reproduction. The residual curve shows the relative magnetization left after the medium (tape) has left the magnetizing field. Just as in the operation of a class-A amplifier, only the flat portion of the linearity curve can be used for hi-fi. Thus, the recording must be limited to the flat portion of the residual magnetization curve. This is accomplished by use of bias to change the relative operating level of the recording signal fed to the recording head. This may be done by application of d-c or a-c (supersonic) bias with the signal to the recording head.

When d-c bias is used, a sufficiently positive d-c field is applied just ahead of the recording head to magnetically saturate the tape medium. Then negative d-c bias is fed with the signal to the recording head, sufficient to center the average recording level in the flat portion of the magnetization curve, as shown around point F in Fig. 4-50.

Supersonic (a-c) bias may be applied instead of d-c bias with the signal directly to the recording head. The supersonic signal is above the audible range, in the neighborhood of 40 to 100 kHz—the higher the frequency, the more fidelity is possible.

The mixing effect of the signal to be recorded and the supersonic bias signal is shown in Fig. 4-51. The total signal variation is within the area of linear operation of curve AB as described. The supersonic bias signal is recorded and reproduced along with the audio signal but cannot be heard by the human hearing system. The supersonic bias arrangement is very popular because it has quieter effects at low volume and silent areas of the program and it is easy to produce and adjust.
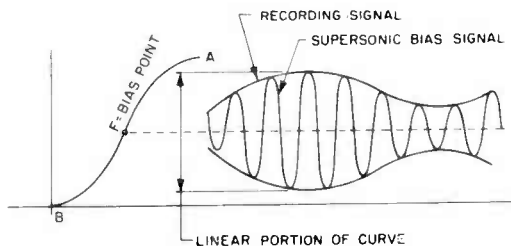
Fig. 4-51. Mixing and applying recording and bias signals.

Erase

Demagnetization principles, which have been used in other fields for years, are applied in a precise manner to erase a recorded signal from a tape. By applying an a-c field to a tape in sufficient strength to completely saturate all particles on the tape magnetically, thereby homogenizing its magnetic pattern, and then to gradually decrease the same a-c field to zero strength, the tape will become demagnetized. For tape recorders this a-c field is obtained from the supersonic bias supply. In addition to being an excellent source for such a purpose, possibility of a beat note from a separate supply is eliminated.

Erasure may be applied in several ways. When a tape is recorded on, it is customary to continuously erase all information and noise just prior to the recording operation as is provided for in tape recorders (see position of erase head in Fig. 4-49.) This removes any former programs or noises that may have been on the tape. *Bulk erasing* may be applied to the whole tape all at once, using the same principle but by immersing the tape in an a-c magnetic field large enough to receive the reel. Bulk erase equipment is more powerful than recorder erase units and is used to erase more quickly and to restore tapes overloaded to near a permanent magnetization condition beyond the erase power of ordinary recorder equipment.

Tape-Drive Motors

Tape-drive motors must have properties similar to the very best motors used for phono turntables. Tape drives, however, do not have the advantage of the flywheel effect of a turntable to damp flutter and other

changes of speed of the motor; therefore, perfection of motor design is even more desirable for tape. The best tape machines use hysteresis motors, which cause the machine to cost considerably more. For quality reproduction, motor flutter should have a rating of not higher than 0.5 percent,which is the audible level. For high-fidelity operation the motor should be able to hold all variations in speed to less than 0.5 percent, and the better units are rated at less than 0.2 percent.

The total number of motors in a recorder may vary from one to three. The better machines usually have three—one for feed and rewind, one for take-up, and one for capstan drive.

Heads

The better machines provide three heads, as has been described: one each for erase, record, and playback. Optimum operation requires a separate playback head to not only play back but to monitor the signal as it is being recorded. This provides a simultaneous check on every link in the recording chain so that if anything is wrong one does not have to do it over or lose the program. The erase, record, and playback heads are similarly constructed. However, separate design of heads is desirable because optimum operation of each requires different gap spacing and inductance characteristics. Heads contain a ring-shaped, high-permeability core with a fine gap formed to allow a ¼ inch wide by 0.002 inch thick tape to pass with minimum clearance. The core has an inductance to provide or pick up signals. Tape contacts the core at the gap either to be magnetized or to induce a field according to the operation desired. Low-cost commercial tape-recorder heads will provide reasonable recording and reproduction at voice frequencies but most professional units require better heads. Hi-fi heads, such as Dynamu heads, are available to replace standard heads on lower-cost units. These can record and play back with full response up to 14,000 Hz ± 3 db if the rest of the elements of the recorder are suitable.

Mechanical Features of Recorders

The transport mechanisms and controls of tape machines can be very elaborate. A tape-transport mechanism draws the tape past the heads at a constant rate of speed, viz., the speed at which the tape is to be played back. The tape is unwound from a reel passed through the heads, and wound on to a take-up reel. A capstan with a roller-tension device to press the tape against it pulls the tape at a constant speed. This may be reviewed in Fig. 4-49.

The feed reel has a torque applied in a direction opposite to the tape travel. The take-up reel has a similar torque in the direction of the tape travel to relieve the capstan of any difficulty with the tape. On rewind, the pressure on the tape is freed to release it from the capstan and the feed reel is speeded up to rewind at high speed.

In the better equipment, brakes are usually provided to quickly stop the reels; various speeds are provided for rewind; interlocks are provided to protect tapes from fast rewind speeds during record or playback; electrical cutouts are provided to prevent erase during rewind or playback; release of tape pressure from heads during rewind is arranged to reduce wear; pressure is released on the capstan roller when not in use to prevent a flat space on the rubber idler wheel; and automatic-protection switches if tape breaks or tangles.

Tape decks as separate machines or complete with amplifiers are available. Most equipment can be operated as a separate unit with an outside amplifier feeding the recording head, or the playback head feeding a separate audio system.

Audio amplifiers and controls to feed a recorder of the constant-current type must provide from one-tenth to several milliamperes output. To provide for constant-current output, the output circuit must contain a majority of the impedance of the total coupling network. The amplifier should also contain equalization networks to compensate for the magnetic recording characteristics. This equalization is built in most recorders, but if separate audio systems are used, similar compensation must be used. Compensation for recorders varies for different machines, but essentially each system only has to be arranged to equalize in playback the emphasis and de-emphasis applied to record any tape. Generally, a recording amplifier is compensated to NARTB tape-recording standards, and regular preamplifiers (for reproduction) with NARTB tape-standard equalization are used.

Recording

Better stereo amplifier and control center combinations have provisions for two-channel output for stereophonic (or monophonic) tape recording. These outputs can be fed directly to stereo tape recorders, and live stereo programs can be recorded for permanent storage.

This material can be obtained from actual two-microphone pickup (live) or live stereo broadcasts, or from stereo records. The broadcasts may be fed from a tuner or a record player and pickup cartridge to the amplifier control center, monitored, balanced, emphasized or de-emphasized according to personal taste, and fed to the stereo tape recorder. These tapes provide an optimum medium for permanent storage.

Prerecorded tapes are also available and have advantages over hi-fi records. An important advantage is permanence of fidelity. The quality of program material applied to prerecorded tape is practically permanent. With normal caution, prerecorded tape will not acquire surface noise over hundreds of plays. There is constant retention of its frequency range and stereo effects in its program material.

Records do not have these characteristics. Discs always develop surface noise from needle wear and handling. Lint, dust, and scratches on records also contribute other undesirable noises and loss of quality and stereo effect on records.

The disadvantage of higher cost per unit playing time of prerecorded tapes is being gradually overcome by reduction of speed in relation to fidelity of response. Cartridge tapes with suitable characteristics and equipment operated at 1⅞ inches per second provide high-fidelity output at price per minute competitive with standard LP records.

Playback

Many other recent improvements in techniques and equipment for tape recording and playback have made tape more competitive with discs as a home playback method. Other relative advantages and disadvantages of tape for playback were discussed in Chapter 3.

There are virtually no tape recording machines that are not also useful for tape playback. Thus a tape machine offers the advantage of allowing one to make his own recordings as well as to take advantage of permanent playback characteristics of high-quality prerecorded stereo tapes.

However, just "any old tape machine" is not enough to ensure stereo reproduction that satisfies the audiophile. The machine must be of better-than-average quality with respect to frequency response, freedom from flutter and wow, low harmonic distortion, and high signal-to-noise ratio.

All good stereo machines are now designed for "four-track" operation. As explained in Chapter 3, this means that stereo sound can be recorded on the full tape length in both directions. The four-track arrangement requires special heads, mounted in a single head assembly over which the four-track tape passes.

Most tape machine manufacturers sell both complete recording and playback systems and separate "tape transports." A *tape transport* is the mechanical assembly of drive motors, drive system, and the heads and bias-generating equipment to derive a signal from or record a signal on the tape. It does not include the equalizing networks, amplifier(s), and speaker(s) that come with a complete recorder-playback unit. The transport can be used to play a stereo tape, with the outputs of the two playback heads fed to the inputs of the preamplifier of a stereo high-fidelity system. Just about all preamplifiers include a position on the input equalized switch for tape playback. However, for recording with your tape transport a separate amplifier may be necessary.

Typical high-quality tape machines for stereo recording and reproduction are illustrated in Fig. 4-52. Most machines operate at either of two speeds, with the speed selectable by the operator. A few operate at either 15 inches per second (ips) or 7½ ips, others at either 7½ ips or 3¾ ips. Still others include a third speed: 1⅞ ips. It is an inherent property of the tape method of recording for frequency response to improve with
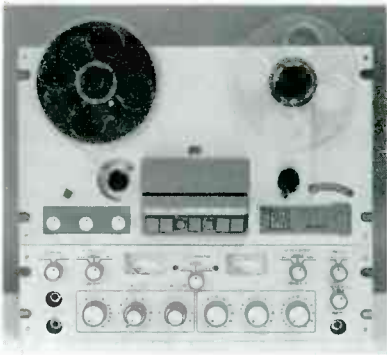
99

(A) *Magnecord 1024.*

(B) *Revox recorder.*
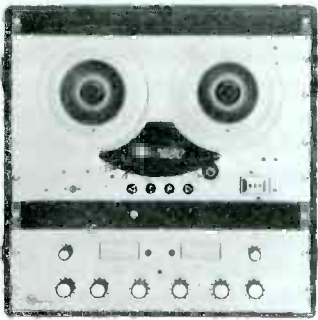
(C) *Viking Model 230 stereo recorder.*

(D) *Tape reproducer for four-track stereo only.*

(E) *Lear Jet eight-track stereo cartridge tape deck.*
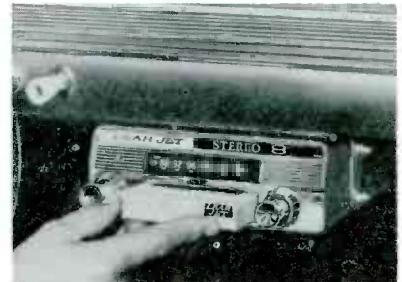
(F) *Lear Jet eight-track stereo cartridge player with f-m radio for car.*

Fig. 4-52. Typical tape machines for stereo recording and reproduction.

speed, and a check of manufacturers' specifications will show this. However, it must be remembered that as speed increases, playing time for a given length of tape decreases. Also, a two-track stereo tape, which plays back only in one direction, provides only half the playing time on stereo as a four-track tape. Table 4-1 shows playing time of tape for several different modes of operation. It is important to keep this information in mind when you are examining claims of playing time.

Table 4-1. Playing Time of Tape in Hours

	Monophonic Playback		Stereophonic Playback	
	3 3/4 ips	7 1/2 ips	3 3/4 ips	7 1/2 ips
Single Track	1	1/2		
Dual Track	2	1		
Twin Track	2	1	1	1/2
Four Track	4	2	2	1
Eight-Track Stereo Cartridge			1 1/3	

Claims as to frequency response at a given speed vary from one machine and manufacturer to another. It should be noted whether the response is claimed with a variation of 2 db, 3 db, or just on some "usable response." Machines using a speed of 15 ips obtain better response with less technical difficulty at that speed than machines using lower speeds. Response at that speed should be from about 30 to 40 Hz to 20,000 Hz or more with variation in the order of plus or minus 2 db. The upper limit of response ($\pm$ 2 db) at 7½ ips is usually about 12,000 to 18,000 Hz, and at 3¾ ips between 7500 and 12,000 Hz. New designs operating at 1⅞ ips are providing comparatively excellent frequency response. Don't let a few hundred cycles of upper-limit frequency response be a final criterion in choice of a machine, because manufacturers differ in the rigidity and manner of making their tests. Listening is the best test.

Total harmonic-distortion tolerances are about the same as those for amplifier equipment alone, and about 2 percent should be considered maximum.

Signal-to-noise ratio is important in tape machines because a relatively high noise level is one of the basic problems tape machine manufacturers have had to overcome. A minimum signal-to-noise ratio acceptable to the high-fidelity enthusiast is at least 40 db, and 50 or 60 db is considered highly desirable.

As explained in Chapter 3, one of the disadvantages of tape machines has been the inconvenience of having to thread the head assembly and set up the reels properly. For this reason, cartridge-type tape machine arrangements have been developed. The cartridge is loaded with stan-

dard ¼-inch tape and plays up to an hour of stereo at 3¾ ips. Threading of the tape is not necessary because the tape is already "set up" in the cartridge. The cartridge is merely inserted or removed as desired. When the end of the tape is approached in playing, the machine is automatically shut off, so the tape does not become detached from the reels inside the cartridge.

Figs. 4-52E and F show the Lear Jet 8 eight-track stereo cartridge players, which have a comparatively flat response at frequencies upwards to 10 kHz. As shown, a model with f-m radio included is available for the car, and another model is available as a tape deck for the home. The car model operates on 12 vdc, and the home model operates on 115 vac. These units accept cartridges which can play continuously, with one hour and 20 minutes of program material which will repeat until turned off. The tape deck may be indexed (advanced) to any one of four portions of the program material by depressing a button on the panel.

MICROPHONES

There are many types of microphones available. Each has certain advantages and disadvantages. The type of material to be reproduced, the placement of the microphone, whether it is to be used indoors or outdoors, the frequency response desired, and a number of other factors, affect the choice of a microphone.

The basic types of microphones, grouped according to their principle of operation, are:

1. Carbon
2. Crystal
3. Dynamic
4. Ribbon
5. Capacitor

Each of these microphone types has its own characteristics with respect to: (1) output level, (2) frequency response, (3) output impedance, and (4) directivity. These characteristics determine whether or not a microphone is suitable for a given application.

Output Level

The output level of a microphone is important because it governs the amount of amplification that must be available for use with the microphone. The output level of microphones is usually given in db preceded by a minus sign. The minus sign means that the output level is so many db below the reference level of 1 milliwatt for a specified sound pressure.

The unit of sound pressure used in rating microphones is referred to as a *bar*. A bar is equal to a sound pressure of 1 dyne per square centimeter. Speech provides sound pressures between 0.4 and 15 bars. For music, the pressure ranges from 0.5 bars to 1250 bars.

102

Microphones are rated in a number of different ways, and this often causes confusion. If ratings are given in any manner other than in bars, it is a good idea to convert their output level rating to db below 1 milliwatt for a sound pressure of 1 bar.

Table 4-2 gives correction factors which, when applied to the corresponding method of microphone rating, will convert it to output level in db below 1 milliwatt for a sound pressure of 1 bar. When a rating has been converted to these terms, it is much simpler to use when calculating amplifier gain requirements and the like.

Table 4-2. Comparison of Microphone Ratings

Rating Given	Correction Factor
db below 1 mw/1 bar	0 db
db below 1 mw/10 bars	− 20 db
db below 1 volt/1 bar	2 db
db below 1 volt/10 bars	− 18 db

A microphone with a low output level necessitates the use of an amplifier with greater gain, which, in turn, increases the possibility of noise and hum. The absolute minimum noise level which can be practically attained at the grid of the input tube of an amplifier is about −125 db. From this, it has been determined that to have a reasonably quiet installation, the microphone level should not be below −85 db.

When very low level microphones are used, it is often necessary to provide a direct-current heater supply for the input tube, in order to eliminate hum which results when an a-c heater supply is used.

Frequency Response

The frequency response of a microphone is a rating of the fidelity of relative output voltage which results from sound waves of different frequencies. The simplest way to find a complete picture of the frequency response characteristics of a microphone is to plot a curve of its output voltage versus input frequency. Since good modern microphones are relatively flat over their range, it is often considered sufficient to specify the range over which their output does not vary more than plus or minus 1 or 2 db.

For ordinary home high-fidelity use, a microphone frequency-response curve should be reasonably flat between 40 and 10,000 Hz. With systems designed specifically for speech reinforcement, a lower limit of 150 Hz and an upper limit of 5000 Hz is entirely satisfactory. Where it is desired to reproduce music with the highest possible fidelity, the frequency response should be flat (within 2 db) from about 40 to 15,000 Hz. Fig. 4-53 shows the response of several types of microphones.

103
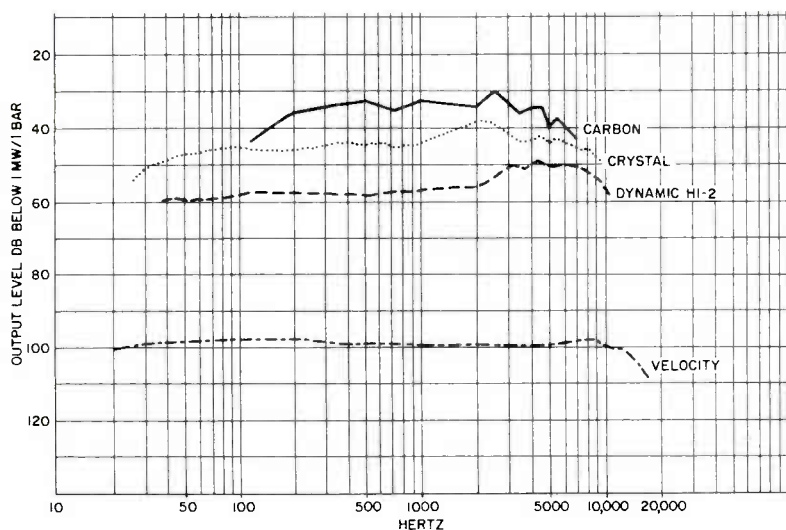
Fig. 4-53. Frequency response curves for typical carbon, crystal, dynamic, and velocity microphones.

Output Impedance

The output impedance of dynamic or velocity microphones requires a transformer or network to match the input impedance of the amplifier. Higher-impedance mikes, such as the crystal types, require only a simple network. Microphones generally employed in public address systems have impedances of from 20 to 500,000 ohms.

Directivity

Microphones do not respond equally to sounds reaching them from all angles. Their frequency response characteristics also vary, depending on the angle at which the sound reaches them. A microphone may respond equally to all frequencies between 40 and 10,000 Hz when the sound is originating directly in front of it, while the high-frequency response falls off rapidly as the sound originates farther to either side. Where it is necessary to pick up sound from all directions, the directional characteristics of some microphones are not suitable.

Fig. 4-54 shows examples of the four important directivity characteristics which can be obtained with the various types of microphones.

The directional characteristics of a microphone can be used to accomplish a number of things. Noise pickup can be reduced by choosing and placing the microphone so that it will not respond to sound originating at the point where the noise is produced. Feedback, which can be very troublesome, can often be completely eliminated by the careful choice and placement of a microphone.

104

NONDIRECTIONAL   SEMIDIRECTIONAL

CARDIOID   BIDIRECTIONAL
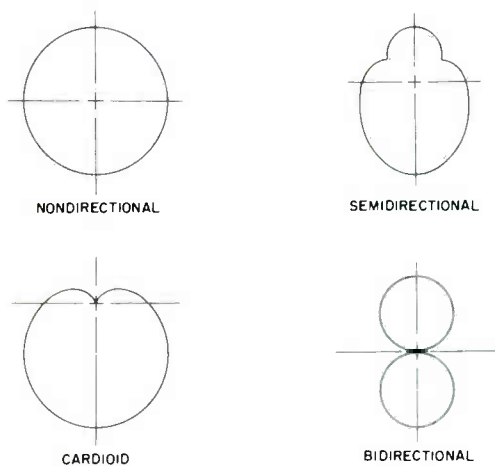
Fig. 4-54. Microphone polar response patterns.

Carbon Microphones

When the maximum output level is required from a microphone, the carbon microphone is often used. While it does have the advantage of high output, the frequency response characteristics of the carbon microphone are poor and cannot be used for hi-fi work.

The carbon microphone consists essentially of a diaphragm and a small cup filled with carbon granules. Fig. 4-55 shows the construction of a typical carbon microphone.

A carbon microphone generates a continuous hiss. This hiss is due to small variations in contact resistance which take place between the carbon granules.
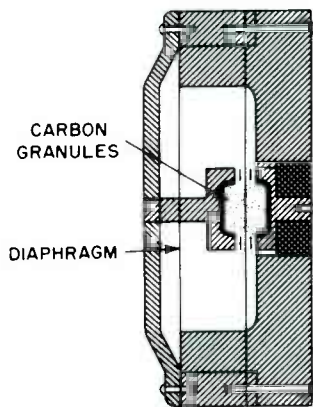
Fig. 4-55. Construction of a carbon microphone.

CARBON GRANULES

DIAPHRAGM

The average output level of carbon microphones is of the order of −30 vu (db). The best carbon microphones have a frequency response of approximately 60 to 7000 Hz. They are substantially nondirectional, although their high-frequency response above 300 Hz usually falls off at angles exceeding 40 degrees from the front of the microphone. Although carbon microphones are not used for high-fidelity work, their ruggedness, low cost, and high output does make them useful in a few cases.

Crystal Microphones

The crystal microphone is the type most widely used in lower-cost installations. The crystal microphone has a relatively high output level and a high impedance. The impedance of the crystal microphone is high enough so that it can be connected through a short cable directly to the grid circuit of a basic amplifier, eliminating the need for an input transformer and preamplifier. A long cable will reduce the output voltage available from a crystal microphone and may effect its high-frequency response.

The most commonly encountered type of crystal microphone employs a diaphragm which moves in accordance with second waves striking it and exerts pressure on the crystal. (See Fig. 4-56.) This type of construction permits complete enclosure of the crystal and reduces the effects of humidity.

The output level of this type of microphone is usually between −48 and −60 db. Their output and impedance is almost always more than 100,000 ohms.

The crystal microphone is normally nondirectional, although a special pressure-gradient crystal microphone which gives a unidirectional
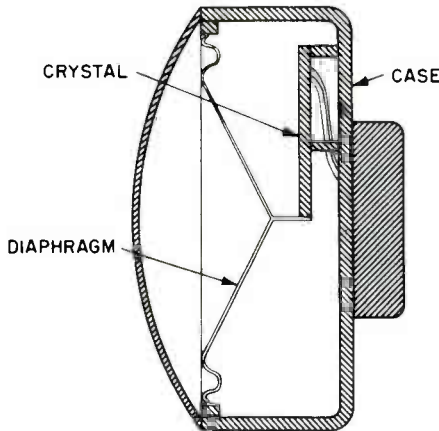
Fig. 4-56. Construction of a crystal microphone.
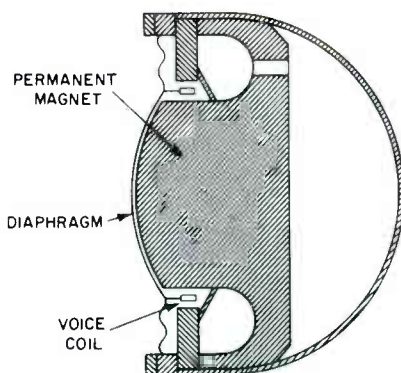
Fig. 4-57. Construction of a dynamic microphone.

response pattern is now being marketed. This microphone gives excellent results.

Good units may have a frequency response substantially flat between 50 and 10,000 Hz. Units are available with slightly wider frequency response ranges.

Rochelle-salt crystal microphones should not be used in locations where the humidity is extremely high. They should never be subjected to high temperatures. If such a crystal microphone is subjected to a temperature of 130 degrees, it will be rendered completely useless. Care must always be taken to avoid exposing a crystal microphone to direct sunlight for any length of time.

Dynamic Microphones

The dynamic microphone consists of a metal diaphragm, a coil which is connected to it, and a magnet. Its construction and operation are similar to that of a dynamic speaker. When sound waves strike the diaphragm, the coil moves. Since the coil is in the field of the permanent magnet, a current is induced in the coil which is directly proportional to the sound waves striking the diaphragm (see Fig. 4-57). This current constitutes the output of the microphone. Dynamic microphones are available with limited or wide-range frequency response characteristics.

The natural output impedance of a dynamic microphone is between 30 and 50 ohms. Very often a transformer is incorporated in the microphone, raising its output impedance to a value between 200 and 25,000 ohms. The average dynamic microphone is simple and sturdy. It is not affected by atmospheric changes, has a long life, and is well adapted to all-around public address work.

The output level of most dynamic microphones is about 55 or more db below 1 milliwatt per bar. The ordinary dynamic microphone is essentially nondirectional, although its high-frequency response falls off

107

rapidly on either side, as shown in Fig. 4-58. To make full use of a dynamic microphone's frequency range, the microphone should face directly toward the source of sound. A special type of dynamic microphone is available for use when high background noise levels are encountered. The response of these units falls off rapidly as the distance between the microphone and the source of the sound increases.
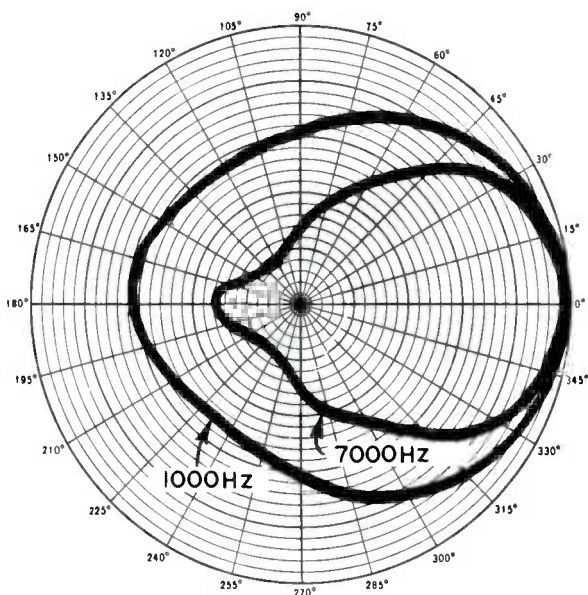
Fig. 4-58. Directivity of a dynamic microphone.

Velocity Microphones

The velocity (or *ribbon*) microphone consists of a very thin ribbon of aluminum foil suspended in the field of a powerful permanent magnet, as shown in Fig. 4-59. The ribbon is corrugated and can move quite freely. The ribbon moves in accordance with the velocity of the sound wave. Response is proportional to the difference in sound pressure between the two sides of the ribbon.

The natural impedance of the ribbon element is about ¼ of an ohm. A transformer is usually mounted within the microphone case, stepping up the impedance at the microphone terminals to a value between 25 and 35,000 ohms. For public address use, the high-impedance units are convenient since they can be connected directly to the grid of an input tube.

The output level of velocity microphones is usually 60 db below 1 milliwatt per bar. Generally, velocity microphones have excellent response characteristics.

108

Fig. 4-59. Construction of a velocity microphone.

ALNICO MAGNETS
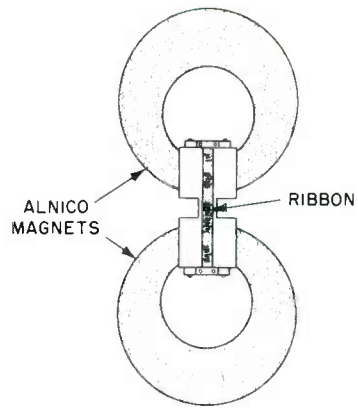
RIBBON

The velocity microphone is bidirectional. Maximum response is to sound reaching the front or back of the microphone at a 90-degree angle to the plane of the ribbon faces. More directional than the crystal and dynamic microphones, the overall response of the velocity microphone falls off as the angle of the sound reaching it varies from 90 degrees to the faces of the ribbon (Fig. 4-60).

The velocity microphone is quite sensitive to the movement of the air surrounding it, and it must be carefully protected from puffs of wind
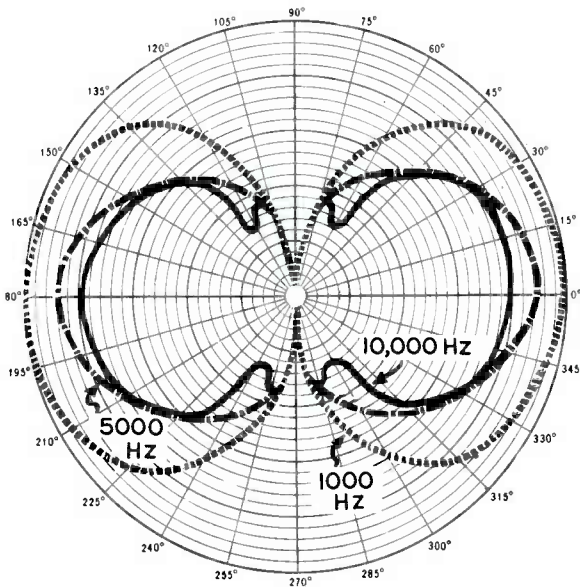
Fig. 4-60. Directivity of a velocity microphone.

when used out-of-doors. A ribbon microphone should be at least 18 inches from the source of the sound.

Capacitor Microphones

The capacitor (or *condenser*) microphone consists of a fixed plate and a diaphragm, as shown in Fig. 4-61. The diaphragm is actuated by the changing pressure of the sound waves striking it, causing the diaphragm to change its position in relation to the fixed plate. This results in a change in the capacitance between the diaphragm and the plate, which is utilized to produce a corresponding voltage drop across a resistor connected in series with the microphone and a charging source.
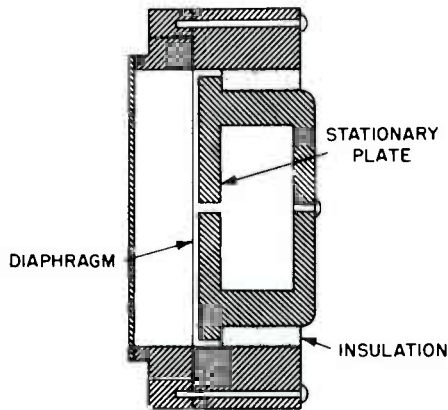
Fig. 4-61. Construction of a capacitor microphone.

The output level of the capacitor microphone is extremely low, and a high-gain amplifier must be used with it. The amplifier should be mounted directly at the microphone, usually right in the microphone case. The capacitor microphone has very excellent frequency response and low distortion. Because of the necessity of mounting an amplifier at or in the microphone case, the microphone is not recommended for ordinary hi-fi work.

Combination Microphones

Microphones are available which make use of two units to secure a particular directional pattern. A dynamic unit is often combined with a velocity (or ribbon) unit. Fig. 4-62 shows the directional pattern which results when a bidirectional velocity unit and a nondirectional dynamic unit are combined. The resultant directivity pattern is known as a *cardioid*, since it is heart-shaped. Other units are also combined to secure similar directivity patterns. Another type of combination microphone, designed especially for stereo recording, is pictured in Fig. 3-4.

110
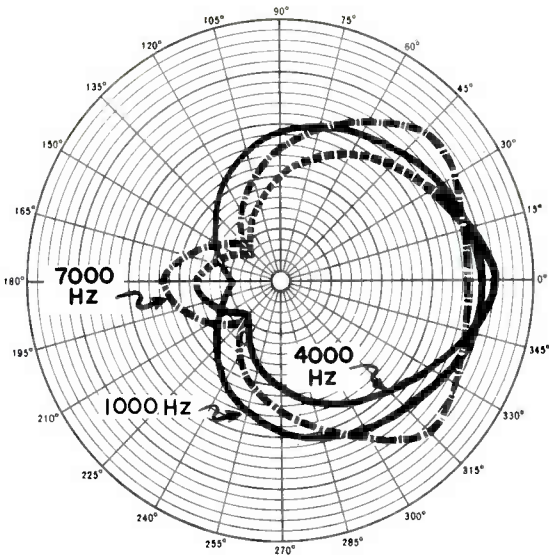
Fig. 4-62. Directivity of a dual-unit microphone.

Microphone Selection

Microphones should be carefully selected in order to utilize the electrical and physical characteristics of the various types, as described previously. There are no particular rules which can be strictly adhered to in the selection of a microphone. There are, however, a few points which should be kept in mind.

A microphone should be selected with frequency response characteristics equivalent to those of the other components in the system. A Rochelle-salt crystal microphone should never be used where it is likely to be subjected to a temperature of more than 120 degrees.

Cardioid microphones should be used when "behind-the-mike" pickup must be eliminated. Fig. 4-63 shows the application of a cardioid microphone on a speaker's platform. The back of the microphone faces the audience. Since the microphone is not sensitive to sounds reaching it from this direction, no audience sounds will be picked up and amplified through the system.

In systems where a speaker must move about a great deal, the lapel microphone is very useful. A number of contact microphones are available for use with string instruments.

Many installations require the use of more than one microphone. An example of this is illustrated in Fig. 4-64. Three microphones are used to pick up sound originating at three different points. Microphone A picks up sound from a soloist, microphone B picks up sound from a chorus, and microphone C picks up sound from an orchestra.

111
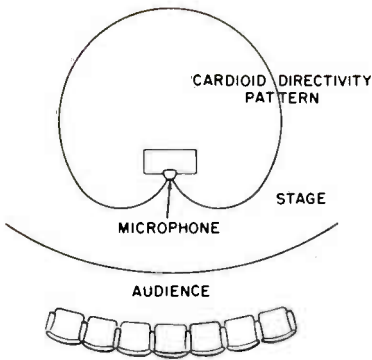
Fig. 4-63. Use of cardioid microphone.

The microphones used have been chosen and placed so that they will pick up the designated sound only. In other words, the microphone in front of the orchestra will pick up sound from the orchestra, but not from the chorus or the soloist. This makes it possible to control the pickup from the three elements; that is, the orchestra, the chorus, and the soloist individually, so that each may be given the proper degree of reinforce-
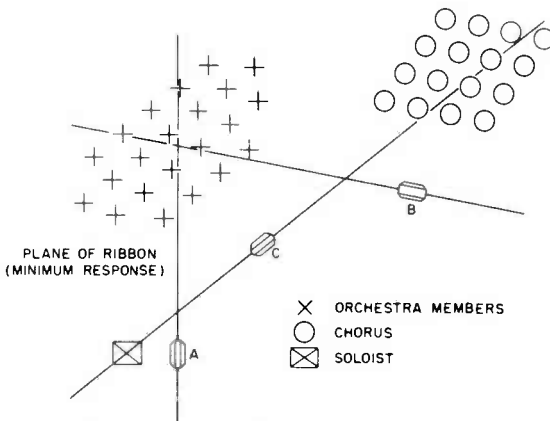
Fig. 4-64. Microphone placement plan.

ment. A microphone setup such as this is particularly useful in adjusting the level necessary for a vocalist, since a vocalist requires a greater degree of sound reinforcement than does an orchestra.

Fig. 4-65 shows another multiple microphone installation. Here, a unidirectional cardioid microphone is used for overall pickup. It faces the orchestra so that the audience is in the dead area of the microphone. A dynamic microphone is used for pickup from a master of ceremonies or vocalist. A third microphone, this one a velocity microphone, is used to pick-up sound from the violin section of the orchestra.

112

Fig. 4-65. Multiple-microphone installation.
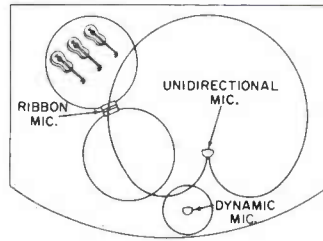
In addition to the differences in pickup patterns, microphone sizes and shapes also vary greatly. In selecting a microphone, consider what uses it will be put to, then select one with the response pattern that will fill the need. Many are very versatile. For example, the unit pictured in Fig. 4-66 can be held in the hand, worn around the neck on a lavaliere cord and clip, or supported by a desk or floor stand.
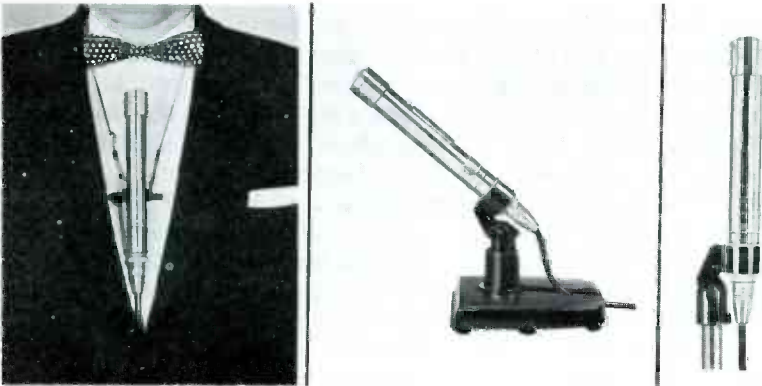
Courtesy Shure Brothers, Inc.

Fig. 4-66. Shure Model 535 dynamic microphone.

TELEVISION

Another source of program to feed a hi-fi system is the audio signal accompanying each television picture signal. The audio equipment in the average television receiver is not of high-fidelity nature. If one wishes, arrangements can be made to pick off the audio signal after detection from the television receiver and feed it through a hi-fi system. The pick-off point should be at a point as near the second detector as a proper impedance match will allow. The lead to the preamplifier should be short.

One can obtain a separate high-fidelity f-m tuner for television stations having features similar to standard f-m tuners already discussed, except that these tuners cover the television frequencies only and the channels are calibrated on the dial.

113

Amplification and Control

The amplifier/preamplifier section of a hi-fi system includes circuits and components necessary to provide power output, inversion, voltage amplification, preamplification, compensation, and a full set of operating controls.

Most commercially available amplifiers and preamplifiers for high-fidelity performance are designed for stereo, rather than monophonic operation. This actually means that they can be used for either. However, because there will always be some monophonic amplifiers and because a stereo system is really a coordinated dual monophonic system with some modifications, we shall discuss these sections first as applied to monophonic form. Then, the methods of combining and modifying for stereo will be given, with examples of actual units.

Preamplifiers are required to bring the level of low- and medium-level signals up sufficiently to provide a useful input to amplifier units. But their most distinctive function is to provide frequency response equalization and controls for volume, loudness, balance, phase, and bass and treble emphasis or de-emphasis. In other words, the modern preamplifier is also an "audio control center." However, some special-purpose preamplifiers, such as those for amplification and equalization of the outputs of variable reluctance cartridges, do not have controls.

The suitability of an amplifier for an installation depends on a number of factors, the most important being its power output, gain, input impedances, output impedances, frequency response, distortions, controls, and the characteristics of the power supply from which it operates. The requirements of the installation and limitations of other components should be considered before the amplifier is chosen. These factors and requirements will be discussed in Chapter 8, on systems design, selection, and installation.

The amplifier section of a hi-fi system includes circuits and components necessary to provide power output, inversion, voltage amplification, preamplification, compensation, and a full set of operating controls. Basic amplifiers are units with power output stages, inverters, and limited voltage amplification only.

Preamplifiers are necessary to raise the level of medium and low input signals such as from reluctance phono pickups and velocity or dy-

namic microphones, to the minimum input level required for full output of the basic amplifier. Preamplifier-equalizers raise the signal and provide for record equalization.

Audio control centers including preamplifiers, equalizing networks, boost, droop, loudness, compensation, filtering, and switching circuits are more elaborate and usually used instead of the plain preamplifier or a simple preamplifier-equalizer. Lower- and medium-cost amplifiers are available which provide in the same unit for all functions just described. However, it is to be expected that the overall quality will have a relation to the price.

The power amplifier unit or section provides amplification and power to actuate the speaker. Average hi-fi speakers for average living rooms require at least 10 watts of audio output power from the amplifier, to handle most conditions. When several rooms, large rooms, or outside areas are to be supplied with the highest fidelity, as much as 60 watts of power might be desirable in a home installation.

This audio output power is provided by the basic amplifier unit or by the amplifier section of a preamplifier and power amplifier combination unit.

From 50 to 100 db of amplification of very low level signals from pickups and other inputs according to the need, plus compensation and controls to obtain various emphasis, is provided by the preamplifier-equalizer part of the system or by a separate hi-fi control-center unit.

Hi-fi systems are designed around either of these two types of amplifier arrangements, that is, the combination of preamplifier, control, and power amplifier, all in one unit such as shown in Figs. 5-1A and 5-1B, or the control center plus basic amplifier units used separately, as shown in Figs. 5-2A and 5-2B.

Circuit arrangements and components may be somewhat similar in both arrangements, but more flexibility and better results can be expected from separate units. It is easier to provide optimum design for separate units, due to differences in conditions of control, amplification, power,

(A) *Scott* Stereomaster *with 50-watt self-contained power amplifier.*

(B) *Heath master audio control with self-contained power amplifier.*

Fig. 5-1. Stereo master audio controls.

115

vibration, shielding, regulation, and noise problems in each of the separate units, such as the tuner, preamplifier, and the basic amplifier. However, it is a matter of practical system design when a tuner is to be used to decide whether the operating controls are more desirable on the tuner panel, in a separate preamp control panel, or on the amplifier unit itself. The answer to this will depend on the system requirements, the location conditions, and the user's preference.

The preamplifier and control section (either as a separate unit or as part of a complete amplifier unit) may provide controls for power on and off, switching the various possible inputs or combination of inputs into the system, volume, record equalization or compensation, bass and treble boost or droop, filtering, and loudness. This section should also provide amplification of the very lowest input signals to be used to a level sufficient to drive the power amplifier section to full output.

(A) *McIntosh stereo preamplifier.*

(B) *Marantz stereo power amplifier.*

Fig. 5-2. Separate stereo amplifiers.

Record equalization is provided to compensate for pre-emphasis and de-emphasis applied to records in their manufacturing process.

Records are made with all high audio frequencies heavily pre-emphasized, i.e., the recorded volume of the highs is proportionately much higher than the normally recorded mid-range frequencies, for the purpose of reducing noise interference effect in playback. On playback, this emphasized signal overrides or masks the undesirable random noise and needle-scratch noises found in the high audio-frequency spectrum reproduction. When the recording is made, the volume level of the high frequencies above 100 Hz (RIAA, AES, NARTB, ORTHO) is increased as shown by the curve in Fig. 5-3. When the record is played back, the preamplifier can be adjusted to the equalization condition to proportionately de-emphasize these signals in an equal but opposite manner, as shown by the curve in Fig. 5-4, developing an audio output very close to the original input to the recorder.

The same technique is used in f-m broadcasting and accounts for much of the superiority of frequency modulation over amplitude modulation when proper compensation is applied in the reproducing amplifier.
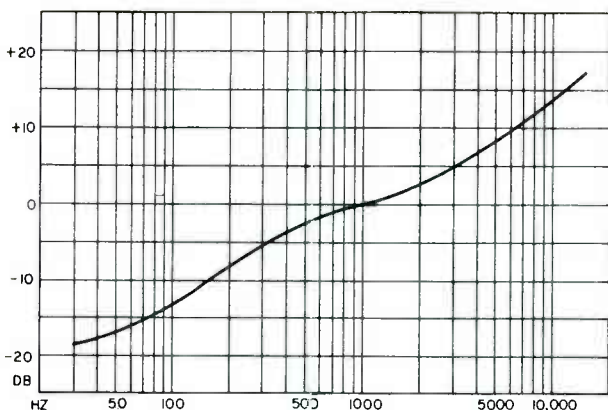
Fig. 5-3. RIAA recording curve.

In a similar but opposite manner the very low frequencies are de-emphasized as shown in Fig. 5-3. This is done to compensate for the wide swings of the record cutter caused by the very low notes. Such wide cuts in the record groove must be reduced or the grooves would have to be made with more spacing, causing less recorded time. Large movements
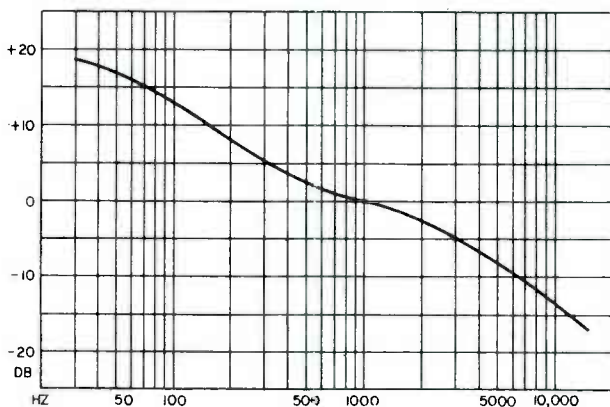
Fig. 5-4. RIAA playback curve.

of the stylus also introduce uncontrollable distortions in the recording head. Emphasis in playback must be provided to compensate for this de-emphasis on the records. Figs. 5-3 and 5-4 also show how this is provided for.

Loudness controls are provided to compensate for the normal variation in response linearity of the ear at different volume levels. It was described in Chapter 1, Fig. 1-1, that as room volume is reduced, reality

117

supposedly becomes more distant because the sensitivity of the human ear to highs and lows drops off more than the "mid-range" frequencies.

The loudness control provides circuits to emphasize the low and high ends of the audio spectrum for the purpose of bringing reality closer as reproduction volume is proportionately reduced. Some amplifiers provide for automatic loudness compensation as volume is varied by use of multiple ganged controls. Other equipment employs switching circuits to provide one or more values of emphasis.

Bass and treble controls provide similar control of emphasis and de-emphasis to that provided by the equalization controls but in a variable manner so that they can be adjusted according to individual taste. These controls, one for treble and one for bass notes, are usually adjustable to provide plus or minus 15 or 20 db variation of either the highs or lows at the ends of the audio spectrum. Advanced equipment provides for roll-off frequency adjustment with constant attenuation slopes and special filters for disturbances such as turntable rumble and objectionable high frequencies.

A balance control provides adjustment of the relative volume output of each of two channels of a stereo amplifier. A phase switch provides for reversal of input of one channel with respect to the other of two stereo channels. This is used to correct the phasing of outputs of pickup cartridges, f-m multiplex outputs, and differences in amplifier output phase if two power amplifiers of different design are used.

There are other controls that may be provided to mix a third channel for three-speaker stereo. These will be discussed later.

How the preamplifier and control sections function will be described in detail following an explanation of the basic power amplifier and its important components.

POWER AMPLIFIERS

Power amplifiers utilize vacuum tubes or transistors to develop the required power to drive the speakers. Vacuum tubes are essentially high-impedance input and output devices. Since speakers are low-impedance devices, a means of transforming the power from a high-impedance output of the vacuum tubes to the low impedance of the speaker is necessary. The device normally used is a transformer, which must be of high quality, or all the care of amplification in a quality manner up to this point in the circuit is lost.

Transistors, on the other hand, have low-impedance characteristics at both the input and the output, and circuits have been devised to direct couple transistor power amplifiers to speakers without the output transformer. (See Figs. 5-18B and 5-18C.) In many cases the output circuits can be designed to closely match a fixed-ohmage load within the output load range that will provide uniform output with extremely low distortion. However, there are applications of transistor audio circuits in which

118

transformers are the best means of matching because of variable load conditions not able to be directly matched properly with transistors.

The most costly and major limiting design factor in power amplifiers is the output transformer. The output transformer is considered to be the most important part of the amplifier, and because of its close relationship to the speaker system we will consider it first. It is one of the most critical components in the high-fidelity system. Modern resistance-coupled amplifier design is such that excellent frequency response and low distortion are inexpensively obtained in the low-level stages preceding the output stage. But at the relatively high power level of the output circuit there are more chances for distortion, and failure to use a high-fidelity type of transformer can nullify all the advantages developed in the careful design of the preceding portions of the amplifier.

In the selection of an amplifier, an amplifier kit, or a circuit for construction, consideration and investigation of the quality of the output transformer is of primary importance. Also, high-fidelity enthusiasts frequently replace output transformers and experiment with their characteristics. Such experimenting often leads to substantial improvement in performance and often allow response adjustments to suit individual tastes. Although the design and construction of these components must of course be left to specialists, the high-fidelity enthusiast will have a keen interest in the factors which make the output transformer so critical and cause its cost to represent a large proportion of his high-fidelity budget. For this reason, the following brief review of the factors most important in the selection of output transformers is included.

Symbolically, and in its basic principle, the output transformer is the same as any transformer designed to couple power to a load from a source. However, unlike ordinary power transformers, output transformers must maintain a high degree of efficiency over a range of frequencies, rather than just at the power frequency, and they must not distort the original signal waveform.

The effects of various factors in the transformer on performance can be most easily visualized by reference to an equivalent circuit, illustrated with the actual circuit, in Fig. 5-5. The equivalent circuit, being in the form of a simple series-parallel connection of inductors and resistors, allows analysis of the effects of any one factor. This equivalent circuit is not complete, because there are also effects from distributed capacitances of the windings, capacitance between windings, and capacitances to ground. Because the output transformer is heavily loaded (low impedance) the capacitances can be neglected unless the design is very poor. The equivalent circuit is made up of two main parts: (1) the primary portion, including effects derived from the primary circuit, and (2) the reflected portion, corresponding to effects reflected from the secondary into the primary. Any impedance so coupled is "transformed" to a new value by the turns ratio; the impedance transformation ratio is

Rp = Resistance of wire in primary winding
Rs = Resistance of wire in secondary winding
Rc = Shunting effect of losses (hysteresis and eddy-current) in the core
RL = Load offered by speaker, including the effects of acoustic loading
Lp = Leakage inductance of primary winding
Ls = Leakage inductance of secondary winding
n = Primary-to-secondary turns ratio
EL = Voltage output at speaker
LM = Primary inductance

Fig. 5-5. Actual and equivalent circuits of an output transformer.

the square of the turns ratio. This is why the reflected inductance and resistance from the secondary are each multiplied by n^2.

Winding Resistances R_p and R_s

As shown by the equivalent circuit, these resistances act in series with the source, adding to its internal impedance. The higher the resistances, the greater is the voltage drop across them at high currents (high output power), and thus the poorer the voltage regulation. Since current through a resistance dissipates power, these resistances represent a power loss in the transformer, reducing its efficiency. The ill effects of excessive resistance dictate that wire as large as possible, consistent with size, weight, and coupling coefficient, be used.

Leakage Inductances L_p and L_s

Leakage inductances also act in series with the source. The reactance they offer to impede the signal depends on frequency, being equal to $2\pi f L$. At low frequencies the leakage reactances are negligible, but at high frequencies they must be minimized, otherwise they interfere with frequency response. Leakage inductance is the inductive effect resulting from the flux of one winding which does not link with the other winding. In good high-fidelity transformers, leakage reactances are minimized by careful attention to the physical shape, mounting, and orientation of the

windings on the core. In some cases, the primary and secondary windings are interwound; that is, a few turns of the primary are wound, then a few turns of the secondary, then some more of the primary, and so on. This is obviously an expensive procedure, and it is one of the reasons for the relatively high cost of a good output transformer.

Core Losses

There are two types of core losses. *Eddy currents* are electric currents generated in the core material due to the fact that it simulates a conductor being cut by lines of magnetic force. The core material has appreciable resistance and thus dissipates power wasted in heating the core. It is because of eddy current that cores are made of many laminations (thin sheets) instead of solid metal. The flux threading through the core is thereby divided into a small portion for each lamination; this, plus the fact that the resistance of the core is increased by the laminations, greatly reduces eddy current losses as compared with those in solid material. The thinner the laminations (maintaining the same total volume of material), the lower are the eddy current losses, but also the more expensive is the construction. The other kind of core loss is that due to *hysteresis*. This is the tendency of the core material to retain residual magnetism and thus resist the positive and negative flux alternations necessary for transformer action. Hysteresis is a function of core material. Much research has been done on determining and developing high-grade core materials. Core materials employed in output transformers for high-fidelity use are of carefully selected materials with minimum hysteresis loss and of a high resistivity as possible to minimize eddy currents.

Both eddy currents and hysteresis losses have the same effect on transformer performance as would be produced by the connection of a shunt resistor across the input to the transformer, as illustrated by R_c in Fig. 5-5. The greater the core losses, the lower is the equivalent resistance of R_c and the greater is its shunting effect.

Primary Inductance

Before a transformer can operate properly, magnetic flux must be set up in the core and be maintained there. This is the job of the primary inductance. It is the inductive reactance of the primary which limits no-load current to a reasonable value. If primary inductance is too low, it shunts the input signal. Since inductive reactance is proportional to frequency ($X_L = 2\pi f L$), its shunting effect is worst at low frequencies, being one of the limiting factors for low-frequency response.

Practical Performance Factors in the Output Transformer

The preceding discussion covers the important factors involved in the performance of the output transformer in general. For high fidelity we are interested in how these factors affect performance in practical,

121

commercially available types. Such effects can be made clearer by re-drawing the equivalent circuit twice, once including only those factors important at high audio frequencies and once for factors important at low audio frequencies. Such equivalent circuits are shown in Fig. 5-6. In transformers of reasonably good quality, core losses become negligible, so they are not shown here. Also, as previously mentioned, distributed and other capacitances, important in the high-impedance interstage transformer, are negligible here because of the relatively low impedance involved.

$R_T = R_p + n^2 R_s$
$L_A = L_p + n^2 L_s$

(A) High frequencies.   (B) Low frequencies.
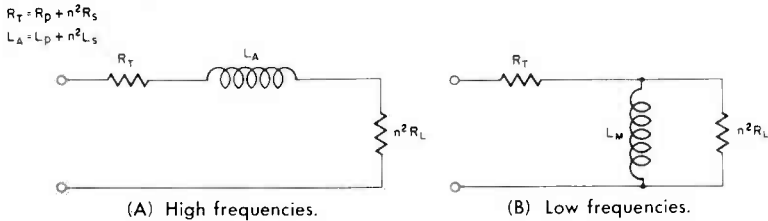
Fig. 5-6. Effective equivalent circuits at high and low frequencies.

In Fig. 5-6A it can be seen that the combined leakage reactances form the important high-frequency factor. If equivalent combined leakage inductance L_A becomes excessive, most of the input voltage at high frequencies appears across it, rather than across the load. At the same time, a large L_A causes phase shift and phase distortion of the output signal. The closer the coupling coefficient between the primary and secondary is to one, the lower is the leakage inductance. Manufacturers of high-fidelity transformers employ special methods of interwinding the turns to minimize leakage inductance.

In Fig. 5-6B it is apparent that the important low-frequency factor is the primary inductance L_M. It was not included in the high-frequency circuit of Fig. 5-6A because its reactance is so high at high frequencies that it appears as an open circuit. At some low frequency, however, the reactance of L_M becomes low enough to cause an important shunting effect across the reflected load. This means a loss of low frequencies and a phase shift at low frequencies. For this reason, manufacturers of high-fidelity output transformers try to use as many turns as possible. But in doing so, they make it harder to avoid more leakage inductance, so a careful compromise between the two must be made. By careful use of interwinding and choice of physical dimensions and shapes, both of these factors are made much more favorable in high-fidelity transformers than in the run-of-the-mill types.

Choosing and Testing Output Transformers

The high-fidelity enthusiast who wishes to purchase an output transformer is faced with an array of commercially available units, ranging

122

in cost from about a dollar to as much as one-hundred dollars or more. Tiny, cheap transformers made to a price for low-cost radio receivers can be ruled out immediately since they can hardly be considered as high-fidelity equipment. Also, all high-fidelity systems worthy of the name use push-pull output; therefore, all single-ended transformers can be eliminated from our considerations.

From here on, it depends on the person himself and the limits he has set on his system as a whole. The speaker system is of first consideration in this respect, since most amplifiers preceding the output transformer are so high quality that they are not a factor. Using a high-grade output transformer with a relatively poor speaker system does not allow full realization of the transformer's capabilities. A transformer with performance comparable to that of the speaker is cheaper and sounds no worse. On the other hand, if you intend to change to a speaker of better quality in the future, it is often wise to get the best possible transformer in the beginning, and thus eliminate the transformer as a serious limiting factor in later changes and tests.

If a top-notch transformer is in order, the purchaser is quite safe in buying one of two or three special types known by high-fidelity men everywhere. Companies such as those whose wide reputation rests primarily on such a unit produce a uniform and dependable product.

Few systems warrant the ultimate in transformer specifications, and few pocketbooks can stand making the whole system the best available. Among more moderately priced output transformers, manufacturers are not always definite about performance. For example, sometimes the drop-off in response over the specified frequency range is not stated. If it is 2 db or less, then the range means something. Examples of good and poor response curves are shown in Fig. 5-7. The curves actually show relative response with respect to response at 400 Hz. In other words, to compare only frequency response, and not total output, the response at 400 Hz (and thus over the entire middle portion of the range) is plotted along the same line, even though output level under
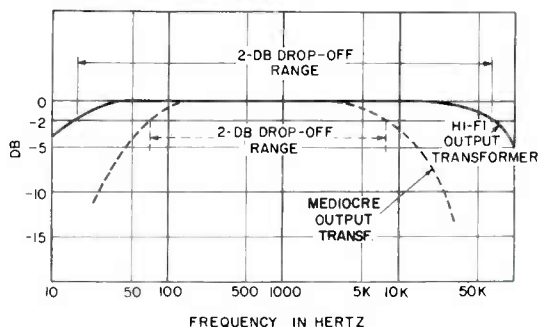
Fig. 5-7. Output transformer response characteristics.

particular actual operating circumstances may be different. Note how much greater a frequency response range could be claimed if the drop-off is taken at 5 db or 10 db.

Some specifications do not take into account the tolerance in the balance between the two halves of the primary. If there is unbalance, distortion is introduced in proportion to the degree of unbalance. Also, unbalance will cause the magnetic flux from the two halves of the primary to fail to balance in the core; unbalance in the fluxes causes possible overload and core saturation, with magnetic overload distortion, hum, and overheating.

Because these things are not apparent in over-the-counter buying, the high-fidelity enthusiast may wish to be prepared to make certain basic tests on a transformer. For this purpose, the following checking and testing procedures are given:

1. *Check the relative weight.* Heavy transformers are not necessarily good ones, but good ones for high-fidelity performance are heavy. The low-frequency response depends on core volume. While it is true that the efficiency of core materials has been greatly improved of late, it still takes a husky chunk of iron to do a good job down to 50 Hz. If you have examined and handled top-notch transformers, you have developed an idea of what the weight must be, and can make comparisons.

2. *Try it in an amplifier.* Listening is, of course, the ultimate functional test, providing conditions are so controlled that any distortion heard can be attributed to the transformer. However, testing equipment is required to make other than an aural test. To make laboratory-type tests, you should know the characteristics of the amplifier used, because such things as unbalance in the output tubes or other inherent amplifier imperfections can introduce distortion which might mistakenly be blamed on the transformer. If these characteristics of the amplifier are known, then you can temporarily hook the transformer into it and make tests as shown in Fig. 5-8. Run tests on amplitude distortion and frequency response. These tests check for unbalance and other deficiencies automatically.

3. *More advanced tests with an impedance bridge.* It is not always handy to install the prospective transformer in an amplifier, or perhaps you are building your first amplifier and do not have a standard amplifier, but you still want to buy and check your transformer before installation. In this case, there are a few tests which, though far from a complete analysis, give a good idea of whether you are buying a well-designed component. These tests call for an impedance bridge. The method of testing is illustrated in Fig. 5-9. It is as follows:

(A) First check for unbalance, because if there is too much unbalance the transformer is unsuitable and further tests are not necessary. Unbalance is checked as shown in Fig. 5-9A. The resistance and induc-
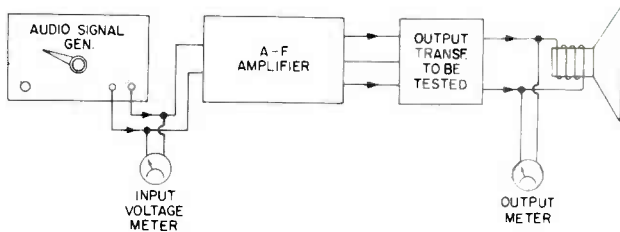
Fig. 5-8. Setup for output transformer tests.

tance of each half of the primary winding, with the secondary winding open, are measured. The percentage difference between them is the percentage of unbalance. Unbalance of 1 percent is considered low enough to be good, 5 percent is only fair, and more is poor.

(B) Now check total primary inductance, as illustrated in Fig. 5-9B. The reason this test is appropriate will probably be more clear on re-examination of Fig. 5-5. If, in this equivalent circuit, the load is removed (to simulate an open secondary), it can be seen that the only inductances in the circuit are the leakage inductance of the primary and the primary inductance. The leakage inductance, even in relatively poor transformers, is very much less than primary inductance, and it is therefore negligible in this test. Thus, the measurement of the inductance across the primary leads with the secondary open is a reasonable check on primary inductance. However, because the bridge measurement is at low signal level, it makes no check on the inductance at high audio levels. How the in-
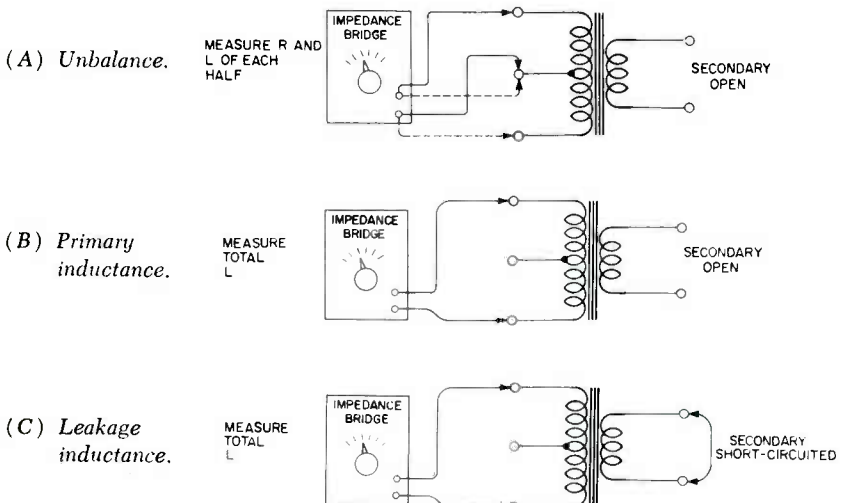
Fig. 5-9. Simple tests on output transformers.

125

ductance holds up under increases of signal amplitude depends on the adequacy of the core. If the weight, volume, and material of the core are sufficient, the bridge measurement of inductance should be a good indication of primary inductance. The minimum acceptable value is about 15 henrys; the method of interpreting the measurement is explained in the following paragraphs.

(C) *Check leakage inductance.* This is done by short circuiting the secondary winding and again measuring the inductance across the total primary winding, as illustrated in Fig. 5-9C. By reference to Fig. 5-5, it can be seen that short circuiting the load shunts the primary inductance with the relatively low value n^2L_S, so effectively there is just the leakage inductance in the circuit. Now our bridge measurement indicates a relatively low value. Maximum leakage inductance allowable in a good transformer is of the order of about 75 millihenrys.

Frequency response is usually stated in terms of the range between the frequencies at which the output falls off ± 1 db. These limiting frequencies are: (1) that low frequency at which the primary inductance has a reactance equal to the load impedance (the load impedance is that offered to the output tubes for proper operation and is listed as such in the tube manuals), and (2) that high frequency at which the leakage reactance becomes equal to the load impedance. For example, if the load impedance is 6000 ohms, a primary inductance of at least 19 henrys is required to reach 50 Hz at low-frequency drop-off ($X_L = 2\pi fL = 6.28 \times 50 \times 19 = 5970$ ohms). For the same impedance, 95 millihenrys leakage inductance would allow response to 10,000 Hz, while 63 millihenrys allows response to 15,000 Hz. For the convenience of the reader, graphs showing upper and lower response limitations, respectively, for typical values of load resistance and a range of inductances are included in Figs. 5-10 and 5-11.

It must again be emphasized that the foregoing relations hold only if the inductances are correct for at least normal operating levels. This is where the power rating of the transformer becomes important. For instance, if a transformer is rated at 5 watts, and is to be operated at 10 watts, distortion is likely to result because the core is overloaded. The core will saturate and the peaks of the audio waveform will be flattened, as illustrated in Fig. 5-12. It is good policy to operate well below the power rating of the transformer to allow for good dynamic range in the system. In other words, although normal operating average power may be about 5 watts in a typical case, the peak power on momentary loud passages may go to several times that.

Output Impedance

The output impedance of an amplifier should equal the input impedance of the speaker system to be driven. For universal application it is

126
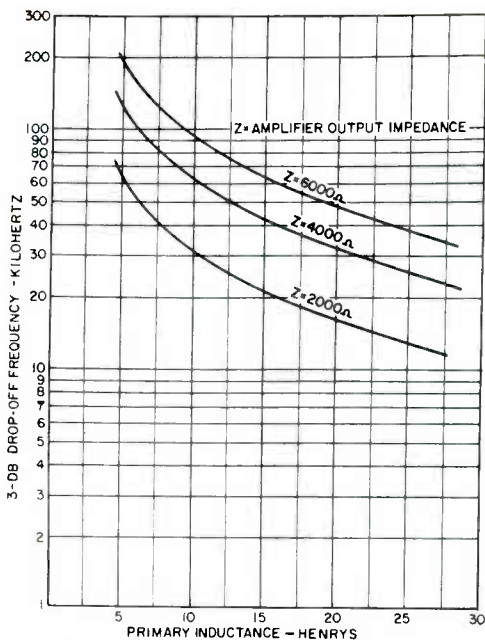
Fig. 5-10. Primary inductance required *vs.* minimum response frequency.
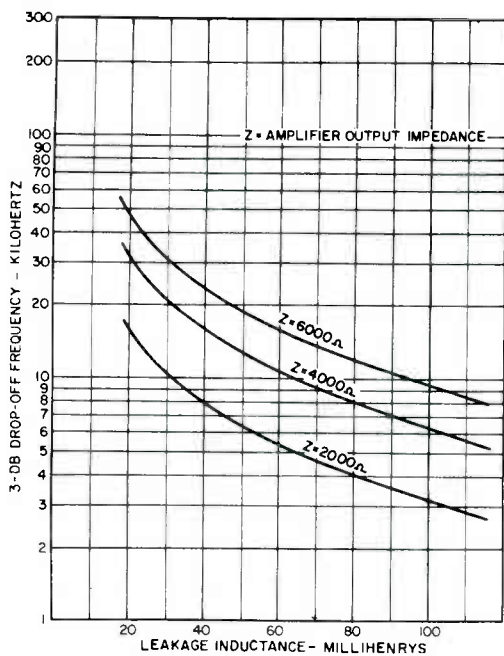
Fig. 5-11. Maximum leakage inductance *vs.* highest response frequency.

desirable to choose an amplifier having a transformer with a range of output impedance including one to match a 500- or 600-ohm line. This facility is useful when it is desired to operate a speaker at some distance from the amplifier, as considerable loss is experienced in long lines unless a matched-impedance feed line of 250 ohms or more is used. If a 500-ohm line impedance is used, it is necessary to use a 300- to 500-ohm transmission line feeding a 500-ohm output-impedance transformer to the speaker. This transformer should be placed at the speaker and should have outputs matching the speaker system. This transformer should be equal in overall quality to the output transformer. Standard impedance output values desirable are: 4, 8, 16, 32, and 500 ohms.

(A)   (B)
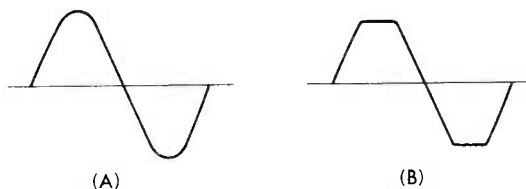
Fig. 5-12. Waveform of a transformer in which the core is saturating.

Characteristics of Tube Versus Transistor Power Amplifiers

The ability of an amplifier to deliver its rated power into a speaker load depends on the impedance match between the two. The advent of solid-state amplifiers, which usually do not use multiple-impedance output transformers, requires proper selection of speakers in regard to impedance.

Vacuum-tube amplifiers, because of the high impedance of the output tubes, require a transformer to match to the typical speaker, and it is easy to include several "taps" for matching speakers of different impedances. Transistor amplifiers, however, generally do not require an output transformer because the efficient matching impedance of a typical transistor output stage is some value between 4 and 16 ohms, close to the nominal impedance of most speakers. A transistor amplifier has a single optimum load impedance determined by its circuit design, and therefore it is not able to deliver full power into a range of fixed load impedances, as is readily obtainable with a vacuum-tube amplifier.

Speakers are not fixed-impedance devices. The impedance of any speaker or speaker system varies over its useful frequency range. The Electronic Industry Association specifies that the rated impedance shall be the minimum impedance over the speaker's useful frequency response. However, even where the manufacturer adheres to this rating procedure, combinations of speakers in systems and the settings of crossover controls for mid-range and treble balance may significantly alter the impedance characteristics. The minimum (rated) impedance may rise to several times the rated value at other frequencies, notably reaching its low-

end peak at the resonant frequency of the speaker enclosure system. Since speaker impedance is not purely resistive, other factors also limit speaker ability to utilize the amplifier rated power. Large multiple-speaker systems may appear as largely capacitive loads, just like a full-range electrostatic speaker, and the electrostatic speaker may drop well below its rated impedance at some frequencies.

Because the load is not a constant impedance, no amplifier can deliver power into any speaker system at all frequencies. Tube amplifiers with multiple-impedance output transformers can match several specific loads (but not simultaneously), and an impedance mismatch has a similar power limitation on any amplifier, tube or transistor. The Institute of High Fidelity has chosen 8 ohms as the standard impedance for rating power amplifiers—the modal nominal impedance, but speakers vary from 4 to 32 ohms.

A transistor amplifier which delivers 60 watts maximum into an 8-ohm load delivers between 30 and 40 watts into a 16-ohm load, and less into a 32-ohm load. Lower load impedances, such as 4 ohms, cause more current to flow in transistors than in higher impedances. Excessive current can lead to rapid transistor failure unless the power output is deliberately limited to prevent excessive current. There, care should be used in application, matching, and operating.

In order to minimize problems of excessive current, some solid-state amplifiers require the use of series resistors with 4-ohm speakers to prevent damage. This soaks up a high percentage of the available power and greatly diminishes the effective damping of the speaker. The same problem exists with speakers (supposedly specially designed for transistor amplifiers) which include series resistors in order to present a more nearly uniform impedance characteristic. Because the majority of 16-ohm speakers are significantly more efficient than the most popular 4-ohm speakers, it is of greater importance to most users to have the best match between 4 and 8 ohms.

It is worth noting that transistor power amplifiers cannot be paralled to obtain higher power output at usable speaker impedances. Because these amplifiers are essentially current or constant-voltage amplifiers, when they are paralleled the same voltage will be available only at an output impedance which is too low for general speaker use, and there is risk of damage to the amplifier through interconnection of the two outputs. For higher-power applications, the preferred procedure is to drive each channel of the amplifier with identical signals, and connect each output to a separate speaker system.

Some of the transistor power amplifiers to date have been deficient in high-frequency performance and temperature stability. The diffused silicon transistors permit good circuit performance at high frequencies. Silicon transistors are desirable for power output stages because of their ability to perform at much higher junction temperatures than can ger-

manium. This means smaller heat-radiating fins can be used for the same power dissipation. On the negative side, silicon often has higher saturation resistance, which gives descreased operating efficiency that becomes appreciable when operating from low-voltage supplies.

POWER AMPLIFIER CIRCUITS

Next to the output transformer and the speaker the power amplifier circuit and the other components used therein are usually the most important factors governing the percentage of distortion present in a system. Since it is usually desired to keep distortion to the lowest value consistent with economy, the power amplifier should be carefully designed. Some form of degenerative feedback should always be used. Feedback reduces harmonic distortion, stabilizes the output impedance, and reduces the effects which resonance in speakers has on the output.

Push-pull arangements will give a low percentage of distortion. Triodes in push-pull give excellent results. The use of beam tubes in ordinary circuits results in distortions higher than is permissible unless sufficient degenerative feedback in used. From 15 to 20 db of feedback should be used in beam-power amplifiers.

Beam-power tubes used in push-pull arrangements in certain modern circuit designs give negligible distortion and are recommended when highest-quality reproduction and high output of music are desired.

Output Circuits

Standard single-ended parallel or push-pull parallel circuit arrangements do not produce sufficient fidelity to be used for hi-fi systems. Push-pull arrangements produce less distortion because even harmonics, plate current effects in the transformer, and hum pickup in the plate circuit normally tend to cancel out.

There are several successful types of push-pull amplifiers in current use in output circuits utilizing triode, tetrode, and ultralinear (hybrid) circuits. Pentode amplifiers have such nonlinear characteristics that they cannot be considered as high-fidelity equipment. Triode arrangements with triode connected tetrodes as shown in Fig. 5-13 offer the simplest approach to high fidelity.

This approach may be used with or without additional overall systems circuit improvements such as negative feedback (to be discussed later). Compared with straight tetrode-connected operation, triode-connected output stages using comparable components provide better stability, noncritical performance, less distortion, and better speaker-damping characteristics.

However, triode amplifiers (compared with tetrodes) have the following disadvantages: (1) Triode amplifiers have the lowest efficiency; therefore they require large and more expensive components, tubes, and power supplies to obtain outputs equivalent to those obtained from tet-

(A) Triode.
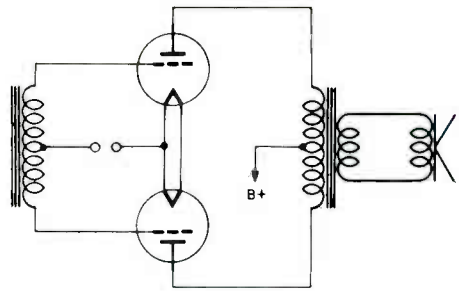
(B) Triode connected tetrode.
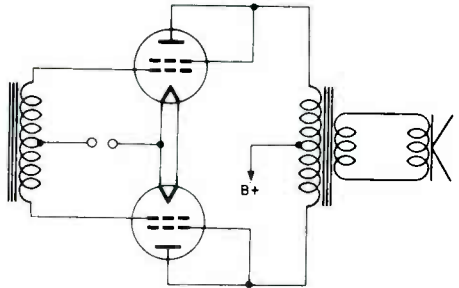
Fig. 5-13. Power amplifier stages.

rode amplifiers. (2) Triode amplifiers require much more driving power. This condition further increases cost. The overall percentage of distortion will be increased by each additional stage required to raise the power sufficiently to drive the triode amplifiers. The higher the power required from any intermediate stage, the higher the distortion of the output of that stage will be. All these distortions are added to the final triode amplifier stage distortion, partially negating the advantages to be gained in the use of the triode output stage itself.

Tetrode Amplifier Circuits

The efficiency of a tetrode amplifier can be as high as 60 percent. Tetrodes operated in certain circuits have very high amplification characteristics, requiring but a tenth of the drive of triode amplifiers. These advantages make possible reduced cost to a practical commercial level for large-scale manufacture, distribution, and enjoyment of high-fidelity amplifier equipment. A tetrode final amplifier circuit is shown in Fig. 5-14.

The disadvantages of tetrode amplifier design have been overcome by circuit, component, and electron-tube improvement over the past few years to an extent that highly satisfactory results can be attained. Technically, the output characteristics of tetrode amplifiers have been so improved that the measurable distortion is negligible. These high-fidelity characteristics in tetrode amplifiers have been gained by tube application

131

improvement, amplifier circuit change, and components improvement. Tubes such as the KT66, 6CA7, 5881, 6550, and 7829 have been designed for improved performance and more power capabilities to handle peak loads and provide greater dynamic range of operation within the flat portion of the tube characteristic curve. These tubes are specially suited to ultralinear operation.

Ultralinear operation involves a compromise circuit arrangement in which a tetrode is neither connected as a triode or a tetrode but wherein special taps on the primary of the output transformer are connected to the screens, thereby providing an amplifier operation having some of the advantages of both tetrode and triode operation. At the same time, the disadvantages of both types of circuits are reduced. This circuit arrangement is more efficient and has more gain than straight triode operation, and at the same time the stability and speaker-damping characteristics are practically as good as in straight triode operation. Most important, distortion in the power output stage is hardly increased over that of triode operation. This compromise design is generally accepted as one of the better contemporary approaches to high-fidelity power amplifier circuits. The circuit arrangement to achieve this is shown in Fig. 5-15. The tap point varies from 20 percent to 40 percent of the winding, with considerable difference of opinion between different manufacturers as to the optimum point. Tap-point design is usually determined empirically by listening tests. Always, when the human ear is involved, measurement by a fixed standard is impossible. Therefore, one must listen to each unit to determine which is preferable to him.

Another design considered optimum by current standards is the patented McIntosh unity-coupling circuit shown in Fig. 5-16A. "Unity coupling" is a name which identifies a group of audio amplifier circuits. These circuits are designed to reduce distortion at high frequencies (the treble range) which conventional push-pull circuits may generate. However, at high frequencies, conventional push-pull circuits lack the coupling necessary to provide low-distortion amplification.

B+
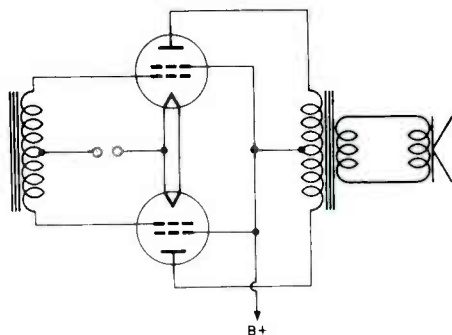
Fig. 5-14. Tetrode power amplifier.

To illustrate this problem graphically, consider a tone waveshape as shown in Fig. 5-16B. One push-pull tube operates from A to B, the other from C to D. Both tubes operate from C to B. The small interval CB can be reduced to zero though it seldom is practical.

One half of a push-pull amplifier ideally produces a wave as shown in Fig. 5-16C. It consists of one-half a sine wave and a flat. Each half-wave is added together to make one full-tone wave in the output transformer.

The conventional transformer consists of two halves wound as shown in Fig. 5-16D. Since the winding direction is the same, current flowing into the center goes in two different directions, synthesizing the whole tone wave. At high frequencies the flat between successive half-waves changes shape. Graphically, the change appears as shown in Fig. 5-16E.
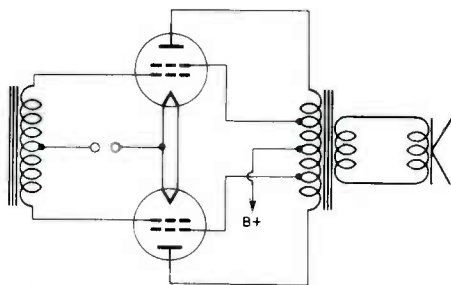
Fig. 5-15. Ultralinear power amplifier circuit.

The new shape from B to C is due to the collapse of the magnetic energy in the output transformer, which is not completely coupled from one half to the other. When two waves of this shape are added together they appear as shown in Fig. 5-16F. The encircled departure from the original tone shape represents the addition of new tones generated by the amplifier itself.

The unity-coupled circuits in their simplest form use only one half of a conventional push-pull transformer, as shown in Fig. 5-16G. The coupling under this condition is nearly 100 percent of unity. However, it is still desirable to use two transformer windings so that only one power supply is needed. McIntosh unity-coupled circuits wind both wires side by side for their entire length, as shown symbolically in Fig. 5-16H. Having such a close physical relation the two wires are within a very small fraction of 100 percent coupling. No notch distortion is developed at high frequencies, as shown in Fig. 5-16B. An additional advantage of this circuit improvement is that more transformer turns can be used at low frequencies to reduce distortion there too. Therefore, unity coupling improves both the high-frequency performance and the low-frequency performance.

133
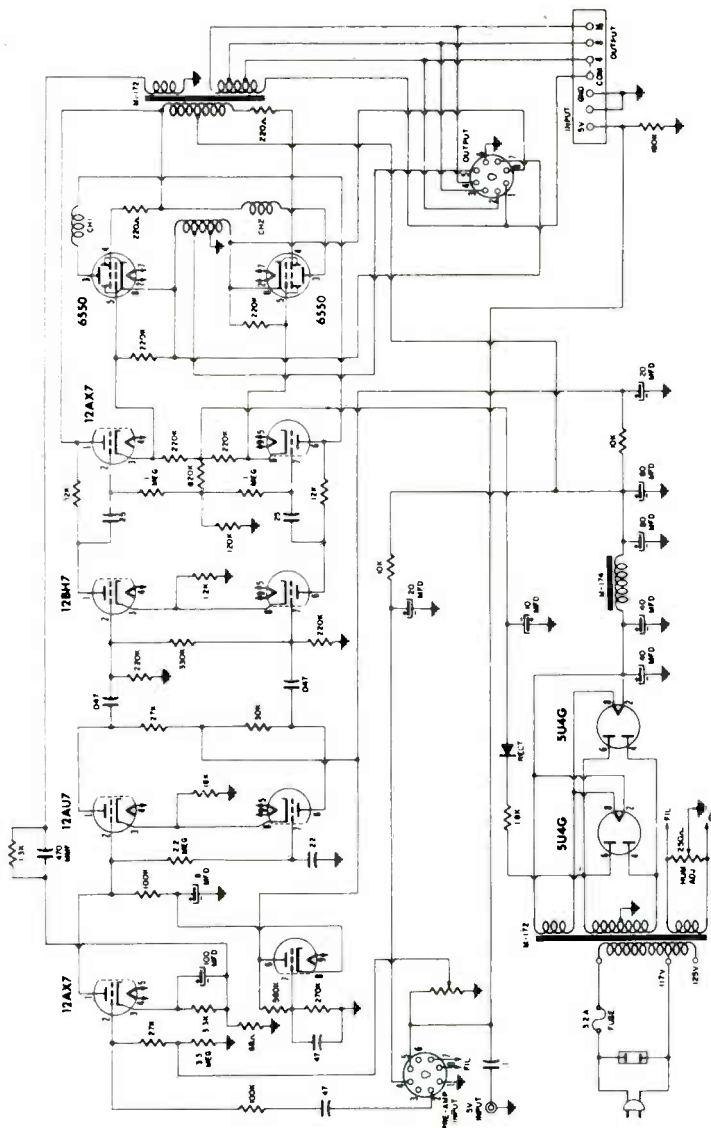
(A) Schematic of McIntosh Model MC-60 amplifier circuit.

Fig. 5-16. The

134
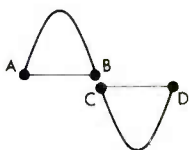
(B) Pure tone waveshape.

(C) Waveform of output of one half of a push-pull amplifier.
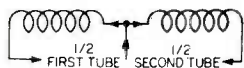
(D) Conventional output transformer primary.
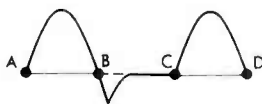
(E) How waveform changes shape in conventional circuit.
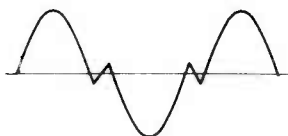
(F) Conventional output waveform with notched distortion.
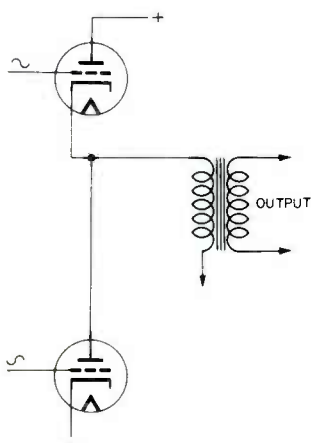
(G) Unity coupling in its simplest form.
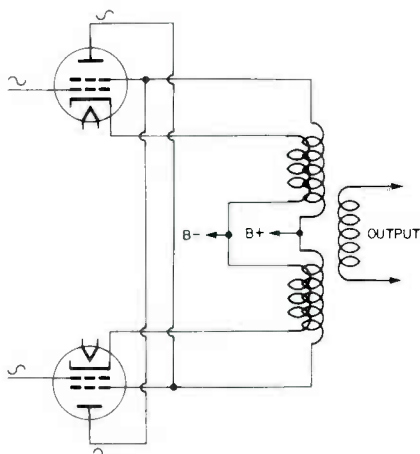
(H) Symbolic schematic of unity coupling and half cathode-follower circuit.

unity-coupling story.
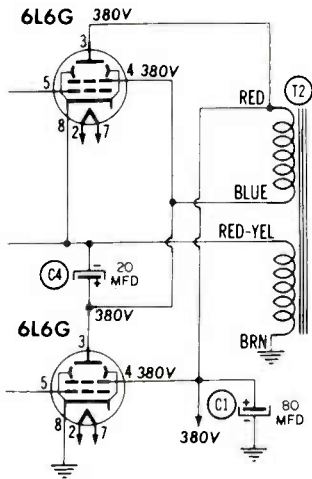
Fig. 5-17. Single-ended push-pull amplifier circuit.

The McIntosh unity-coupling circuit is unique in one other respect. Both output tubes are used in semi–cathode-follower circuit arrangements, providing better stability in the output circuit.

There are other variations of circuits for connecting the output tubes to the speaker, such as that shown in Fig. 5-17. This circuit is essentially a single-ended push-pull stage. The output tubes are in series in regard to the plate current swing, and the grids are in push-pull. When the current rises in one tube, the current in the other falls. This gives the same result as the operation of a conventional push-pull output stage. The primary advantage here is the reduction of switching transients and a high degree of linearity without special transformer windings.

Fig. 5-18A shows the application of power transistors to high-fidelity amplification in a simplified class-B circuit. Class-A amplifier circuits are usually utilized for preamplifiers with transistors, but a disadvantage of class-A operation in transistor power amplifier circuits is the requirement that collector current flow at all times. When two transistors are connected in class-B push-pull, one transistor amplifies half the signal and the other transistor amplifies the other half of the signal. These signals are combined in the output circuit to restore the original signal in an amplified state.

Fig. 5-18B shows a transistor amplifier with the output transformer eliminated. The advantages of class-B operation utilizing transistors can be obtained without the use of an output transformer. In this circuit the secondary windings of driver transformer T1 are phased so that when there is a negative signal from base to emitter of transistor Q1, there is a positive signal from base to emitter of transistor Q2, and vice versa. The negative signal to the base of Q1 causes Q1 to draw current, and the positive signal to Q2 cuts Q2 off. When the condition is reversed,
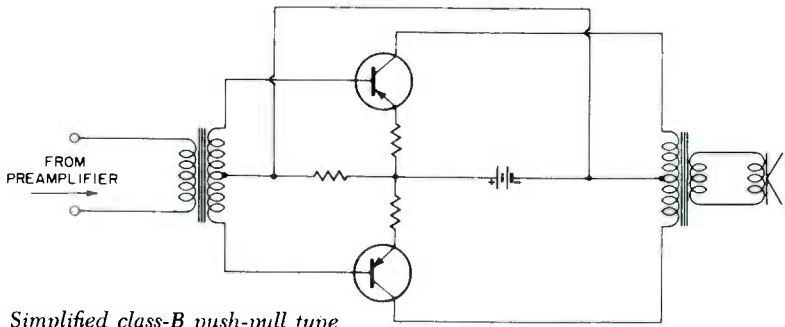
(A) Simplified class-B push-pull type.
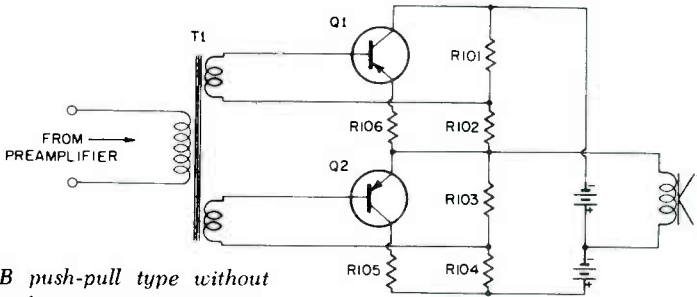
(B) Class-B push-pull type without transformer output.
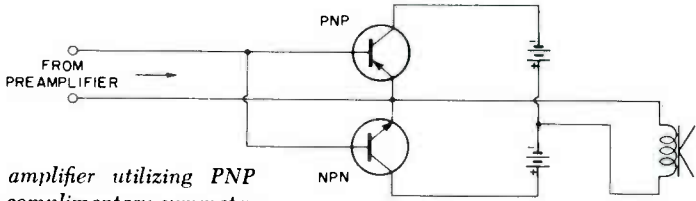
(C) Power amplifier utilizing PNP and NPN complimentary-symmetry circuit.

(D) Direct-coupled transistor power amplifier.

Fig. 5-18. Transistor power amplifiers.

137

transistor Q2 conducts and Q1 is cut off. Resistors R101, R102, R103, and R104 provide d-c bias to reduce crossover distortion and to keep the transistors a little above cutoff for no-signal conditions.

Fig. 5-18C shows a simplified complementary-symmetry circuit utilizing PNP and NPN junction transistors together in a class-B push-pull amplifier with direct coupling to a speaker. These two transistors are connected in a single stage so that the d-c current path in the output circuit is completed through the collector-emitter circuits of the transistors.

In the circuits shown in Figs. 5-18B and 5-18C practically no direct current flows through the speaker voice coil, making direct connection possible.

Fig. 5-18D is a direct-coupled transistor power amplifier with excellent low-frequency response. It has the advantage of d-c feedback for temperature stabilization of all stages. This feedback system stabilizes the voltage division across the 2N553 power output transistors, which operate in a single-ended class-B push-pull arrangement. These transistors also operate class-B in the Darlington connection to increase the current gain. Using an NPN transistor (2N696) for Q5 gives the required phase inversion for driving one of the output transistors and also has the advantage of push-pull emitter-follower operation from the output of 2N1925 input transistor Q1 to the load. Emitter-follower operation has lower inherent distortion and low output impedance because of the high percentage of voltage feedback.

The output transistors have a small forward bias of about 15 ma to minimize crossover distortion and also to operate the output transistors in a more favorable beta range. This bias is set by the voltage drop across R1 and R2, which shunt the input to the intermediate amplifiers. The intermediate amplifiers are biased at about 2 ma (to minimize crossover distortion) with the voltage drop across silicon diode D11. Junction diodes have a temperature characteristic similar to the emitter-base junction of a transistor. Therefore this diode also gives compensation for the temperature variation of the emitter-base resistance of three of the transistors. These resistances decrease with increasing temperature; thus the decrease in forward voltage drop of approximately 2 millivolts per degree centigrade of the diode provides some measure of temperature compensation.

The input transistor is a class-A driver with an emitter current of about 4 ma. Negative feedback to the base of this stage lowers the input impedance of this stage and thus requires a source impedance that is higher, so that the feedback current will flow into the amplifier rather than into the source. Resistor R10 limits the minimum value of source impedance.

About 10 db of positive feedback is applied by way of C101 and R101. This action helps to compensate for the unsymmetrical output circuit and permits the positive-peak signal swing to approach the am-

138

plitude of the negative peak. This positive feedback is offset by about the same magnitude of negative feedback.

A ½-ampere fuse is used in the emitter of each output transistor for protective fusing of the output transistors and also to provide local feedback since the ½-ampere type 3AG fuse has about 1 ohm d-c resistance. This local feedback increases the bias stability of the circuit and also improves the declining frequency response of the output transistors at the upper end of the audio spectrum. Because of possible lower transistor efficiency above 10 kHz, care should be taken when checking the amplifier maximum continuous sine-wave output at these frequencies. If continuous power is applied for more than a short period, sufficient heating may result to raise the transistor current enough to blow the ½-ampere fuses. Since there is not sufficient sustained high-frequency power in regular program material to raise the current to this level, actual performance of the amplifier does not suffer, because the power level in music and speech declines as the frequency increases beyond about 1 to 2 kHz.

The speaker system is shunted by 22 ohms in series with 0.22 mfd to prevent the continued rise of the amplifier load impedance and its accompanying phase shift beyond the audio spectrum.

The overall result, from using direct coupling, no transformers, and ample degeneration, is an amplifier with output impedance of about 1 ohm for good speaker damping, low distortion, and good bandwidth. The power response at 1 watt is flat from 30 Hz to 15 kHz and is down 3 db at 50 kHz. At this level the total harmonic and intermodulation distortion are both less than 1 percent. At 8 watts the intermodulation distortion is less than 2½ percent and the total harmonic distortion is less than 1 percent, measured at 50 Hz, 1 kHz, and 10 kHz. The performance of the amplifier of Fig. 5-18D is about the same for both 8- and 16-ohm loads.

Stereo Power Amplifier Circuits

The design features of power amplifiers for stereo are similar to those for monophonic operation. The use of two electrically separate power stages is still considered the best all-around arrangement. However, efforts have been made to design single stages that can handle both stereo signals at once.

An example of "two-way" amplifier, sometimes called *biortho* or *simplex* system, is one developed from the long-used telephone "phantom" circuit. Fig. 5-19 shows the original phantom* circuit, and Fig. 19B how its principle can be applied to a push-pull amplifier. The basic part of the circuit is a conventional push-pull amplifier as shown in the upper portion of Fig. 5-19B. The push-pull amplifier uses input and output

---

*Do not be confused by the use of this word in connection with a third, "middle" speaker and its associated circuit, discussed later.

transformers. A second channel is added by inserting additional input and output transformers in series with the return leads from the center taps of the transformer windings. The signal voltage from the extra input transformer is applied through the halves of the regular input transformer to the grids of the tubes in phase. This signal does not interfere with the regular signal because it cancels out in the primary winding of the regular output transformer. However, this extra signal does change the instantaneous voltage on both grids in the same way at the same time, thus causing both tube plate currents to do the same thing. Although the plate current fluctuations from this source cancel out as far as the regular output transformer is concerned, they do cause the same fluctuations in the total plate current into its center tap. This total current passes through the extra output transformer, whose secondary winding delivers the output from the extra channel.

The circuit of Fig. 5-19 could be used for stereo by applying the L signal to the regular push-pull channel (E_1) and the R signal to the extra circuit (E_2). However, there are serious disadvantages to this. One is that the extra circuit operates as a single-ended (rather than push-pull)
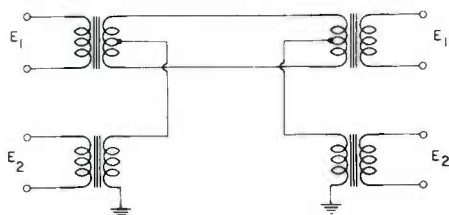
(A) *Basic phantom circuit.*
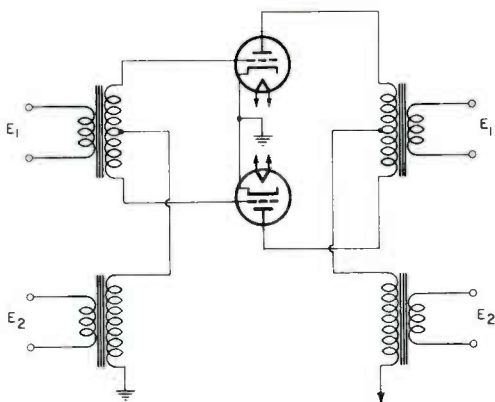
(B) *Phantom circuit used with push-pull amplifier.*

Fig. 5-19. How a two-channel power amplifier can be derived from the phantom-type telephone circuit.

amplifier with the tubes in parallel. This type of circuit is much more subject to distortion than is the push-pull amplifier, and its power-handlng capability with tolerable distortion is much lower. Since the L and R signals usually do not differ very much in amplitude, the lop-sided arrangement would be inefficient.

To overcome this objection, the circuit is rearranged to use sum (L + R) and difference (L − R) signals. The sum signal carries most of the power of the stereo information. Since the L and R signals are usually nearly equal, the difference (L − R) signal is relatively small—it is the difference between two signals which are almost the same in amplitude. This small signal is applied to the extra channel, and does not tax its relatively low, distortion-free, power-handling capability.

The new arrangement of the circuit is shown in Fig. 5-20. No input transformer is necessary, we just apply −L to one grid and R to the other grid. The negative L signal is obtained simply by passing the L signal through an odd number of conventional grounded-cathode amplifiers, each of which provides a phase reversal; or, the signal can be passed through an even number of grounded-cathode amplifiers and one cathode follower.

In the circuit of Fig. 5-20, each tube inverts its signal, so the relative polarities of the signals across the entire transformer primary and each half of the transformer are as indicated. The result is that the signal coupled through the main transformer (T1) is equivalent to L + R with polarities as indicated by the arrows at the secondaries.

Now consider the other channel. The primary winding of transformer T2 carries current from the center tap of the primary of T1. Therefore, this current is the sum of the plate currents of the two tubes, and the
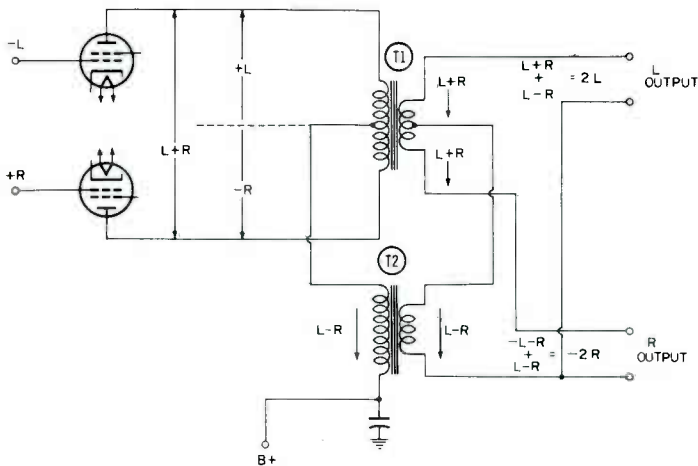
Fig. 5-20. Improved combination output circuit.

voltage has the relative polarity shown. The output of T2 is the L − R signal. This output is combined with the L + R outputs of T1 to provide the overall L and R outputs of the combination circuit. Although the L − R output enters into the derivation of both the L and R outputs, its amplitude is relatively small, so it can be handled as an extra output with little distortion.

Whether the simplified circuit just described or a dual power amplifier is used, there are certain switching functions usually provided in a stereo power amplifier. One of these is to provide for parallel operation for the monophonic mode. A switch is provided simply to connect the L and R outputs in parallel. If you happen to have an amplifier not so equipped, or have two separate amplifiers which you are combining, you can connect the outputs together yourself. Two things are important to remember:

1. The phasing must be correct. This is indicated by the connection that gives the most output.
2. The rules of impedance matching must be observed for most efficient operation. For example, if the speaker has an impedance of 8 ohms, each of the two parallel-connected outputs should be of 16 ohms impedance.

Actually, there is some doubt as to real need for such a parallel connection during monophonic operation. Many audiophiles with good stereo systems say that they find playback of monophonic recordings and broadcasts very pleasing through two channels, even though there is no stereo effect involved. Of course, paralleling does provide a single output single with the power capabilities of both channels and this may be useful under some special circumstances.

Another switching function sometimes found in the power amplifier portion is the "quasi-stereo" switch. This switch permits either the L or the R signal alone to be fed to both channels.

Some output circuits also include a channel-reversing switch, which might more accurately be called a "channel-swapping" switch. Operation of this switch causes the speaker that was receiving the L signal to receive the R signal, and vice versa. This is useful when the listener is checking to see if his L and R connections through the system are correct. For example, if he makes his own tape recordings, he may have an error in the connection of tape-machine outputs, and this can be quickly corrected by the "reversing" switch. Otherwise, once your pickup cartridges and tuner outputs have been properly connected, there is no further use for this switch.

Class of Amplifier Operation

Amplifiers in general are classed by letters A, B, and C, according to the method of adjusting the tubes. Audio power output stages are gen-

erally adjusted for class-A or class-AB types and sometimes as class-B amplifiers—never as class-C. Preamplifier stages are usually voltage amplifiers and are adjusted to operate in the straight class-A region.

A class-A amplifier circuit has a fixed grid-bias voltage, and applied signal voltages so arranged that plate current flows at all times.

A class-B amplifier is one in which the operating fixed grid-bias voltage is adjusted approximately equal to the cutoff voltage of the tube so that plate current is proportionately low with no signal and so that plate current flows for each half-cycle of an applied a-c signal.

A class-AB amplifier is adjusted so that the fixed grid-bias voltage and an alternating signal voltage applied to the grid cause the plate current to flow for appreciably more than half but less than the entire alternating signal voltage cycle.

Class-C amplifiers are so arranged that plate current flows for less than half-cycles of an alternating signal voltage. Class-C circuit arrangement are not used in audio work.

Class-A and class-AB arrangements are well known for their high-fidelity characteristics. In years past, triodes arranged to operate in class A were the only acceptable amplifier circuits for hi-fi equipment, but today the best amplifiers operate in class AB with special compromise arrangements (ultralinear and unity-coupling designs) to produce audio reproductions with distortions practically unmeasurable and with excellent response and high outputs.

Driver Amplifier and Phase Inverter Stages

The amplification factor of the most sensitive power amplifier stage is insufficient to provide the necessary amplification to give full output from the input of the average input device such as a phono pickup, microphone tuner, or others. Additional amplifier stages are added to all power amplifier stage designs to provide the required gain. The low driving power requirements of tetrode tubes operated class AB, as is standard in modern amplifier design, allow use of a voltage amplifier, a driver, and inverter stages with low-power outputs. These can be operated on a linear portion of the tube characteristic and produce sufficient gain per stage in such a manner that usually two to four stages, according to the particular design, can produce more gain than required.

The voltage amplification is usually accomplished in one, two, or three of these stages, and then the signal is fed to inverter and driver stages to provide two components of input signal 180 degrees out of phase with each other to drive either side of the push-pull power output stage.

There are various arrangements for providing linear, undistorted, low-level power for driving final amplifier. Block diagrams for various arrangements are shown in Fig. 5-21.

Sensitive, high-gain, tetrode power-amplifier stages require comparatively small driving power and are usually connected directly to a simple

cathodyne phase-inverter stage as shown in Fig. 5-21A, even though the gain of the cathodyne phase inverter is less than unity. In this case design requirements are simple as the cathodyne is fairly foolproof and develops negligible distortion.

Where the power amplifier has a low amplification factor, as when straight class-A operation is used, more power must be provided, and a circuit such as in Fig. 5-21B may be used. The advantage of this will be found in the power amplifier operation only because each added push-pull stage must be balanced, stable, and most carefully designed to eliminate distortion to feed the final amplifier stage properly.

The circuit shown in Fig. 5-21C is used in one of the top-brand amplifiers and has been designed with considerable care to achieve balance throughout the push-pull voltage amplifier stages. Such a design is practical for use in laboratory-type equipment.

(A) *Simple phase-inverter stage.*

(B) *Added push-pull driver stage.*

(C) *Addition of push-pull voltage amplifier stages.*
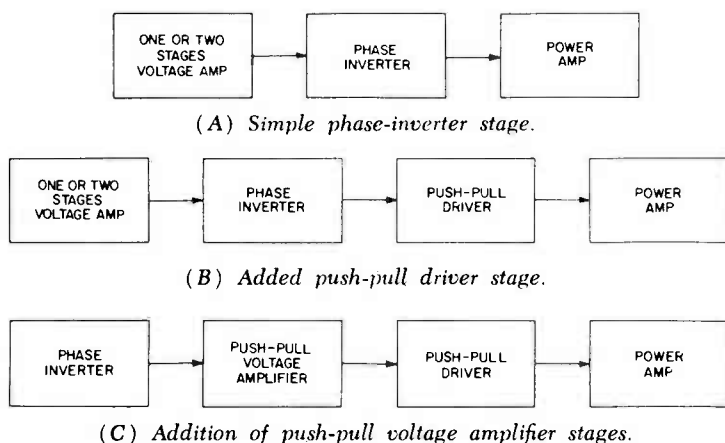
Fig. 5-21. Block diagrams of various systems arrangements.

Phase Inversion

The signal at each grid of the power stage must be as near to 180 degrees out of phase and equal in amplitude as possible at all audio frequencies. Since the input and preamplification stages are almost always of single-ended design, it is necessary to split the amplified signal into two equal components, one lagging the other by exactly one half-cycle (180 degrees).

The simplest device for splitting the phase is a transformer with a single input winding and a center-tapped output winding as shown in Fig. 5-22. Such a transformer will provide the required phase split and equality of amplitude but at a considerable expense. An input transformer of this type must be specially designed and constructed. It is large

144

and heavy and, generally speaking, impractical as compared with the simplicity and low cost of resistance-coupled electron-tube phase-inverter types of circuits shown in Fig. 5-23.

Fig. 5-23 shows various commonly employed types of electronic phase inverters. The circuit shown in Fig. 5-23A utilizes a self-balancing arrangement. The incoming signal drives the grid of the upper tube, V1. The output of V1 drives the upper tube (V3) in the power amplifier, and a portion of this same output feeds through to drive the grid of V2. The output of V2 is 180 degrees out of phase with the input; therefore, the phase has been inverted as required.
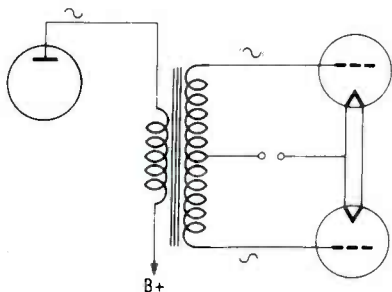
Fig. 5-22. Phase inversion, using a transformer.

The cathodyne circuit shown in Fig. 5-23B is the most commonly used phase inverter because of its simplicity and because it is self-balancing. The signal fed to the grid of V1 is reversed 180 degrees at the plate but in phase at the cathode, thereby splitting the phase in accordance with the requirements for driving the power amplifier stage. The same value of current flows through R1 and R2 (equal values of resistance), therefore causing balanced output to each tube in the power amplifier.

The circuit shown in Fig. 5-21C is called a cathode-coupled phase inverter operating in a similar manner but with two tubes. This circuit has provision for equalizing the output from each side by adjustment of R1 (with signal applied). This latter circuit is generally used when one or more push-pull voltage amplifiers follow to drive the power amplifier.

Voltage Amplifiers

The gain of power amplifier stages is relatively low and phase inverter stages usually have less output than input. The average amplifier requires several additional stages with high gain per stage to bring the signal up to sufficient level to obtain full output according to the power stage capabilities. Further, basic power amplifiers usually require a separate preamplifier with output voltages of 1 to 2 volts to drive the amplifier to full output.

These stages are operated as low-power, high-gain voltage amplifiers. Voltage amplifiers are almost always resistance coupled, operating on a
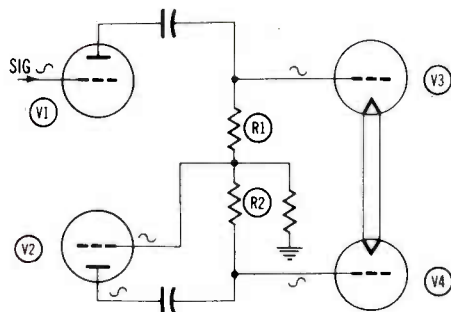
(A) Self-balancing arrangement.
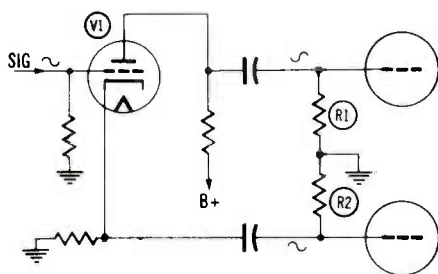
(B) Single-triode arrangement.
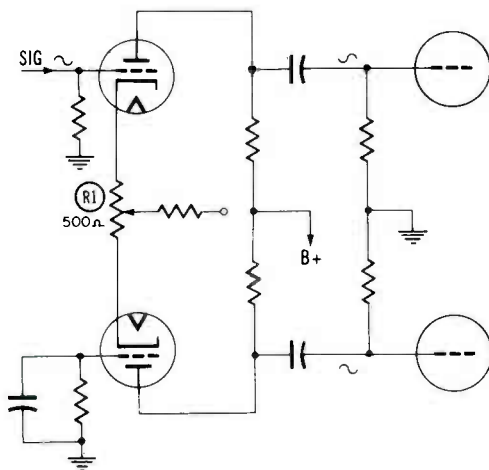
(C) Cathode-coupled phase inverter.

Fig. 5-23. Phase-inverter circuits.

small and flat portion of their characteristic curve and providing considerable voltage gain with negligible distortion. Pentodes are used for highest gains, and triodes are used for greatest stability. One to three voltage amplifiers are usually contained in a basic amplifier, according to the sensitivity and gain requirements.

Good voltage amplifier design can be achieved with negligible distortion of the signal, but it is not so easy to keep down internal noise and hum pickup. Special design arrangements have been made to reduce noise and hum. Hum can be reduced by use of special isolated filament circuits with short and shielded leads carrying alternating currents. Another attack is to use direct-current filament supplies. Low-noise design has also been improved by use of special input tubes and circuits and other components, such as low-noise transistors, resistors, and controls.

Negative Feedback

In the dynamic operation of a modern basic amplifier there is usually provided feedback of a portion of the output signal to the input stage. This feedback is negative, or, in other words, a cancelling signal fed in opposite phase with respect to the input signal at the feedback point. This feedback signal is superimposed on the original signal to reduce distortion and noise originating from within the amplifier. Negative feedback also lowers the effective output impedance and provides better speaker damping. The only disadvantage of negative feedback is that it reduces the overall amplifier gain considerably. However, this loss of gain can be recovered by use of an additional voltage amplifier stage as found in modern arrangements.

Fig. 5-24 shows a block diagram arrangement using a negative-feedback loop. Fig. 5-25 shows a feedback circuit arrangement in one stage only. Negative-feedback voltage may be fed back over one or any odd number of stages.

With application of negative feedback, noise and distortion are reduced by cancellation. With no signal input any internally generated noise has a signal component fed from the output back to the input as shown. Since this feedback signal is opposite to the internally generated noise and when the feedback is adjusted so that the noise portion of the feedback signal is proportionately equal to the average generated noise over the feedback loop, a large portion of the noise will be cancelled out.
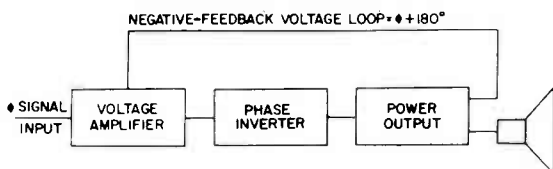
Fig. 5-24. Negative-feedback loop.

NEGATIVE–FEEDBACK VOLTAGE

B+

POSITION A PROVIDES POSITIVE–
FEEDBACK CURRENT WITH
RESPECT TO SIGNAL AT CATHODE.

A   B

POSITION B PROVIDES NEGATIVE–
FEEDBACK CURRENT WITH
RESPECT TO SIGNAL AT CATHODE.
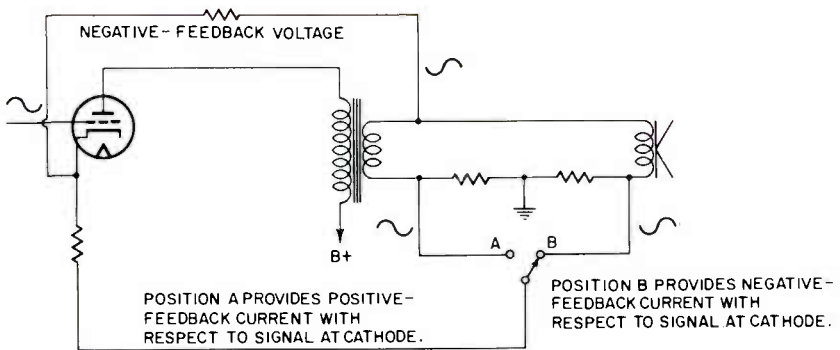
Fig. 5-25. One-stage voltage and current feedback circuit.

The same is true for hum internally picked up, providing the phase of the hum is suitable for suppression.

The same principle of cancellation applies to distortion of the original signal originating within the amplifier at any point within the feedback loop. An internally distorted signal waveform will be pulled back toward the original signal shape. The original signal will be reduced, but additional components added internally to the original signal will tend to cancel out.

The effective output impedance of the amplifier is lowered by means of the variation in amount of negative feedback applied in relation to the output load. The negative feedback is taken off the output of the amplifier in the same manner as the speaker takes power. When the speaker is not taking power the relative output signal voltage rises and consequently there is proportionately more negative-feedback voltage. More negative feedback reduces the gain and lowers the output signal, tending to compensate for the unloaded condition of the output. In this manner negative feedback has the same effect as lowering the output impedance of the system.

Negative feedback of an amplifier is rated in db. The feedback db rating is equal to the reduction in gain of the amplifier caused by the negative feedback.

Positive Feedback

Just as negative voltage feedback lowers output impedance of an amplifier system, positive-feedback voltage will increase the effective operating output impedance. Positive-feedback voltage, if applied in the same manner but opposite in phase to negative feedback, will cause system oscillation at random audio frequencies.

Current feedback applied from the output of the amplifier to the cathode of an intermediate amplifier stage will cause opposite effects on output impedance. Positive-feedback current will lower effective output

148

impedance, and negative-feedback current will raise the effective output impedance of an amplifier. Positive-feedback current will also increase distortion in a manner opposite to the cancellation principle of negative feedback.
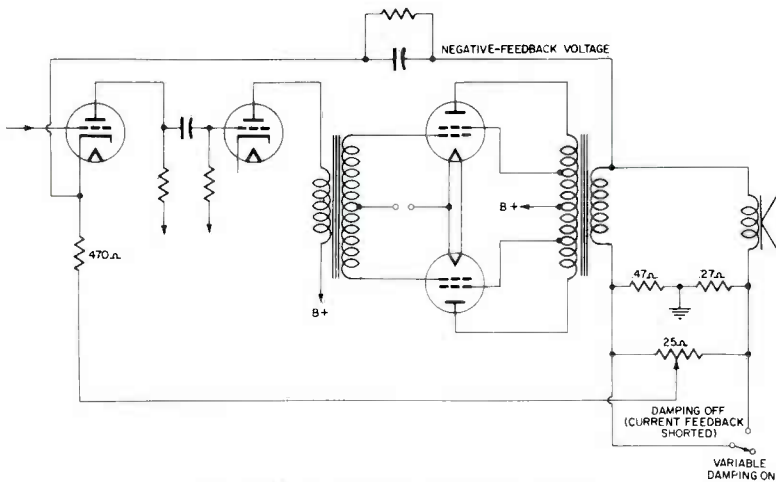
Fig. 5-26. A variable damping circuit.

Variable Damping

This feature of amplifier design does not improve amplifier performance. It is incorporated in some amplifiers to provide better speaker operation. Variation of the effective output source impedance of an amplifier, as can be provided by voltage and current feedback controls, will provide an adjustment to best load and damping for any particular speaker arrangement. Fig. 5-25 shows how simple combinations of voltage and current feedback circuits with an amplifier can be arranged to obtain any value of output impedance and damping factor desired. In Fig. 5-25, when the switch is at position A, positive-feedback current (below) and negative-feedback voltage (above) are fed back to the input to lower the output impedance and increase the damping factor of the system. When the switch is at position B, negative-feedback current and negative-feedback voltage are used to raise the output impedance and lower the damping factor. Fig. 5-26 shows a circuit arrangement used in a popular modern amplifier kit to provide continuously variable damping designed to provide complete dynamic matching of the amplifier to any speaker according to individual taste.

PREAMPLIFIERS

Basic amplifiers usually require about 1 volt of signal into a load resistance of 100,000 to 300,000 ohms to drive them to full output. Tuners

149

usually provide this amount of drive, so that a preamplifier is not required to increase gain when only tuners are used.

High-output phono cartridges and microphones will provide sufficient output (up to 1 volt) to drive some basic amplifiers, but high-fidelity phono pickups and microphones of the magnetic or dynamic type give considerably lower outputs. The lower the output, the easier it is to design quality characteristics into these pickup units.

To develop sufficient signal to drive a basic amplifier to full output, additional amplification or preamplification is required before feeding the basic amplifier. This gain is usually provided by a preamplifier, preamplifier equalizer, or control center. Preamplifier equalizers and control centers are also designed to provide all kinds of control adjustment, compensation, and filtering to cover about every condition one would desire, as has been described.

A low-output phono pickup feeding a simple preamplifier with one or two tubes provides sufficient gain to drive a basic amplifier which needs up to 3 volts for full output, but when various arrangements of compensation filtering networks and feedback circuits are employed to develop special effects these features have circuit requirements that reduce the overall gain by an amount greater than the compensation provided. For instance, if a range of 20 db of treble boost and 20 db of treble attenuation are provided, a total of 40 db range of control is applied. If the amplifier is to provide a gain of 50 db under flat operating conditions, then it must provide for an output level of 50 db at the reference frequency, which is usually around 500 Hz, plus 20 db at boost frequencies. The full attenuation of the highs or lows will also cause reduction of the overall gain at the reference frequency so that more than 50 db of gain at the reference frequency at flat setting is required—an average design might require 10 db of additional reference-frequency gain to cover all conditions of treble and bass attenuation curve variations and overlap. Therefore, in order to provide for bass and treble boost of 20 db, that much more gain (10 db) must be added. If 20 db of negative feedback is collectively applied, then the gain of the amplifier should be at least 50, plus 10 for overlap, plus 20 for boost, plus 20 for negative feedback as described, or a total of 100 db of amplification gain to provide 50 db of effective gain because of these provisions. There are other provisions in the more complex control units that require even more gain; thus arises the reason for more stages and complexity as more features are incorporated in the unit.

Compensation

There are several forms of compensation provided in preamplifiers and control units. One form of compensation is equalization of the highs (attenuation) and lows (boost) to compensate for emphasis and de-emphasis respectively of the highs and lows in the processing of high-fidelity

records, as described previously. Current hi-fi records are cut with a curve like that shown in Fig. 5-3, as has been described. Equalizing circuits in preamplifiers and control centers must have provision to compensate for this by amplification compensation curves of closely equal and opposite form, as shown in Fig. 5-4. These curves show equalization for most all records cut today as per Record Industry Association of America (RIAA) standards. However, there are several other curves to which many popular records are still being made from old master recordings. These curves are shown in Fig. 5-27. Equalizers will have to provide for these as long as the older recordings are played, and provision must be made in equalizers to compensate for them. This requires more switching circuits. Another form of compensation is to equalize for emphasis of highs used in f-m broadcasting techniques and to compensate for the loudness effect of the human hearing system, as described previously.

All these compensating effects are usually provided by various arrangements of high-pass and low-pass filters. These filters are placed in series with some channel of the preamplifier according to their use, or they are inserted in negative-feedback loops to attenuate by discrimination of feedback at certain frequencies and at the same time to reduce overall distortion.

Record equalization is usually selected by a single rotary switch control. More elaborate systems provide additional finer adjustment of roll-off and turnover frequencies to suit personal taste.

Equalization roll-off points and slopes are controlled by switching-in varying amounts of capacitance, resistance, or inductance in the high- and low-pass filtering circuits as described. Almost all modern records are being cut to one standard, so this provision is mainly useful for compensating for odd records and to give considerable flexibility for experiment or to compensate for other conditions according to the critical listener's taste.
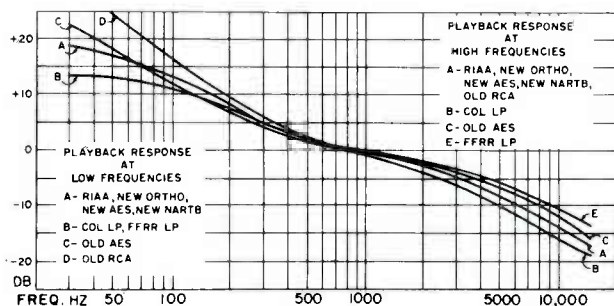
Fig. 5-27. Comparison of standard RIAA playback curve with some curves used before standard was adopted.

A modern high-fidelity system may have many sources of program material: phono pickup, a-m or f-m tuners, television, tape, microphone, and others. Control centers are available with any degree of complexity one might desire to conveniently provide for the many kinds of inputs, to switch from one to the other, apply power to several program sources, control or alter the signal, mix programs, and in general do the many operations used in advanced high-fidelity techniques.

PREAMPLIFIER CIRCUITS

A simple preamplifier is shown in Fig. 5-28. The plug-in unit shown is the simplest form of preamplifier including only an input matching circuit, a straight amplifier working on the linear portion of its curve, and an output to match the power amplifier input.
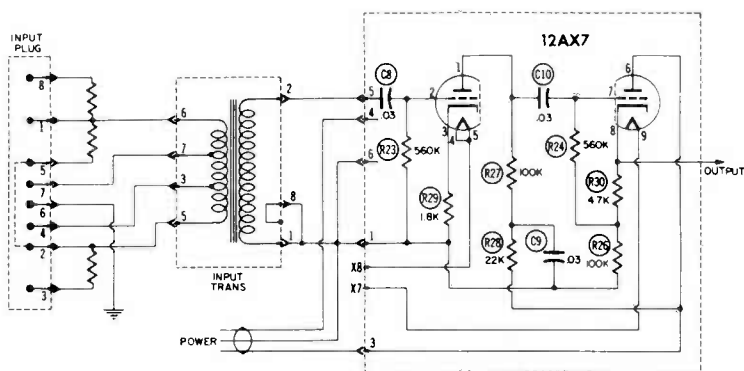
Fig. 5-28. A simple preamplifier circuit.

The purpose of this unit is to match the input device, such as a phono pickup or microphone, to the input circuits of the amplifier and to increase the input signal from such low-output devices as a dynamic microphone sufficiently to drive the amplifier to full output. The prime advantage of such a unit is its obvious simplicity.

The transformer provides as a multimatch for input devices having impedances from 50 to 20,000 ohms. The output matches a 100,000-ohm or high-impedance input of a basic amplifier.

Where no control, compensation, or filtering is desired this unit will provide all that is needed. This is especially suitable to microphone application.

Fig. 5-29 shows a circuit of a simple preamplifier specifically designed for tape. This preamplifier includes (NAB) variable equalization for tape speeds from 1⅞ ips to 15 ips. See Fig. 5-30. The maximum output into a high-impedance load is 6 volts a-c rms. This unit must be fed to an amplifier with a 10-foot or shorter shielded lead. There is a similar
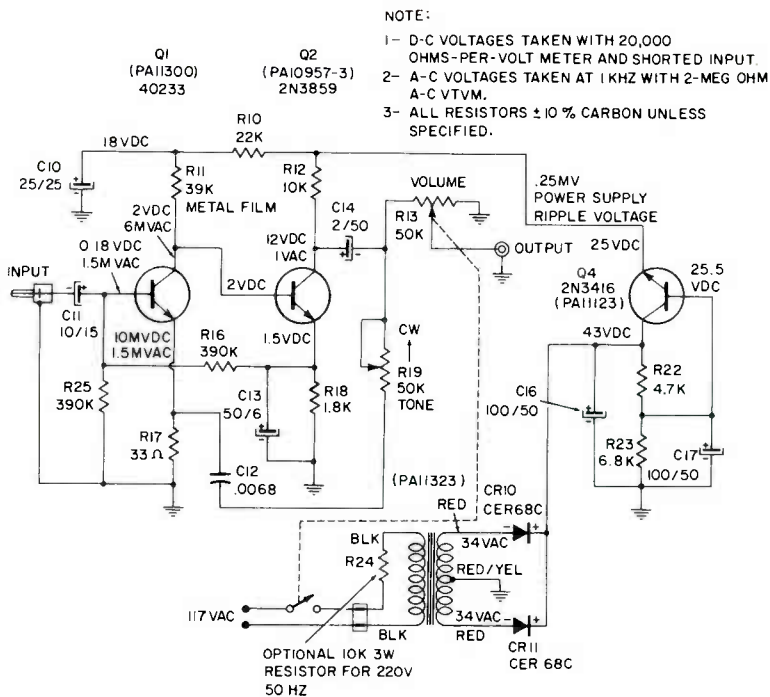
Fig. 5-29. Viking PB10 tape preamplifier circuit.
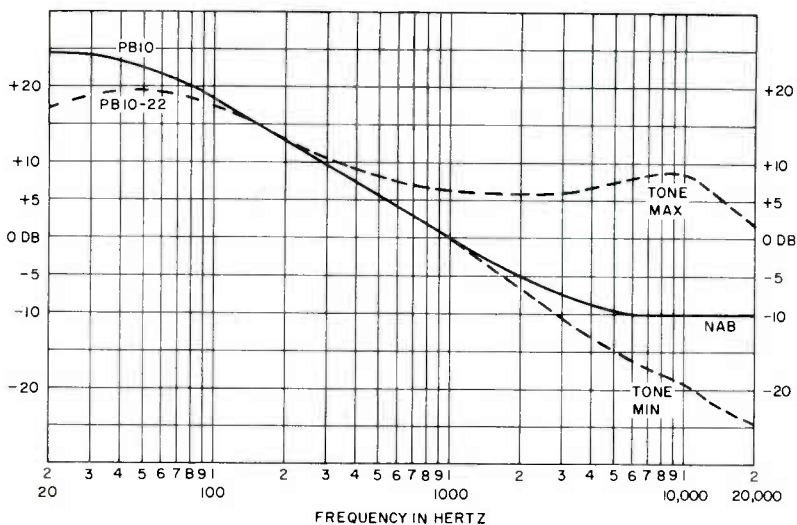
Fig. 5-30. PB10 frequency response in series with head.

153

model with one stage added and an output transformer to match a 600-ohm balanced line and taps for 1500-, 150-, and 6-ohm loads. A 600-ohm line, up to several hundred feet long, can be used. Fig. 5-31 shows the front of the PB10.

Figs. 5-32, 5-33, 5-34, and 5-35 show three preamplifier circuits of simple design but each successive unit has progressively more complex circuitry. Fig. 5-32 is a two-stage amplifier with impedance-matching
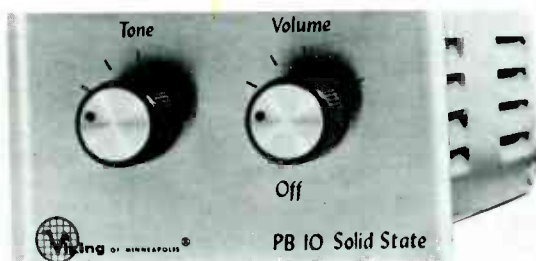
Courtesy Viking of Minneapolis, Inc.

Fig. 5-31. Viking PB10 preamplifier.

and gain characteristics similar to the unit in Fig. 5-28 but in addition has fixed equalization to compensate for average pre-emphasis and de-emphasis of recordings. The purpose of this unit is to match the output of high-fidelity, low-output magnetic phono pickup cartridges, to increase the signal, compensate for record playback, and match a basic amplifier input. Equalization is achieved by use of R1 to roll-off the highs, and C3 and R4 provide low-frequency compensation. The feed-back loop is negative and tends to remove any internally generated distortion or noise as well as provide attenuation for compensation.
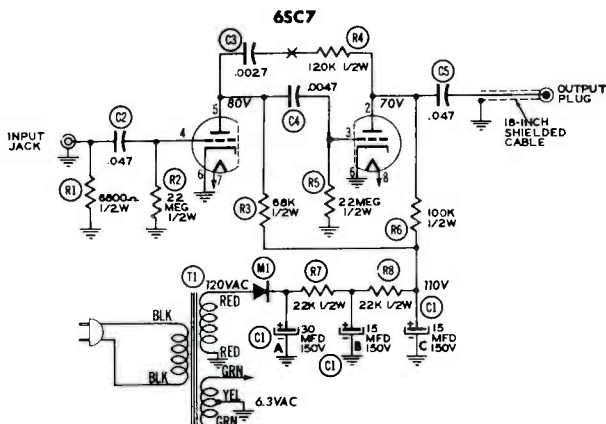
Fig. 5-32. A preamplifier with fixed equalization.

154
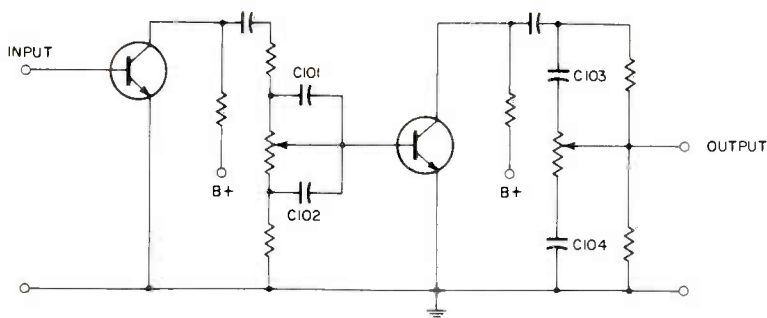
Fig. 5-33. Two-stage preamplifier circuit with separate bass and treble controls.

This unit also can be used for straight public-address work or microphone input by removing the equalization circuits.

While older records had many variations in emphasis roll-off frequencies, the variations are small enough that the circuit shown in Fig. 5-32 will do a reasonable job on all of them. (See Fig. 5-27.) However, if it is desired to be able to compensate closely for the original emphasis conditions, more elaborate units with several equalization filter circuits and components that can be switched in are available.

Fig. 5-33 shows a circuit with the same functions but with controls added in a transistor circuit. The tone-control network has two stages,
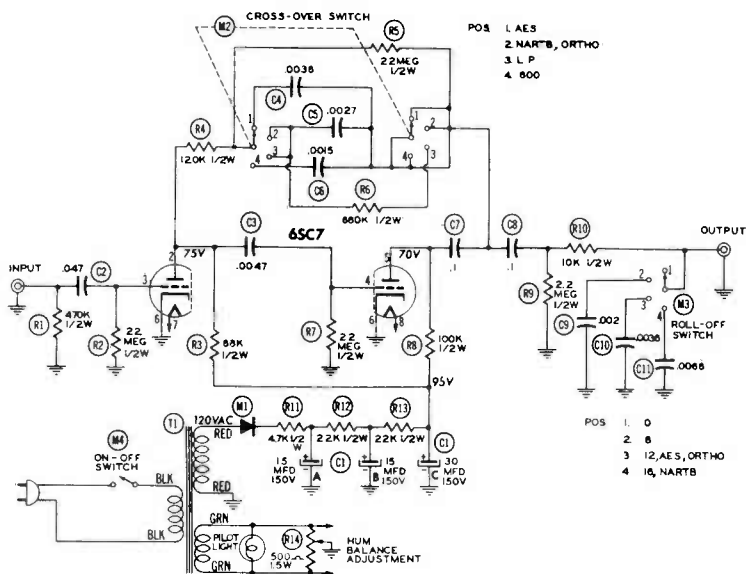
Fig. 5-34. A preamplifier with variable equalization.

with separate bass and treble controls. The frequencies that boost and cut are controlled by the values of C101, C102, C103, and C104. Loading of output and input must be compensated for when added according to value.
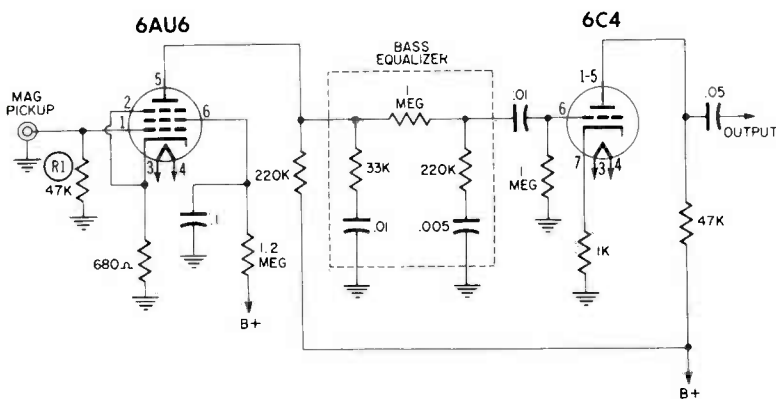
Fig. 5-35. A preamplifier with series equalization-attenuation.

Phono Preamp and Control

The unit shown in Fig. 5-34 is similar to and used primarily for the same purposes as the unit shown in Fig. 5-32. This unit has hi-fi magnetic pickup signal amplification and equalization but it has the added features of equalization selection. Examination of the circuit reveals additional switching for control of low- and high-frequency compensation. Low-frequency compensation circuits are adjustable by switch M2. High-frequency roll-off is accomplished by switching capacitor C9, C10, or C11 across the output with switch M3.

The method of attenuation in both circuits shown in Figs. 5-32 and 5-34 is degenerative. The component frequencies that are desired to be attenuated are fed back negatively, and those that are not desired to be attenuated are fed back in such a manner as to achieve proper emphasis and de-emphasis.

Equalization is also achieved in a simpler but effective manner by high- and low-pass filter arrangements inserted in series with the amplification circuits as is used in the circuit shown in Figs. 5-33 and 5-35.

The degenerative compensation circuit has advantages in the distortion and interference cancelling effects of negative feedback which is used over the entire audio spectrum; that is, if the signal is fed back so as to provide 20-db treble attenuation, the treble is actually attenuated 30 db and mid-range 10 db, so that the negative feedback affects the whole spectrum.

The circuit of Fig. 5-36 shows a transistor arrangement which meets the requirements of increasing the signal level from a pickup device to

156

about 1 volt rms. It also meets the requirement of providing compensation, if required, to equalize the input signal for a constant output with frequency when the pickup device is a tuner, magnetic microphone, phono cartridge (monaural or stereo), or tape head.

The preamplifier will match most magnetic pickup impedances. Input impedance to the preamplifier increases with frequency in switch positions 1, 2, and 3 because of the frequency-selective negative feedback to the emitter of the first stage. The impedances of magnetic pickups will also increase with frequency, but they are below that of the preamplifier.

The first two stages of this circuit have a feedback bias arrangement, with R101 feeding bias current to the base of the first stage that is directly proportional to the emitter current of the second stage. The output stage is well stabilized with a 10-kilohm emitter resistance.

The a-c negative feedback from the collector of the 2N508 (Q2) in the second stage to the emitter of the first stage (Q1) is frequency selective to compensate for the standard NAB or the standard RIAA recording characteristics for tape. The flat response from a standard NAB recorded tape occurs with treble control R91 near mid-position. There is about 6 db of treble boost with the control at 30K, and approximately 10 db of treble cut with it at zero. Mid-position of the treble control also gives flat response from a 7½-ips tape. This treble equalization permits adjustment for variations in input and output components.
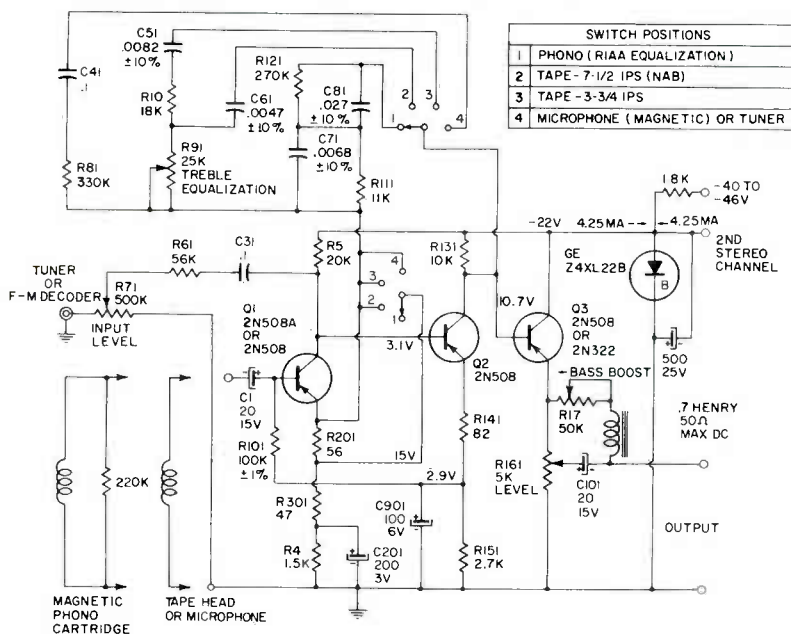
Fig. 5-36. Phono, tape, microphone, and tuner preamplifier.

A 3¾-ips recording tape head gives an equalized response with ± 1 db variation from 60 Hz to 12 kHz. Noise level is 76 db below reference-level output.

A good signal-to-noise ratio (S/N) can be realized with a tape head inductance of 0.4 henry. The S/N and dynamic range are improved by R141 in the emitter of the second stage, which reflects a higher input impedance from this stage and thus increases the gain of the first stage.

The voltage feedback from the collector of the second stage decreases at lower frequencies because of the increasing reactance of the feedback capacitor in series with the treble control. In the last switch position, capacitor C41 is large enough with R81 to make the voltage feedback (and thus the gain) constant across the audio spectrum. This flat preamplifier response can be used with a tuner, f-m decoder, or microphone. The input impedance to the preamplifier in switch position No. 4 is about 5.5 kilohms, and a 300-microvolt input level gives a 1-volt output (70-db gain). This sensitivity and input impedance gives excellent performance with low- and medium-impedance magnetic microphones. The noise is 65 db below the 1-volt output level. A magnetic pickup should be left connected at the preamplifier input while using the tuner or f-m decoder. This tuner input has a sensitivity down to 250 millivolts.

The RIAA feedback network first switch position has capacitor (C71) for decreasing the amplifier gain at the higher frequencies. This eliminates the need to load a magnetic cartridge with the proper resistance high-frequency compensation. An input level of 7 millivolts gives a 1.5-volt output.

The emitter-follower output stage of the preamplifier gives a low-impedance output for a cable run to a power amplifier (transistor or tube) and acts as a buffer so that any loading on the preamplifier will not affect the equalization characteristic. The preamplifier output should not be loaded with less than 5 kilohms, preferably about 15 kilohms.

Control-Center Preamplifier Circuit

It has been shown that to obtain more control features and flexibility, more gain is required and therefore more stages, circuits, and components. Quality design becomes increasingly difficult, especially at the input stage of a high-gain preamplifier. This stage is probably the most critical because of noise conditions. Any noise additional to the original signal produced or picked up in the first stage will be amplified in direct proportion to the total amplification of the input signal. If the input signal is low, as it is from magnetic and dynamic phono pickups, the first-stage noise and hum pickup must be proportionately very low to not cause noticeable interference. Hum pickup from a-c–operated filaments is such a problem when low-output magnetic pickups are used that several manufacturers of high-grade preamplifiers have a special rectifier and filter to provide current for the first or more stages in the preampli-
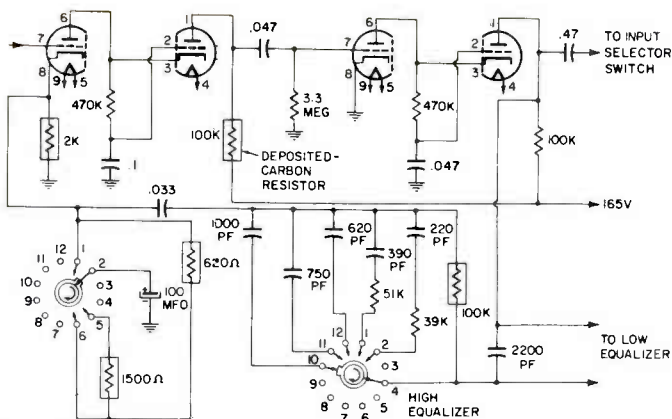
Fig. 5-37. Application of low-noise deposited-carbon resistors to reduce input-stage noise.

fier. Still further precautions are taken in designs by having a separate chassis for the power supply and a provision for d-c filament supply. The real advantage of this is again experienced in the input stage. Further noise reduction is attempted by use of special low-noise resistors in loading-circuit controls and networks where carbon resistors do develop and add noise to the signal as it passes through. Circuits where these special resistors are applied—plate, cathode, and compensating networks—are shown in the schematic diagram of Fig. 5-37 from a Bogen preamplifier design. The rectangular resistor symbols are low-noise deposited-carbon resistors.

The input section of preamplifiers and controls also contains the switching circuits for changing various input sources as desired. One interesting arrangement having considerable flexibility is found in the master control as shown in Fig. 5-38. This unit provides for seven inputs. The input stages provide the proper impedance match to the input unit. How this is accomplished is also shown in Fig. 5-38.

This unit has been conservatively designed on several principles: one, that an equipment having higher than necessary output capabilities will give better fidelity when working on a smaller portion of its overall response curve, and, two, that process quality control by use of close-tolerance components in all critical circuits will ensure that each input produced will be very similar in quality of operation to a laboratory model.

Naturally this type of unit will cost more than an ordinary model, but it has features and specifications as follows:

Hum: inaudible at full gain (20 volts output, whereas usually less than 5 volts is required to drive most basic amplifiers).
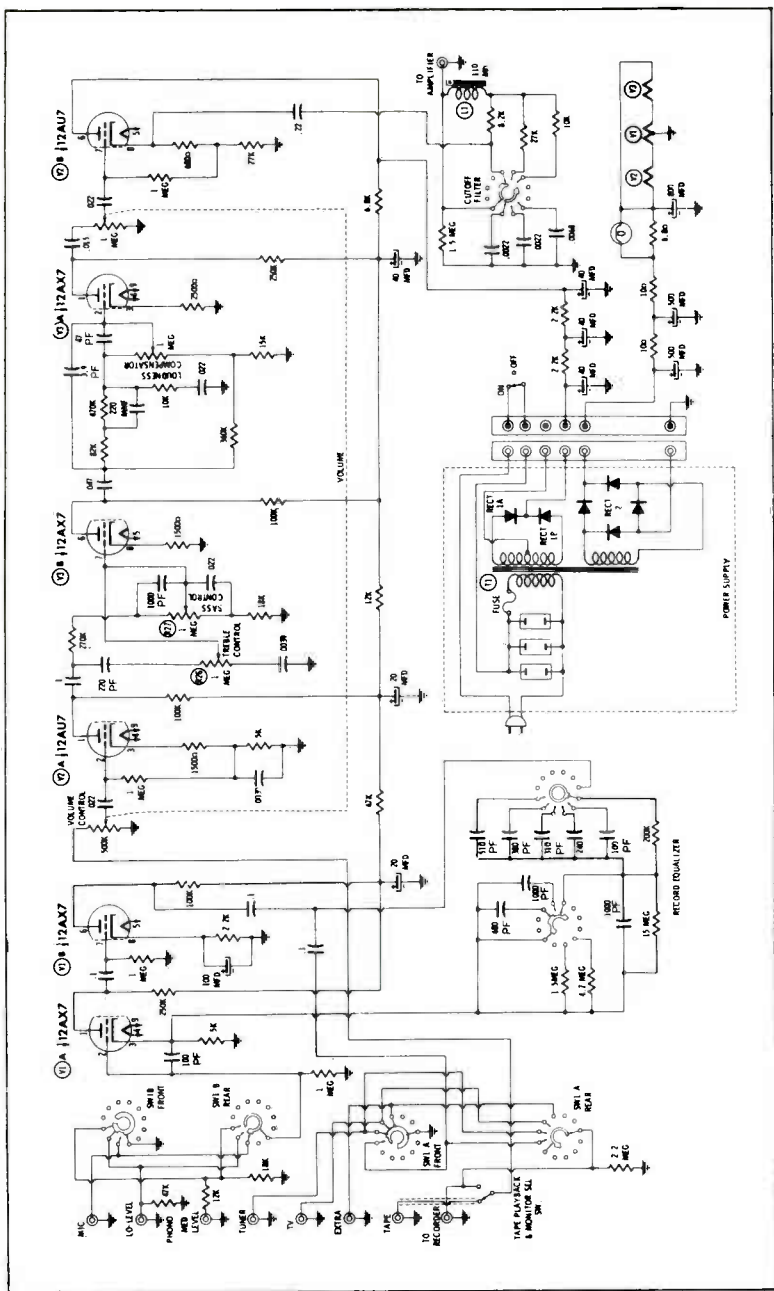
Fig. 5-38. A single-channel laboratory-type preamplifier equalizer.

160

Noise: 70 db or lower than average low-level signal input from magnetic phono pickup (10 mv).

Intermodulation Distortion: less than 0.1 percent at normal output (2 volts).

Frequency Response: ±1 db from 20 Hz to 60,000 Hz.

Controls:
A. Switching
   (1) Low-level microphone
   (2) Two low-level magnetic cartridges
   (3) F-m or a-m tuner
   (4) Television
   (5) Ceramic phono pickup cartridge
   (6) Tape playback or monitoring (monitor as you record)
B. Equalization compensation. Six accurate equalizer turnover positions. Six accurate treble roll-off positions
C. Loudness compensation. A continuously variable loudness control which is so designed as to compensate for the Fletcher-Munson hearing characteristics at various levels. (See Figs. 1-1 and 5-39.)

To obtain consistency and quality, over a dozen accurate low-noise resistors are used, a dozen low-tolerance silvered mica capacitors, and a high-Q toroidal coil for cutoff filter circuits. Filaments operate on direct current which has three stages of hum filtering. A separate plug-in mini-
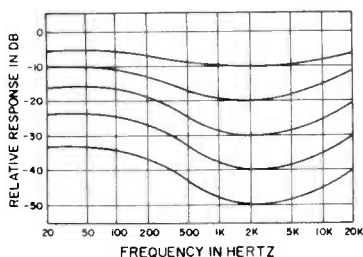
(A) *Maximum position.*
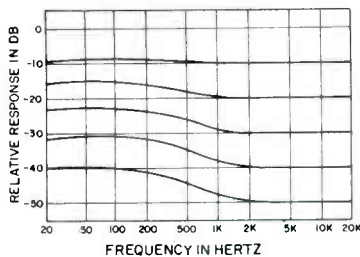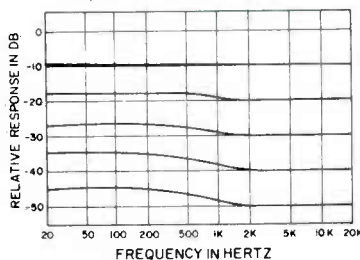
(B) *Second position.*

Fig. 5-39. Response of the amplifier shown in Fig. 5-38 at various positions of the loudness control.

(C) *First position.*

161

ature power supply contributes to reduction of hum to below the thermal noise level so as to be inaudible and not measurable at full gain and with full bass boost. A variable high-frequency cutoff filter is provided for use with tuners to eliminate a-m interstation beat whistle (10 kHz). This circuit also provides positions to quiet undesirable record-surface noise, and tape and f-m tuner hiss.

There are several circuits common in principle to almost all the foregoing control centers. One such circuit is the bass and treble controls designed to provide from 15 to 20 db boost or attenuation of bass or treble frequency components separately. This kind of circuit is shown in Fig. 5-38. Treble control R26 (Fig. 5-38) and associated components provide adjustment of the high frequencies by variable high- and low-pass filtering. Bass control R27 and associated components provide a similar effect in the bass regions. These controls are arranged in high- and low-pass filter networks and made variable about a design center which constitutes flat position when the controls are on dead center. Turning to the left droops and turning to the right boosts high- and low-frequency notes for each control, respectively.

Another feature common to most advanced control centers is the convenience of two outputs: one to feed a tape recorder for recording one's own program material, and one to feed the amplifier system. These outputs are usually fed through a cathode-follower circuit arrangement in order to reduce the output impedance. The advantages of the high-impedance input and low-impedance output characteristics of a cathode-follower output circuit is that the low-impedance output characteristic permits the use of long connecting lines between the preamplifier and the basic amplifier without danger of hum pickup or loss of high-frequency components of the signal.

In preamplifiers having direct-coupled output from the plate of the last stage, a short shielded lead, preferably less than a foot or two feet at most, is desirable because this lead will be sensitive to surrounding magnetic fields which may induce undesired noise components. Also the capacitance of the lead may attenuate the highs.

A loudness control is usually furnished as a switch for a common setting or as a variable control to compensate for different degrees of human hearing response. Refer to the Fletcher-Munson curves at different audio volume levels, as shown in Fig. 1-1.

Common to all units, either coupled to a loudness control or separate, is the volume control. This control is usually inserted in either one or two grid circuits or on the output side of a cathode-follower output stage. The latter has the advantage of simple uncritical design in that it does not affect frequency response in any way.

The grid-connected ganged volume control as used in the circuit of Fig. 5-38 prevents overloading by too strong a signal before it reaches the last stages of the system.

162

STEREO AMPLIFICATION

The overall amplifying functions of a stereophonic high-fidelity system can also be divided into two or more channels, each having three main parts, as illustrated in Fig. 5-40. The first part, frequently physically separate, is called the *preamplifier* and is similar to single-channel systems. Although it amplifies the signal, its primary jobs are not amplification but: (1) to compensate for frequency characteristics of source material (mainly discs and pickups), (2) to provide adjustment of balance among the signal components of high, low, and medium frequencies (and for speaker compensation and listener taste), and (3) to provide control of loudness and other effects.
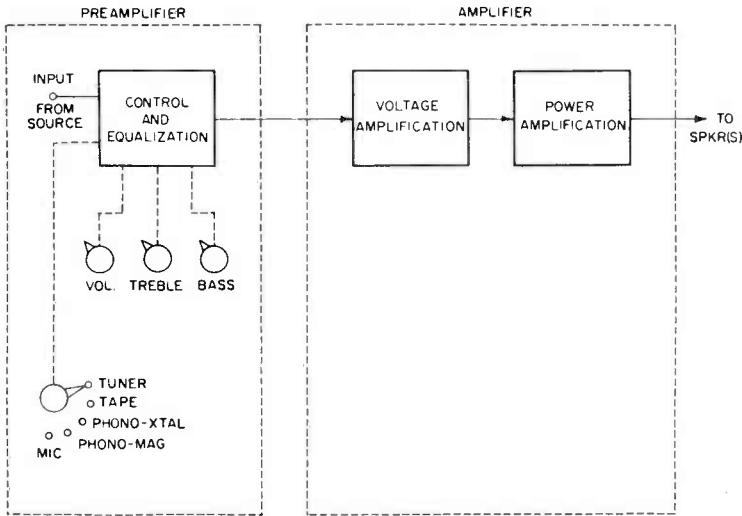
Fig. 5-40. Diagram showing the major divisions and functions of one channel in a stereo amplifying system.

The second and third amplifier parts in Fig. 5-40 are usually combined physically into what is known as just "the amplifier." The voltage amplifier brings the signal voltage up to a level sufficient to drive the power amplifier. In addition it provides phase splitting so that two equal signal components, 180 degrees out of phase with each other, are available to drive grids of the push-pull power amplifier. The power amplifier is very similar to the ones found in single-channel systems.

Basically, stereo amplifiers are different from ordinary amplifiers because they must amplify two or more separate channels rather than just one. The first stereo systems used two complete single-channel amplifiers, separate physically as well as electrically. The block diagram in Fig.

163

5-41 illustrates how two amplifiers, or amplifier channels, are combined for stereo. Special features differing from monophonic arrangements may be summed up as follows:

1. Use of two amplifying channels, and combination of these channels physically into relatively compact arrangements.
2. Provision of additional controls, to allow adjustment of relative output amplitudes for optimum stereo effect. In many cases, volume, loudness, and tone controls for the two channels are ganged to simplify operation.
3. Arrangements for blending the outputs and for providing output for a third ("middle") speaker when desired.
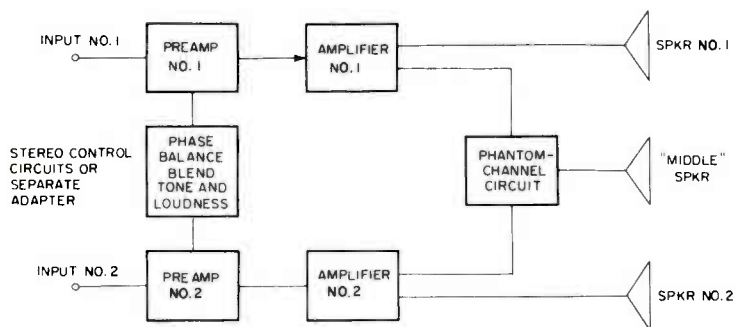
Fig. 5-41. How two amplifier controls are combined for stereo.

Physical Arrangements

The simplest way to provide two complete electrical channels is to provide two complete amplifiers and speaker systems, and in some cases this is done. However, this makes the system bulky and costly. Accordingly, various ways of physically combining parts or all of the amplifiers physically for reduction in size and cost have been devised.

Just mounting two complete amplifiers on one chassis is sufficient to reduce weight and size considerably. A power supply of twice the current capacity is considerably less than twice as large or heavy, and the use of one, instead of two chassis (even though it is a little larger than either of the two it replaces), makes the weight and size much less.

One interesting way that has been used to combine the two channels physically is the use of dual tubes for all stages except the power amplifier. Just as many functional tube sections are used, but because each pair of voltage amplifier tubes is in the same envelope, space is conserved.

Control Functions

When two amplifying systems are used together for stereo, they cannot provide the desired stereo effect unless their operation is coordinated.

164

Although the pioneers of stereo employed two completely separate amplifiers, this required operation of two complete sets of controls, including loudness, treble, bass, and equalization selection. Having all these controls separate makes a flexible setup for controlled laboratory experiments, but is not practical for the home stereo system. Therefore, in most commercial stereo equipment, at least some of these controls are combined for greater ease of operation. Let us review the controls having special significance in stereo.

1. *Balance Control*—This is a combination of the volume controls of the two channels. It has additional significance compared with monophonic volume controls because in stereo not only the volume of the sound but also the relative volumes of sound from the two channels are important. A balance control for stereo is usually a ganged potentiometer with one section for each amplifier. When the shaft is rotated clockwise, the output of one channel increases while that of the other channel decreases; at about mid-position the outputs are the same. Adjustment of this balance control takes care of differences in source equipment outputs, gain of the two amplifiers, difference in efficiencies and directive properties of the two speaker systems, speaker locations, room acoustics, etc. If a stereo program is being reproduced, adjustment of the balance control should make the overall source location appear to move toward the left, right, or center, depending on whether the left or right speaker (or center) output is being increased.

In some cases, both balance and individual volume controls are included. The individual controls are "one-time" adjustments to set the balance control. A monophonic signal is fed through the amplifiers and the listener stands at an appropriate location for listening; then, with the balance control set at mid-position, the individual loudness controls are adjusted until the sound seems to come from midway between the two speakers, and the volume is at the desired level. The volume controls can then be left set at this level and the balance control used to compensate for different types of program material.

2. *Master Gain Control*—Even though the volume controls are adjusted for comfortable volume during the balance adjustment, the listener may still want to adjust overall volume without disturbing balance. This is the purpose of a master gain control. One way of providing such control is to gang the two loudness controls through a mechanism that allows either individual or ganged operation. Such a mechanism is now used on some amplifiers. The shafts for the two controls are concentric, like those on some television controls. By pulling out on the inner shaft, the operator unlocks the controls and allows independent adjustment. After the balance adjustment is completed, the shaft is pushed in again, and the combination used as an overall loudness control.

3. *Blend Control*—This is used to "dilute" the stereo effect. It allows some of the L signal to be fed to the R or center channel or some of the

R signal to the L or center channel. This control is useful mainly when pickup microphones are spaced far apart and there is too strong a left, right, or "hole-in-the-middle" effect in playback.

4. *Quasi-Stereo Switch*—This allows switching either the L or the R signal alone into both channels. The purpose of this is to use both channels, and thus both speaker systems, for a monophonic source; those who have tried this say it makes for more pleasing listening than with a single channel. However, with matrixed (sum and difference) connections, such as those used in tuner multiplex, this is automatically done with monophonic material. Therefore, the main use of the quasi-stereo switch has been with a conventional tuner or monophonic pickup or tape head wherein a single channel is connected to only one of the amplifier's input circuits.

5. *Cartridge Paralleling*—This is very important for playing monophonic discs with a stereo cartridge. As explained earlier in this chapter, rumble in a turntable has a strong vertical component which can be picked up by the channels of a stereo cartridge. However, the rumble signal of one channel is out of phase with that of the other channel. Connecting the two channels in parallel for monophonic playback causes the rumble signal to cancel out. Thus, although in monophonic playback both channels reproduce the same monophonic signal, they should be paralleled for rumble reduction. In addition, the signals from the two channels combine to produce a higher-energy output.

6. *Phase-Reversing Switch*—This reverses the phase of one of the stereo signals with respect to the other by simply reversing connections of the two leads carrying the signal of one channel at the input. Although this should seldom be necessary after initial setup of the stereo system, some audiophiles like to be able to change phase. They feel that in some studio programs and in some home recording experiments errors in phasing will crop up.

7. *Input Equalization*—Both stereo channels require the same input equalization for record and pickup characteristics as do single-channel systems. For this reason, input selectors are usually ganged, with one control for both channels.

Stereo Adapters

The controls described in the foregoing discussion for the most part require connection into the circuits of both channels of the stereo system. This means that for the use of two separate amplifying systems not designed for stereo the owner must break into the circuit somewhere and insert these controls.

To simplify use of separate monophonic systems for stereo, several types of adapters have been developed and are available commercially. A block diagram of such an arrangement is illustrated in Fig. 5-41. The adapter connects to both preamplifiers, either at the outputs or at a jack

166

called "Tape Output." This jack ordinarily allows the signal from the disc or tuner into the preamplifier to be tapped off and fed to a tape machine for recording. In this case, the output can be applied to the stereo adapter.

An adapter contains the essential controls that are not used in a monophonic (single-channel) system and also provides for coordination of the two signals. The two stereo signals leave the adapter unit and are fed to two standard single-channel amplifiers arranged for stereo. A typical adapter is pictured in Fig. 5-42.

Courtesy Lafayette Radio Corp.
Fig. 5-42. Low-cost compact-type stereo adapter.

Although a few compact low-cost stereo preamplifiers are available (such as the one shown in Fig. 5-43), most stereo preamplifiers are of the more sophisticated type illustrated by the examples shown in Fig. 5-44. To illustrate the features of a typical preamplifier control center and clarify the meanings of the various controls, let's take the example of a rather elaborate unit, illustrated in Fig. 5-45.

The Selector control in the upper left corner selects the input and impedance required for proper load for the source device. It has positions for microphone, tape head, phono pickup cartridge, f-m and a-m tuner, tuner with multiplex, and tape recorder inputs. The distinction is made between a direct connection of the tape head for tape playback, in which proper loading and equalization for the head itself is necessary, and the input to the preamplifier from the output of the playback amplifier of the complete tape recorder.

The Mode Switch is in the lower left corner of the unit in Fig. 5-45. This switch allows manipulation of the two channels of the preamplifier

Fig. 5-43. A stereophonic preamplifier.

so they can be used for normal stereo operation (Normal), with the two inputs swapped (Reverse), with just channel A, or with just channel B.

Just to the right of the Mode switch is the Master Volume control. This is a dual control, which simultaneously adjusts the gain of both channels. Because the Master Volume control adjusts volume without allowing for the loudness characteristic of the human ear, a Loudness control (just to the right) is also included. This is really a switch, providing stepped increases or decreases of volume in both channels simultaneously. At each step, an equalizing network compensates for the response of the listener's ear (see Fig. 5-39).

To the right of the Loudness control is the Stereo Balance control. In its center (up) position, it allows approximately equal gains in the two channels. As it is turned toward the right (clockwise) it increases the gain of channel B and reduces the gain of channel A until, at its extreme position, output is obtained only from channel B. When it is turned to

(A) Marantz stereo preamplifier and control center.

(B) McIntosh preamplifier control center.

Fig. 5-44. Examples of preamplifier units designed for stereo.

Fig. 5-45. A typical stereo preamplifier and control unit.

the left (counterclockwise) it increases gain of channel A and decreases that of channel B in the same manner. This control is used for adjustment for optimum stereo effect, which may be realized at different adjustments for different types of program material and different stereo systems.

To the right of the Stereo Balance control are the tone controls for adjusting the degrees of bass and treble boost. This unit differs from many others in that the Bass and Treble controls are step switches instead of continuous controls, and in that here the equalization for different playback characteristics (LP, NAB, RIAA, and AEB) is provided by adjustment of the boost controls instead of at the input switch. In other units the equalization characteristics are selected by the input selector, which will have such labels as ceramic, crystal, RIAA, etc., and provide continuous adjustment of the bass and treble boost.

Two output-level meters are shown in the upper middle portion of the panel in Fig. 5-45. These are used to check the balance between the two signals during either playback or recording. The switch located between the meters transfers the meters to register for recording or playback, and the controls adjust recording gain and playback gain independently. The control in the upper right-hand corner is used to turn the unit on, off, or provide connection for automatic shut-off. The latter is used when the phono turntable mechanism is interconnected with the preamplifier to shut off all power when the records of a stack have all been played.

Amplifiers

Power amplifiers for stereo reproduction have the same requirements (electrically) as amplifiers for monophonic reproduction. They must have sufficient output to operate the speaker system at a desirable level, low harmonic and intermodulation distortion, frequency response from 20 or 30 Hz to 20 kHz or more, and good response to transients. If any of the requirements of high fidelity for monophonic amplifiers are compromised, the full benefit of stereo reproduction cannot be realized.

The primary differences between stereo amplifiers and high-fidelity monophonic amplifiers are physical in nature. First, there must be two or more separate amplifier circuits or amplifier channels. These are usually combined on one chassis. Of course, there is nothing to stop one from buying two or three separate amplifiers, but the compactness of combination units is an advantage.

Because all of the manual controls can be provided in the preamplifier which feeds the amplifier, most amplifier units are designed with only a limited number of controls, and these controls need adjustment only once, or at the most, just occasionally. Some examples of stereo power amplifiers are illustrated in Fig. 5-46. Each can be concealed completely in a cabinet, with all the controls located on the preamplifier panel.

Courtesy Marantz Co., Inc.

(A) *Marantz stereo power amplifier only.*

Courtesy McIntosh Laboratory, Inc.

(B) *McIntosh stereo power amplifier only.*

Fig. 5-46. Typical stereo power amplifiers.

Transistor Stereo Control Centers and Amplifier Units

The trend in hi-fi stereo design is toward multiple units on one chassis, and compactness without loss of quality. For this reason, transistors are well suited for high-fidelity control-center amplifiers. Hum pickup from wiring, transformers, microphones, and other components is reduced because transistors do not have filaments. Circuit arrangements and layouts can be made more efficient, due to elimination of wiring and due to the transistor's small physical size. Shielding requirements are reduced for the same reasons. Since transistors are low-impedance devices, they, compared with tubes, more nearly match magnetic pickups, dynamic microphones, tape decks, speakers, and lines, with increased efficiency in power transfer and reduced component requirements.

Arrangements utilizing transistor application and transistor/tube combinations with nuvistor input stages, field-effect transistors, and silicon and silicon-planar transistor types have made many new designs possible, and the state of the art continues to advance. Transistors allow reduction in size. In some cases, transformers can be eliminated.

Properly designed circuits for transistor use or transistor/tube combinations with negative feedback can provide for high-fidelity equipment the required wide frequency response and low distortion, equaling that of circuit arrangements utilizing only vacuum tubes.

It is difficult to attain faithful reproduction of all signals with transformer-coupled circuits. High-quality transformers are required, and, to have a good response at low frequencies, they are physically large and expensive.

Transistors have made possible practical development of transformerless circuits for audio frequencies, since the transistors for audio are basically low-voltage, high-current devices. The emitter-follower stage, in particular, offers interesting arrangements, since it has low inherent distortion and low output impedance.

A hi-fi stereo preamplifier-amplifier utilizing transistors throughout is shown in Fig. 5-47A.
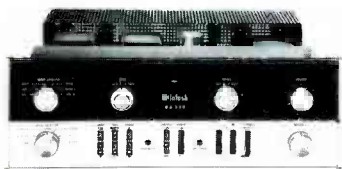
Courtesy McIntosh Laboratory, Inc.

(A) *McIntosh MA230 transistor/tube preamplifier/amplifier (60 watts).*

Courtesy Lafayette Radio Corp.

(B) *Lafayette LA-90T transistor amplifier with controls.*

Courtesy Heath Co., a subsidiary of Daystrom, Inc.

(C) *Heathkit solid-state amplifier Model AA-22.*
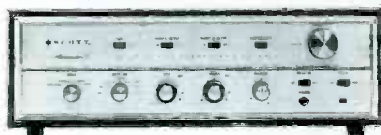
Courtesy H. H. Scott, Inc.

(D) *Scott Stereomaster 260 audio control with 100-watt power amplifier self-contained.*

Fig. 5-47. Transistor amplifiers.

Combination Units

The trend toward combining all units except speakers into one compact cabinet is reducing cost and increasing the adaptability of hi-fi equipment to all sizes of homes.

For the most exacting requirements in hi-fi, separate units still have some advantage, but the latest designs of stereo "compacts" are hard
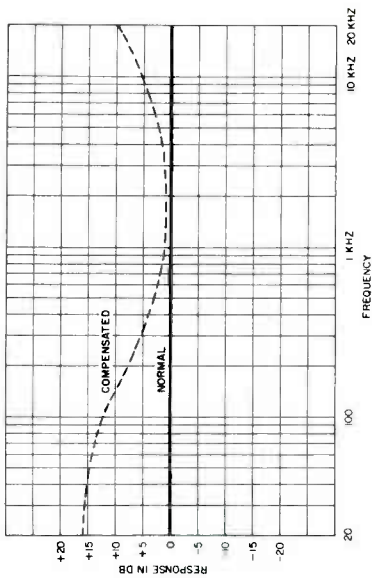
(A) Loudness control response.
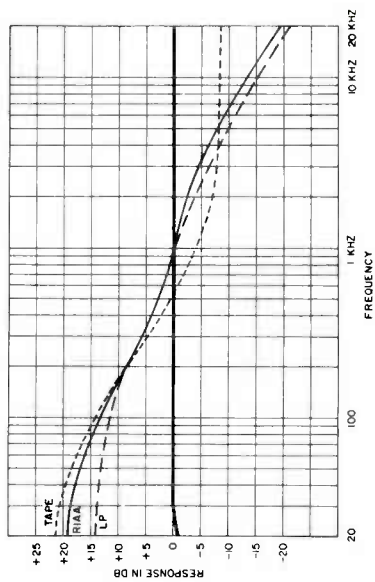
(B) Equalization curves.
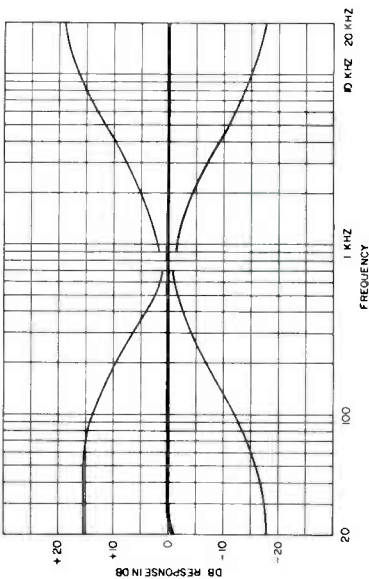
(C) Bass and treble control response.
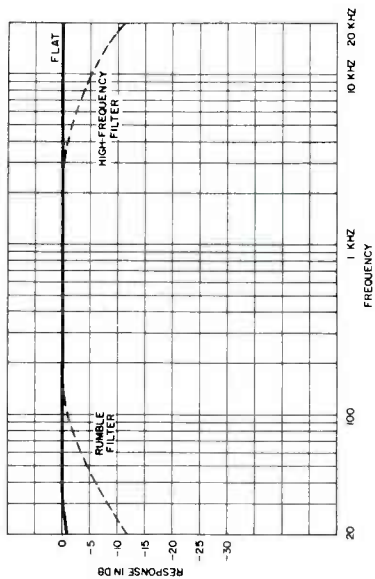
(D) Rumble and high-frequency filter response.

Fig. 5-48. Response curves for MAC 230.

(A) *McIntosh 85-watt MAC 1500 receiver.*

(B) *Altec 711A preamp, 100-watt amp, and tuner.*

Fig. 5-49. Stereo receivers.

to beat. For average use, they are more than adequate, and it is expected that in time they will equal the best of separate units.

Fig. 5-47 shows stereo preamplifier power-amplifier combination units of high quality. These preamplifier power-amplifier combinations are all solid-state, except for the McIntosh MA 230, which is solid-state except for the output stages and drivers. These stages utilize vacuum tubes and unity coupling to reproduce the finest quality currently available—with universal matching.

Overall response curves of a preamplifier and power-amplifier combination in one unit (McIntosh MA 230) are shown in Fig. 5-48.

Figs. 5-49A and 5-49B show two tuner stereo control-center amplifier compacts. Fig. 5-50A shows a combination record changer, tuner, control center, and amplifier. Fig. 5-50B shows a similar compact combination with $3\frac{3}{4}$-ips cartridge tape deck added.

A high-quality tuner-preamplifier power-amplifier arrangement is the MAC 1500, shown in Fig. 5-49A. The block diagram for the MAC 1500 is shown in Fig. 5-51. This unit is complete, compact, and contains three basic separate sections integrated into a compact, easy-to-operate stereo

(A) *Scott stereo compact combination system.*

(B) *Bogen MSC compact with 8-track tape cartridge.*

Fig. 5-50. Compact stereo systems.
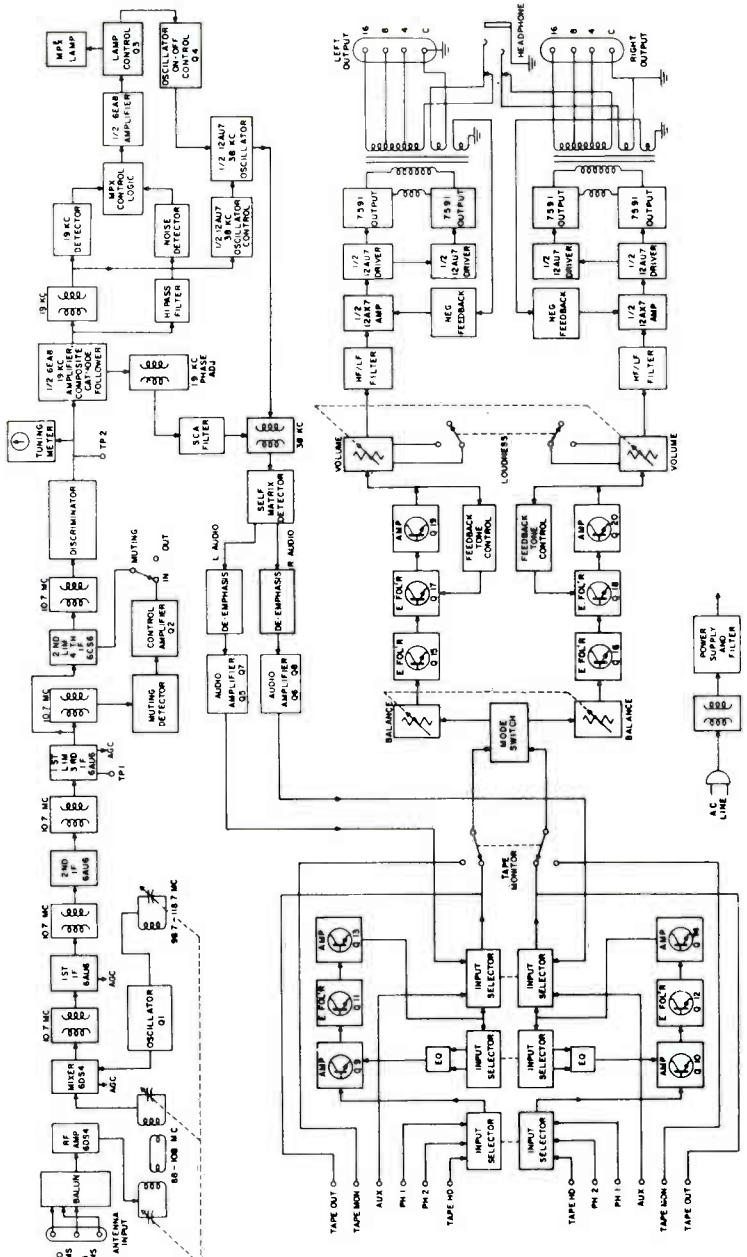
Fig. 5-51. MAC 1500 stereo receiver.
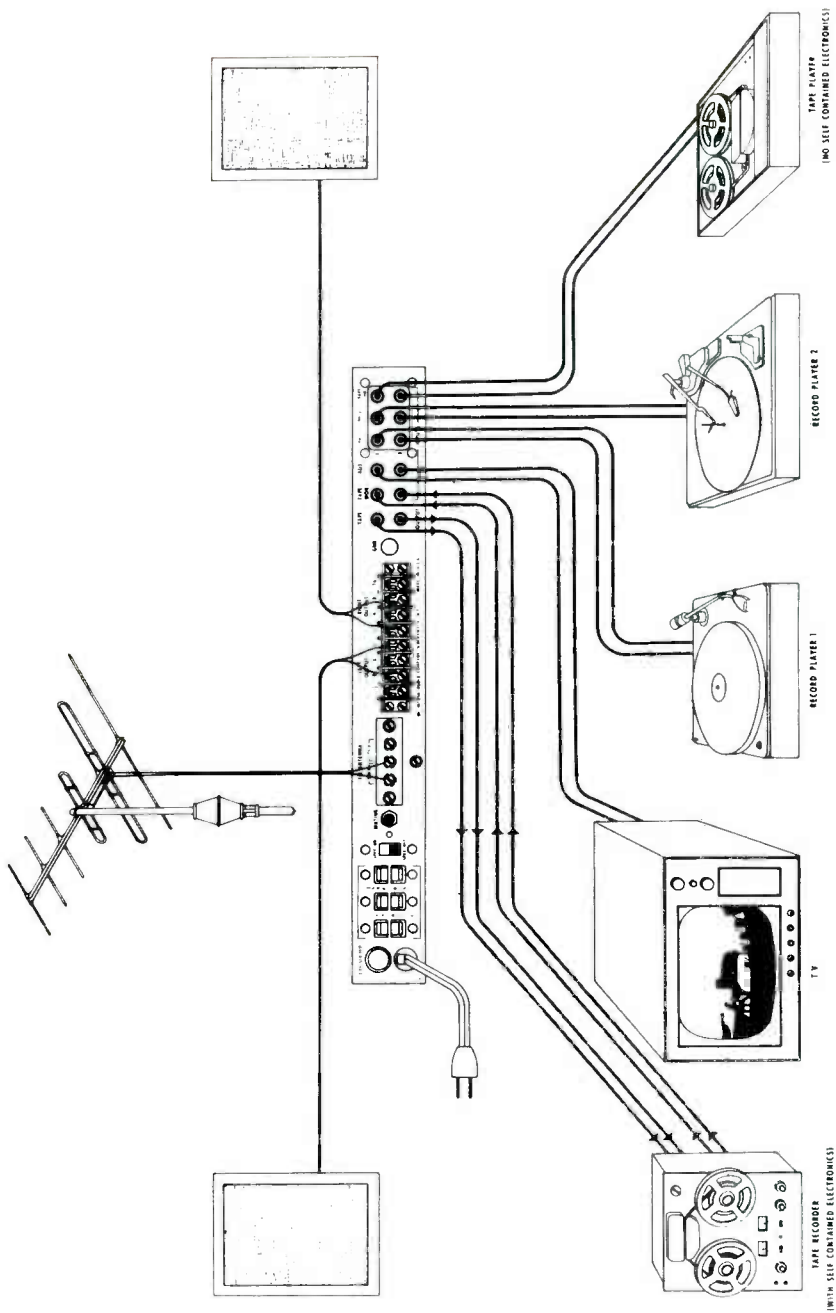
Fig. 5-52. Typical stereo arrangement (MAC 1500).

FM TUNER SECTION:

USABLE SENSITIVITY (at 100% modulation): 2.5uV (I.H.F. Standards)

SIGNAL TO NOISE AND HUM RATIO: 65db.

HARMONIC DISTORTION: Mono, less than .5%, Stereo, less than .8%

DRIFT: Less than 25KC

FREQUENCY RESPONSE: Flat from 20 cps to 20KC with standard 75 μ second deemphasis and 19KC pilot frequency filter.

CAPTURE RATIO: Better than 2.0db.

IMAGE REJECTION: Better than 60db.

STEREO MULTIPLEX SEPARATION: Better than 30db at 1KC.

SPECIAL FEATURES:

a) Automatic stereo switching.

b) Muting: IF Injected circuit with at least 50db quieting between stations.

c) Antenna inputs for 300 ohm balanced (for twin lead) and 75 ohm unbalanced (for coaxial cable).

d) Nuvistor RF amplifier, Nuvistor mixer.

e) Four stages of IF amplification, with AGC used to insure that limiting occurs only in the limiter stages.

f) Two limiter stages used for exceptional capture ratio and smooth muting operation.

g) Multiplex filter and SCA filter, to suppress 19KC and 38KC signal components at least 40db below program and to suppress 67KC SCA by 60db.

h) Noise immune logic circuit used to activate MPX stereo light and automatic stereo switching on 19KC stereo pilot only.

i) D'Arsonval tuning meter for accurate center of channel tuning.

j) Flywheel tuning for ease of operation and precise tuning

AMPLIFIER SECTION:

POWER OUTPUT: 60 watts continuous, 30 watts per channel, 85 watts total music power (I.H.F. Standards).

HARMONIC DISTORTION: Less than 0.5% 30 cps to 20KC, both channels operating at rated output at the same time.

INTERMODULATION DISTORTION: Less than 0.5% for any combination of frequencies from 30 cps to 20KC at rated output (equivalent RMS watts). I.M. decreases as output power is reduced.

FREQUENCY RESPONSE: ±0.5db 20 cps to 20KC. (Power amplifier response is 2 cps to 150KC at −3db; power bandwidth is 19 cps to 30KC.)

OUTPUT IMPEDANCE: 4, 8, and 16 ohms. Rated output is delivered by any of these impedances.

DAMPING FACTOR: Greater than 10.

INPUT SENSITIVITY AND IMPEDANCE:

Tape Head—2.8MV, 1 megohm
Phono 1—5.6MV, 47K ohms
Phono 2—2.8MV, 47K ohms
Aux—400MV, 500K ohms
Tape Monitor—400MV, 500K ohms

TAPE OUTPUT LEVEL: Tuner—1.2 volts, for other inputs 400MV at rated sensitivity.

HUM AND NOISE:

Power Amplifier, −90db
High Level Inputs, −75db
Low Level Inputs, −60db

TONE CONTROL RANGE:

Bass control + 15db to −18db at 50 cps.
Treble control + 15db to −15 db at 10KC.

L.F. and H.F. FILTER:

L.F. filter, cutoff frequency = 50 cps, attenuation rate = 12db per octave

H.F. filter, cutoff frequency = 5KC, attenuation rate = 12db per octave

SPECIAL FEATURES:

a) Loudness control for full fidelity listening at low levels.

b) Tape monitor switch for listening to recorded tape program while recording.

c) Tone controls are "clutched" for tracking operation of left and right channels or independent operation.

d) Headphone output fed from special tap on output transformer for maximum signal to noise ratio. Speakers may be automatically muted when headphone plug is inserted, by use of rear mounted switch.

e) Silicon transistors used in all preamplifier stages for low noise and hum free operation.

f) High quality epoxy circuit boards for reliable long life performance.

g) High quality conservatively operated components are used throughout for long life.

h) Zener regulated power supply is used for critical circuits.

i) Equalization and tone control circuits are feedback type for lowest distortion and greatest accuracy.

MISCELLANEOUS

FINISH: Gold anodized panel and knobs, black finished chassis.

DIMENSIONS: 16" wide, 5½" high, 16" deep.

WEIGHT: 40 lbs., shipping weight 54 lbs.

POWER CONSUMPTION: 180 watts, 120 volts, 60 cps.

TRANSISTOR AND TUBE COMPLEMENT:

19 silicon transistors
1 germanium transistor
15 tubes
16 diodes, rectifier, and zener diode

Fig. 5-53. Example of technical features and specifications (MAC 1500).

unit. The best advantages of tubes, solid-state devices, and nuvistors have been obtained and combined into a practical size about the same as other individual units of equal quality and function. Fig. 5-52 shows a typical interconnection arrangement for a hi-fi system built around this unit. Technical specifications of this unit (MAC 1500) are shown in Fig. 5-53.

Fig. 5-54. Remote-control unit used with stereo preamplifier to control volume and balance from the listening position.

Many listeners like to be able to control volume and balance from the listening position since this is the only place adjustment can be made for the proper effect. To provide for this need, manufacturers are making remote-control units available. A typical remote-control unit is illustrated in Fig. 5-54. This unit has a 30-foot cable, the other end of which plugs into the tape input and output jacks of the preamplifier. Thus when the controls are set for tape record position, the outputs of the amplifier must pass through the cable to the remote-control unit and back to the preamplifier before passing on to the amplifier. In the process the volume and balance may be controlled by the remote unit.

Balance and Tuning Instruments

One of the functions important for the maintenance of good stereo effect is the preservation of balance in the system. If you have a preamplifier like the one shown in Fig. 5-45, the panel meters can be used to check balance when a test tone is applied to both input channels at once. However, most systems do not have built-in meters like this. Therefore, an external metering device which will indicate when balance is achieved is useful. Two examples of commercially available balance meters are shown in Fig. 5-55. These meters can be attached to the outputs of the amplifiers and adjustments made until the meter needle is centered—indicating the two signals have exactly the same amplitude. It should be noted that this balancing is not normally done with the balance control; rather, fixed-setting adjustments (screwdriver adjustments) are set for balanced output with the balance control in its center position. Then, the balance control can be adjusted later, during actual operation of the system, to compensate for such things as speaker types and placement, room acoustics, etc.

177
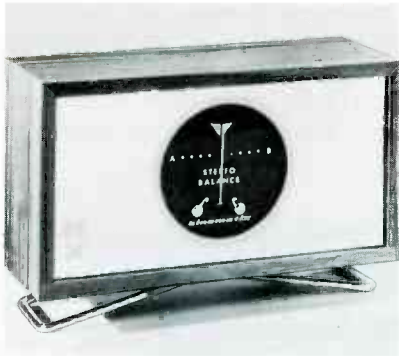
(A) *Kinematix stereo balance.*
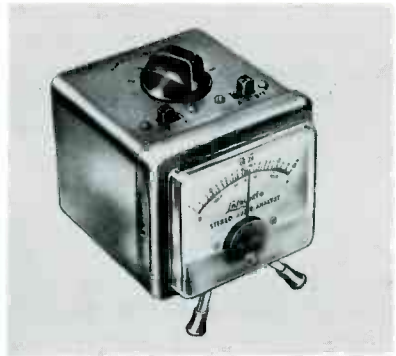
(B) *Lafayette stereo audio analyst.*

Fig. 5-55. Typical stereo balance meters.

A common cause of distortion is multipath reception of the f-m signal, which is caused by reflected signals from the same transmitter arriving at the receiving antenna slightly later than the direct signal. This can usually be corrected by antenna rotation or replacement. To check this condition, an indicator using a 3-inch cathode-ray tube is available. This instrument is shown in Fig. 5-56. The instrument is made by McIntosh. It will also check signal strength and indicate correct tuning position more accurately than a meter; it will indicate the nature of the signal, viz., monaural or stereo, the relative phase of the two channels, which

Fig. 5-56. McIntosh maximum performance indicator.

channel is being used if only one is used, channel balance, and separation. For those who want the ultimate in f-m stereo reception, this instrument is a valuable tool. There are other receivers available with similar features built into one integral unit.

Speakers

The speaker system constitutes several links in the overall high-fidelity equipment chain. These links are illustrated in the block diagram of Fig. 6-1. Each link must be considered not as a separate entity but in its relationship to the links which precede and follow it.

The amplifier output stage may be considered the energy source which supplies the driving power to the speaker system. Because most speaker electrical input circuits have a low impedance (2 to 20 ohms), most systems must employ an output transformer to match this low impedance to that of the output amplifier, which is ordinarily about 1500 to 6000 ohms. As will be explained, the transformer works both ways. The load on the speaker system is reflected back to the amplifier stage and influences its operation, and the amplifier impedance is reflected forward to contribute to the load on the speaker driver and radiator.

The speaker system itself can be divided into three functional parts:

1. The electromagnetic part, consisting of the voice coil and field magnet. Audio-frequency electric current in the coil causes mechanical motion of the cone or diaphragm on which it is mounted. This part is often referred to as the *driver* or *motor* of the system.
2. The mechanical part, on which the driving coil is usually mounted and which is set into mechanical motion by the audio-frequency electric current in the driving coil.
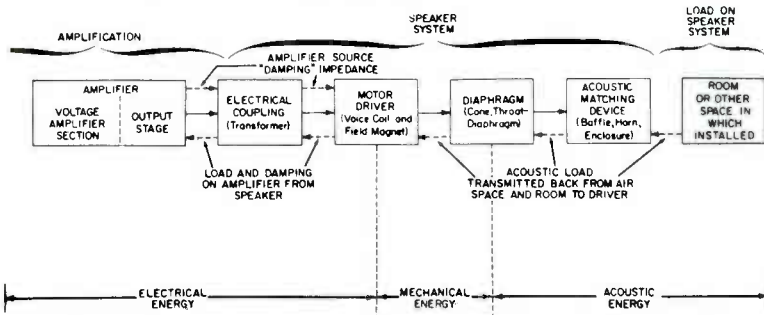
Fig. 6-1. Block diagram showing detailed sections and functions of the output of a high-fidelity system.

3. The acoustic part, which transmits the sound energy developed by the mechanical part to the room or other area served by the system, in the most efficient and faithful manner possible. This takes the form of a baffle or enclosure, with a horn being a form of enclosure.

A complete understanding of the operation of speaker systems requires a sufficient view of the whole flow of sound energy from the output amplifier stage to the listener, as depicted in Fig. 6-1.

SPEAKER DRIVERS

This section will cover the definitions and types of drivers. The speaker driver is that portion which converts electrical energy from the output transformer to mechanical energy in the diaphragm or cone radiator. The driver is also sometimes called the *motor* because, like electric motors, its input is electrical and its output mechanical.

A number of different types of speaker drivers have been tried during the history of sound-system development. Those sufficiently successful to be commercially available include the following:

1. Moving-coil dynamic driver
2. Crystal drivers
3. Capacitor drivers

Of these, by far the most popular and useful in fidelity applications is the dynamic moving-coil type; so our discussion will be primarily about that type. Capacitor and crystal drivers are sometimes employed in high-frequency (tweeter) portions of dual systems, and these are also given brief mention.

It should be mentioned here that the performance of direct-radiator speakers is very importantly influenced by the type of baffle or enclosure used. Since baffles and enclosures are a subject in themselves and must be discussed after speakers, the explanation of these factors must await a later chapter. All comparative data in this section will assume that direct-radiator speakers are employed with an infinite baffle. Definition of an infinite baffle is given in the chapter on baffles and enclosures.

Moving-Coil Dynamic Drivers

The principle of the dynamic speaker driver is based on the interaction of two magnetic fields. One field is relatively strong and steady; the other, developed by the passage of an audio-frequency signal current through the voice coil, varies with the instantaneous amplitude of the sound to be reproduced.

Electrodynamic Type—The basic construction features of an electrodynamic moving-coil type of driver are shown in Fig. 6-2. The strong, steady magnetic field is generated by a large field coil wrapped around the core. This core is mounted in a frame of magnetic material. The shape of

the core and frame is such that magnetic flux is concentrated in the annular gap in the front of the structure. The voice coil, which is wound on a cylinder of fiber or aluminum, fits into the annular air gap in the core and frame structure. The a-f electrical signal output from the output transformer is applied across the voice coil. The a-f current resulting in the voice coil generates a varying magnetic field which works against
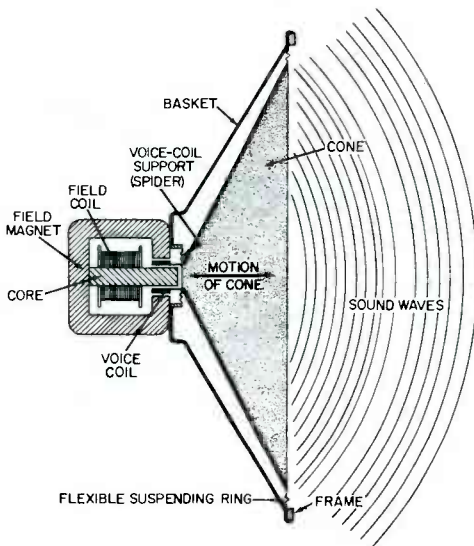
Fig. 6-2. Construction of a moving-coil dynamic speaker.

the strong static field of the field coil, and the resultant motor force produces mechanical motion of the voice coil. The voice coil is mounted to the cone-type radiator; hence the cone also moves with it and radiates the sound. The use of the dynamic driver is by no means limited to cone radiators; it is also frequently employed with horn-type radiators, as illustrated in Fig. 6-3.

For minimum distortion and maximum frequency range, it is important that the voice coil and radiator or diaphragm have a minimum mechanical mass. Excessive mass in this structure would result in inertial effects becoming worse at high frequencies, and the cone or diaphragm would have a tendency to distort physically in an attempt to follow the rapid variations of high-frequency sound components. For this reason, the voice coil is made as small and light as possible. It usually consists of a single layer of fine enameled wire about ½ to 2 inches long wound along the outside surface of the voice-coil form. Because the voice coil must be so compact and light, its impedance is necessarily low; this is why output transformers must be used to couple efficiently between the

relatively high impedance output amplifier tubes and the dynamic speaker.

Leads from the voice coil are usually cemented to the middle portion of the cone surface, then brought out to terminals mounted on the basket of the speaker structure.

The impedance rating of the voice coil, in commercially available units, may be any value from 2 to 16 ohms. This impedance is not that of the voice coil alone but includes the effect of acoustic loading on the cone or diaphragm and mechanical effects in the structure. These factors tend
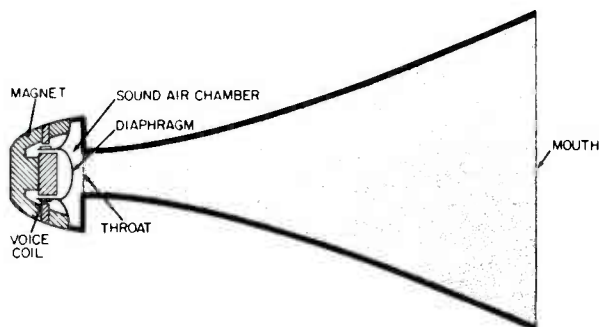
Fig. 6-3. Cross-sectional view of a horn-type reproducer.

to resist motion of the voice coil and thus raise the impedance "looking into" the voice-coil terminals. These latter effects are the largest part of the rated impedance, and the effects of the self-inductance and resistance of the voice coil are relatively small. The impedance rating is also specified for some standard frequency, usually either 400 or 1000 Hz. For the foregoing reasons, a simple resistance test of the voice coil will show a relatively low resistance, compared with the impedance rating of the speaker as a whole. Further details about the acoustic and other loading effects which help make up the impedance looking into the voice coil are discussed later, under "Enclosures and Baffles."

Permanent-Magnet Type—The foregoing discussion was devoted to the dynamic driver which employs a field coil, as shown in Fig. 6-2, to derive the strong steady flux required in the gap containing the voice coil. This type of driver is referred to as the electrodynamic type. Instead of a coil, the magnetic-field flux can be provided by a permanent magnet in the arrangement known as the permanent-magnet dynamic. The constructional features of a permanent-magnet dynamic speaker are illustrated in Fig. 6-4.

Practical permanent-magnet dynamic speakers were made possible by the development of high-grade magnetic materials, particularly alnico. Very powerful magnets, which hold their magnetic properties indefinitely with little loss, can be made from this material. A round piece of this

magnetized material is mounted between the core and the frame of the magnet structure in the dynamic-driver unit, as shown in Fig. 6-4. It is thus effectively in series with the other iron in the magnetic circuit, producing flux in the same way as the many turns in the field coil of the electrodynamic type. The obvious advantage of such an arrangement is that no field-coil current supply is required. Also, because the permanent magnet is lighter than a field coil providing the same amount of field, the overall weight of the speaker is reduced below that of an equivalent electrodynamic unit.
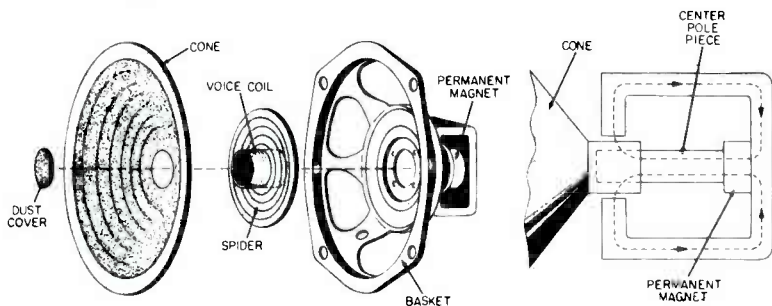
Fig. 6-4. Permanent-magnet speaker construction.

Since the permanent magnet supplies the fixed field for operation of the driver, the output power of the speaker is limited by the available flux and the size of the permanent magnet. In addition to power-handling capability, the size of the magnet also affects frequency response because the lower-frequency components contain most of the power in the audio signal; and they are attenuated in driver units with small magnets. Permanent-magnet speakers in general contain magnets weighing from 2 to more than 20 ounces. Types desirable from a high-fidelity standpoint are those with 6-ounce or heavier magnets, depending on the power requirements for the speaker.

Magnetic-Armature Type

At one time, magnetic-armature speakers were the most popular type in use, but they have now been almost completely superseded by the more efficient and better-quality dynamic type. In the magnetic type, the coil is fixed and consists of many turns of wire around a soft-iron core. The armature, which is either the radiator or a diaphragm fastened to the radiator, is also of soft iron and is mounted in a gap in the iron-core magnetic circuit around which the coil is wound. One of the prime advantages of this arrangement is the fact that the coil, being fixed, can be large enough to match the impedance of the output tube or tubes without an output transformer. Inertial and mechanical-motion limitations and nonlinearity of flux distribution have all but made this type

183

obsolete, and it is mentioned here only for the purpose of completeness and perspective.

Crystal Drivers

Rochelle-salt crystals have the property of becoming physically distorted when a voltage is applied across two of their surfaces. This property is the basis of the crystal type of speaker driver. The crystal-driver type of speaker is illustrated schematically in Fig. 6-5. The crystal is clamped between two electrodes across which the audio-frequency output voltage is applied. The crystal is also mechanically connected to a diaphragm. The deformations of the crystal caused by the audio-frequency signal across the electrodes cause the diaphragm to vibrate and thus to produce sound output.
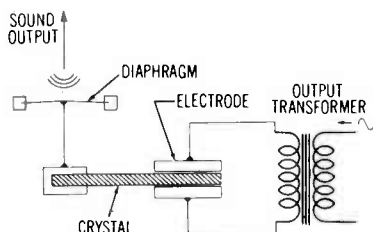
Fig. 6-5. Schematic representation of a crystal-type speaker.

In general, crystal speakers have been impractical for reproduction of the full audio-frequency range because the input impedance is almost completely capacitive. Thus it is difficult to couple power into them. At high audio frequencies, the reactance becomes lower and the relative amount of power smaller. Consequently, crystal units have found some use in tweeters (the high-frequency portion of dual-speaker units to be discussed later). Even as such, they offer no obvious or outstanding advantages over moving-coil dynamic types and thus they are rarely encountered.

Capacitor Drivers

Another principle which has been applied to speaker drivers is that of electrostatic force. When a potential is applied between two metal plates, the resulting electrostatic field produces a force that tends to pull the plates together (because opposite charges attract). If like charges are applied to the plates, a force which tends to push the plates apart is created.

The principle is usually applied in a push-pull arrangement, as illustrated in Fig. 6-6. A pair of plates is connected to a balanced audio-frequency output source as shown. Another plate, mounted between these two plates, is free to move and drive a diaphragm. The movable plate is polarized with a positive charge, as shown in the diagram.

184
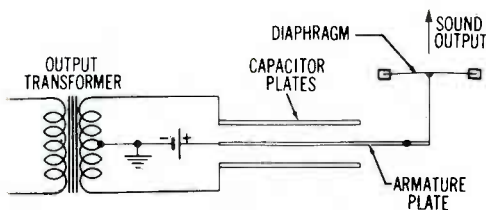
Fig. 6-6. Schematic representation of a capacitor-type speaker.

When the polarity of the audio-frequency signal is such as to make the top capacitor plate negative, the positive armature plate is attracted to it and is repelled by the positive bottom capacitor plate. The use of push-pull arrangement reduces harmonic distortion, which is inherent in most capacitor-type speakers.

As is the case of the crystal speaker, the input impedance to the capacitor speaker is almost a pure capacitance; accordingly, similar problems of coupling power into it are encountered. Although a few are in use, they are very much in the minority.

CONE-TYPE RADIATORS

Of the functional blocks of the speaker system illustrated in Fig. 6-1, we have discussed the driver only. The next step is the diaphragm which converts energy of mechanical motion into energy of air motion, called *acoustic energy*. There are two commonly used forms of diaphragms: (1) the cone type and (2) the horn type. The cone type acts as both diaphragm and radiator because it not only converts the mechanical energy of the driver into acoustic energy but also at the same time couples this energy into the room or area where the listeners are located. On the other hand, the horn-type diaphragm provides only mechanical-to-acoustic conversion; its acoustic energy output must be fed to the throat of a horn which couples it to the listening area. First, we will consider the cone-type diaphragm and radiator, which is more common than the horn-type diaphragm.

It is the purpose of the cone or any diaphragm-radiator combination to convert mechanical energy from the driver into acoustic energy in the listening area. The conversion must be such as to provide the greatest amount of acoustic power output for a given electrical power input, with a minimum of distortion of the output sound waveform. Although the speaker cone is a power transducer, output quality is more important than output power. Although efficiency (ratio of output acoustic power to mechanical power input) should be as high as possible, the modifications necessary to keep distortion to desirable low levels make the majority of speakers inherently low-efficiency devices. Most speakers have overall efficiencies ranging from 5 to 15 percent; some very elaborate systems approach 40 percent.

For higher efficiency in transfer of the mechanical energy to the air, it is desirable that the greatest possible area of contact be made between the radiator and the air. Since it is desired that the air mass be alternately moved forward and backward (and not up and down or sideways), it is natural to envision a large flat sheet driven by the speaker driver, as shown in Fig. 6-7A. The greater the area of contact, the better the air mass loads the driver unit. Unfortunately such a flat structure is not mechanically practical, because when it is constructed light enough for good high-frequency response it does not retain rigidity over its entire surface. To retain better overall mechanical rigidity with comparable large-area air contact, the cone type of radiator has been used; this type is shown in Fig. 6-7B. It has been found that, with such a shape, a relatively large area of air may be activated with a relatively high ratio of strength to weight.

Treated paper is universally used in cone construction. The more rigid the paper, the greater is the sound output obtained, but the poorer is the frequency response. Soft, blotterlike cone materials improve uniformity of response in the low- and medium-frequency ranges but give poor response at high frequencies. Soft cones are also better for transient-response rejection. High-fidelity speakers often use a two-piece cone of different materials, as will be explained later.

The size of the cone is important because it influences both the low-frequency response and the power-handling capacity. The larger the cone diameter is, the greater is the power capacity for all frequency components combined, and the better is the low-frequency response. However, such improvements are not necessarily derived from larger cones unless the voice coil is appropriate. The acoustic impedance offered to the cone rises as the cone is made larger; the voice-coil impedance must then also be made larger for proper energy transfer and efficiency. The larger the cone is, the lower is the lowest useful frequency of operation. But frequency range is not the only factor improved by increase of cone size. Because the major portion of the ordinary a-f signal power is in the low-frequency components, the overall power-handling capacity is also improved, as mentioned previously. The increase in the frequency range

(A) Flat-sheet radiator.   (B) Cone-type radiator.
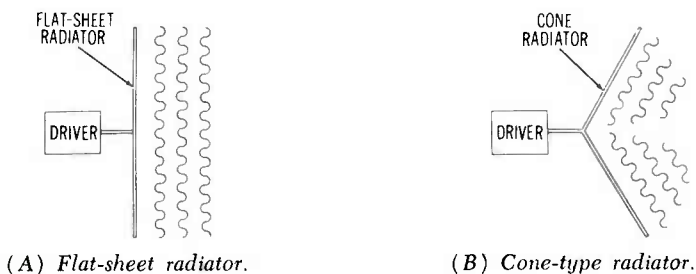
Fig. 6-7. Schematic representation of two types of radiators.

at the low end of the spectrum by an increase of cone size is illustrated in Fig. 6-8. Note that these curves show a response peak just before the response falls off at the low-frequency end of the range. This peak occurs at the resonant frequency of the speaker, which will be explained later.
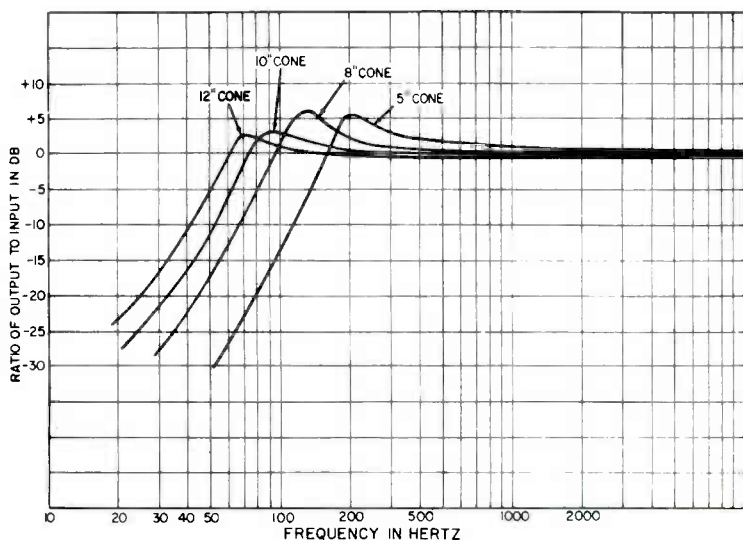
Fig. 6-8. Effect of cone size and resonance on low-frequency response.

The designation of the cone diameter is one of the most important in speaker specification. For this reason, the definition set up by the Radio Electronics Televisions Manufacturers Association* (RETMA) for the designating size of a speaker is of special interest. This definition is as follows:

> The designating size of a loudspeaker employing a circular radiator (cone) shall be twice the maximum radial dimension, measured to the nearest eighth inch, of the front of the speaker, except that the designating size shall not exceed the maximum diameter of the unsupported portion of the vibrating system by more than 25 per cent.

The size of the cone is also important in the choice of an enclosure in which the speaker is to be mounted. Enclosures are designed to operate with speakers of specified characteristics which depend mostly on size; that is why a given enclosure is stated to be used only with a speaker (or speakers) of a given size. Of course, it is assumed that the overall design of the speaker is consistent with high-fidelity performance with the

*Now Electronic Industries Association (EIA).

187

nominal cone size. A large cone with a small voice coil is not considered adequate for high-fidelity output.

The shape of the cone also influences performance. It has been found that a circular cross section as illustrated in Fig. 6-9A gives the best performance. Elliptical cones (Fig. 6-9B) tend to have a lower acoustic impedance than those of a circular cross section, and it is thus more difficult to couple power into them with good efficiency. For this reason, elliptical cones are not used in high-fidelity equipment. Also important is the shape of the flare of the cone. Straight sides are most common, but they tend to concentrate high-frequency sound components in the small area surrounding the axis of the cone. Better distribution of the high-frequency components is obtained by use of a curved flare (illustrated in Fig. 6-9C) in the cone sides, and some speakers are manufactured

(A) Straight-sided circular.

(B) Straight-sided elliptical.

(C) Cross section of cone with flared sides.
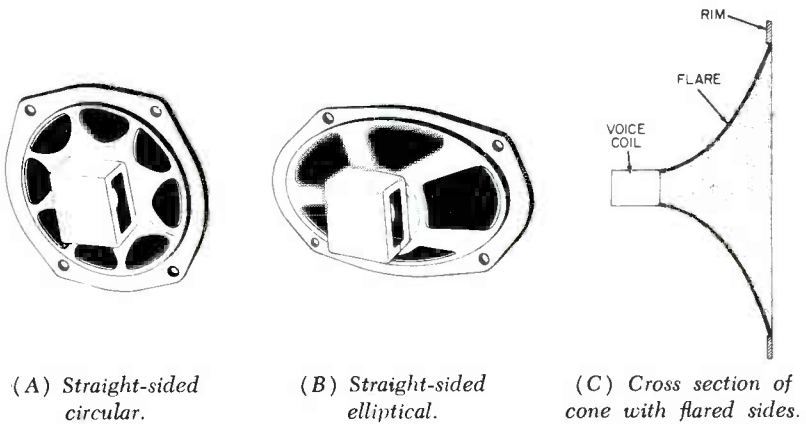
Fig. 6-9. Three variations in cone shapes.

with this shape. However, this is a more difficult manufacturing process than that of the nonflared speakers, and these speakers are therefore more expensive than straight-sided versions. In dual-speaker arrangements in which a separate high-frequency driver unit is employed, the operation of the large cone at high frequencies is not so important, and the straight-sided cone may be as good as the flared one. In speakers in which the response of a single cone is extended over the full desired range, the curved, flaring shape shown in Fig. 6-9C is often employed.

The higher the frequency is, the smaller is the portion of the cone around the center which is used for radiation. In fact, for many cones, the highest frequencies in the audible range are radiated only by the voice coil itself. The portion useful at a given frequency is approximately a fixed quantity independent of the overall size of the cone; therefore, the larger the cone, the smaller the percentage of area employed at high frequencies. This accounts for the general fact that the larger the speaker
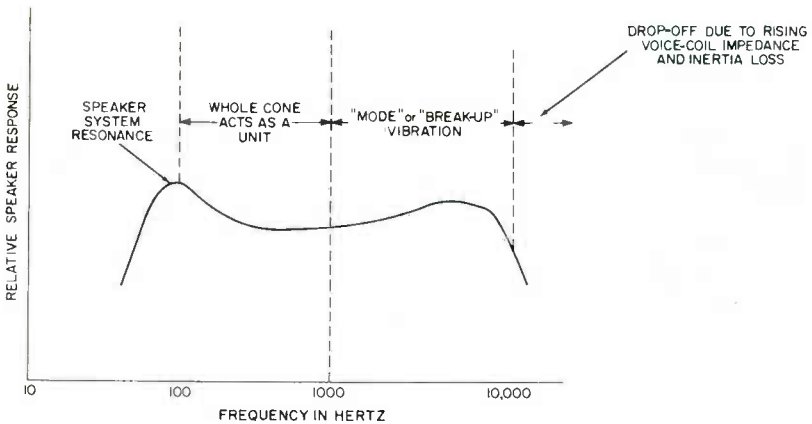
Fig. 6-10. Typical cone-radiator response curve.

is, the poorer the relative high-frequency response is and the better the relative low-frequency response is.

A generalized speaker response curve is shown in Fig. 6-10. First, there is the resonant peak at about 100 Hz (often at lower frequencies). Below resonance, response falls off rapidly. Above resonance and up to about 1000 Hz, the whole cone acts as a unit, all parts of it vibrating in phase. Response in this region is about constant. Above 1000 Hz, break-up occurs; that is, parts of the cone vibrate independently of each other, as shown in Fig. 6-11. In this portion, the response increases gradually until losses and impedance in the system increase sufficiently to cause final drop-off at the high-frequency end of the range. In some speakers, efficiency at high frequencies is improved by incorporating into the cone a more flexible material in the form of circular rings or corrugations

Courtesy North American Philips Co., Inc.

Fig. 6-11. Typical cone breakup at high frequencies.

189
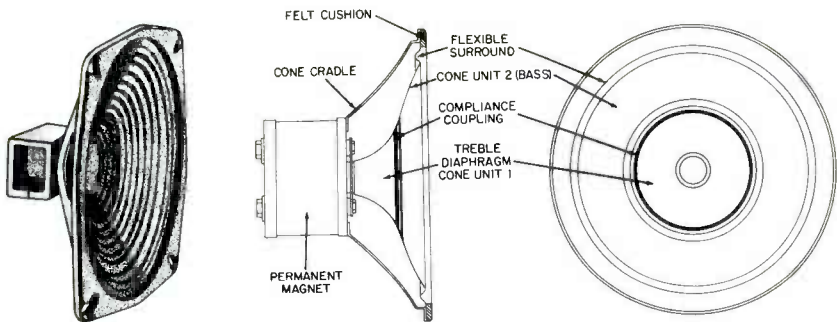
(A) Multiring unit.   (B) Hartley Turner two-unit speaker.

Fig. 6-12. How corrugations are put in cone to improve efficiency at high frequencies.

coaxial with the cone to allow greater flexing (see Fig. 6-12). This makes it easier for the small inner portion of the cone to operate independently at high frequencies, but the structure will also transmit low-frequency vibration so that the whole cone will act as a unit at the lower frequencies.

The cone is mounted to the speaker frame in two places: (1) at the outer edge or rim and (2) at the center, near the voice coil. These mounting agencies are called *suspensions* of the cone assembly. The stiffness of the suspension affects the frequency response and other performance features, as is explained later. The inner suspension near the voice coil is often referred to as the *spider* because of the physical resemblance of some versions to one. Spiders may be divided into two main groups. One group employs a phenolic or plastic sheet cut out so that the voice coil will be suspended by relatively narrow cross members, as illustrated in Fig. 6-13A. The other type, now more common, is a piece of solid, flexible material with circuit corrugations, as illustrated in Fig. 6-13B. The outer suspension is sometimes just an extension of the cone struc-
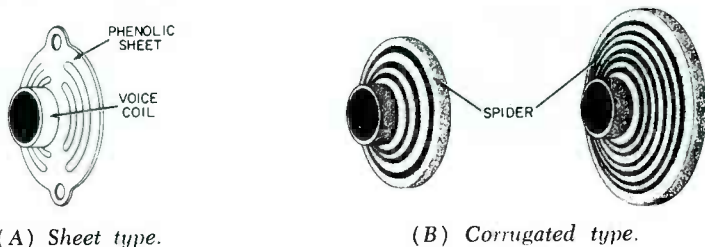
(A) Sheet type.   (B) Corrugated type.

Fig. 6-13. Two types of spiders.

ture itself, where it fastens to the metal rim of the basket. In other cases, the material at the outer edge of the cone is feathered (made thinner) or corrugated to provide increased flexibility. Some speaker units have been manufactured with suspensions of leather or other damping materials. This soft, flexible material is excellent for minimizing transient distortion at low frequencies.

SPEAKER IMPEDANCE

It has been previously pointed out that the impedance looking into the voice coil is not only the self-impedance of the coil itself but a combination of the self-impedance and the more important reflected acoustic impedance. A parallel may be drawn with a transformer or electric motor. Each of these devices draws a small current when operating unloaded, indicating a relatively high input impedance. When the transformer secondary is loaded by an electrical resistance or the motor shaft is coupled to mechanical load, the input current rises and the input impedance of each device is lowered in proportion. In other words, the load impedance has been reflected into the input circuit in each case, whether it is an electrical load in the transformer or a mechanical load in the motor.

The voice-coil winding is similar to any other coil in that it has resistance (of the wire used in the winding), inductance, and a small amount of capacitance (distributed, between turns). The resistance and reactance of the coil combine to form the self-impedance of the winding, without any impedances coupled into it from its association with the other parts of the speaker.

In a speaker, the principle of total impedance is the same as in our motor and transformer analogies. The self-impedance of the voice coil (in a vacuum) is modified by the reflected impedance of the load on the diaphragm. It may be difficult at first to think of a diaphragm having an impedance; this will be made more clear if the following analogies between mechanical and electrical systems are considered.

Mechanical-Electrical Analogy

Mechanical inductance is called *inertance*. When the diaphragm starts to move some air, that air resists the force tending to set it in motion, due to its inertia. After the air is in motion, it tends to stay in motion when the diaphragm stops or reverses its motion. The degree to which the air tends to stay at rest or in motion is a measure of its inertance. In the electrical analogy, it is the inductance of a circuit which provides electrical inertia; and it is the current which tends to stay at rest or in motion in proportion to the amount of inductance present. Mechanical inductance, when applied to the air, is also referred to as *acoustic inductance*. The term "inertance" is more especially applied to the acoustic system and the air in contact with the diaphragm. The mechanical inductance of

191

the cone and voice-coil structure and of its suspensions is also a factor in the input impedance to the voice coil, and is reflected back to it with the acoustic inertance.

Mechanical capacitance is called *compliance*. This is the "springiness" or "give" of the mechanical assembly or the air. The best example of a mechanical capacitance is a spring. Force applied to a spring stores work (force times distance) in the spring. Then, when the spring is released, the stored energy is released. This is exactly what happens electrically in a capacitor in which energy is stored by the flow of current into the capacitor by application of a voltage. The applied voltage is analogous to the applied mechanical force, and the resulting current is analogous to the motion or change of displacement of the spring. When speaking of mechanical systems, we call this effect *mechanical compliance*. Although the speaker does not contain a spring, the cone suspensions do act as springs and offer to cone motion resistance which increases as cone displacement increases. The suspension compliance is the main capacitive effect, although the "springiness" of the air load and the cone and voice-coil structures during flexing add other capacitive factors. When applied to the air, this effect is called *acoustic compliance*.

Mechanical resistance is friction. It is the resistance force developed between two surfaces, two layers, or two or more groups of particles within a material when they rub together. In a speaker of the dynamic type, there are no material surfaces which rub together (under normal operating conditions). Purely mechanical resistance arises in the friction within the cone and suspension materials when they flex during operation. The useful resistance component is that of the acoustic load. The latter is developed by the friction of the particles and layers of the air surrounding the cone or diaphragm when they bear upon each other or along the mechanical surfaces of the speaker assembly when motion is imparted to the air in the form of acoustic vibrations.

Mechanical components of impedance (explained in the foregoing) have the same relationships among themselves as exist among their counterparts in the electrical circuit. Power is dissipated only by the resistance component. The inertance and compliance produce mechanical reactance which varies in direct proportion to frequency in the same way that electrical reactance varies.

ACOUSTIC IMPEDANCE AND RESONANCE

The resistances and reactances of the system (including acoustic, mechanical, and electrical effects) combine in the effective impedance looking back into the output transformer. This combination is best visualized by means of an equivalent circuit, illustrated in Fig. 6-14. The diagram of the corresponding portions of the system in the upper portion of the figure help symbolize the corresponding physical locations in which the impedance factors appear. The efficiency of power transfer can be seen

192
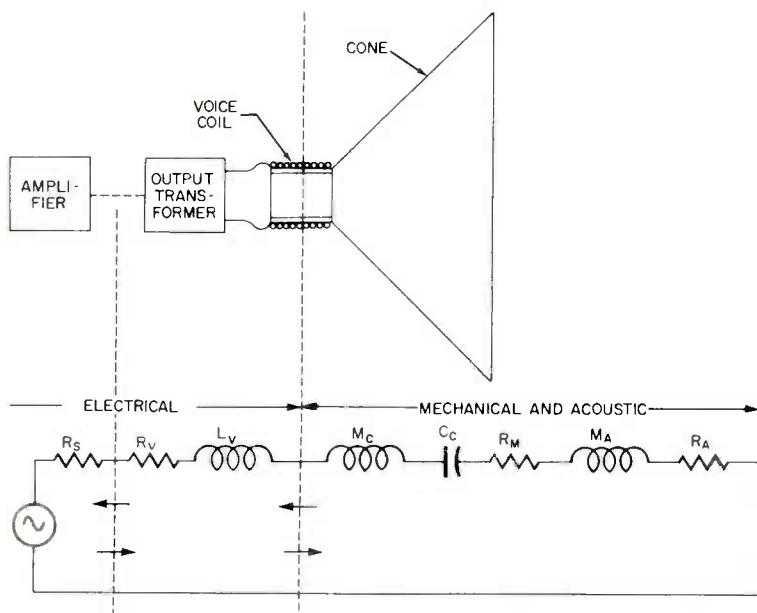
R_S = Equivalent resistance of source (including plate resistance, and transformer winding resistance).

R_V = Equivalent resistance of voice-coil.

L_V = Inductance of voice-coil plus leakage inductance of transformer.

M_C = Mass of cone and voice-coil assembly.

C_C = Compliance of suspension.

R_M = Mechanical resistance of cone and suspension.

M_A = Mass of air load.

R_A = Frictional resistance of air load.

Fig. 6-14. Equivalent circuit of a speaker.

to be dependent on the proportion of the impedance represented by R_A, which represents actual acoustic power dissipated in overcoming air friction and in radiating the acoustic power.

We have observed that there are two types of mechanical and acoustic reactance in the speaker system. They are mass or inertance, corresponding to electrical inductance, and compliance, corresponding to electrical capacitance. As in a purely electrical system, the capacitive reactance which is the compliance C_C resonates with the combined inductive effects L_V, M_C, and M_A at some frequency. This frequency is known as the *resonant frequency* of the speaker. At the resonant frequency, all reactance is cancelled out of the system, and output and efficiency increase greatly over what they are for other frequencies. Rather than being beneficial, such an increase in efficiency is actually detrimental because it occurs only in the vicinity of the resonant frequency. If the effect of speaker

resonance is not reduced considerably or if the resonant frequency is made lower than the lowest frequency to be employed, extremely annoying frequency, amplitude, and transient distortions result. The system is then highly sensitive to signals at or near the resonant frequency. Every time changes in signal amplitude occur rather suddenly, the system tends to self-oscillate at the resonant frequency even though it is excited by sound-signal components of other frequencies. The reader has probably heard this effect in sound systems which, when listened to from a distance, gives the impression of producing nothing but a "booming" noise. The "boom" is the reaction of the speaker system at its resonant frequency.

The curves of Fig. 6-8 show the effect of speaker resonance on response. The response rises to a peak at the resonant frequency; then it falls rapidly at lower frequencies. The shape of the response curve shows how the resonant frequency can be used as an indication of the limit of low-frequency response. Undesired sharpness of the resonant peak can be lessened by electrical, mechanical, or acoustic damping. Damping is the addition of a resistive load.

One method of providing damping is through design of the output stage of the amplifier. The latter should have as low a source impedance (R_S in Fig. 6-14) as possible. The amplifier impedance is reflected through the output transformer and is effectively connected across the voice coil. If this reflected impedance (resistance) is low enough, it reduces the Q of the resonance of the speaker and thereby reduces the severity of the response peak at the resonant frequency and the sharpness of the drop-off below it. The reduced Q also minimizes transient distortion because the tuned circuit represented by the speaker has less "flywheel effect." The impedance of the output amplifier stage is lowered by use of low-impedance tubes and by the use of negative feedback. Triodes have a much lower plate impedance than tetrodes and beam tubes, and that is one important reason for the preference of some for triodes.

Another method of providing speaker damping is through design of the speaker enclosure, as will be explained in the chapter on baffles and enclosures.

HORNS AND HORN DRIVERS

A horn is a tube so flared (tapered) that the diameter increases from a small value at one end called the *throat* to a larger value at the other end called the *mouth*. A basic horn-driver combination is illustrated in Fig. 6-3. Horns have been used for centuries for increasing the radiation of the human voice and musical instruments.

The horn does acoustically what the cone does mechanically. It couples the small voice-coil area to a large area of air. In this way, the horn acts as an acoustic transformer and converts the relatively high impedance at the throat and driver. The horn is a fixed physical boundary for

its enclosed column of air and does not vibrate itself. Acoustic energy fed to its throat must therefore be obtained from a vibrating diaphragm which converts mechanical motion from the driver voice coil (or other armature) to acoustic energy. Although the cone-type radiator acts as both diaphragm and radiator and transduces from mechanical to acoustical energy, the horn acts only as a radiator, with both input and output energy being acoustic.

We have seen that the high-frequency response drop-off of a cone-type radiator is caused by the inertial effect of the mass of the cone. Because the transformation in a horn is through an air column rather than through solid material, the high-frequency response of the horn is much better than that of the cone. The overall efficiency at all frequencies is better.

In spite of these advantages, straight (unfolded) horns are not commonly used for general-frequency coverage systems or low-frequency units in homes because of their bulk and their relatively high cost of manufacture. They do find wide use, however, in two forms, as follows:

1. In straight form. In dual speaker systems (to be discussed later) and in connection with the tweeter speakers, which reproduce only the high-frequency portion of the audio-frequency range.
2. In modified folded form. In special speaker enclosures (to be discussed later).

The high-fidelity sound enthusiast should therefore be familiar with some of the fundamentals of horns. The following basic information about horn design is included to clarify these fundamentals rather than to act as constructional information, although a few audiophiles have constructed their own horns with good results.

The most common type of horn design is the *exponential horn*. Its distinguishing characteristic is the fact that the increase of its cross-sectional area is proportional to its length. In other words, if the cross-sectional area 10 inches from the throat is twice the cross-sectional area at the throat, the area 20 inches from the throat is four times that at the throat; at 30 inches, the area is eight times that at the throat; and so on. The basic equation defining an exponential horn shape is the following:

$$\frac{A_x}{A_T} = e^{mx}$$

where,

A_x is the cross-sectional area at distance x from the throat, in square inches,

A_T is the cross-sectional area of throat, in square inches,

e is the natural logarithm base, 2.7183,

m is the flare constant of the horn, in inverse inches,

x is the distance from throat, in inches.

This equation is of importance primarily in defining the flare constant, m. The greater the flare constant is, the faster the diameter of the horn increases.

The flare constant determines how long a horn with a given mouth-to-throat area must be, but it is of much greater importance in another connection. Each horn has a cutoff frequency below which no sound energy can be coupled through it. Below the cutoff frequency, the throat area offers to the driver a pure acoustic reactance and no resistance; thus, no power can be transmitted. The cutoff frequency is dependent solely on the flare constant for a given horn in air of a given temperature and humidity. The relation between them is as follows:

$$f_c = m \frac{V}{4\pi}$$

where,
    f_c is the cutoff frequency, in Hz,
    m is the flare constant, in inverse inches,
    V is 13,500 inches per second, the velocity of sound in air at 20°C.

Linear measuring units must be consistent throughout and in this case are inches because they are the units most appropriate to horn structures of practical size.

We are primarily interested in the type of flare required for a given cutoff frequency. A convenient and commonly used method of specifying flares is that of stating the distance along the axis of the horn over which the cross-sectional area doubles. Doubling the area is the same as multiplying the diameter by 1.414 (the square root of 2); consequently, the shape of the desired horn can be laid out if a series of diameters at the proper distances from the throat are plotted.

The relation between the area-doubling distance (x_D) and the cutoff frequency for exponential horns is plotted in Fig. 6-15. Notice that a horn must have a doubling distance of 25 inches for a 30-Hz cutoff frequency. Another horn, which need only be used for frequencies above 300 Hz, can double its area each quarter inch! This indicates why tweeter horns are relatively short and flare out very rapidly, especially if designed for a high-frequency crossover such as 3000 Hz.

If the cutoff frequency were the only consideration, the throat and mouth diameters could be made so nearly the same that even with a small flare (large x_D) for low frequencies the horn could be made short. For the best frequency response, however, there are important reasons for keeping the throat small and the mouth large; they are as follows:

1. When the horn is to be used for full-frequency coverage, the throat must be small to couple properly to a small diaphragm. If the diaphragm is not kept relatively small, it suffers attenuation at the high frequencies because of its mass, just like the cone. If the throat diameter is not nearly

the same as the diaphragm diameter, there is loss of energy in the acoustic transfer from the diaphragm to the horn. When the horn is to be used only at low frequencies such as for a woofer in a dual system, then the throat can be very large.

2. If the mouth is not made large enough, the sound tends to be reflected back toward the throat and serious attenuation of the low-frequency components takes place. For this reason, the diameter of the
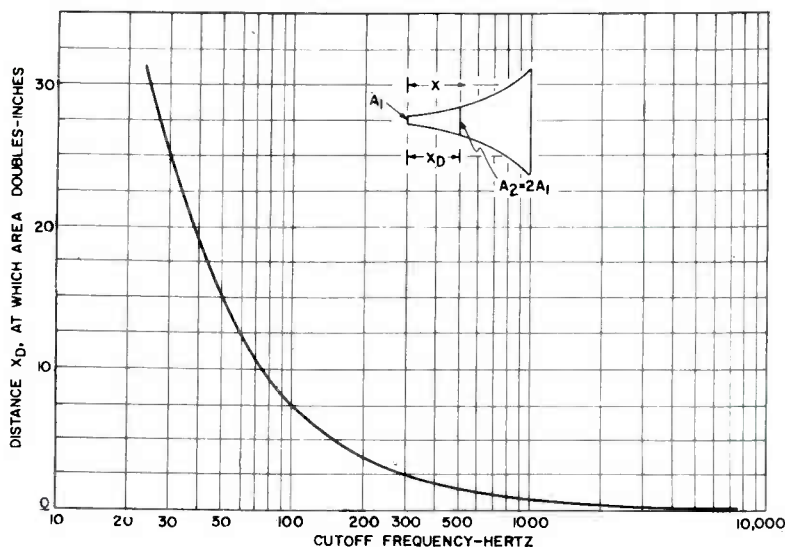
Fig. 6-15. Relation between cutoff frequency and distance in which cross-sectional area doubles for exponential horns.

mouth should be kept to a minimum of a half-wavelength at the lowest frequency to be reproduced. A wavelength is equal to the distance the sound wave travels during the period of one cycle. It is thus equal to 13,500 inches per second (the velocity of sound) divided by the frequency. Since we are interested in a half-wavelength and this is our suggested minimum size for the mouth, the following relation is pertinent:

$$D_M = \frac{6750}{f_l}$$

where,

D_M is the diameter of the mouth, in inches,
f_l is the lowest frequency to be reproduced, in Hz.

For example, to maintain good operation down to 67.5 Hz, the mouth diameter should be 100 inches. This gives an idea of just how bulky horns with good low-frequency response could become.

197

All of the foregoing discussion of horns has implied the use of a circular cross section, but a square cross section can also be used. The same relation holds as far as cross-sectional area is concerned, except that the dimensions of the sides instead of the diameter are given.

When a low-frequency horn is designed, the cutoff frequency closely approximates the lowest frequency to be reproduced. However, a tweeter horn must have a cutoff frequency appreciably lower than the crossover frequency at which the tweeter stops operating. This is because crossover should be gradual rather than sharp, as explained further in a later discussion of crossover systems.

A horn may be driven by any of the previously described driver types. Tweeter horns sometimes feature crystal or capacitor drivers, but in general the moving-coil dynamic driver is most common. This arrangement is illustrated in Fig. 6-3, which shows the magnet, voice coil, and small diaphragm of a typical unit. As previously explained, the diaphragm is small enough to vibrate with efficiency at the highest frequencies in the reproduced range. If the diaphragm were open to the free air, it would have very poor efficiency at low frequencies because of the small area of contact; however, because of the transformer action of the horn, the large area of air at the mouth of the horn is effectively coupled to the small diaphragm area in the driver.

The diaphragm is closed-in on all sides, except for the port which accommodates the throat of the horn. The space between the diaphragm and the throat of the horn is known as the *sound (air) chamber*. In air chambers of simple annular shape, attenuation at high frequencies is sometimes encountered. At high frequencies, the wavelength is small and the different portions of the diaphragm are at different distances from the mouth of the horn. This means that appreciable phase differences appear and resultant cancellations occur between the high-frequency sound components coming from different parts of the diaphragm, as illustrated in Fig. 6-16. To overcome this phase problem, special chamber designs like those of Fig. 6-16B are often used. These make it necessary for all the sound energy to flow through ports of roughly equal length to the horn mouth, thus minimizing high-frequency phase differ-

(A) *Cancellation of out-of-phase components.*   (B) *Two types of construction to overcome cancellation.*
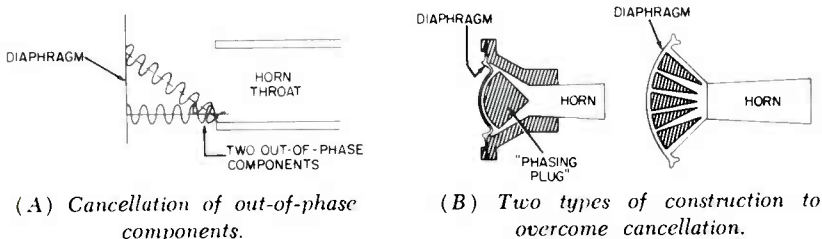
Fig. 6-16. Out-of-phase cancellation of high-frequency components in the air chamber of a horn driver.

198

ences and cancellation. This path equalization is also aided by use of a curved diaphragm like that shown.

Up to this point, we have been discussing straight horns, that is, horns whose axis is a straight line. The advantages of horns can also be obtained by using the same flare and by curving or folding the length of the horn to save space. Folded horns in high-fidelity systems are most often employed as, or in conjunction with, speaker enclosures. They are discussed later in the chapter on that subject.

DIRECTIVITY OF BASIC UNITS

It is desirable that listeners in any part of a listening area receive sound of the same quality. This ideal can be approached but never quite reached because speakers have a directivity characteristic. As could be expected, greater volume of sound is obtainable from the front of a cone or the mouth of a horn than from other parts of the radiator. However, overall volume loss with direction is not so important as long as there is a reasonably low level of distortion and good balance of frequency components. It is the change of directivity of a speaker with respect to frequency which constitutes an important problem.

Radiators of both the cone and the horn types tend to concentrate radiation of the high-frequency components of sound in a narrow cone about the axis of the radiator. The degree of directivity of a speaker is indicated by a directivity pattern, the basic function of which is indicated in Fig. 6-17. The axis of the radiator is considered the reference line with an angle of zero degrees. Directivity patterns are normally shown as a top view in a horizontal plane through the radiator axis. A cone, a circular or square horn in free space should have the same pattern in a vertical plane; but, of course, room reflections and speaker mounting may cause it to be different. The pattern line in Fig. 6-17 indicates the relative sound intensity radiated in any direction by its distance from reference point O in that direction. For example, the line OA indicates by its length that the sound radiated along it is a maximum compared with that in any other direction. At a 45-degree angle, the line OB is a measure of the relative sound intensity in that direction. Since OB is only about half as long as OA, a listener along that line would hear only about half the volume that a person along OA at the same distance from O would hear. At angles near 90 degrees, the pattern indicates zero radiation; of course, in any practical setup, such a zero area would not exist because sound would reach there by reflection.

Because directivity normally varies considerably with frequency, a complete diagram must show separate patterns for each of at least several frequencies. Typical variation of directivity with frequency for a 12-inch cone is illustrated in Fig. 6-18. It is assumed that the speaker is mounted in an infinite baffle (baffles are discussed later). Notice how much narrower the radiation pattern is at highs than at lows.

199
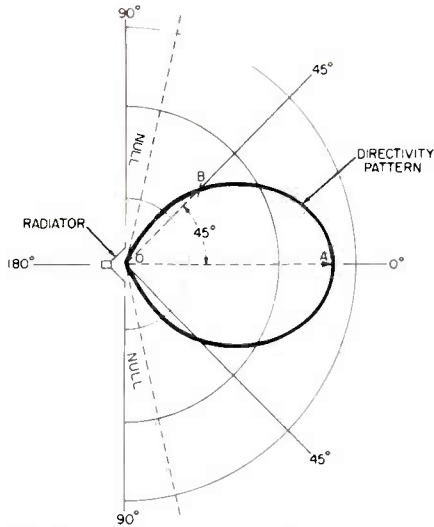
Fig. 6-17. Radiation pattern for a typical cone at one frequency.

The directivity of horns is not much different from that of cones when the cone diameter is approximately the same as the horn mouth diameter. However, at low frequencies at which the wavelength approaches the mouth diameter, the horn directivity becomes much broader. Contrary to what might be expected, directivity for these frequencies becomes broader as the mouth diameter decreases. Then at higher frequencies at which the mouth diameter is several times a wavelength, the directivity narrows slightly as the mouth diameter becomes smaller. All in all, for cones and horns of practical sizes, it may be said that the direc-
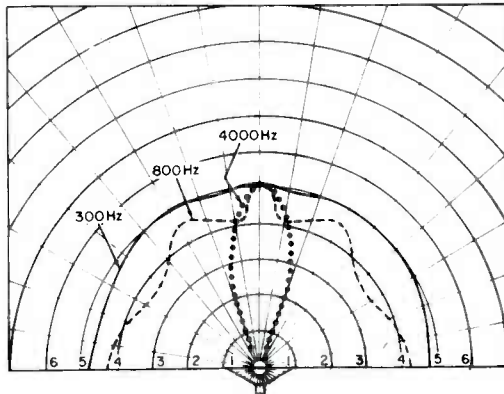
Fig. 6-18. Variation of directivity with frequency for a typical cone radiator.

tivity of the cone is little different from that of the horn, except at the highest frequency, at which the horn gives wider distribution.

The limited directivity of speaker radiators at high frequencies is a factor given considerable attention in speaker system designs. One approach is to use more than one speaker, pointing each of several units in a different direction. This is more frequently done with horns than with cones for the following two reasons:

1. Because of the small throats of the horns, they are much easier to mount at an angle and close together than the cones.
2. Horns suitable for high frequencies are more compact and give slightly better distribution.

A typical example of how a number of horns may be combined to give better high-frequency distribution than is possible with one horn is shown in Fig. 6-19. This type of structure is very popular for tweeters in dual systems and is frequently referred to as a *multicellular horn*. With such an arrangement, distribution of high-frequency sound components can be made almost as broad as that of the low-frequency components. In most cases, the throats of all the horn units are fed by the same driver unit; in a few more elaborate installations, separate drivers are used.
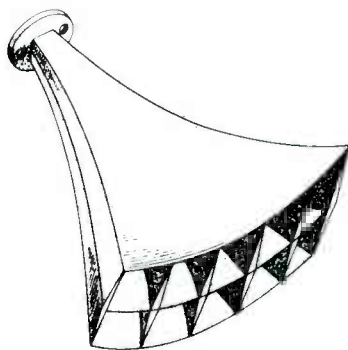
Fig. 6-19. A multicellular horn.

DUAL AND MULTIPLE SPEAKERS AND SYSTEMS

It has previously been explained that a simple single-cone speaker has definite limitations as far as frequency range is concerned. For good low-frequency response, the cone should be relatively heavy, its suspension should be as soft as possible, and its area should be as great as possible. At the highest audio frequencies, these measures are all detrimental and the cone should be as light and small as possible with a stiff suspension. Thus, we have the generally accepted conclusion that a cone of conventional design cannot produce acceptable high-fidelity response (from 60 to at least 12,000 Hz). This conventional cone, if designed for reasonable low-frequency response, ordinarily becomes unacceptable

at about 8000 Hz. In some cases, the fall-off at the high end is such that it can be at least partially compensated for by treble boost in the amplifier, but, in many speakers, "hitting the highs harder" leads to annoying distortion.

All speaker designers agree about this limitation, but they do not agree about the best way to overcome it. There are three main approaches to the problem of extension of frequency response of the conventional speaker to satisfy high-fidelity requirements:

1. Special design of a single cone to extend its response.
2. Combination of two radiators, one for high frequencies and one for low frequencies, into one physical assembly, or closely attached to each other usually along a common axis (exemplified by the coaxial type of construction). Sometimes three radiators are used in the same arrangement. In some units, the radiators are coupled through mechanical compliance between them; in others, separate voice coils are employed.
3. Use of two or more completely separate speakers, each designed to reproduce only a specified portion of the frequency range.

No one of these approaches is universally recognized as best. The proponents of each approach present convincing arguments, but the subjective nature of any final test has prevented any obviously conclusive choice. It is this which helps lend fascination to the pursuit of high fidelity, and the reader can expect to enjoy many hours of speculation concerning his own choice of a speaker system. Rather than favor one method over another, we present the most common arguments for each system, and this should equip the reader to form his own opinions.

Single-Cone Radiators

This type is favored by its proponents not only because of its relative simplicity but also because it is claimed that separate tweeters (high-frequency radiators) have a tendency to become "fuzzy" because of a phenomenon called *rim resonance*. The contention is that the rim of the tweeter cone or horn resonates at some high frequency and oscillates at that frequency, causing interference when high frequencies are being reproduced. It is claimed that a single cone can be so designed that a high-frequency portion in the center will operate independently at high frequencies and that it will be loaded by the outer portion of the cone to prevent rim resonance. It is also claimed that no dual arrangement can make the high- and the low-frequency sounds appear to the listener to be coming from the same source, and that the difference in construction of the high- and low-frequency portions cause detectable quality differences (coloration) to the trained ear. In the single-cone arrangement, both high- and low-frequency sounds emanate from the same cone; this is supposed to eliminate the problem of duality.

Coaxial Arrangement

Proponents of this type point to the limitations of response of a single-cone unit because of the conflicting requirements for size, mass, and suspension at the two ends of the audible frequency range. This means that some additional unit must be introduced to divide the frequency range into two or more parts; and with each part designed for optimum operation in its own range, the best response and minimum distortion are obtainable. The coaxial enthusiast, although favoring a dual arrangement, eliminates the completely separate systems because of the danger of phase differences and the tendency claimed that the high- and low-frequency sounds seem to come from different sources. (This is a similar argument to that used by single-cone proponents.) To minimize such spatial distortion, this design has the tweeter radiator right inside the low-frequency (woofer) cone coaxial with it; therefore, the apparent source of both frequency-range components is the same.

Separate Woofer-Tweeter Arrangement

Those who favor the woofer-tweeter arrangement argue that interaction and loading resulting from the placement of the woofer and tweeter together in the coaxial arrangement cause distortion not present in separate arrangements. They state that there is also a rather narrow distribution of the high-frequency components of reproduced sound from the coaxial type, and this distribution is due at least partly to the action of the woofer cone as a wide-mouth horn at the high frequencies. This can be overcome by separating the tweeter so that its energy distribution will not be influenced by the woofer.

As can be concluded from the review of pros and cons, each approach has inherent potential advantages and weaknesses. However, the designers and manufacturers of the better speakers of each type have taken measures to minimize each weakness, and speakers of high quality can be obtained in any of the three categories. We reiterate that the choice is the buyer's and should be exercised only after careful consideration of all claims, plus his own application problem. Even better, listen to each type under as well-controlled conditions as possible; but unfortunately it is seldom practical to find these conditions.

CONSTRUCTION FEATURES OF SPEAKERS

To help in a study of the various models of speakers in all categories, let us now consider some of the constructional features which are used to ensure high-fidelity performance.

Extended-Range, Single-Cone Type

Some of the measures taken to extend the response of a single cone have already been mentioned. One of the most important is the division

of the cone into two parts: (1) one part which resembles a small cone and which is the center portion of the main cone, and (2) the second part, which is coupled to the first (the remainder of the main cone) by a compliance which extends this second part to its full dimension. This is illustrated in Fig. 6-20. The high-frequency portion of the cone is connected to the remainder of the cone through a mechanical compliance which is a ring of softer material than the cone. This compliance material
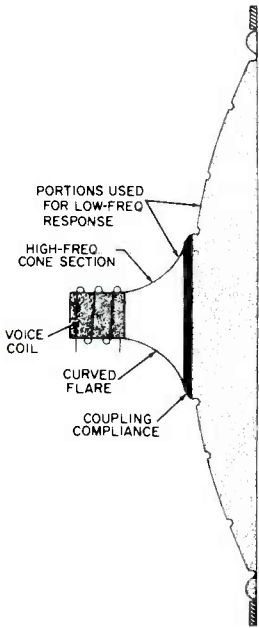
Fig. 6-20. Construction of a typical single-cone, extended-range speaker.

allows the center high-frequency portion to operate as a separate unit but transmits low frequencies to the remainder of the cone with a blending action in mid-range. The whole cone assembly acts together as a low-frequency radiator.

The cone, especially its center portion, of this type of unit is made with a curved flare. As previously explained, this helps high-frequency response. Frequently, the center portion of the cone is also made of harder material than the outer portion; this helps the center portion to operate independently at the high frequencies. An example of another type of extended-range speaker is shown in Fig. 6-21.

Coaxial Type

The coaxial principle is probably exemplified in more different commercial models than any other. We cannot review all combinations and

Fig. 6-21. Photograph of a typical single-cone, extended-range speaker.

types, but a few representative ones will give the reader sufficient general information to recognize the others. First, consider the generalized diagrams of the two main types of dual coaxial units shown in Fig. 6-22. Fig. 6-22A shows the type employing a single voice coil for both high and low frequencies. The voice-coil form is fastened rigidly to the tweeter radiator so that the high-frequency components will be efficiently transmitted to it. It is fastened to the woofer cone through a soft mechanical compliance which transmits low-frequency components but which tends to reject high-frequency components and keep them out of the low-frequency radiator. In this way, the physical construction causes the unit to function as a mechanical divider network which automatically separates high- and low-frequency components. It will be noted that actually the extended-range, single-cone speakers described previously are similar to the coaxial type because the tweeter section operates separately in its range, and because the total cone and the high-frequency portion are mounted coaxially with each other.

The other general type of coaxial unit is shown in Fig. 6-22B. In this type, each radiator has its own voice coil. Because the voice coils are separate, an electrical dividing network must be employed; therefore, only the high-frequency a-f currents are fed to the tweeter voice coil, and only low-frequency currents are fed to the low-frequency voice coil. If appreciable signal power at frequencies outside its intended range is applied to either voice coil, distortion and overloading will result. Divider networks, also called crossover networks, are discussed in a later chapter.

205

When a horn is used for the tweeter, a separate diaphragm and voice coil must be used to excite it, as illustrated in Fig. 6-22C.

Some representative coaxial speakers as they appear in models commerically available are illustrated in Fig. 6-23. The speaker in Fig. 6-23A may be considered a transition between the extended-range single-cone and the duocone (single voice-coil) coaxial types. Small conical

(A) Single voice-coil type.

(B) Separate voice-coil type.

(C) Tweeter horn with separate diaphragm and voice coil.
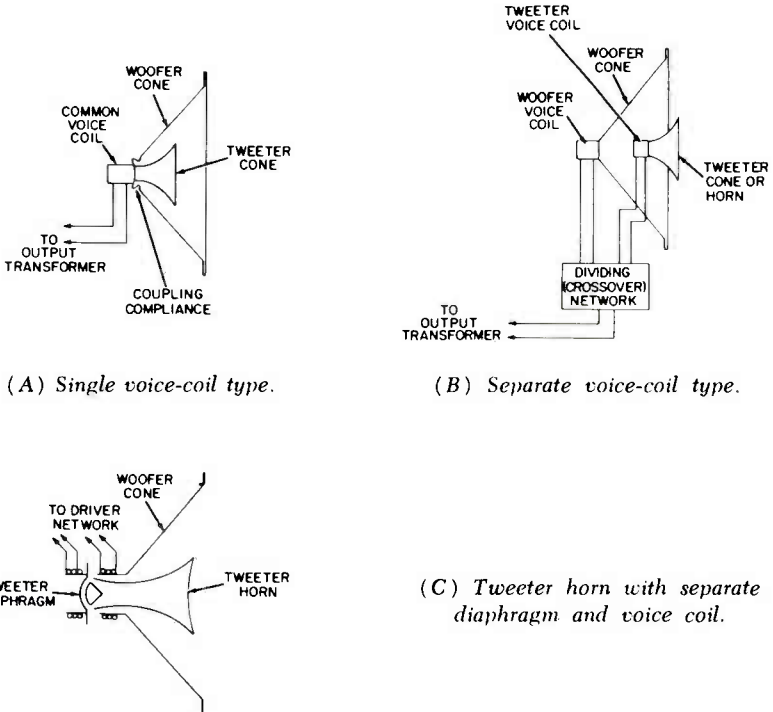
Fig. 6-22. Basic arrangements employed in coaxial speakers.

domes are fastened to a special corrugated single-cone structure, and a small radiator is added in the center. The domes break up the surface of the cone for the high-frequency components, reducing losses and helping to distribute radiation better.

The type of coaxial unit in Fig. 6-23B employs a separate high-frequency cone coupled to the common voice coil through a mechanical connection. Besides its function as a high-frequency radiator, the small cone is said to improve low-frequency response by addition of its mass to that of the large cone, and to act as a diffuser for the large cone in the middle range of frequencies.
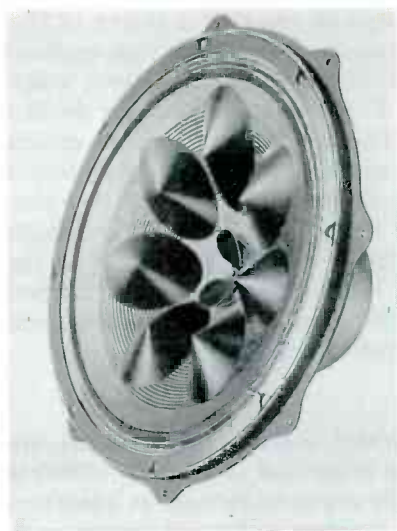
(A) *RCA* Duocone.

(B) *Philips Type 971 OM.*

(C) *Quam coaxial speaker.*

(D) *Altec Type 604C.*

Fig. 6-23. Various types of coaxial speakers.

A coaxial unit employing a separate cone-type tweeter is shown in Fig. 6-23C. It is mounted on brackets fastened to the metal rim-support frame of the woofer.

The problem of proper distribution of high-frequency sound components in coaxial speakers has led some manufacturers to the use of multi-

cellular horns for the tweeter. An example of this type is shown in Fig. 6-23D. The tweeter has a completely separate driver of the type required for excitation of a horn. This driver fits inside the woofer driver, which has a voice coil 3 inches in diameter. The horn is a single unit divided into sections by baffles at the mouth. This division into sections directs the high-frequency components over a wider radiation angle than would be obtainable without such division. In the unit illustrated, the flare cut-off of the horn (which is exponential) is 1800 Hz, which is far enough below the 3000-Hz crossover frequency to ensure smoothness in the transition between woofer and tweeter. Fig. 6-24 shows another method of achieving better high-frequency distribution with coaxial design and the use of an acoustical lens in the high-frequency horn.

Separate Woofer-Tweeter

Systems employing physically separated woofers and tweeters are commonly custom-built or at least are composed of units by different manufacturers. A few systems are sold by one manufacturer as integrated units, but we shall concentrate on the separate woofer and tweeter units and how they are combined.

Because the tweeter takes over above the crossover frequency, the woofer does not have to have extended range in the high-frequency direction as is the case with full-coverage, single-cone speakers. Full-coverage, single-cone speakers are frequently good woofers but are overly expensive for the purpose because the special design effort expended in improvement of their high-frequency response is wasted in this application. On the other hand, just any good-sized cone speaker is not necessarily a good woofer because, in spite of their large size, single-cone speakers do not always have good low-frequency response. Low-frequency response is doubly important in a woofer: (1) because this is the primary function of the woofer, and (2) because reproduction does not seem balanced if high frequencies are well reproduced by the tweeter without the low frequencies, or vice versa. The woofer should therefore have a resonant frequency as low as possible, with 50 Hz or lower being a rough guide. As will presently be explained, the resonant frequency of the low-frequency system (as opposed to the speaker unit itself) depends on the type of baffle or enclosure used. However, the speaker should have as low a self-resonance as possible in any event.

Although any good conventional speaker with the aforementioned features will do as a woofer, greatest economy can be effected in the purchase of a unit expressly designed as a woofer. The economy arises from the fact that most conventional speakers when designed for better-than-average low-frequency response are also designed with some improvement in high-frequency response in mind. The latter is, of course, not needed in the woofer application; however, the woofer must have good response well beyond the crossover frequency to ensure good transitional

operation. This is no problem at all when low crossover frequencies of 300 to 1000 Hz are used, but some speakers may start to show some drop-off or other poor response characteristics when the crossover frequency is as high as 3500 Hz (in which case the woofer should have good response to about 4500 Hz).

As mentioned, speakers designed expressly and solely as woofers in dual or multiple systems are few and far between, so no representative

(A) General appearance.

HIGH-FREQUENCY
DIAPHRAGM

PHASING
PLUG

RING MAGNET

ACOUSTIC LENS SYSTEM
IN HIGH-FREQUENCY HORN

LOW-FREQUENCY CONE
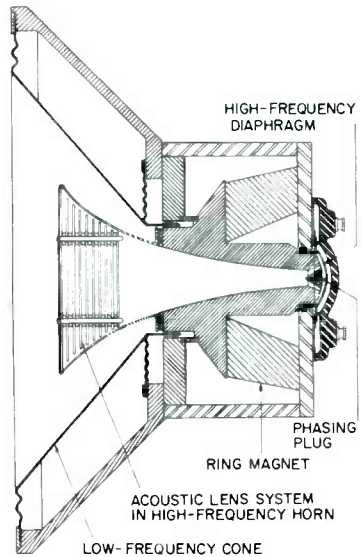
(B) Internal construction.
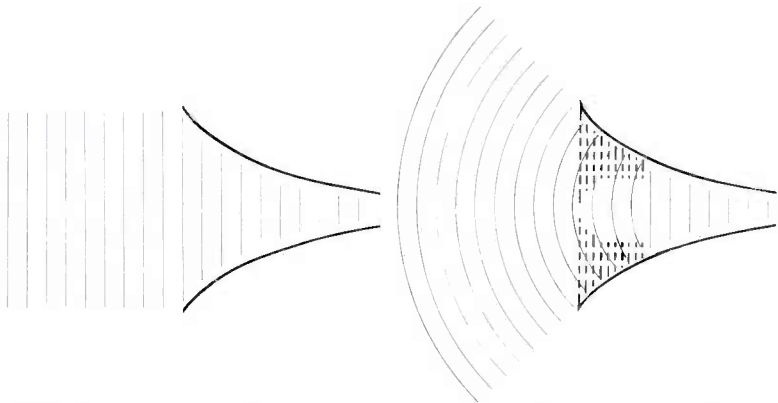
(C) Comparison of distributions without and with the acoustic lens.

Fig. 6-24. A coaxial speaker with a horn-type tweeter and an acoustic lens.

types are illustrated here. What many audiophiles do is to build up their speaker systems gradually. For example, a first step could be a full-range, single-cone speaker with carefully checked low-frequency resonance. This unit can be used alone for very good high-fidelity reproduction until a tweeter and suitable divider network can be purchased. With the addition of the latter, the speaker system is extended to the dual type.

When a speaker is to be used only for reproduction of low-frequency components, in other words as a woofer, it is sometimes mounted in back of a slot instead of a circular opening. The slot is usually rather narrow, and its length is less than the diameter of the speaker cone. (Such a slot is shown schematically in Fig. 7-2C in the next chapter.) Such a slot acts as an acoustic low-pass filter. It attenuates all sound components above a certain frequency. It also improves the loading and impedance match to the speaker cone at low frequencies.

Unlike woofers, tweeters cannot normally be used alone. All types therefore have the same special purpose—to reproduce only the higher-frequency portion of the audio-frequency range. One of the most important characteristics of a tweeter is its low-frequency cutoff point. The minimum frequency of operation of the unit must be below the crossover frequency of the system so that it will overlap the range of the woofer. If a low cutoff frequency (300 to 1000 Hz) is employed, the range of the tweeter from the crossover frequency to the limit of audibility for full-range fidelity in the system makes it difficult to obtain a unit with uniform response, minimum distortion, and wide-angle distribution over that range. On the other hand, relatively economical and simple driver and radiator arrangements will handle the range neccessary with a high crossover frequency such as 3000 or 3500 Hz. The choice of a crossover frequency is further discussed under "Crossover Networks."

The construction of separate tweeter units is much like that of the tweeter portions of the examples given of coaxial units in the previous section. Horns are more popular than cones because of their greater efficiency and greater potential frequency range. A typical tweeter assembly is shown in Fig. 6-25.

Standard stock models of cone-type speakers can sometimes be used as tweeters. However, it must be emphasized that just because a speaker is small it is not necessarily a good tweeter. In fact, most ordinary small speakers are not good tweeters and often have no better high-frequency response than an average woofer. Obviously, such speakers would provide no improvement at all over just a woofer and would be likely to add considerable distortion resulting from application of high-frequency signal components (from the divider network) to which it does not properly respond. If you are able to hear tones up to 15,000 Hz very well,*

---

*Children and adolescents can often hear sounds of frequencies up to 18,000 Hz, but each person's limit of audibility descreases with age and often drops to 10,000 Hz or lower.

Fig. 6-25. Typical tweeter-unit construction.

it is advisable to test a prospective tweeter speaker by applying a signal from an audio signal generator or oscillator and by noting how well the output holds up to the limit of your own audibility. In making the test, it is important to remember that as frequency rises, directivity sharpens; so, be sure to stay directly in front of the cone when listening for the upper-limit signal.

Multiple Combinations

Some audio engineers believe that the audio-frequency spectrum should be divided into more than two parts with speakers for each part for proper full-range reproduction. For example, systems employing three or four speaker radiators are commercially available—some in the co-axial form, others in the separate form. The use of three or more ranges will reduce the width of each range so that uniformity of coverage for each unit is much more easily obtained. For example, the woofer need cover only from the low limit, say about 30 to 60 Hz, to about 500 Hz or 1000 Hz. For this range, it is certain that the woofer cone will operate as a whole and that no attenuation due to breakup will occur. Another separate radiator is employed for the middle range of frequencies, from 500 to 1000 Hz to about 3000 to 8000 Hz, with the exact limits depending on individual design. This middle-range speaker unit is sometimes referred to as the *squawker*, and the three-speaker system is known as a *woofer-squawker-tweeter* combination. Because of the extended range of the squawker over the normal top frequency of a woofer, the tweeter

211

can start at a relatively high frequency, and its design requirements are not so rigid as for a tweeter in a two-way system. An example of a three-way coaxial or triaxial speaker is shown in Fig. 6-26. The woofer and squawker cones are mechanically connected (or divided) in a duocone arrangement. The tweeter, with its own separate driver, is mounted inside the squawker. An electrical crossover network is used to divide the amplifier output between the tweeter and the woofer-squawker unit.

Commercial units are also available for systems employing four sections. Typical frequency ranges are: low-bass section, 35 Hz to 200 Hz; mid-bass section, 200 Hz to 600 Hz; treble section, 600 Hz to 3500 Hz; very-high-frequency range, 3500 Hz to the limit of audibility.
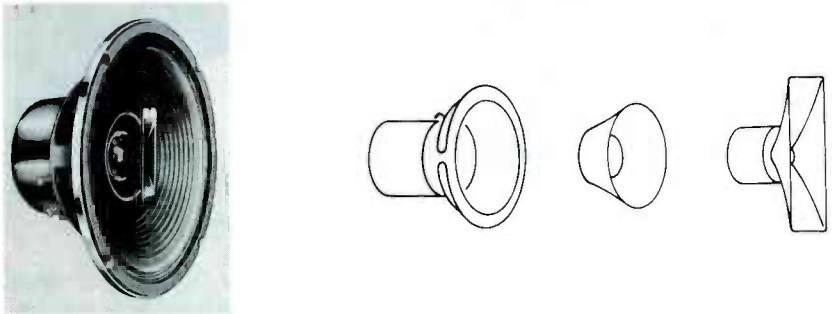
Fig. 6-26. A three-way coaxial speaker.

Multiple systems have the advantage of allowing relatively simple driver-radiator units to be used for each range. Since the unit for each range does not have to work as hard as in single-radiator systems, uniform frequency response should be and usually is easily obtained. However, in the rush to a dual or multiple system, the reader should not forget that there are other things of equal and sometimes even greater importance. The changes in directivity and in fine shadings of harmonic content between different components in a multiple system sometimes cause a consciousness in the listener of the takeover from one unit to the other. Some critics claim the apparent change of source location, particularly between the high and low notes of a single instrument, is evident in some systems. Then, in addition, if there is even a slight slip-up in the design or fabrication of the crossover network or the assumed takeover frequencies, severe distortion may result. In multiple systems, there are more parts which can self-resonate and cause trouble. These things are brought up not as pure criticism but to give the reader as much perspective as possible. While the dual or multiple system may be the complete answer for some enthusiasts, it is by no means a panacea in general. It should be remembered that any system—single, dual, or multiple—can give excellent results only when carefully designed and installed.

ELECTRICAL DIVIDER (CROSSOVER) NETWORKS

In dual or multiple speaker systems, the audio-frequency energy from the amplifier must be divided so that only the appropriate frequency components are fed to each unit of the system. In most cases, the individual parts of the system, although designed for optimum operation in their specified respective portions of the frequency spectrum, are subject to distortion and sometimes even overheating if they are driven to full power rating at frequencies outside their normal range. This is an important factor adding to the more obvious one: overall efficiency is substantially reduced by feeding too much low-frequency energy to a high-frequency unit and high-frequency energy to a low-frequency unit.

In dual systems employing a common voice coil, the mechanical compliance between the high- and low-frequency radiator sections of the cone divides the energy after it has been converted to mechanical motion of the voice-coil form. The compliance acts as a low-pass filter, eliminating most of the high-frequency components from the woofer or larger section of the cone. Low-frequency energy is fed to the high-frequency portion of the radiator, but not to it alone, because at low frequencies it acts only as part of the total mass composed of both low- and high-frequency portions.

When each of the units in a multiple system has its own voice coil or at least two have separate voice coils, the division of low- and high-frequency energy must be done electrically by divider networks.

The simplest type of divider network consists merely of a single capacitor, as illustrated in Fig. 6-27. The fact that the reactance of a capacitor is inversely proportional to frequency is employed to distribute the audio signal. In the arrangement of Fig. 6-27A, the tweeter and woofer voice coils are connected in series and a capacitor is connected across the woofer. The value of capacitance is made such that at frequencies above the desired range of the woofer the reactance of C becomes so low that it shunts the woofer, which acts as a bypass capacitor. Low-frequency components can be kept out of the tweeter if a parallel connection of voice coils is used with capacitor in series with the tweeter circuit, as illustrated in Fig. 6-27B.

Inductances can be used with capacitances to make the divider network more complete. For example, in Fig. 6-27B the inductance (L) can be connected in series with the woofer leads as shown. The inductance, the reactance of which increases with frequency, chokes the high-frequency components out of the woofer, and the capacitor (C) blocks low-frequency components out the tweeter. The values of C and L must be such that the reactance in each case is about equal to or a little lower than the voice-coil impedance in the frequency range to be attenuated.

A capacitor or inductor provides gradual attenuation with frequency, as the range of undesired frequency components is approached. Al-

though the crossover range should not be too narrow, simple reactance circuits as in Fig. 6-27 are ordinarily too broad in the changeover region. Instead, a combination low-pass (for the woofer) and high-pass (for the tweeter) filter circuit is usually employed. With this type of circuit, much more rapid attenuation can be made near the crossover frequency
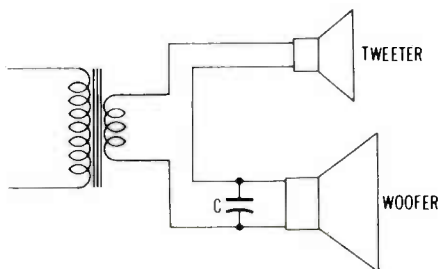
(A) *Series connected.*
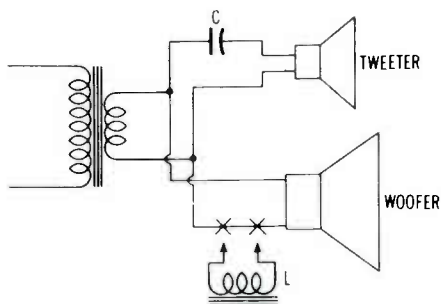
(B) *Parallel connected.*

Fig. 6-27. Simple divider circuits employing single reactances.

than is possible with simple capacitor and inductor arrangements as illustrated in Fig. 6-27. Attenuation of about 12 db per octave* is considered proper in most applications. Gradual crossover arrangements attenuate at about 6 db per octave. Filters with sharper cutoff than this can be constructed by use of additional components, but power losses in the filter become excessive, and the additional sharpness is not necessary anyway. A typical divider-network response graph is shown in Fig. 6-28. The curve of woofer output crosses the curve of tweeter output response at the crossover frequency. This intersection is also at the 3-db or half-power level; at the crossover frequency, half the output power is being fed to each unit. From this, it can be seen why the respective individual response characteristics of the woofer and tweeter units must overlap

---

*An *octave* is a range in which the frequency doubles.

214
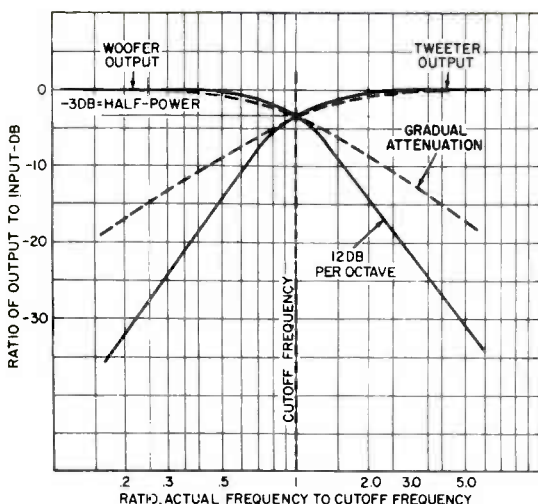
Fig. 6-28. Divider-network response curves.

substantially. If the crossover level were lower, there would be a lessening of total output in the crossover region; and this would result in frequency distortion of the system. The dashed-line curve of Fig. 6-28 represents a gradual crossover attenuation of 6 db per octave, compared to the more commonly encountered solid-curve value of 12 db per octave.

The construction of divider networks is not as simple as the schematic diagrams may indicate. Some of the reasons are:

1. The capacitors ordinarily require fairly accurate odd values that are hard to obtain without several components being connected together.
2. There is no polarizing voltage, applied voltages are purely a-c voltages at audio frequency, and electrolytic capacitors cannot be used. At the values necessary, other types of capacitors are relatively bulky and expensive.
3. Current at low impedance is appreciable; therefore, fine wire cannot be used for the coils, which have hundreds of turns for the values required. Again, the values are odd, so standard units are not applicable.

To illustrate the problem, the schematic diagram of one of the more popular types of divider networks is shown in Fig. 6-29, with formulas for calculating the values required for any crossover frequency f_c and speaker impedance Z and an example of its use in a practical application.* The values obtained in the solution of this example illustrate the

*For those familiar with filter theory, these are m-derived half sections with equations simplified for $m = 0.6$.

previous statements about the odd values and sizes of the capacitances involved. The circuit shown in Fig. 6-29 can be expected to give an attenuation of about 12 db per octave.

Although the difficulties mentioned must be taken into consideration, they are not insurmountable, and some audiophiles prefer to construct their own divider networks. If the reader wishes to do so, it is suggested that he consult some handbook for data as to diameter, number of turns, etc., for his calculated inductance values. Although the number of turns and overall size of the coils can be reduced by use of iron cores, this is not recommended, because the latter introduce nonlinearity. The capacitors can be of the oil-filled variety and can be made up from standard sizes connected in series or parallel.

For those who prefer to buy their crossover networks ready-made, the latter are available from a number of manufacturers of audio-frequency equipment. The schematic diagram, response curve, and physical construction of a commercial three-way network are shown in Figs. 6-30A and 6-30B. Note from the response curve that the crossover between the woofer and squawker is 500 Hz and that the crossover between the squawker and the tweeter is 5000 Hz. This model employs
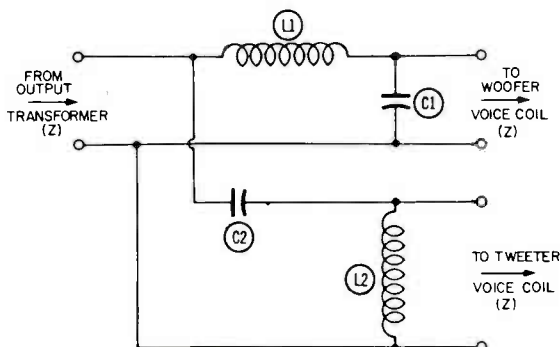
$$L1 = \frac{0.255\,Z}{f_c}\ \text{henrys} \qquad\qquad C1 = \frac{0.15}{f_c\,Z}\ \text{farads}$$

$$L2 = \frac{0.15\,Z}{f_c}\ \text{henrys} \qquad\qquad C2 = \frac{0.10}{f_c\,Z}\ \text{farads}$$

Example: Find component values for $f_c = 500\ \text{Hz}$ $Z = 8\ \text{ohms}$

$$L1 = \frac{0.255 \times 8}{500} = 4.08 \times 10^{-3} = 4.08\ \text{millihenrys}$$

$$L2 = \frac{0.15 \times 8}{500} = 2.4 \times 10^{-3} = 2.4\ \text{millihenrys}$$

$$C1 = \frac{0.15}{500 \times 8} = 37.5 \times 10^{-6} = 37.5\ \text{mfd}$$

$$C2 = \frac{0.10}{500 \times 8} = 25.0 \times 10^{-6} = 25.0\ \text{mfd}$$

Fig. 6-29. A filter-type divider circuit, formulas, and sample calculation.

Courtesy Klipsch & Assocs., Inc.

(A) Construction.
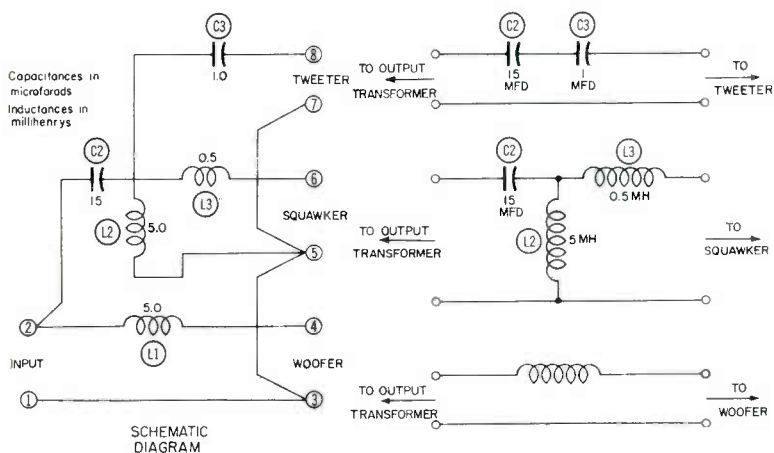
(B) Circuit.
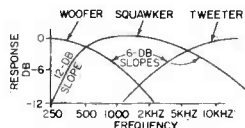
(C) Response.

Fig. 6-30. Designs features of a commercial three-way divider network.

gradual attenuation of 6 db per octave except of the low end of the squawker response, where it is 12 db per octave. The input and speaker impedances are 16 ohms. The schematic diagram is analyzed in Fig. 6-30C. The tweeter portion includes simply the two series capacitors, C2 and C3, for a total capacitance of slightly less than 1 mfd. The squawker circuit employs a series capacitor, a shunt inductor, and a

217

series inductor to form a high-pass filter. The woofer portion of the circuit is simply a series inductor (L1) which is small enough to pass the low frequencies but at the same time large enough to attenuate all but the lowest frequency components in the woofer circuit. The tweeter and woofer circuits are of the single-reactance type and cause the slow rate of
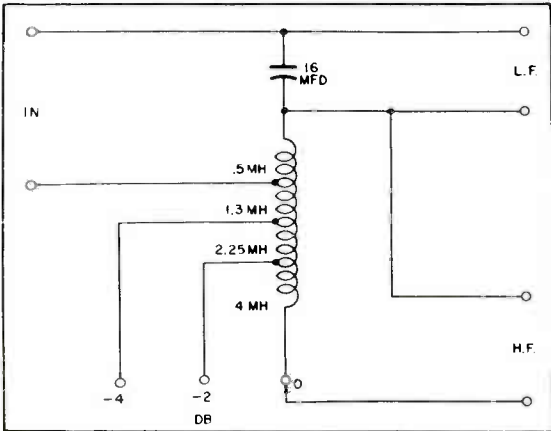
(A) *Circuit.*
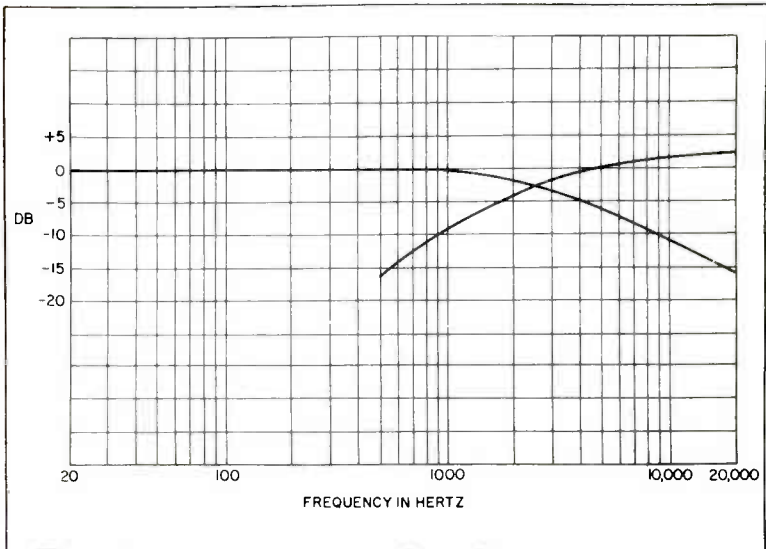
(B) *Response characteristics.*

Fig. 6-31. A commercial divider network which provides adjustment of the tweeter output to compensate for tweeter-horn efficiency.

218

attenuation (6 db per octave). The squawker circuit has more rapid attenuation because it is of the composite filter type.

When a horn-type tweeter is used with a cone-type woofer, there is a tendency toward energy unbalance between the high and low frequencies because of the higher efficiency of the horn. Some divider networks are designed to provide adjustment of the output of either section, relative to the other, to compensate for the difference in reproducer efficiencies. A typical commercial crossover network of this type is illustrated in Fig. 6-31A. The response characteristic of the network is shown in Fig. 6-31B. The response shown applies for the 0-db position. Note that the tweeter response can be reduced either 2 db or 4 db with respect to the woofer response by moving the tap.

When divider-network response is considered, it should be remembered that the actual acoustic attenuation at the crossover frequencies may be much greater than that indicated by the electrical circuit. This is because, even when separate voice coils are employed, there may be between units mechanical compliance which acts as an attenuator.

The divider networks described thus far are connected between the output transformer and the speaker units. Because the power level is high and the impedance low, the inductors must be capable of handling fairly high current and are therefore bulky. Because of the low impedance, the capacitors must also be large, as has been explained. For these reasons, some systems provide frequency division in the amplifier ahead of the power output stage. At that point, the power level is so low and the impedance so high that simple resistance-capacitance–type filters can separate the high-frequency components from the low-frequency components. From this dividing point to the speakers, two separate channels are provided. There are separate output amplifiers and separate output
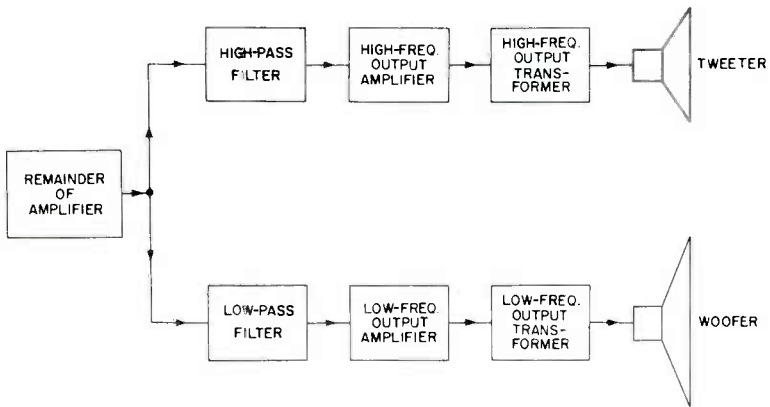
Fig. 6-32. A dual-speaker system in which the high- and low-frequency components are separated at a low power level in the amplifier.

transformers, one for each channel, as illustrated in Fig. 6-32. This is not as expensive an arrangement as might at first seem because, since each channel need have only a limited frequency response, the output transformers can be much less expensive.

Fig. 6-33. A typical dual–voice-coil unit for use as middle speaker in a stereo arrangement.

Courtesy University Loudspeakers, Inc.

CENTER SPEAKER FOR STEREO

Individual speaker requirements for stereo reproduction are substantially the same as for monophonic reproduction. The same low-distortion and wide-frequency-range requirements must be met in each unit of the stereo system.

One special feature developed especially for stereo is the dual voice coil. A typical speaker of this type is illustrated in Fig. 6-33. Two voice coils are wound on the same form on the cone structure of the speaker. This arrangement forms a convenient method of mixing the left and right stereo signals for the center speaker. Since the center speaker is usually required only to reproduce the lower frequencies, which are not as directional, the dual voice-coil speaker is usually a woofer. It is normally used with additional left and right speakers, which must reproduce only the frequency components of frequencies above approximately 200 Hz.

Speaker Baffles and Enclosures

In our discussion to this point, we have considered cone-type radiators as though all the sound energy were released from the front of the cone. This is essentially what we meant in an earlier statement that direct-radiator speakers would be considered as being mounted in an infinite baffle. Now that we have considered the design features of speakers, we shall proceed to describe the effect of their accessory equipment.

BAFFLES

Actually, sound energy is released from both sides of a cone. This is natural because there is air on both sides of the cone and the cone moves as a unit; however, when the cone moves forward, the air in front of it is compressed and the air in back of it is rarefied. The sound released from the rear of the cone is of opposite phase to that released from the front of the cone. If the sound from the rear is allowed to flow so that it meets the sound from the front, cancellation takes place and the response of the speaker drops off sharply. Such cancellation is substantial only when the paths to the meeting place are short compared to a wavelength and maximum when the total path length from the front of the cone to the back is exactly equal to zero or one wavelength. Sound waves from the rear change 360 degrees in phase in one wavelength and therefore oppose front waves. The wavelength of sound becomes longer as the frequency decreases; consequently, front-to-back interference is worst at the lowest frequencies and ordinarily marks the cutoff frequency of the speaker mounting. Such interference is not appreciable at higher frequencies at which the wavelength is small compared to the path length between the front and back of the cone. At these frequencies, the compressions and rarefactions are so closely spaced that there is no definite general cancellation action as at the low frequencies.

The longer the path length between front and rear, the lower is the frequency at which interference can take place. By extending the edges of the cone with some rigid flat material, we make it necessary for sound waves from the rear to travel out to the edges of the material before they can meet the sound waves from the front and interfere with them. The added material is called a *baffle*, and its principle is illustrated in Fig. 7-1. With the speaker alone (Fig. 7-1A), the front and rear waves must

221

travel only along one side of the cone to meet at the edge. This path is so short that a speaker alone without baffle will usually not reproduce much below about 350 Hz. In Fig. 7-1B is shown the situation when a baffle is added. The length of the interference path is increased by the width of baffle material on each side. The reader can clearly demonstrate this effect by operating a speaker connected to a radio receiver or record player. First, listen to the speaker alone; then place it against a temporary, improvised baffle. The latter can be a large piece of cardboard or corrugated carton with a hole cut in it. The increase of low-frequency response will be very clearly noticeable.

Baffles should be made of good sound-insulating material and should be soft enough to prevent rattle. Soft woods are satisfactory, but material like *Celotex* is more appropriate. The speaker must be securely fastened to the baffle, and the baffle must be rigidly mounted to prevent rattle.

If a baffle is to be of limited size, the speaker should not be mounted in the center. The center is a bad position because the path lengths to all four edges are the same, and the frequency components at which the path length (one side) is one wavelength are severely attenuated. The center position is illustrated in Fig. 7-2A. If the speaker is moved toward one corner, as in Fig. 7-2B, the uniformity of response is much better because the path length to each edge is different and the interference attenuation is distributed. The interference does not have to take place around the edge of the baffle but sometimes is purposely made to take place through a port, as illustrated in Fig. 7-2C. The design principle of such an arrangement is to equalize the response peak due to speaker resonance by the sharp attenuation around that frequency by spacing the port so that the sound travels a half wavelength. Some au-

(A) *Speaker alone.*   (B) *Speaker and baffle.*
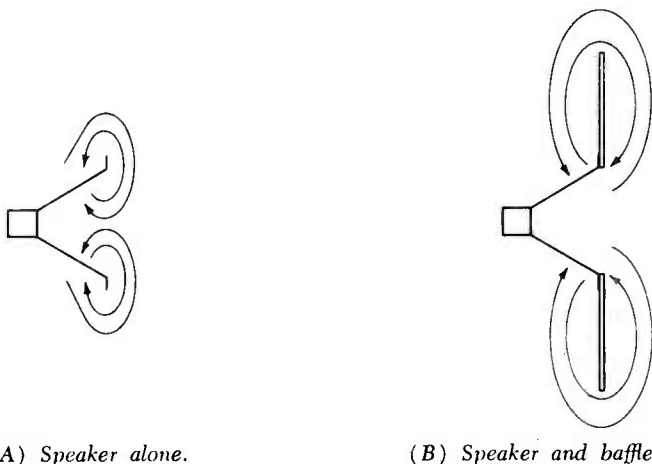
Fig. 7-1. Increasing the front-to-back interference path length by adding a baffle.

222

(A) Symmetrical
mounting.

(B) Offset mounting
on baffle.

(C) With Compensating
port.
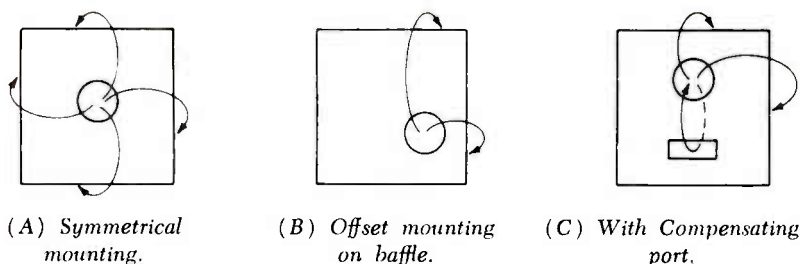
Fig. 7-2. Mounting a speaker on a flat baffle.

diophiles adjust the size and shape of the port until it balances out the resonant peak of the speaker.

For a theoretically perfect speaker, the ideal baffle is one which has infinite lateral dimensions. The interference path lengths are then infinite, and no matter how low the frequencies of the sound components, cancellation or reinforcement due to baffle limitations cannot take place. Obviously, an infinite baffle in the fullest sense cannot be realized. However, if the baffle dimensions are sufficiently large that the frequency at which the path length is a half wavelength is far below the range of frequencies to be used, the baffle is referred to as *infinite*. For example, a speaker mounted in a hole in the wall of a house and placed at least 6 feet from the nearest opening, with its back opening on one room and its front on the adjacent room or outside is, for all practical purposes, an infinite baffle. For use as an infinite baffle, one or two identical wide-range speakers with essentially flat response over the desired frequency range are recommended to be placed at ear level or directed toward ear level. Speakers for this application should have very low resonance characteristics because there will be no provision for eliminating such defects. In the previous discussion of speakers, it was stated that each direct-radiator speaker was assumed to be mounted in an infinite baffle because this removes the effect of front-to-rear interference and allows us to consider the inherent effects of the speaker.

SIMPLE ENCLOSURES

An infinite baffle, or an approximation of it, is one of the best speaker mountings. However, its size is a disadvantage in an ordinary home. For example, to reduce the frequency of interference to 50 Hz, the baffle must be at least 12 feet square! Because of this size problem, various arrangements have been developed in an attempt to get the same effect without the use of so much space. This has led to the design of speaker enclosures.

The evolution from a flat baffle to a simple enclosure is illustrated in Fig. 7-3. The simple flat baffle is symbolized in Fig. 7-3A. To reduce the maximum dimension of the baffle, the outer portions can be bent back

223

at the four edges to form an open box, as indicated in Fig. 7-3B. The path length is as great as for the flat version, but the lateral dimensions are smaller. In Fig. 7-3C the process is carried one step further, and the back of the enclosure is closed; the back prevents any sound from the rear from getting to the front.

The open-back cabinet arrangement of Fig. 7-3B is the one commonly employed for radio and television receivers. A glance at the size of midget radio cabinets and a quick estimate of their path lengths will quickly show why low-frequency response is lacking in this type of receiver. The small speaker used is also a low-frequency limiting factor, but the cabinet is usually the important limitation. Large console models employ larger speakers and larger cabinets, but the path length is seldom sufficient to allow reproduction as low as 150 Hz, unless some special cabinet design other than the simple open-back box is used.

Another disadvantage of the open-back box is the fact that it acts as a resonant tube at some frequency well within the operating range, unless it is very large. Sounds of frequency near resonance are reproduced with annoyingly excessive volume relative to other sound components. Any sudden sound peaks of any frequency or sounds of low frequency can shock-excite the box into oscillation at the resonant frequency. Low-frequency sounds which are not attenuated by the interference path all seem to sound the same, because of shock excitation at the resonant frequency. This accounts for the fact that many console radio receivers and some of the earlier juke boxes emitted a constant booming during reproduction of music.
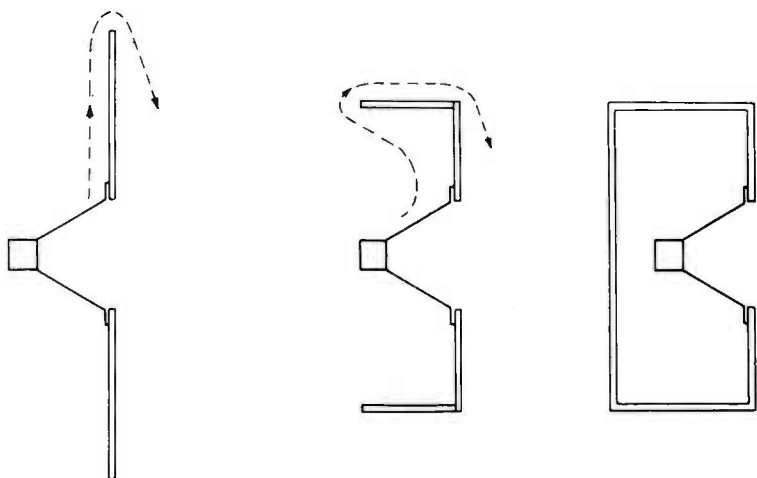
(A) Speaker with simple flat baffle.

(B) Open-back enclosure.

(C) Closed-back enclosure.

Fig. 7-3. Evolution from a flat baffle to a simple enclosure.

It would seem, then, that simply closing the box as in Fig. 7-3C would be the answer; and the box could be as small as desired as long as it holds the speaker because the sides and back would block the rear-to-front interference path. Unfortunately, this is not so. As soon as the box is closed up tight, as in Fig. 7-3C, the air in it is no longer free to move in the open as in the open-back cabinet. Instead, the action of the cone causes pressure changes in the cabinet rather than a combination of pressure and velocity. This means that the springiness or compliance of the air is an important factor. Compliance is acoustic capacitance and combines with the compliance of the cone suspension in such a way as to raise the resonant frequency of the system to a value higher than that of the speaker alone. The reason for this effect can be noted from the equivalent circuit for the closed box, given in Fig. 7-4. The mass (inductive effect) of the cone, the compliance (capacitive effect) of the cone suspension, and the compliance (capacitive effect) of the air in the box are all effectively in series with each other. The smaller the box, the smaller is the acoustic capacitance it simulates. Use of a small box lowers the capacitance connected in series with the series-resonant circuit of the speaker and thus raises tthe overall resonant frequency.

An important feature of the closed box is the fact that, because there is no motion of air in and out of it, there is no inertance or inductive effect. There is only compliance, or capacitive effect. Accordingly, the box does not resonate itself as do some other types which we shall consider later. All it does is enter the resonant circuit of the speaker unit and raise the resonant frequency of the system above that of the speaker.

Compliance and thus the capacitive effect of a closed box increases with size if the box can be made large enough so that the equivalent air compliance capacitance is large compared with equivalent speaker-suspension compliance capacitance. Then the box will raise the resonant frequency only a negligible amount above the resonant frequency of the speaker. On the other hand, because the only effect of the box is to raise

C_E = COMPLIANCE OF AIR IN ENCLOSURE
C_C = COMPLIANCE OF CONE SUSPENSION
m_c = MASS OF CONE AND VOICE-COIL ASSEMBLY
R = RADIATION RESISTANCE
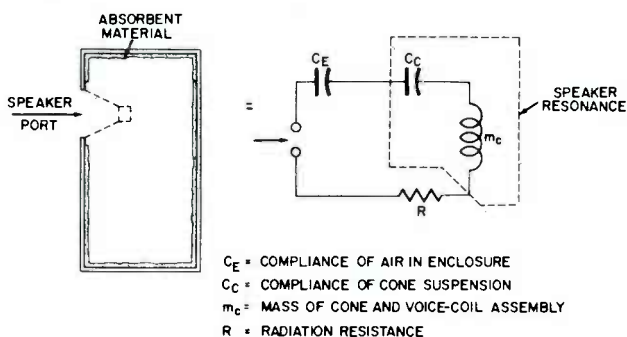
Fig. 7-4. Equivalent circuits of closed-box enclosure.

225

the resonant frequency, a speaker with a very low self-resonance could be put into a relatively small box. Then the system resonance would still be low enough for good results, even though it is raised above the resonant frequency of the speaker alone. Generally, a closed box of a given volume will raise the resonant frequency of the system a given percentage above the resonant frequency of the speaker.

Because the closed box keeps back radiation from getting around to the front and interfering, it is frequently referred to as the *infinite-baffle enclosure* and sometimes even as *infinite baffle.* As should be clear from the preceding discussion, this closed-box enclosure is not at all equivalent to a true infinite baffle unless it is so large that its effect on the resonant frequency is negligible.

Although the primary consideration in connection with a closed box is its effect on system resonance, this is not by any means the whole story. Conditions are such that as frequency increases above the resonance value, the system goes through a series of minor resonances which, if not counteracted, cause substantial irregularities in the response characteristic. There is also an effect due to reflection of back-radiated energy from the inside of the back and walls of the enclosure, with irregularities resulting from phase variations between cone vibrations and such reflected energy reaching it. These effects are minimized in practice by padding the inner surfaces of the cabinet walls with sound-absorbent material such as heavy felt, glass fiber, rock wool, or cellulose. This padding material acts as an acoustic resistance, lowering the Q of the box at high frequencies and tending to smooth out the response.

The *boffle* is a development of the foregoing, fitted with a system of screens which act as two-stage acoustic filters damping out cabinet and air resonances.

The compliance of the box increases with the size of the speaker cone. Therefore, the larger the cone, the greater is the percentage increase of speaker resonance caused by the enclosure. The resonant frequency of larger speakers is ordinarily so much lower than the resonant frequency of smaller speakers that the compliance increase is overcome, and the system resonance is lower with a larger speaker. Practical designs are such as to limit the increase of resonance due to the cabinet to 20 percent over the speaker resonant frequency. Authorities do not agree on any standard cabinet volume necessary for a given size of speaker, probably because of the wide variations of speaker resonance and suspension compliance in different models of speakers of the same cone size. Data from different sources specify that from 400 to 700 cubic inches of cabinet volume per inch of nominal speaker diameter should be used as a minimum. Since it is safer to stay on the large side, let us assume that 600 cubic inches per inch of nominal speaker diameter is a good minimum value. On this basis, some typical speaker sizes and minimum required enclosure volumes would be as given in Table 7-1.

226

One significant design feature in many of these infinite-baffle types of enclosures is the breakup of the rear wall to minimize internal reflection effects which get bad at higher frequencies. These breakup arrangements vary from two-surface double panels to complicated assemblies of small flat surfaces all at different angles. Another method of minimizing these reflections is to make the whole rear portion of the enclosure semispherical, to break up reflections into an infinite number of angles. Representative of this type is the "kettledrum," which looks like the instrument after which it is named, with the speaker unit mounted in the diaphragm portion of the drum.

Table 7-1. Minimum Enclosure Volume for Typical Speakers

Speaker Size (inches)	Volume	
	(cu in)	(cu ft)
6	4800	2.8
10	6000	3.5
12	7200	4.2
15	9000	5.2

One manufacturer makes use of the infinite-baffle box but makes the rear wall with a flexible mounting to the remainder of the box. This makes the cabinet give under sound-pressure waves, reducing the stiffness of the enclosure acoustically. The acoustic impedance of the box thus becomes equivalent to that of a much larger box. This enclosure is used with an extended-range, single-cone speaker to give excellent performance in a realtively small space.

In the infinite-baffle enclosure, all radiation from the rear of the speaker is eliminated. This minimizes interference but also makes systems of this type less efficient than others which utilize the sound from the rear.

One way to overcome this loss of efficiency is to use two or more identical speakers in the enclosure. When two or more speakers operate on the samel audio-frequency signal and are mounted close to each other, they interact. Sound energy from one speaker reinforces the vibration of the cone of the other speaker, and vice versa, in what is known as mutual coupling. The action is similar to that in a multielement antenna array. The result is that the efficiency of the combination is much better than the efficiency of either speaker alone. Another advantage arises from the fact that even speakers of the same model are seldom exactly alike, because of tolerances in manufacture. These tolerance are actually an advantage because the difference in the resonant frequencies of the respective speakers tends to distribute the speaker resonance effect and make low-frequency response more uniform.

This idea is frequently used in infinite-baffle enclosures but is just as applicable to other types of enclosures. An example is given under the topic of bass-reflex enclosures.

BASS-REFLEX ENCLOSURES

The infinite flat baffle and the closed box eliminate back interference by isolating the rear of the cone from the front. In another, more popular arrangement, radiation from the rear of the cone is used to reinforce the front radiation rather than cancel it. This arrangement is illustrated in Fig. 7-5 and is known as the *bass-reflex enclosure*. It is the same as the closed box except that an opening is cut below the speaker. This
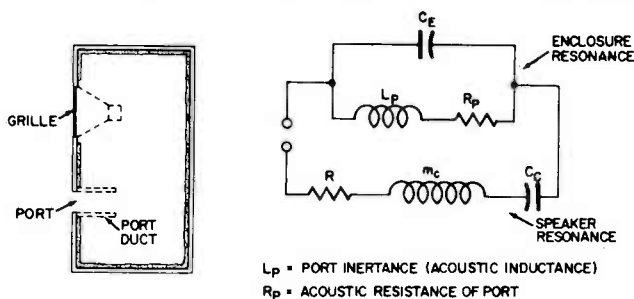
Fig. 7-5. The bass-reflex enclosure and its equivalent circuit.

opening, or port, allows air to flow in and out of the box as the speaker cone moves back and forth. The basic design principle is that the back radiation is fed through a path, including the box as a whole and the port, such that the back radiation emerges and reinforces radiation from the front.

Addition of the port adds acoustic inductance (inertance) to the load offered to the speaker by the cabinet, because of the motion of air through the port. Although the stiffness of the air in the enclosure is not so great as in the closed box, it is still appreciable. The flow of air through the port is also accompanied by friction, which is an acoustic resistance effect. As illustrated in the equivalent circuit in Fig. 7-5, the port inertance, friction, and enclosure compliance form a parallel-resonant circuit. This circuit is connected effectively in series with the series-resonant circuit of the suspension and mass of the cone of the speaker.

Because of this more complex acoustic-impedance pattern, the design of a good bass-reflex enclosure involves a little more than simple path length. The effective length of the path of the sound from the rear of the cone is primarily dependent on the nature of the impedance offered by the enclosure itself, as expressed in the equivalent circuit. The design is simply a matter of keeping the resonant frequency of the enclosure in proper relation to the resonant frequency of the speaker. Because the

enclosure offers parallel resonance, and the speaker series resonance, one of these resonances can be made to offset the other. Accordingly, one of the most popular methods of design places cabinet resonance right at the speaker resonance frequency. Others believe in making the enclosure resonate slightly lower in frequency than the speaker, feeling that, although this does not completely compensate for speaker resonance, it extends the overall response to a lower frequency.

Placing cabinet resonance exactly at the speaker resonant frequency does not eliminate the bump in frequency response due to speaker resonance, unless the Q's of the two resonant frequencies are approximately equalized. Such equalization is accomplished by loading the enclosure with acoustic resistance until the overall response is smoothest. Acoustic resistance is supplied by absorbent material on the inner walls but more efficiently by placing grille cloth or other meshlike material over the port. When air is forced through small holes, friction, which is acoustic resistance, is developed.

The port cannot be placed too close to the speaker because then the path from the rear of the cone starts to become direct, like that in a small flat baffle, rather than including the impedance of the box as a whole, as it should. To keep the overall cabinet size to a minimum, some designs add a duct in back of the port, as shown by the dashed lines in Fig. 7-5. This allows maintenance of a minimum direct path length from the rear even for relatively small spacing between the port and the speaker and keeps down the overall volume. There is a limit to volume reduction by this method because, as the port becomes larger, the effective volume of the cabinet (which must not include port-duct volume) becomes smaller inside. The end of the port duct must also not come too close to the rear wall of the enclosure and should not exceed one-tenth of a wavelength at the resonant frequency.

The cabinet resonance is determined by the well-known Helmholtz resonator equation:

$$f = 2070 \sqrt[4]{\frac{A}{V^2}}$$

where,

f is the resonant frequency of cabinet, in hertz,
A is the area of port, in square inches,
V is the volume of cabinet, in cubic inches.

Calculations of the design parameters of the bass-reflex enclosure are beyond the scope of this book, and our approach has been to explain which factors are important and why. Dimensional information has been carefully worked out and put into graphical form, an example of which is shown in Fig. 7-6. The cabinet volume must have a definite value and should not be as large as possible, as occurs with the flat baffle and the closed box.

Because of variations in speakers and tolerances in cabinet construction, the actual cabinet and speaker resonant frequencies seldom coincide exactly, even though designed to do so. Because the area of the port influences the cabinet resonant frequency, this area may be varied after construction and assembly are complete, to trim the cabinet resonance
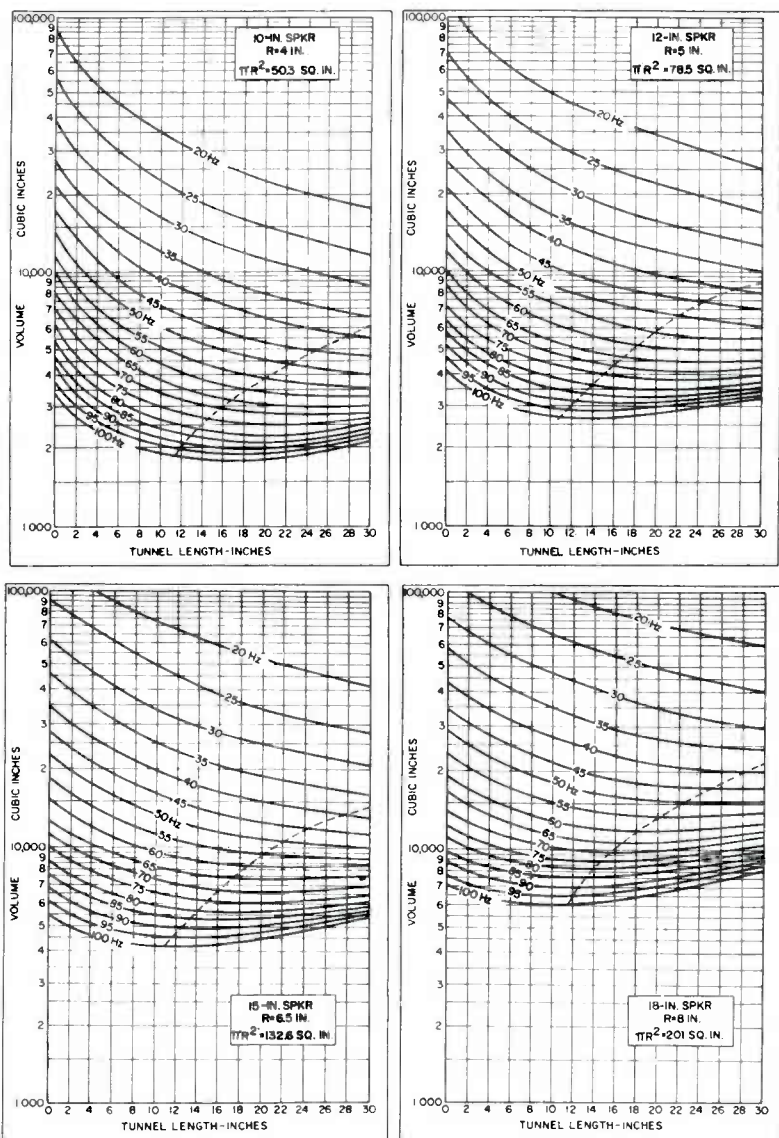
Fig. 7-6. Design graphs for bass-reflex cabinets.
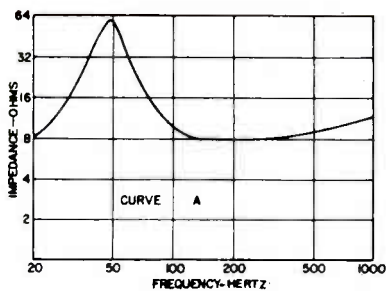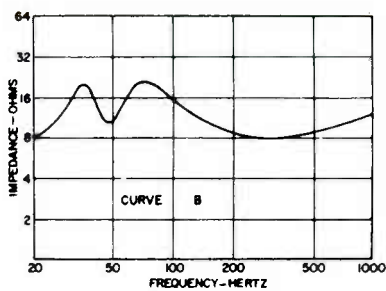
(A) *For speaker only.*   (B) *Speaker with ported enclosure.*

Fig. 7-7. Speaker impedance versus frequency curves.

to match the speaker. The port can be covered up in part by a book or other flat, rigid device, and the open area can be varied.

An audio-frequency signal generator (audio oscillator) of reasonably constant output is useful in checking for proper port-area adjustment. The front of the speaker cone mounted in the enclosure is watched as the generator output frequency is passed through frequencies from 30 Hz upward. As the resonant frequency of the speaker is reached, cone excursion increases greatly to a maximum and subsides to fewer deflections at higher frequencies. The audio signal is adjusted for resonance. Then the port area is varied until cone excursion becomes minimum. The speaker resonance is then being damped by cabinet antiresonance.
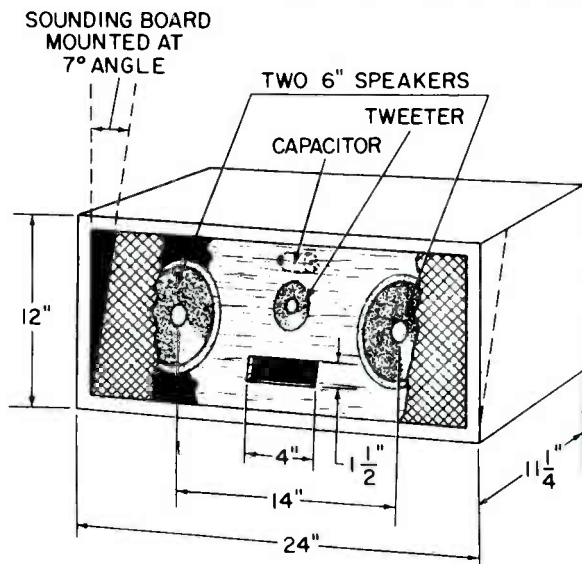
Fig. 7-8. The multiple-speaker arrangement.

Because the bass-reflex enclosure reverses the phase of the back radiation and makes it additive to the front radiation at low frequencies, it is often referred to as an *acoustic phase inverter*.

Some designers prefer to make the bass-reflex type of enclosure with distributed ports. Instead of one port of calculated size, a number of smaller ports are used. The areas of these ports add up to the proper calculated port area. It is claimed that the different positions of the different holes prevent dips in the response curve at frequencies at which the interference path length becomes one wavelength.

The effect of the bass-reflex type of enclosure in damping the response peak of speaker resonance is illustrated in Fig. 7-7.

Sometimes more than one woofer or tweeter unit is used in a bass-reflex cabinet to take advantage of mutual-coupling efficiency. An example of a small bass-reflex enclosure employing two 6-inch woofers and a tweeter is shown in Fig. 7-8.

ACOUSTIC LABYRINTH

Another method of acoustic phase inversion is exemplified by the acoustic labyrinth, depicted in Fig. 7-9. The cabinet is divided into parts by a series of baffles in such a way that the spaces between the baffles will form a lengthened passage, or duct, of approximately constant cross section between the back of the speaker and the front of the cabinet. The labyrinth thus feeds the back radiation around to the front.

Besides the acoustic inversion effect, the labyrinth has another very important design consideration. The pipe or tube simulated by the space between the baffles acts as a tuned line when it is exactly a quarter-wavelength long at the resonant frequency of the speaker. It simulates a parallel-resonant circuit which equalizes the series resonance of the speaker unit in the same manner as in the bass-reflex cabinet. The labyrinth has two main beneficial actions: (1) It equalizes the resonant bump in the speaker response and spreads out low-frequency response, and (2) it provides reinforcement of the sound in the range near the frequency at which the duct length is a half-wavelength (twice the resonant frequency of the speaker).
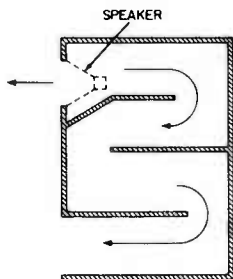
Fig. 7-9. An acoustic-labyrinth cabinet arrangement.

HORN-TYPE ENCLOSURES

Audio engineers have always been attracted to horns because of their high efficiency and good frequency response above cutoff frequency; but, as was brought out in our previous basic discussion of horns, a conventional horn structure for a low enough cutoff frequency for high fidelity (say 50 Hz or lower) is so large at the mouth as to be prohibitive for ordinary use. To overcome this problem, a number of designs have been developed to simulate the performance of a large horn without the large dimensions necessary in the conventional type. In home installations, the use of horn-type enclosures is confined to the low-frequency range, and they are ordinarily employed in conjunction with a separate tweeter (which may itself have a separate horn), or they are driven by a coaxial or extended-range, single-cone driver. In the home, cone-type drivers are almost universally used to drive the horns at low frequencies.

One of the first measures taken to reduce size is the folding of the horn. Folded or rolled exponential horns have been used for centuries in musical instruments, and these look like that shown in Fig. 7-10A. Of course, this type has a small throat and in electrical sound systems would be used with a diaphragm-type driver. In high-fidelity systems, horns are driven by cone-type speakers (for low frequencies) and must therefore have large throat diameters.

A low-frequency horn application more typical of high-fidelity equipment is shown in Fig. 7-10B. The front of the cone faces away from the listener, but its output is directed around between the baffles toward the
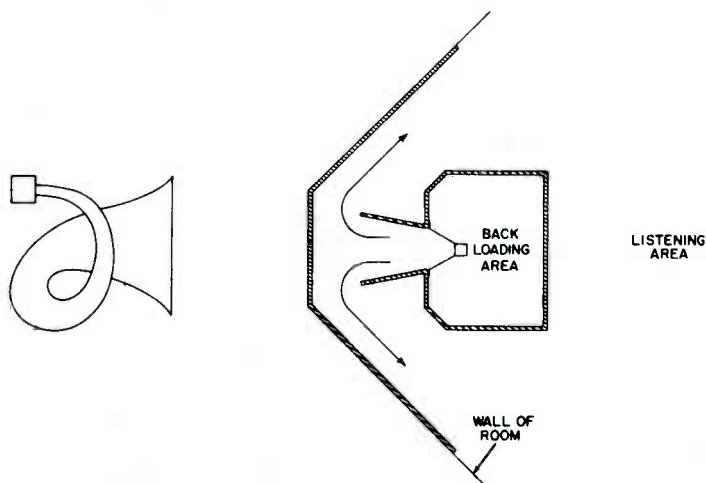
(A) *Musical instrument type.*   (B) *Low-frequency horn.*

Fig. 7-10. Folded horns.

233

listening area. The baffles are so constructed and mounted that the closed area leading from the speaker cone to the room roughly approximates an exponential horn of the low-frequency cutoff desired. This type of horn in practical design cannot in itself have the full mouth area necessary for theoretical cutoff in the desired range of approximately 50 Hz.
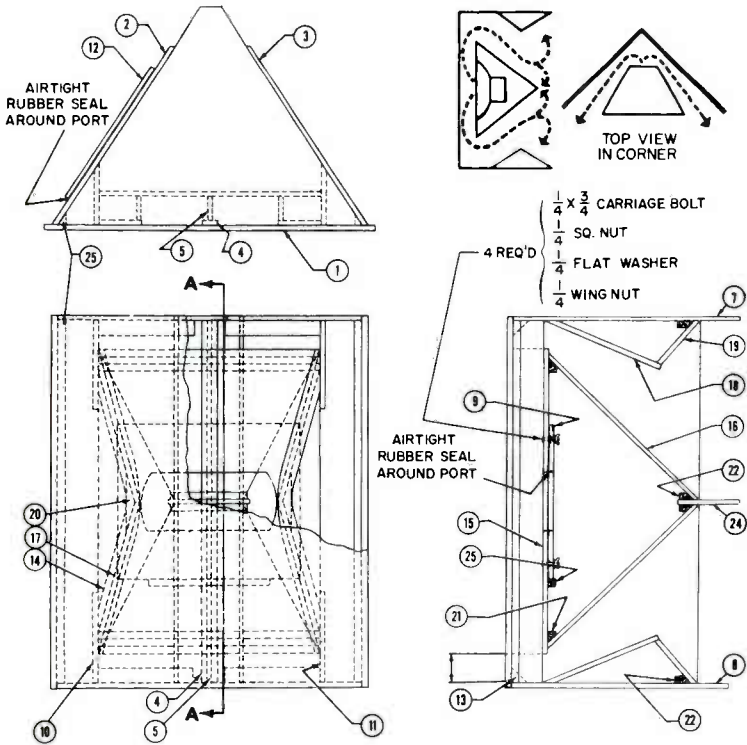
Fig. 7-11. Klipschorn arrangement used in Electro-Voice *Georgian* baffle assemblage.

The structure is normally designed to be placed in the corner of a room, and the walls of the room become continuations of the sides of the horn, as illustrated.

Notice the back-loading area in the structure of Fig. 7-10B. This is considered necessary for cone drivers because without it the loading on the cone is not symmetrical and distortion results. This is why in practically all low-frequency horn structures some balanced loading is provided. The back loading is not always a close area but in some cases constitutes slots or ports.

234

One of the best known of the better reproducing systems is the Klipschorn arrangement. A layout of the interior of a typical commercial type of the Klipschorn is illustrated in the diagrams of Fig. 7-11. The principle is the same as that of Fig. 7-10B. A cutaway view of the original Klipschorn is shown in Fig. 7-12. An enclosure in the center of the cabinet back-loads the woofer. The front of the woofer cone opens into a narrow front channel, which is the throat of the horn. From there, the woofer output is lead around the center enclosure through ducts of a
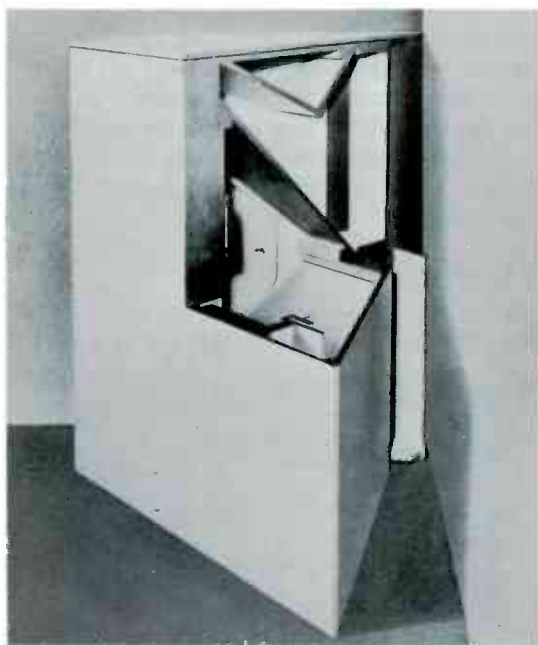
Fig. 7-12. Cutaway view of Klipschorn woofer horn.

gradually increasing cross-sectional area. These ducts open out to the back of the cabinet. A vertical baffle at the back leads the sound to emanate from the sides through vertical openings between it and the remainder of the cabinet. The sides of the cabinet are at such an angle with respect to the walls that the space between them and the walls forms a continuation of the horn. As the sound waves pass the front of the cabinet, they are still enclosed in the continuation of the horn formed by the walls in the corner of the room. A cutaway view of the low-frequency horn and the complete system, with tweeter horn mounted on the top, is shown in Fig. 7-13. Another horn-enclosure arrangement utilizing a labyrinth back-loading is shown in Fig. 7-14.

The Empire speaker with acoustical suspension and adjustments (Fig. 7-15A) has the following features (Fig. 7-15B):

1. A 12-inch mass-loaded woofer with floating (acoustical) suspension, a 4-inch voice coil, and a large (18 lbs) speaker ceramic structure.
2. Sound-absorbent rear loading.
3. Die-cast mid-frequency/high-frequency full-dispersion acoustic lens.
4. Imported marble top.
5. Ultrasonic domed tweeter.
6. Full-presence mid-range direct radiator.
7. Totally damped acoustic fiber enclosure.
8. Exclusive dynamic reflex stop system for enriched bass response; can be adjusted to suit room acoustics.
9. Front-loaded horn with 360-degree-aperture throat.
10. Terminals concealed underneath.

Compact Hi-Fi Speaker Systems

Compact high-fidelity speaker systems in beautiful wooden rectangular cabinets suitable for placing on a bookshelf are now designed and made

Courtesy Klipsch & Assocs., Inc.

Fig. 7-13. Rear view of Klipschorn Style 7.
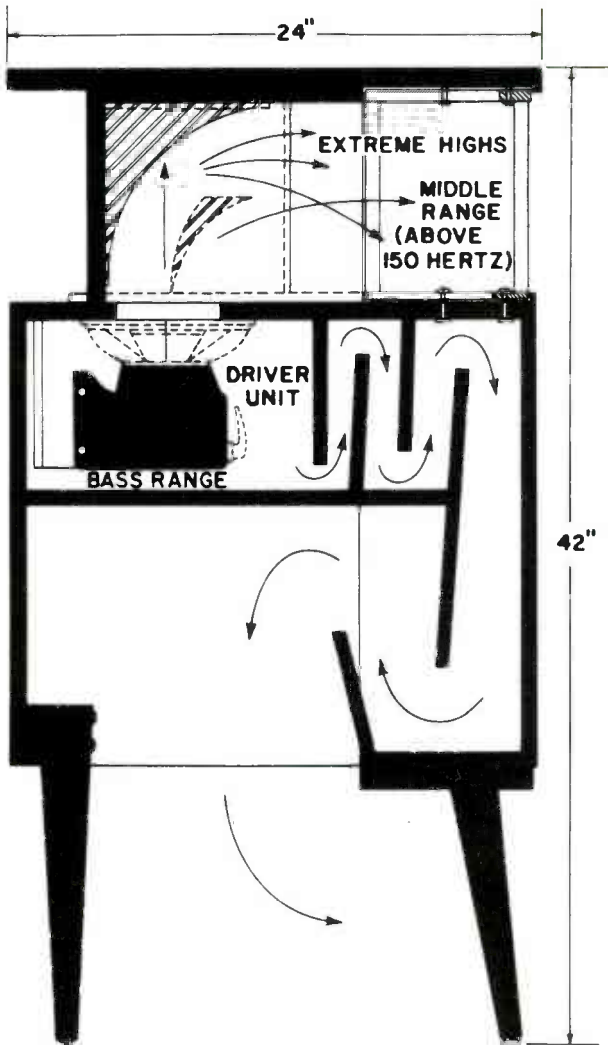
Fig. 7-14. Cross section of the Brociner Model 4 horn.

to reproduce as accurately as any other type of system available—and at reduced prices over larger systems. These systems will reproduce accurately from 40 to 20,000 Hz and can be purchased in the price range of $50.00 to $500.00. See Fig. 7-16.

Exponents of larger speakers and systems point out that low bass response requires movement of large amounts of air and therefore hi-fi reproduction requires a large speaker cone. However, new designs and

materials and the application of the natural laws of physics have made possible the movement of larger amounts of air with small cones by the use of loose mountings (in acoustical suspension) allowing greater lateral movement, and by adding to the effective mass of the moving material. The increased travel of the cone moves the same amount of air as a larger cone with normal travel, and it provides more accurate control and reproduction. The latter is accomplished by the use of heavier and stronger magnets.

Courtesy Empire Scientific Corp.

(A) *Photograph.*
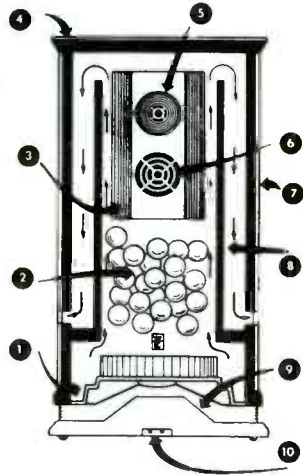
Courtesy Empire Scientific Corp.

(B) *Cutaway view.*

Fig. 7-15. The Empire *Grenadier* 800P speaker.

Acoustical suspension is provided by the reduction of forces on the cone at the mounting ring by the use of soft materials such as polyvinyl chloride (PVC) painstakingly applied to provide greater stretch with dynamic concentric alignment and negligible resistance to wide lateral movement. The force necessary to dampen the movement of the cone is provided by the elastic air sealed inside the enclosure (thus it is said that the cone is in acoustical suspension) or suspended on pressurized air and the elastic mount.

In these designs the compression and decompression operate in a nearly linear manner, reducing much of the distortion inherent in larger speaker designs. In addition, tweeters and mid-range reproducers have been improved in quality and width of dispersion, creating overall im-

238
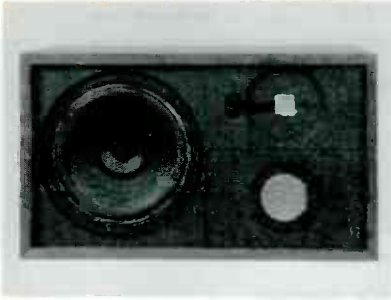
(A) *Acoustic Research AR2ax with front cover removed.*
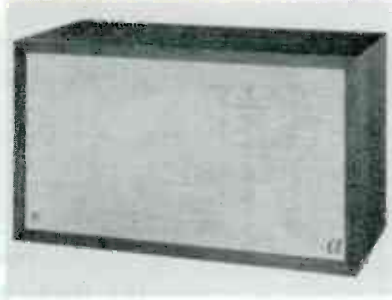
(B) *Acoustic Research AR2ax with front cover in place.*
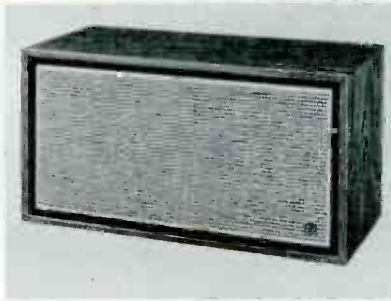
(C) *Electro-Voice EV7.*

(D) *University UR-4.*

(E) *Empire Model 8500 wall-mount system.*

(F) *University Mediterranean-style end-table system.*

Fig. 7-16. Compact and end-table speaker systems.

239
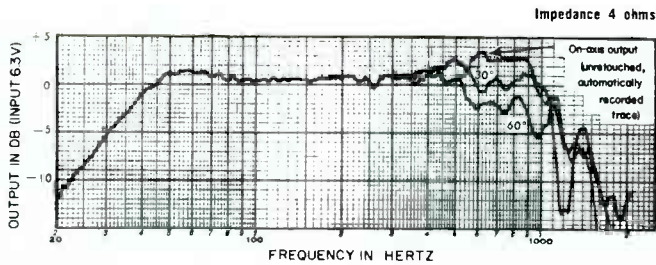
(A) Woofer frequency response.
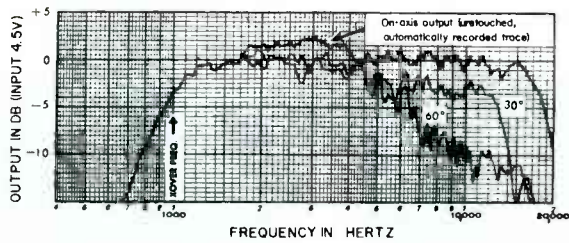
(B) Mid-range frequency response, measured in test baffle.
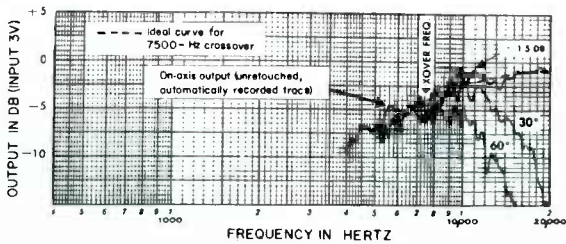
(C) Super-tweeter frequency response, measured in test baffle.
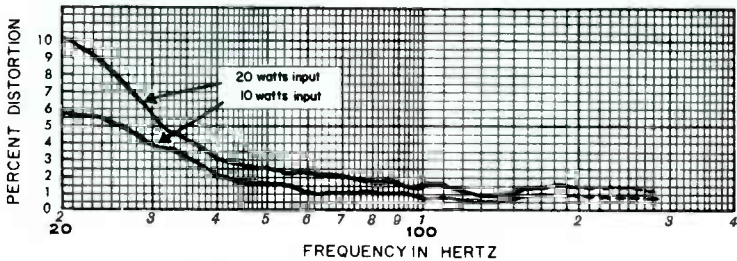
(D) Woofer harmonic distortion.

Fig. 7-17. Output characteristics of Acoustic Research AR-3 speaker.

provements that widen the utility and application of lower-cost high-fidelity systems.

For output measurements of Acoustic Research speaker AR-3 see Fig. 7-17. In this figure, the response curves of each radiator are shown separately so that interference patterns—which reflect the particular position of the recording microphone rather than reproducing quality—are eliminated. The speaker provides level controls for independent adjustment of the mid-range unit and super-tweeter.

STEREO SPEAKER SYSTEMS

The basic design principles of speaker enclosures are the same for stereo as for monophonic reproduction. However, the fact that two or three sound reproducing sources must be used in stereo has led to variations in how these enclosures are arranged.

The simplest approach is two complete reproducing systems, one to handle the left signal and one for the right signal. Stereo speaker-enclosure systems have evolved along the lines indicated by the three arrangements illustrated in block form in Fig. 7-18. Many arrangements take advantage of the fact that the stereo effect is obtained predominately with higher-frequency components of the sound, because at low frequencies a wavelength is so great that phase differences and depth distinctions between spaced objects are small. There is no definite frequency at which transition takes place, but there is a gradual change effect. Authorities place the dividing line for practical purposes between 150 Hz and 400 Hz. Frequency components in this range and below may come from a single source since little stereo effect can be produced under normal conditions.

In Fig. 7-18A the existing system has been expanded by addition of a limited-range (not much low-frequency response) compact enclosure to the full-range system previously used for monophonic reproduction. The added speaker provides all but the lowest frequency components and is the right speaker for the stereo effect. The full-range system provides the left stereo speaker and reproduces all of the lowest-frequency nondirectional components.

Two compact wide-range enclosures and a low-frequency center speaker are illustrated in Fig. 7-18B. The center reproducer need only be a woofer, responsive to frequencies up to 500 Hz. A dual voice-coil speaker, like that shown in Fig. 6-22, would be a logical choice for the center speaker. However, if the system is being evolved from a former mono system, the center speaker could be the full-range reproducer from that system.

The advent of stereo stimulated the trend toward smaller enclosures. Therefore, special techniques have been developed to reduce the size and price of full low-frequency response units. A number of compact single-channel reproducers have been developed with excellent characteristics

for arrangements like those of Fig. 7-18A and B. They are available in all finishes and styles. Some are floor models; others are placed on a bookshelf or table. Some contain single speakers and have response from
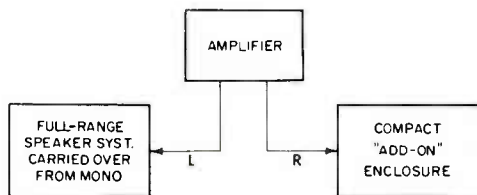
(A) *Addition of compact second-speaker enclosure to existing monophonic system.*
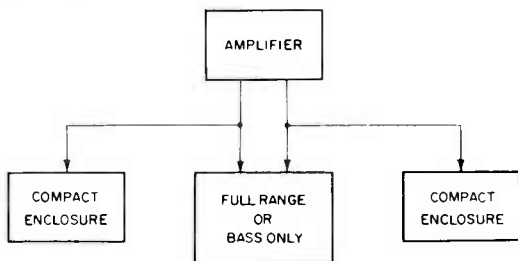
(B) *Use of existing monophonic system as a woofer in center, compact enclosures at left and right.*
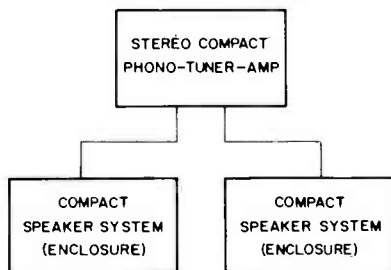
(C) *Compact stereo system.*
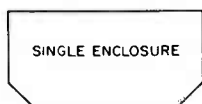
(D) *Use of one large cabinet for complete system.*

Fig. 7-18. Popular types of enclosure arrangements for stereo.

(A) *Bogen stereo system.*

(B) *Electro-Voice system with EV4 speakers.*

Fig. 7-19. Use of compact enclosures.

about 50 Hz up, so they can be used with a center woofer with no low-frequency loss; others are full-range arrangements, with claimed frequency response extending from 40 to 20,000 Hz. The latter could be used without center speaker, to provide left and right sources for a two-source arrangement. Most of the compact enclosures do not exceed 2 feet in their maximum dimension. Two typical complete arrangements using compact enclosures are shown in Fig. 7-19.
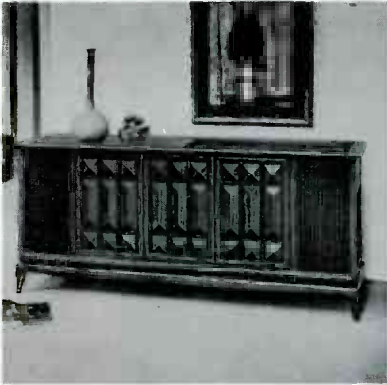
Courtesy The Magnavox Co.
(A) *Magnavox* Astro-Sonic *radio phonograph.*

Courtesy The Magnavox Co.
(B) *Magnavox* Agean, *with color television.*

Fig. 7-20. Single-cabinet stereo systems.

There are still a large number of people who would prefer to have the complete stereo system enclosed in one cabinet to simplify furniture arrangement. Because the separation between left and right reproducers must be at least 5 feet (effectively) this means that the cabinet must be either 5 feet long or must have some arrangement by which the sound can be projected so as to simulate the proper spacing. Two examples of single-cabinet stereo-system reproducer enclosures are shown in Fig. 7-20.

A very elaborate stereo reproducing system is pictured in Fig. 7-21. The design of the reproducer is illustrated in Fig. 7-22. Basically it consists of two three-way (woofer, mid-range, and tweeter) speaker systems, one mounted on each side of the enclosure structure. Their sound outputs are directed against a large round surface which reflects them to the listening area in such a way as to distribute the sound yet maintain its essential directivity for stereo effect.

Stereo Headphones

A good pair of stereo headphones is a worthwhile accessory to any high-quality music system. An increasing number of amplifiers and re-

Fig. 7-21. A full-range stereo horn enclosure.

ceivers include a stereo-headphone jack on their front panels, and, if no jack is available, connection to the speaker terminals is a simple matter. If a tape recorder is part of the music system, then any serious attempt at live recording, as well as sound-on-sound recording techniques, requires headphone monitoring for proper balance and level adjustment.

A type of headset suitable for home listening is shown in Fig. 7-23. Each earpiece contains a 1-inch minature dynamic speaker and is filled with plastic foam for resonance damping. The front of the driver cone is also loaded with plastic foam, and its enclosed rear is vented to the interior of the earpiece through two small holes. These design features are intended to give a wide, smooth, frequency response with a minimum of peaks and holes.

The liquid-filled vinyl cushions mold themselves firmly but gently around the wearer's ear. External sounds are almost totally excluded, and it is said that the padded headband and soft ear cushions permit the phones to be worn for hours without discomfort or fatigue.

The 8-foot plastic-covered cord has four conductors so that the two earphones can be electrically isolated if desired. They are fitted with a
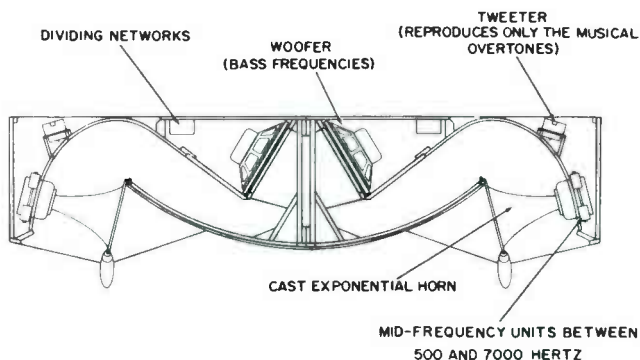
Fig. 7-22. Speaker arrangement in the enclosure of Fig. 7-21.

245

Fig. 7-23. Koss professional stereo phones.

three-circuit phone plug that fits the stereo phone jacks of most amplifiers.

The frequency response of the phones shown in Fig. 7-23 is rated at 30 to 20,000 Hz. While stereo headphones provide a very different listening impression than do speakers, earphones have a wide-range response, smooth and with low distortion. High-frequency hiss is somewhat noticeable. Fig. 7-24 shows a record changer and built-in amplifier that is specifically designed for headset listening.

Fig. 7-24. Shure SA-10 record changer and amplifier for headset stereo listinging.

CHOOSING A SPEAKER SYSTEM

We cannot be too specific with specifications for speaker systems, because their performance, or rather the judgment of it, must be in good part subjective. A number of system designs are available with excellent characteristics. Differences in the performance of these systems is such that the listener's taste rather than known measurable technical factors provides the basis of a choice. This is partly because the listener does not always conform to the ideal of wanting reproduction exactly like that which he would hear when present at the location of the perform-

ance. Various factors have conditioned him to tend to like a few deviations from realism, although constant experience with reproduction that is nearly perfect can educate one. There are a few people who just do not have a hearing system adequate to hear more than a limited degree of the audio range and who, either physically or by training, cannot distinguish between the performance of a poor radio receiver and that a well-designed high-fidelity system. Then there are all the gradations in between. Keeping these facts in mind, we can note the following points in the selection of a high-fidelity speaker system.

1. There is no set formula for high quality, for reasons already explained. Each type of well-known systems has its enthusiastic supporters, but the best audio engineers recognize both the strength and weakness of each system. Although they may favor one over the others for themselves, they realize that the type of listener and his situation influence things greatly. You cannot find the best system offhand; you must figure it out for yourself.

2. You cannot separate the choice of the speaker from the choice of the system as a whole. When you first decide you want high fidelity, you must also decide the optimum compromise between your ear and the cost. Optimum high fidelity costs money. Speakers cost the most of any part of it. The speaker system is thus very likely where the limit of cost will enter the choice. The first thing to do is determine the highest quality which really means something to you. The only way to do this to shop around. Visit the stores where systems are sold, and, even better, get acquainted with folks who have high-fidelity systems of various degrees of quality and different types of arrangements, and see which ones you like. More important, determine where you stop appreciating increments of increase in quality and cost.

A vast majority of people are very happy with a system having response only to about 6 or 8 kHz, providing other distortions are low. A limit can almost be defined in terms of dollars because performance generally follows cost to some degree, assuming that parts of the system are properly matched. Once you have made your choice of quality level, the specifications of your whole system including input devices, preamplifier, as well as the speaker system, should be chosen to match. This is important because two of the easiest ways to waste money are: (1) to have major parts of a system not matched in performance and cost requirements, and (2) to pay the high cost of quality increments above your natural taste requirements.

3. Frequency range limitation is not as annoying as other forms of distortion. When we speak of the response of a component or system, we assume (and thus probably take too much for granted) that the harmonic and intermodulation distortion over the entire range is low. It should be kept in mind that a limited frequency range in itself minimizes

harmonic distortion because it cuts off components of harmonic frequencies above the range. Listener tests in general have definitely shown that a good, clean, low-distortion frequency response up to 4000 Hz is much preferable to a frequency response from 10,000 or 15,000 Hz when only slightly greater harmonic distortion is present. The prospective purchaser of a speaker system should therefore not place too much emphasis on nominal claims of frequency response but should assure himself as to how low all forms of distortion remain over the entire range.

4. The listener's ear is the final test. All specifications, charts, and data should be considered in their relationship to the sound to the listener's ear, rather than to any technical tests; however, data from a valid technical test may be criteria as to whether the listener may some day acquire a more discriminating taste.

5. Although there is much to be said for the wide range of some of the single-cone, extended-range systems, the greatest range systems are generally those of the multiple type. The purchase of a multiple system does not in itself guarantee high fidelity. Such things as divided presence (consciousness of dual source locations), takeover, and fuzzy highs are common complaints. Many feel that a dual system is worthwhile in a higher-priced system but that the single-cone type is appropriate for lower-cost systems.

6. In separate-unit multiple systems, watch the phasing. The path length between the listener's ear and the source should be the same for both woofer and tweeter. If this requirement is not adhered to, the high- and low-frequency components join in the wrong phase relationship at many frequencies, and substantial distortion can result.

SPEAKER LISTENING TESTS

As has been explained, the final test of a speaker system is how it sounds. The listener who does not know what to look for in such tests is at a disadvantage. He may overlook factors which later, after he has purchased the system, may assume new importance. Experience has shown that proper attention to certain features will go a long way toward assuring continued satisfaction.

There are no standards of performance. Most prospective purchasers of speaker systems do not have equipment to make the tests necessary to check a system against proper standards anyway. So listener tests must be made as personal comparisons between different available systems.

The following suggestions should tell the reader what to look for in making listener tests on a speaker system. Each feature should be compared from one system to the next in an attempt to evaluate the one most pleasing. In any tests, every effort should be made to eliminate other components of the whole high-fidelity equipment by using the same or similar auxiliary equipment with each different speaker system. If the

reader has not had experience in distinguishing between audio-frequency tones of different frequencies, it is suggested that he obtain an audio signal generator or test record made for this purpose, connect it to a speaker, and run through the audio range, familiarizing himself with the sound of audio tones of low, middle-range, and high frequencies. He will then be better prepared for the listening tests which follow.

1. Look for distinctions between various kinds of bass notes, such as those from different musical instruments. If a bass drum, plucked strings, or bass horn all sound alike, then there is probably excessive bass resonance effect in the speaker or the enclosure.

2. Compare performance on voice with performance on music. Voices should be crisp, natural, and highly intelligible with the same amplifier adjustments used for music from the same source.

3. Listen for sudden crescendos of percussion instruments and to pizzicato passages of music. Check for evidences of hangover transient distortion which would make it difficult to distinguish one pluck from another.

4. If possible, apply to the system a high-frequency tone of, let us say, about 10,000 Hz. If this is not possible, listen to needle scratching or interstation noise from an f-m tuner. Then walk around in front of the reproducer, checking relative output directivity. The broader the output distribution at high frequency, the better is the system.

5. Tap the sides of the enclosure with your knuckles and note whether they are deadened properly or respond with undesirable vibration. All parts of an enclosure structure should be solid and dead.

6. Be sure that, for each speaker system checked, the amplifier boost controls are adjusted for the most pleasing performance. Also check speaker balance control (if any) for best balance between woofer and tweeter output. This control is sometimes referred to as the *treble* adjustment of the speaker system. It is unfair to compare systems in which all controls are not adjusted for optimum performance.

Systems Design, Selection, and Installation

The assembly of a number of components to create a complete sound system is not difficult if some general principles of system design are observed. The major units which make up a sound system are record players, tuners, tape equipment, microphones, television, preamplifiers, amplifiers, speakers, enclosures, baffles, horns, and other acoustical aids. Other parts, such as switching circuits, pads, microphone and speaker stands, cabling, and the like are also to be planned. In order to assemble a sound system which will perform in a desired manner, it is necessary to be familiar with all the components used, and what is required of each of them.

REQUIREMENTS

The most important physical and electrical design characteristics of a complete sound system involve size, power, fidelity, gain, compensation, filtering, number and type of controls, number and type of input (program) sources, and number and type of speakers. The design objective and limiting factors are elements which determine these characteristics. Each must be carefully considered before the components of a sound system can be specified. All of the components which make up a complete system must be carefully coordinated if the system is to operate properly. No component should be chosen without regard to the other parts of the system with which it is to be used.

AUDIO POWER

Before the components of a system can be selected, the power necessary to supply the required sound volume must be determined. The absolute unit of power measurement as applied to physical sound, is the *acoustic watt*. The acoustic watt is a unit of power of the physical activity of air at sound frequencies. It is not convenient to think in terms of the acoustic watt when designing a sound system, because the actual requirements are related to too many variables and the net results are extremely difficult to measure. The *audio watt* is more convenient to work with. The audio watt referred to in this respect is the unit of electrical power

at audio frequencies as measured at the output terminal of an amplifier. The power requirement of a specific installation can be expressed in terms of the number of watts of audio power required from the amplifier to be used if the efficiency of the speaker system is taken into consideration.
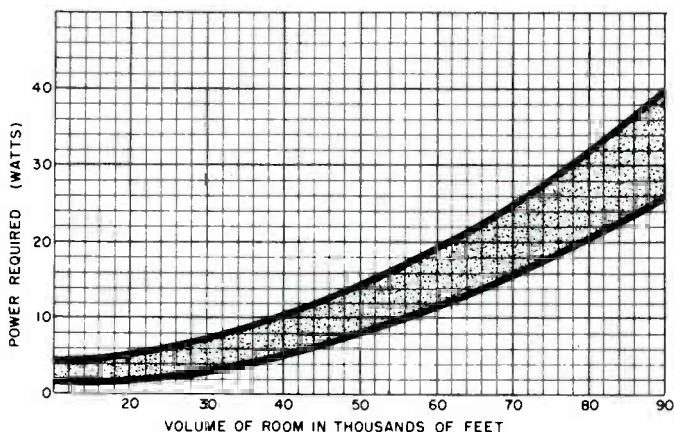
Fig. 8-1. Audio power required for various room sizes.

The efficiency of good speakers used with sound systems runs from 5 to 15 percent. Assuming that the speakers used have an average efficiency of 10 percent, it is possible to estimate the power requirements of a particular installation in terms of watts of audio power.

Fig. 8-1 is a chart that can be used for rough estimating, showing the approximate power required for adequate coverage of rooms of various sizes. The size of a room is based on its volume in cubic feet.

The shaded area indicates the approximate limits within which the power requirements fall. The range of selection is given to provide for various possible uses of the system. The wider the range of reproduction required, the greater is the power required.

Determining Power Required

The exact audio power required from the amplifier used with an indoor installation is governed by several factors. The most important of these are: the cubic volume of the room or rooms to be served; the acoustic characteristics of the walls, ceiling, and floor; the average noise level prevailing; the frequency range of the system and of the material to be reproduced; and the efficiency of the speaker system.

The chart of Fig. 8-2 can be used to estimate detailed power requirements, giving the amplifier power capacity required for rooms of various volumes. Curve A is for speech reinforcement systems using high-effi-

ciency horn-type speakers when the prevailing noise level is low. If the noise level is high, curve B applies, providing horn-type speakers are used. Curve B also applies when cone speakers are used and the noise level is low. When the noise level is high and cone speakers are used, curve C applies. Curve C should also be used for the average music reproducing system when the noise level is low. For very high quality, wide-range reproduction, curve D should be used.
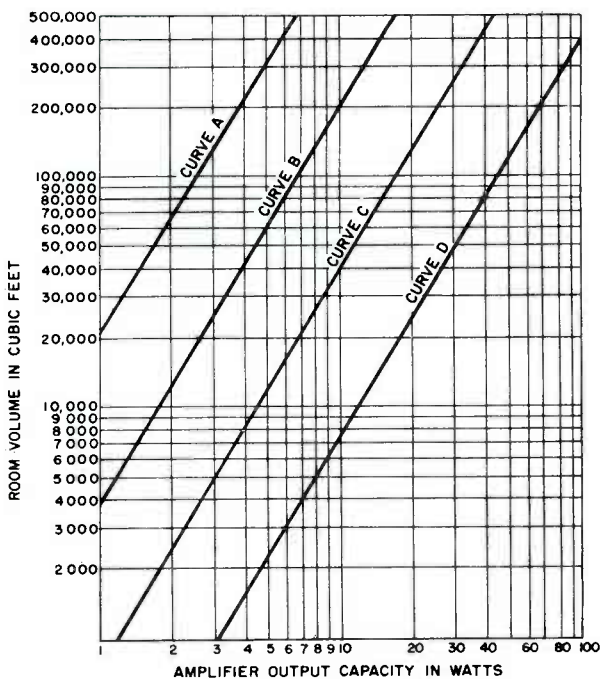
Fig. 8-2. Amplifier output required for different enclosures under various conditions.

The curve of Fig. 8-3 gives the amplifier power capacity for outdoor reproduction. The power required for outdoor installation depends on the distance between the speakers and the farthest point to be covered. Curve A is for a system using a single horn-type speaker covering an angle of 30 degrees. Curve B is for a system covering an angle of 60 degrees.

In multiple-room installations, i.e., installations where several areas are to be served, it is necessary to determine the total power requirements of all the locations which are to be equipped with speakers, and then arrange the distribution of power to the various speakers in accordance with the requirements of its individual location.

252
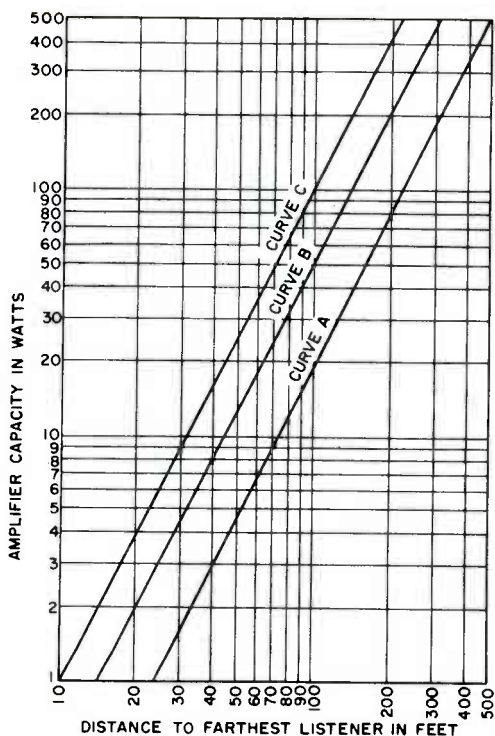
Fig. 8-3. Volume requirements of outdoor reproduction.

For top-quality installations there are two reasons why it is advisable to select an amplifier with higher wattage rating than is actually required. One reason is that the higher-power-rated amplifiers with the same ratings in regard to distortion and frequency response as the lower-power amplifiers will provide better operating characteristics at lower-power operation than the lower-rated units. The other reason is that greater dynamic range is available to handle extra loud crescendos, as may occasionally drive a smaller amplifier beyond its rated output, thereby causing noticeable distortion.

FIDELITY

The term "fidelity" includes a number of considerations. By definition, fidelity is the degree with which a system accurately reproduces without distortion at its output the essential characteristics of the signal that is impressed on its input. The fidelity requirements of a system depend largely on the type of material which is to be produced and on the listener's taste.

Frequency response is an important characteristic which helps to de-

termine the fidelity of a system. Frequency response is a rating which indicates the range over which a system reproduces all frequencies uniformly; thus, a system may be said to have a flat response between 30 and 20,000 Hz, meaning that a curve of its output plotted against frequency is flat from 30 to 20,000 Hz. The term "flat" is generally understood to mean a deviation of less than ± 0.5 db, which is not noticeable to the average human hearing system.

Although human hearing is limited to about 20,000 Hz at best, modern equipment is often rated up to 100,000 Hz. The implication and fact is that if an amplifier has X distortion at higher than audible range, it should be able to produce X minus a reasonable factor or less distortion at audible frequencies. The same principle is applied as that applied to the linear operation of class-A amplifiers. The smaller the operating portion used of the overall response curve, the flatter the operation will be.

FREQUENCY REQUIREMENTS

For systems used predominantly for music, the range of equipment should extend from at least 30 to 15,000 Hz. When a system is used for a direct pickup, and optimum fidelity is desired, a frequency response of from 35 to 20,000 Hz is desirable. This means that if the sound to be reproduced is being picked up directly from an orchestra or a singer, a wide range should be used. If, however, the source of sound is from recordings or electrical transcriptions, there is no necessity for an extremely wide range. A frequency response of from 30 to 15,000 Hz is adequate for such a system. This takes into consideration the average ear and all available components in the hi-fi chain.

Both the upper and lower limits of the frequency response of a system are important. It has been conclusively determined through subjective tests that the upper and lower limits are related, and that when the upper limit is raised, the lower limit should be lowered. It is a fairly well established rule of thumb among audio engineers that the product of the upper and lower frequency limits should be about 640,000 Hz^2.

This means that if the upper limit is 10,000 Hz, the lower limit should be 640,000 $\div$ 10,000, or 64 Hz. When the upper limit is 20,000 Hz, the lower limit should be around 32 Hz. The center point of the audible sound range is usually regarded as 800 Hz ($800^2 = 640,000$).

For best results, the response of a system should be the same number of octaves above 800 Hz as it is below. (An octave is the difference between two tones whose frequencies are related by a factor of 2. For example, one octave below 800 is $800 \div 2$, or 400 Hz. One octave above 800 Hz is 1600 Hz.)

The wider the frequency response of a sound reinforcement system, the more important it is to keep distortion content to the lowest possible percentage. Distortion can completely negate the advantage of a wide frequency response characteristic.

254

If a sound installation is to be used for a-m radio reproduction or if other limted-response programs are to be used that are confined to frequencies between 100 and 8000 Hz, designing a system whose upper limit is 8000 Hz and whose lower limit is 100 Hz will reduce to a considerable extent the amount of power and response requirements without reducing the efficiency of the system. When it has been determined that a system is to be used for limited response only and reproduction of highest-fidelity components is not considered worth maximum investment, the system components should be selected so as to have a narrow frequency response range throughout. This will save considerable money, and modern limited-range equipment will still sound good.

DISTORTION

Fidelity also includes the distortion characteristics of a system. For systems of the highest quality, the total cumulative distortion as fed to the speaker should not exceed 1 percent at normal operating level.

Distortion refers to the presence of components in the output which were not in the input, but were generated in the system itself, and which may be in the form of changes in waveshape or new components harmonically related to the input frequencies. For a detailed description of distortion see Chapter 1. If the equipment were to be used for speech reinforcement only, distortion content up to 3 percent might not be objectionable, and the highest frequency needed to be reproduced would be limited to approximately 7000 Hz. Overall system harmonic distortion becomes increasingly unpleasant to the ears as the upper frequency limit is raised.

GAIN

The overall gain of the amplifier parts of a system is important and must be carefully considered. The total gain of the amplifiers must be sufficient to drive the final stages to full output with the lowest-level input to be used. Gain is measured in *decibels*, or *db*. A decibel is an expression of a ratio of power or a ratio of voltage; thus, if we know the input signal voltage (E_i) of a preamplifier and the output voltage (E_o) of the same preamplifier, the overall gain requirements of this unit may be determined by using the following expression:

$$\text{Gain (in db)} = 20 \log\left(\frac{E_o}{E_i}\right) + 10 \log\left(\frac{Z_{\text{in}}}{Z_{\text{out}}}\right)$$

where Z_{in} and Z_{out} are the input and output impedances respectively.

If the input power (P_i) and the output power (P_o) are known, the gain may be determined by the following formula:

$$\text{Gain (in db)} = 10 \log\left(\frac{P_o}{P_i}\right)$$

As a matter of convenience, specifications of hi-fi preamplifiers and amplifiers are usually given in overall ratings in a more practical manner. Specifications for basic amplifiers are described as a certain voltage input across a definite input load impedance necessary to produce full output according to power-output rating.

Standard input requirements for basic amplifiers may vary from ½ volt to 3 volts of input across input load impedances varying from 100 to 500 kilohms.

Voltage input specifications must always be related to the load. A preamplifier or tuner rated to deliver 1 volt across 100,000 ohms will not drive a basic amplifier requiring 1 volt across 250,000 ohms to full output without more than rated preamplifier distortion. In some cases, input load condtions of basic amplifiers are not given and ample drive should be provided to cover all contingencies. Also, basic amplifiers should have less than 2500-pf input capacitance to avoid distortion in the coupling network.

Preamplifiers are usually rated in db of gain or input voltage level for full output. Fifty to sixty db gain will usually provide sufficient output (5 volts or more) across standard input impedances from the lowest-level output program sources, such as dynamic phono cartridges or microphones. Outputs from these sources may be as low as 5 millivolts across 170 ohms. Input devices having lower outputs than this fall out of the practical application range as noise becomes too great a factor.

If all units are rated in db, decibels may be added or subtracted; an amplifier always adds a certain number of db, and an attenuator subtracts a certain number of db. Thus, if a preamplifier with a gain of 50 db is used with a power amplifier with a gain of 60 db, the total gain will be 50 + 60, or 110 db.

"Dbm" is a term used to indicate the volume level of constant tones. It means that the level, or sound volume, of a constant tone is the specified number of db above 1 milliwatt. These units are used in specifications of professional equipment and apply only to microphone levels as far as high fidelity is concerned.

VOLUME UNIT

The *volume unit* is another term commonly encountered in use of pickup units. The volume unit is similar to the dbm in that it is used to indicate the level of a signal in db above or below 1 milliwatt. The volume unit indicates that the measurement was made on average program material rather than on a constant tone as does the dbm. The volume unit is abbreviated "vu."

After the power output which will be required from the amplifier has been determined, and the microphones and other input sources have been selected, the gain which will be required from the amplifier can be determined. The first step in determining the gain is to find the volume

level in vu to which the power output of the amplifier corresponds. The curve in Fig. 8-4 gives a number of typical values of output power and the volume levels in vu (db above 1 milliwatt) to which they correspond.

It is then necessary to determine the output level of the lowest-level microphone to be used. Since microphones are rated in a number of different ways, their ratings should be converted to db below 1 mw/1 bar, as described previously. The output level of microphones is usually a number of db below 1 mw. Therefore, the output level must be added to the power output of the amplifier (in vu) to obtain the gain in db required from the amplifier.
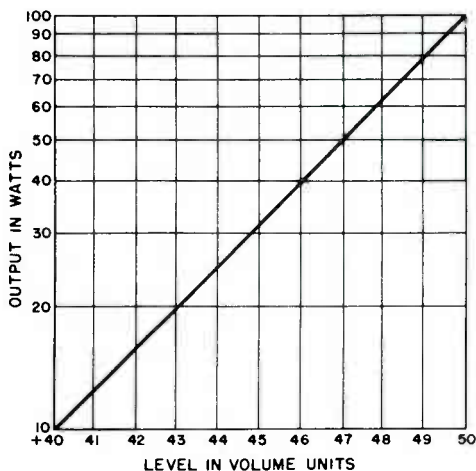
Fig. 8-4. Power versus volume units (db above 1 milliwatt).

The following is an example of the procedure followed: Assume that the power required from the amplifier is 15 watts, which is equivalent to + 42 vu (see Fig. 8-4). If the microphone to be used with the amplifier has an output level of 53 db below 1 mw, the gain required from the amplifier in this installation would be 42 + 53, or 95 db.

In practice, it is usually wise to add a safety factor of about 10 percent. In the preceding case, it would bring the gain required to 105 db. If an amplifier is to be used with a number of input sources, the gain required for use with each microphone should be determined.

If high-fidelity recording of one's own program material is desired, gain becomes an important factor. Since the average basic amplifier provides only a 40- to 75-db gain, a preamplifier to cover all conditions of input and output combinations should provide for a distortionless gain of 70 db or more. Many preamplifiers are only rated at 50 to 60 db gain; so if hi-fi recording is a must, one must either get a basic amplifier

257

with sufficient gain to add to the preamplifier gain to get a total of more than 100, or one must get a preamplifier with more gain.

These high-level gains are only required for high-fidelity microphone pickup. Ordinary magnetic or crystal pickups for phono or microphone applications require less than 100 db total gain for up to 60 watts full output. Under all conditions, 100 db total gain is recommended for average applications, barring the lowest-output hi-fi microphones, because it is easier and therefore less costly to produce high-quality low-gain amplifiers and preamplifiers.

BUILDING UP A HIGH-FIDELITY SYSTEM

Some of us may be fortunate enough to be able to go "all out" for high fidelity with our first system, but most of us cannot. In fact, the reader should not feel discouraged if his budget is limited, because he will probably learn more and derive greater enjoyment from building up his system in stages, starting with something relatively modest. Let's consider now some typical systems and also how an elaborate system can be evolved from a simpler one.

First, consider the simple breakdown illustrated by the block diagram of Fig. 8-5. It's convenient to think of a system as broken down into its three main parts: (1) a program source device, such as a record player or tuner, (2) amplifier and controls, and (3) speaker system. This arrangement is a minimum complement for either monophonic or stereophonic reproduction. The differences in stereo are only that provision must be made for a minimum of two channels and their related controls in the amplifier section, and that two speaker systems must be used. In addition, the source material must provide the two required stereo signal outputs.

If you're starting from scratch, you should keep in mind that you'll eventually want a stereo system. No modern high-fidelity system can be complete without stereo. However, we must consider here the possibility that you have a monophonic system now that is "left over" from pre-stereo days, or that you are starting from scratch and want to keep cost down in the first stage by limiting it to mono. However, in either case, with a little careful planning, the monophonic system can be integrated into a stereo system later.

There is one part of the high-fidelity system in which you can save by starting with mono, with no compromise in future use for stereo: that is the speaker system. A good wide-range speaker system will cost you from $50 to $750, and of course you can go a lot higher. This means

Fig. 8-5. Three main parts of a high-fidelity system.

that you need to double this cost with stereo. If the rest of your system is designed for stereo, you can buy one speaker system first and operate with monophonic reproduction until you can afford the other speaker.

Actually, the cost of amplifiers, changers, and tuners is not much greater for stereo than for mono, if you insist on good high-fidelity reproduction. For example, a good stereo record player can be bought for approximately $75, and buying a record player of equal quality for monophonic reproduction won't save you more than $20. A good stereo amplifier, with two complete channels and all the controls you need, is not more than 30 percent higher in cost than a similar monophonic type.

At this point, individual needs must be analyzed—balanced design applied in relation to price and quality. The greatest attention should be given to the speaker. Fifty or one-hundred dollars more invested in this unit will give many times more return than the same amount spent in the amplifier. The next important unit where most return will be had per extra dollar spent is in the record player. Even a few dollars may make a great difference in this unit. Amplifiers are now so well developed that even the lower-cost combined units perform well. There is little noticeable difference in amplifiers until you double or triple the investment, and then the difference is small, compared with the differences between the lower- and higher-cost speakers.

Enclosures cannot be selected from a technical standpoint alone as they must be selected to suit the overall room layout and associated furnishing of a room. In relation to investment, it is again recommended that emphasis should be placed on the speaker rather than the enclosure. A good speaker will sound good in a poor enclosure, but a poor speaker will still sound poor in a good enclosure. Furthermore, a good speaker will sound good with any reasonable baffling arrangement.

Fig. 8-5 shows a minimum-complexity arrangement which can still produce continuous-program hi-fi reproduction of excellent quality. With a record changer as the program source, this arrangement will be entirely suitable for playing medium-fidelity records and also give excellent performance on hi-fi records. Substituting a tuner (either a-m, f-m, or both) provides us with another starter arrangement, designed for radio reception. If the tuner has high output, and if volume and compensation controls are provided in the tuner, only a basic amplifier need be provided. If the tuner does not possess a high output, then the pre-amplifier-amplifier unit will be required for equalization and volume control.

A good starter arrangement is given in the block diagram of Fig. 8-6A. When you purchase the speaker, you should keep in mind that you'll want to add another speaker later, to provide for stereo reproduction. Therefore, the first speaker system might be something in a small enclosure, so that when the other speaker system is added the total space taken up will not be excessive.

The next step in the evolution is to add the second speaker system, as illustrated in the middle block diagram (Fig. 8-6B). Now, with this one addition, you have stereo performance, using the stereo record player and amplifier.

The next logical addition is a tuner, so you can hear stereo broadcasts (Fig. 8-6C). Your stereo amplifier should have inputs for both the record player and tuner, and a selector to choose either. You are now ready to play either records or listen to stereo broadcasts.

Perhaps you prefer a tape machine to either the record player or tuner; it can be purchased in place of either one, or in addition to them.

The bottom diagram (Fig. 8-6C) exemplifies a fairly complete stereo system, but you can still build from there. Fig. 8-7 illustrates a more advanced system. Here, the preamplifier is separate and is used with a basic amplifier unit. Record player, tuner, and tape input devices are used. A third, "middle" speaker has been added, to minimize "hole-in-the-middle" effects. The same third-speaker circuit can be used to supply remote speakers. However, it should be noted that going from an integrated amplifier-control unit to a preamplifier and basic amplifier is

(A) Starting arrangement.
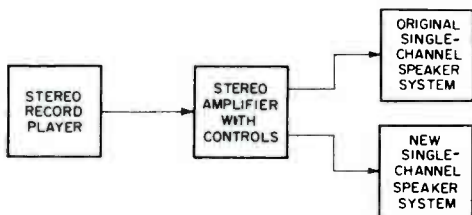
(B) Adding second speaker.
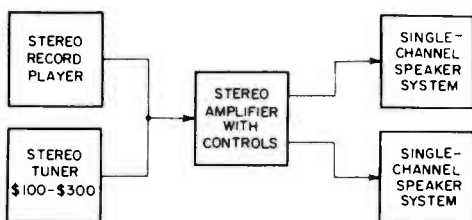
(C) Adding tuner.

Fig. 8-6. Starting with a modest system and gradually adding to it.

a major change in system design because many of the control facilities of the original complete amplifier are duplicated by the preamplifier. The original integrated amplifier could still be used, but it might be economically desirable to replace it with a basic amplifier. The separate preamplifier will normally provide much greater flexibility than the original integrated amplifier.
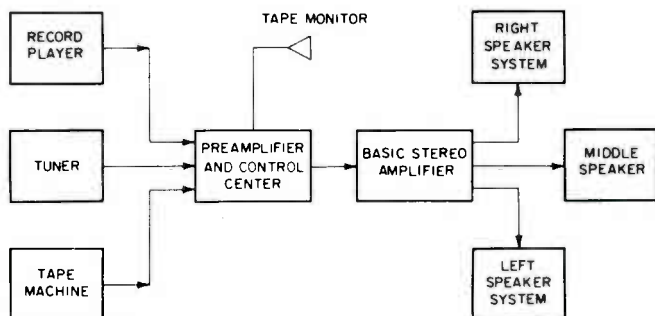
Fig. 8-7. An advanced stereo system.

This system does not provide for control of the center channel because its origin in this arrangement is at the final output point of the other two channels. Control of the center channel is essential to true stereo effect. An arrangement that provides complete control of center-channel blending and volume is shown in Fig. 8-8. This particular arrangement utilizes a combination stereo preamplifier and power amplifier, with center-channel mixing controls and a low-level center-channel output. The center-
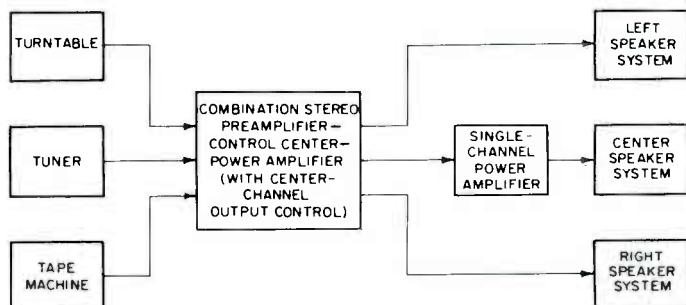
Fig. 8-8. Three-speaker output with controlled middle channel.

channel output is fed to a separate single-channel amplifier which drives the center speaker. An additional feature of this arrangement is that an old single-channel amplifier may be utilized for the center channel; thus the stereo effect can be improved at low cost. The two-channel stereo power output is fed to the left and right speakers in the same manner as shown in previous arrangements.

261

There are numerous stereo-preamplifier control-center and preamplifier power-amplifier control-center combinations providing a low-level third-channel output that may be used in this arrangement.

ELABORATE ARRANGEMENTS

A complete hi-fi system that does most everything one would ordinarily want with highest-quality reproduction could have many units. A fairly complete arrangement is shown in Fig. 8-9. It is expected that the speaker system, basic amplifier, control, and other units would be top quality, perhaps costing as follows:

(a)	Main speaker system	$250 to $700
(b)	Auxiliary speaker system (2)	$100 to $300
(c)	Speaker switcher and controls	$20
(d)	Basic amplifiers	$100 to $200
(e)	Control center	$100 to $150
(f)	Record changer with reluctance cartridge and diamond stylus	$85
(g)	A-m/f-m tuner	$100 to $150
(h)	Turntable with dynamic cartridge and diamond. Hysteresis motor	$150 to $200
(i)	Tape recorder	$200 to $400
(j)	Television chassis	$150
(k)	Mike	$50

One can spend several thousand dollars for top-quality components such as the preceding. In addition to these there are such items available as: automatic time (on and off) switching controls, disc-recording equipment, intercom, dual systems for binaural operation, telephone pickups, and others.

SYSTEMS LAYOUT

There are aspects of high-fidelity systems design that may be related to all the members of a family. Sometimes it is better to sacrifice highest fidelity of sound reproduction to gain better family relationship. A compromise in placement of the units may bring better harmony everywhere both in music and family relations. For instance, one never knows where a wall speaker will sound best until the unit is tried in the wall, but once the hole is made in the wall it is difficult to move.

Problems in physical layouts of hi-fi equipment break down into three groups:

1. Speaker placement
2. Operations and control position
3. Units that can be concealed
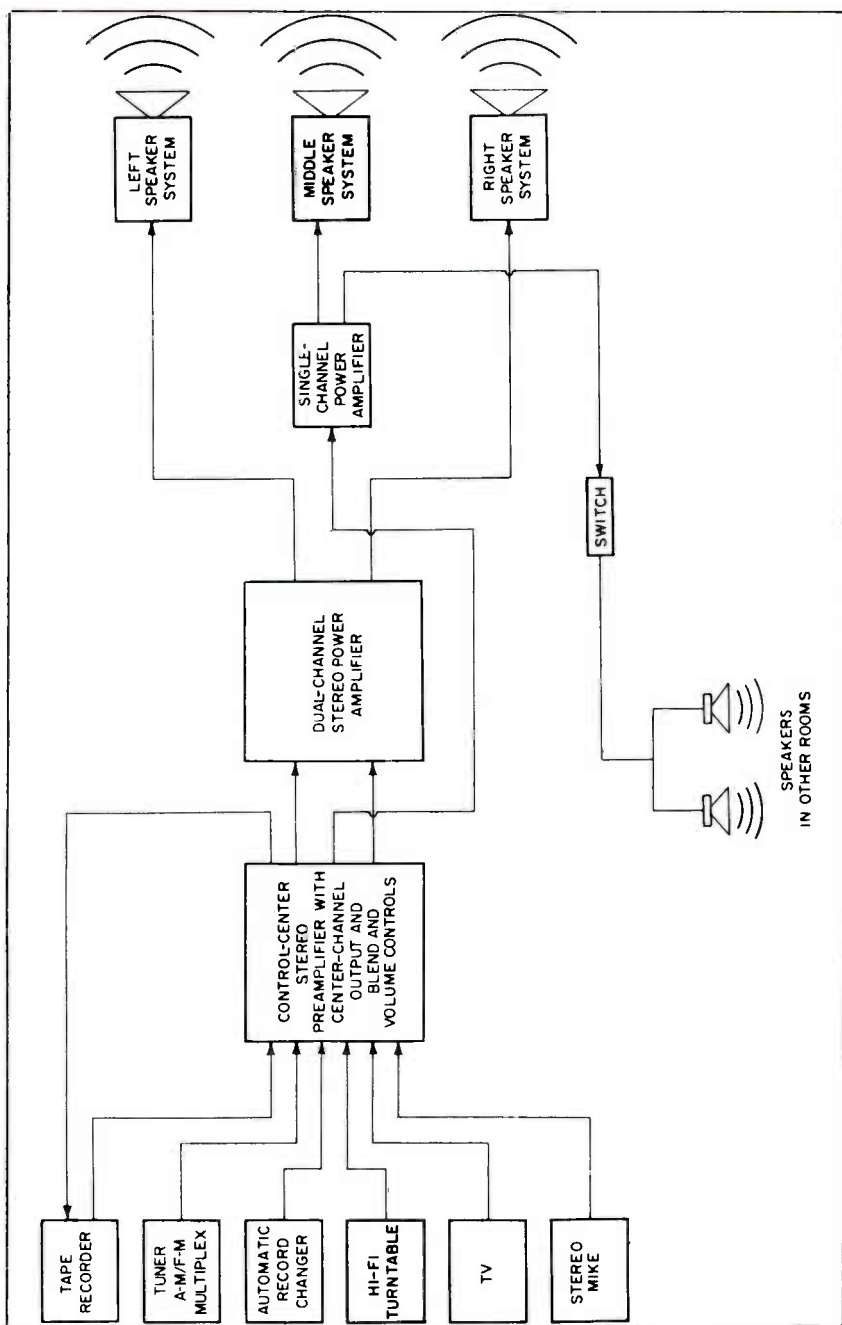
Fig. 8-9. An elaborate high-fidelity system.

The last is no problem anywhere. The first two problems may be solved in a combined manner or separately. The controlling factor is the room acoustics, and so we will begin with speaker placement.

Speaker Placement in Single-Channel Systems

The placement of the speakers is a very important problem, because proper speaker positioning is necessary in order to obtain adequate distribution of the highs and at the same time make adequate provision for bass reinforcement. If microphones are involved, improper speaker placement may cause feedback. Added to this is the problem of minimizing interference between speakers.

There are no rules which can be applied to all installations, but there are a number of points which should be observed when selecting speaker locations.

Speakers should never be placed at two ends of a room. As a rule, they can be mounted on one wall or in one corner of a room. Most of the sound heard by a listener should come from one point or from speakers which are equidistant from the listener.

A number of typical examples will serve to point out the procedure used in locating speakers. Fig. 8-10 shows two methods of mounting a speaker in a square room. In Fig. 8-10A, the speaker is mounted in one corner of the room. This gives the best distribution of sound when a single speaker is used. As previously described, the maximum high-frequency distribution can be expected only over 90 degrees; consequently, this is the only method of obtaining full coverage. In addition, when a corner folded horn is used, the walls serve to act as a part of the system and additional advantage of the corner application is gained. To give the proper assist to the folded corner horn, there should be wall lengths of at least four feet beyond the enclosure. In Fig. 8-10B, the speaker is mounted on one wall (or in it), giving less than full coverage of the highs. This same arrangement, as shown in Fig. 8-10C, for rectangular rooms, will give better distribution because of reduction of reflection.

Fig. 8-11 shows the right and wrong ways to position speakers in an L-shaped room. Fig. 8-11A shows the wrong way, in which a speaker is mounted at the end of each leg of the L. In Fig. 8-11B, the correct
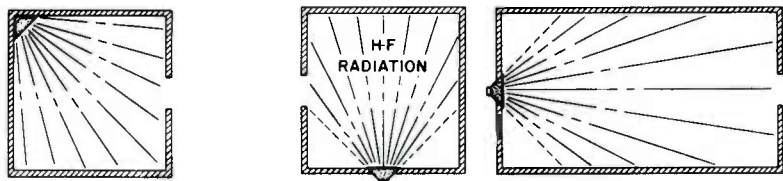
(A) *Speaker in corner of room.*   (B) *Speaker in only one wall.*   (C) *Placement for rectangular room.*

Fig. 8-10. Speaker placement in square rooms.

(A) Wrong way.   (B) Right way.

Fig. 8-11. Speaker placement in L-shaped room.

position is shown. The speaker system is arranged at the junction of the legs of the L, bisecting the angle to obtain even distribution of sound.

Fig. 8-12 shows a two-room installation with adjoining rooms. A single speaker is mounted in each room. Located in this way, the speaker lines are kept quite short, and if a listener is able to hear lows originating from two speakers, the speakers will be almost equidistant from him. This condition will exist near the openings between rooms.

Using an inside wall as an infinite baffle arrangement may turn out in some instances to be an excellent low-cost arrangement, but there are a lot of factors involved that may reduce the effectiveness. In an average home, a speaker may be mounted in a wall between two rooms or between a room and a large closet. In any case, the baffle room must be in proper acoustic relation to the speaker to act as a first-class baffle. Room resonances in either room can be reduced by drapes on opposite walls, but this may not suit the decor. Severe reflections and other interference from the baffling room back to the speaker will cause distortion. A baffling room or closet must be acoustically treated to be effective. Chances are great that the acoustic nature of the room requires a big compromise from optimum location of the speaker to get it placed in a possible spot where infinite-baffling technique can be used to best advantage. Then the compromise may be such that the distribution of the highs only covers half or less of the room. Experience has shown that cutting up a house
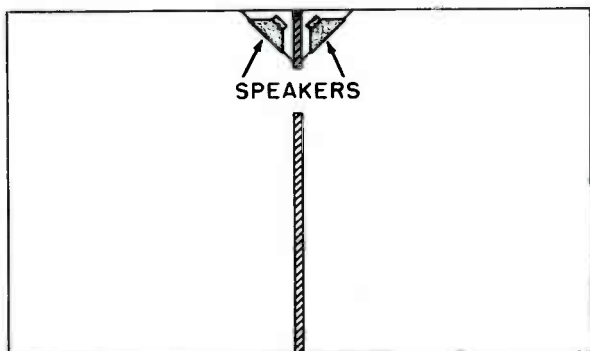
Fig. 8-12. Locating speakers in adjoining rooms.

265

to provide for infinite baffling as a practical matter is not advisable on either a short or a long-term basis. However, excellent results can be obtained if you persist and have a little luck.

In almost all indoor installations, the speakers are mounted from a normal sitting ear-level on up to near the ceiling. This allows a clear path for the sound to listener in any part of the room. Floors and ceilings are not recommended for speaker mounting as the effects seem unnatural.

In outdoor installations, the speakers should be mounted 10 feet or more above the ground. All speakers should be located at the same point. This point does not necessarily have to be at the point of direct pickup, since in many outdoor installations sound from the point of pickup is a negligible factor.

Speaker Placement in Stereo Systems

As in the case of monophonic reproduction, the most critical link in the reproducing chain is the speaker system. The requirements for stereo are even more rigid because proper speaker placement and balance are important for production of the stereo effect.

The use of all of the principles discussed in Chapters 6 and 7 for speakers and enclosures is more important in stereo systems than mono systems. The reader is therefore referred to these chapters for fundamentals which concern the design of each individual speaker system. In stereo we are primarily concerned with the use of at least two of these systems and how they are coordinated for best enjoyment of stereo effect.

The exact requirements of speaker placement, relative frequency response, and relative power-handling capabilities of the speaker systems to be used for stereo are matters on which even the most respected authorities differ. This is probably because the placement of the microphones at the source, the relative size of the source (orchestra, chorus, etc.), and the nature of the sounds involved differ widely in the different programs to be reproduced. One set of conditions cannot be optimum for all of them. So it is not possible to give positive single answers to most of the questions which the reader will have about acoustic reproduction for stereo. We shall therefore concentrate on reviewing some of the different ideas that have been set forth on the subject, aware that the listener is best qualified to make the final choice for himself.

Matching Speaker Systems—It seems generally agreed that the ideal situation is two or three completely matched full-range reproducing systems. Some have suggested that "reasonably good" stereo reproduction can be had with one elaborate system of speakers (for center, right, or left) and one or two cheaper systems which can have limited frequency response (high) and perhaps a small degree of distortion. However, this suggestion is usually made in connection with a transition from a monophonic system to a stereo system, and this arrangement is not considered a good substitute for two top-notch speaker systems.

In some compromises, cognizance is taken of the fact that the stereo effect is obtainable only with the higher-frequency signal components. Because of the large wavelengths of the lower frequencies when compared to the spacing between speakers and the distance to the listener, the low-frequency sounds seem to come from a wide area. (For example, at 50 Hz, the wavelength is over 20 feet.) Also, differences in phase due to sound-source location differences are not noticeable at low frequencies and the stereo effect is lost (although presumably there is still an intensity difference). The fact that stereo effect is not as prominent at low frequencies has led to the idea of concentrating the low frequency components (which contain the most power) in one high-power woofer located in the center. The left and right speakers then need only reproduce the relatively low-power higher-frequency components. The left and right speakers should be carefully matched, but lower-priced speakers than usually employed in a full-range system can be used. The use of a middle speaker also minimizes the "hole-in-the-middle" effect.

The left and right speaker systems must also be balanced as to phase. If the connections to one of the voice coils are reversed, the sound from one speaker tends to cancel that from the other, leading to distortion and lowered output. Correct phasing can be recognized as that resulting from the connection that gives the greater output. Which polarity of each speaker is not important as long as both are phased the same way. Phase checking of speakers is discussed later in this chapter (see "Speaker Phasing," pp. 276-279).

Placement—The ideal speaker placement is the same as that of the microphones where the sound is picked up. Unfortunately, this cannot usually be arranged, because microphone spacing differs for different source material, and moving speakers around to match the program is not practical. Also, the layout and decor of the room must be considered, because, of course, the average home cannot be "designed around" the high-fidelity systems (although in isolated instances this has been done!). Probably the best general guide to placement is the triangle illustrated in Fig. 3-1. First determine approximately where the listeners will sit, then draw an angle of from 30 to 45 degrees between there and the wall where the speakers are to be located. The points where the two sides of the angle intersect the wall are the two approximate locations for the speakers.

As illustrated in Fig. 8-13, the speakers may be placed along either the end wall, the side wall, or in two corners (for which special "corner speakers" are available). As indicated by the shaded areas, the stereo effect is obtained over a much greater area if the speakers are placed at the end of a room. Only those areas within the coverage of both speakers can provide stereo effect. Notice also that it is not good to plan on being too close to the speakers (between them) because then, too, you will be out of the coverage areas.

Another thing to keep in mind is the fact that at the high frequencies, at which the stereo effect is the greatest, the radiation beam of the speakers is the narrowest. This is something to keep in mind especially when you are using single speakers at the left and right. Most tweeters in dual speakers are now designed to spread the high-frequency energy over most of the beam width of the middle-frequency range.
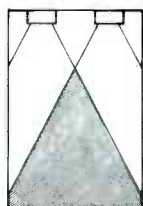
(A) At end of room.
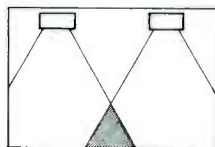
(B) Along long wall.
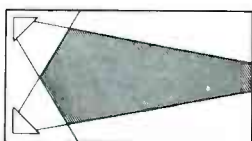
(C) In corners at end.

Fig. 8-13. Three ways of placing stereo speakers.

Where space is limited, and the speakers cannot be spaced far enough apart, various methods are employed to orient the speakers so they radiate as though they were further apart. In one such arrangement, the speakers are mounted at the ends of a rectangular cabinet and doors in the end are opened part way so they deflect the sound outward to the listener.

The Middle Speaker—As has been mentioned, many audiophiles believe in the use of a "middle" speaker, that is, a speaker located between the left and right speakers and reproducing a subdued combination of the right and left signals. The idea of this arrangement is to overcome what is known as hole-in-the-middle effect and provide a wide curtain of sound. Hole-in-the-middle effect arises when the stereo effect is so pronounced that the listener begins to distinguish two separate sources. This can be partially overcome by diluting the stereo effect with a blending control (previously discussed). However, a middle speaker is best for removing hole-in-the-middle effect without loss of stereo effect. Signals from both channels are coupled to the middle speaker, which is operated at a lower level than either the right or left speakers.

The optimum arrangement for providing center-speaker operation is by mixing (blending) the outputs of the preamplifier and feeding the combined signal to a third amplifier channel for the middle speaker, as
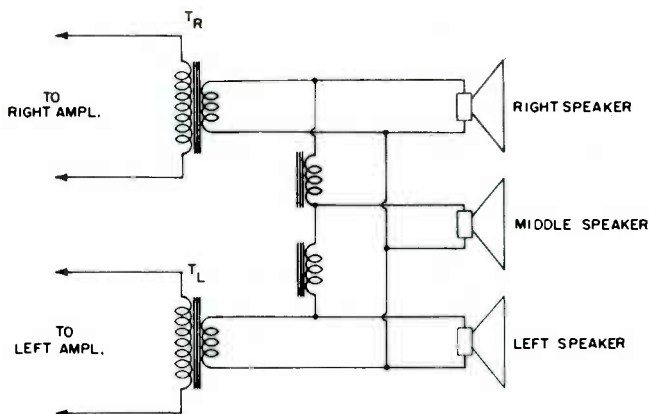
Fig. 8-14. Combining L and R outputs for a middle speaker.

shown in Figs. 8-8 and 8-9. The preamplifier outputs must be combined in such a way that they do not cross over and produce interaction in the left and right amplifiers. A separate blend volume control will provide optimum results.

A lower-cost way to provide a signal for the middle speaker by combining the signals at the amplifier outputs is given in Fig. 8-14.

The L and R signals are fed to the middle speaker through choke coil filters that are designed to pass low-frequency signal components and block the middle- and high-frequency components. However, the filter chokes are far from perfect filters and interaction between the two signals is fed back through them. A more efficient arrangement is given in Fig. 8-15. The middle speaker is of special design, having a dual voice coil;
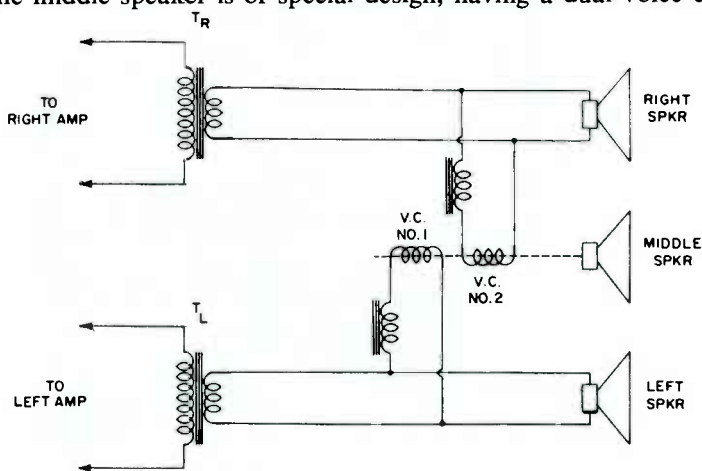
Fig. 8-15. Combining L and R outputs for a two–voice-coil middle speaker.

that is, a separate voice-coil winding is employed for each circuit. Although the filtering is similar, the voice coils are so small that the coupling between them is negligible, and separation between left and right circuits is good. Dual voice-coil woofers, such as shown in Fig. 6-33, are now made by nearly all speaker manufacturers.

Location of Program Sources, Amplifier, and Control Equipment

The units involved here may include everything in the system, or may include all units but the speaker system, or may include all units but the speaker system and the basic amplifier. A basic amplifier completely controlled from a preamplifier could be kept anywhere in the building as long as it has sufficient ventilation.

Wall Cabinets—Sectionalized wall cabinets of all types are available. These cabinets are usually of modern design with spaces to fit any conceivable arrangement, including television and speaker with a full-sized bass-reflex enclosure. One such arrangement is shown in Fig. 8-16. Many others are available. Provision can also be made for extras such as books and ornamentation if desired.

Record player, tuner, controls, and amplifier are all located near each other and the point of operation is at the nearest point to the listener that it can be without remote control. Speaker systems should be mounted at ear level or above and slightly off center of the room. The best position for such a system mounting is on the narrow wall nearer one corner. Make all connections short, shock mount the pickup units and preamplifier, and ground everything metal.

Courtesy Jensen Mfg. Co.

Fig. 8-16. A typical home stereo layout.

The same ideas can be used in novel home-modified break fronts and other odd furniture pieces for matching period decorated or early American homes. Uusually the speaker is best mounted separately to gain advantage of a properly designed enclosure.

Packaged units are commercially available at prices from $160 to $1800. The advantage of these is that one gets a complete set of matched components with no further engineering or construction required—just plug it in. The limitation is that one has no choice in variations of arrangement to suit his individual taste.

Units in Separate Furniture Pieces—Several pieces of furniture may be separately set up to house the various components that must be operated. Table equipment and consoles are available for individually holding record players, tuners, and amplifiers. It is often convenient to have the operating position of the player, tuner, and controls located at an accessible spot convenient to the listening area, to save steps. Here again custom installation in separate furniture pieces to match the decor can be beautiful and practical.

Wall Units—If mounting the pieces in furniture is not desired, it is modern to mount operation control units in the wall. Several units are made to suit this application with modern and antique panels. The type of unit most suitable for this kind of installation are tuners with all controls built-in to provide a control center—otherwise several holes in the wall may be necessary. This may make a very convenient and space-saving operation, but it has disadvantages. The equipment requires a good-sized aperture in the wall, possibly with equipment protruding through the other side and internal wiring. The equipment is generally required to be within 4 feet of the phono pickup. These problems may be easily worked out when constructing new houses, but in old houses they may present many difficulties.

A variety of equipment for wall mounting is available. One example is shown in Fig. 8-17. The advantage is that valuable room space is conserved because a substantial portion of the equipment is located in the wall. The record-player unit swings outward during use and remains closed into the wall like a door when not in use. Stereo speakers are mounted on another wall.

The equipment may in time become obsolete, and if it is removed, the new equipment may not fit in the same place. Holes in the wall are inconvenient to patch and new plaster causes decorating problems. With these cautions in mind, it is suggested that one get the best equipment available and be sure the installation is the one that will be desired permanently before proceeding.

Closet Arrangements—Only shelves and enough room to move around in are required in a closet. If an available closet is located conveniently near the principal listening area, one may save considerable money in furnishing and matching, and the equipment will be protected from

mechanical disturbance, prying fingers, and other problems. Certain types of home-built equipment or less expensive kits that "don't go" in the living area may be kept out of sight here. The wiring will be short and convenient and all units will be located in one spot for convenience of operation and control. While speakers may be mounted on the same closet doors, this partly defeats the purpose of unit isolation and the speaker system thus located does not compete with a good folded corner horn arrangement.

Combinations of all of the foregoing are permissible for any arrangement desired except that one should keep in mind that the phono lead to the preamplifier should not be over 4 feet long.

Courtesy Nu Tone, Inc.

Fig. 8-17. Wall-mounted stereo components.

SYSTEMS INSTALLATION

After the components of a sound system have been selected, and the position of all the components and speakers have been determined, the actual job of making the installation begins. The work of installation consists mainly of mounting the equipment, connecting the control center, the amplifier, and the record player to the source of power, and connecting the speakers and program source equipment to the amplifying

equipment through suitable lines and cables. There are a few problems peculiar to each hi-fi installation which must be individually solved at the time the installation is being made.

Installing Program Source Equipment

When leads from record players, tuners, tape equipment, and mikes are installed, care must be taken to avoid hum pickup, cross talk, and losses in frequency response and level.

High-impedance circuits are very suspectible to hum pickup. Correct grounding of the turntable and pickup arm is exceedingly important in preventing pickup of hum. Connections carelessly made can negate all other precautions in design of equipment. The turntable frame and pickup should be connected by a flexible wire from the turntable frame to the pickup-arm base to the shield of the lead to the preamplifier. Be careful to avoid ground loops in between all equipment. Interunit cables provide a complete ground system. Alternate ground wires create ground loops which will usually increase hum level.

Keep the magnetic phono pickup or microphones more than 2 feet away from power lines and transformers to prevent induction of hum.

When long lines are used with high-impedance sources, with the exception of the crystal type, loss of high frequencies results. When low-impedance microphone circuits are used, there is very little frequency discrimination or loss of level in cabling. These facts should be kept in mind when one is designing any system in which the input circuits must be run a considerable distance.

The cables used with high-impedance equipment usually consist of a single conductor with an overall shield and rubber covering. The shield acts as one of the conductors in the microphone circuit. The shield is grounded and is connected to the microphone case and microphone stand.

Either two- or three-wire shielded cable is often used. When two-wire shielded cable is used, the shield is connected to ground at the amplifier, and to the program source equipment chassis at its other end. The shield does not act as one of the microphone circuit conductors, and the likelihood of hum pickup with two-wire cable is less than with single-wire shielded cable.

Where microphone or other low-level circuits are carried for long distances in flexible cables, three-wire shielded cable is most effective. Two of the wires serve in the input cable circuit; the third is used as a ground lead. The shield is connected to ground at the amplifier end of the cable only. The source equipment case is grounded through the third conductor.

When microphone or other input cables are installed permanently in floors or walls, lead-covered twisted or parallel pair may be used. The capacitance of lead-covered cable is quite high; therefore, it can only be

installed in short lengths when high-impedance ribbon or dynamic pick-ups are used. Recommended overall length for low-level phono pickups is 4 feet or less. The capacitance of the long cable also impairs the high-frequency transmission. Since crystal pickups are not affected by capacity, longer lengths of lead-covered cable may be used with them.

Permanently installed input cables can be plug-in connected to the control center and switched so that only those cables in use at any time are fed to the amplifier.

Control-Center and Amplifier Installation

When installing the controls and amplifier, care should be taken to be sure that they will receive proper ventilation; otherwise, components may be damaged by excessive heat.

When an amplifier must be located where vibration from speaker or other sources may cause microphonic noises, it should be mounted on shock-absorbing mountings or rubber pads. The mountings are available in various sizes and thicknesses, based on the number of pounds the mounting will support while operating normally. To determine the size of the mountings required for an amplifier, the corners of the amplifier should be weighed separately. This is necessary because many of the heavy components are usually located near one end of the amplifier chassis with the result that the weights at the different corners vary. When the approximate weights have been determined, mountings of the proper sizes can be chosen. When the amplifier is mounted in this fashion, it should be grounded through a length of heavy braid.

The amplifier should be suitably fused. If the amplifier is not equipped with fuses, a fused power receptacle should be provided. To protect the amplifier components, a 500-ma fuse can be connected between the center tap of the high-voltage winding of the power transformer and ground, as shown in Fig. 8-18. The entire hi-fi equipment is best powered from a single line from the electric power distribution board, with a 5-ampere fuse in the circuit.

Speaker Installation

When the speakers for a system are installed, the problems of impedance matching and power distribution must be solved. If the speakers are not properly matched, considerable loss in output will result. When a system uses more than one speaker, it is often necessary to have each speaker in the system dissipate a different amount of power. Output impedances are usually 4, 8, 16, and 32 ohms to directly match a speaker or combination of speakers. Sometimes 500- or 600-ohm outputs are provided to match long lines to speaker at distances of several hundred feet or more.

Generally, hi-fi speakers are rated at 8 or 16 ohms. Speaker-matching networks and crossovers are generally 16 to 32 ohms. It is important to

determine the speaker system's specified impedance (usually marked on the speaker, the speaker box, or the speaker literature) and to connect the speakers' or speaker system's terminals to the matching terminals on the output of the amplifier.

Two speakers of 8 ohms may be connected in series to match a 16-ohm output. Two 16-ohm speakers may be connected in parallel to match an 8-ohm output. Any combination of arrangements of this nature may be used. When connecting two or more speakers, check speaker phasing as described in following paragraphs. Typical multiple speaker connections are shown in Fig. 8-19.
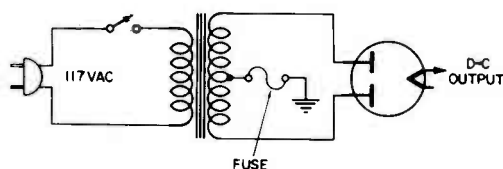
Fig. 8-18. Fusing power amplifiers.

In stereo systems, the amplifiers usually come with speaker connections clearly indicated. However, sometimes connections for combining the two channels into one for monophonic performance must be done by the owner. Fig. 8-20 gives both the parallel and series methods of connecting the two channels. The impedances must be matched by following the basic rules of series and parallel circuits.

Another situation in which the connections of speaker leads are involved is the addition of a third, or "middle," speaker to a two-channel stereo system. One simple way to do this is by connecting the middle speaker to taps on the two transformers, as indicated in Fig. 8-21. This method is not ideal, however, since there is likely to be considerable interaction between the left and right channels, thereby diluting the stereo effect. Other more efficient methods were discussed earlier in this chapter.

Speaker Lines—The wire used in speaker lines must be of sufficient size according to the impedance and power to keep the line losses within tolerable limits.

When speaker line impedance is under 60 ohms, only short lines, not to exceed 100 feet, should be used. When a line impedance of 60 ohms or more is used, lines may be run for a considerable distance, providing wire of sufficient size is used. The curves of Fig. 8-22 give recommended wire sizes for lines of various impedances and lengths.

Speaker Switching—In many installations it is necessary to have facilities for controlling auxiliary speaker volume and for switching speakers in and out of the system. When more than one speaker is used and one or more speakers must be separately controlled, a T-pad variable

275

that may be placed in the circuit to control the speaker volume and to provide a substitute resistance when it is turned off all the way in order to maintain constant load impedance to the system. Fig. 8-23 shows a circuit of an available unit which can be used to accomplish this. The substitute resistor should have a resistance equal to the impedance of the speaker or the group of speakers being disconnected from the circuit. To cut out or in speakers in various rooms, speaker switches able to carry up to 10 watts are available. These will provide for up to three speakers and will switch on any one or any combination of them.

Speaker Phasing—When a group of speakers are mounted at one point or near together so that they cover the same area, it is necessary
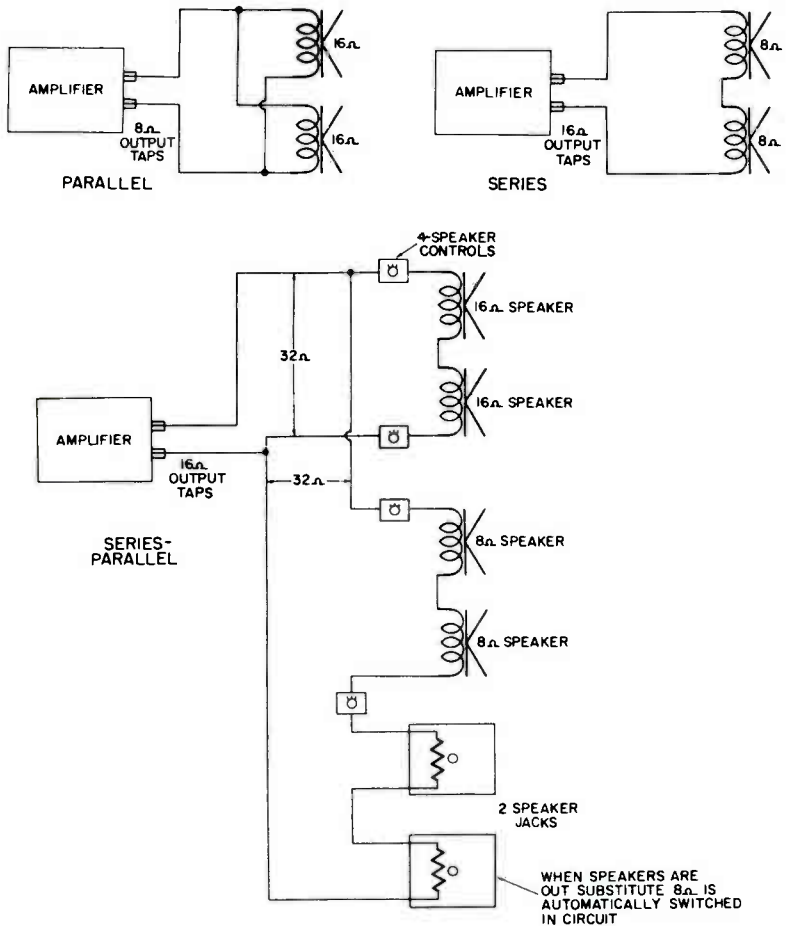
Fig. 8-19. Typical multiple speaker connection circuits.
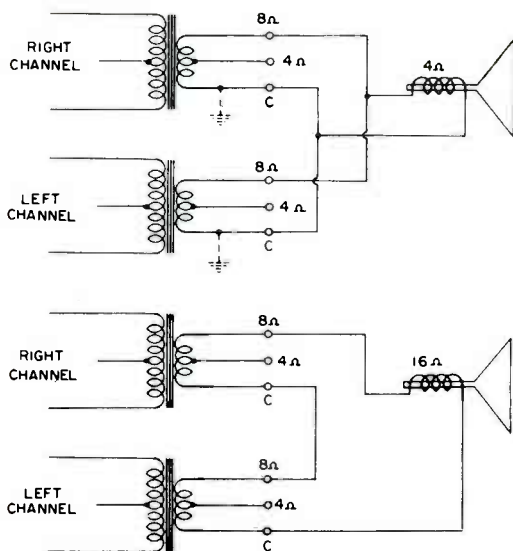
Fig. 8-20. Parallel and series methods of connecting stereo outputs for single-speaker monophonic operation.

that they be correctly phased. If the speakers are not phased properly, the sound waves from the improperly phased speakers will tend to cancel out, reducing the effective output of the system.

There are a number of ways to check the phasing of speakers. Two speakers may be connected to an amplifier in the same manner. They will be connected when installed, and placed close together, facing each other. A low-frequency signal from an audio generator or a record should
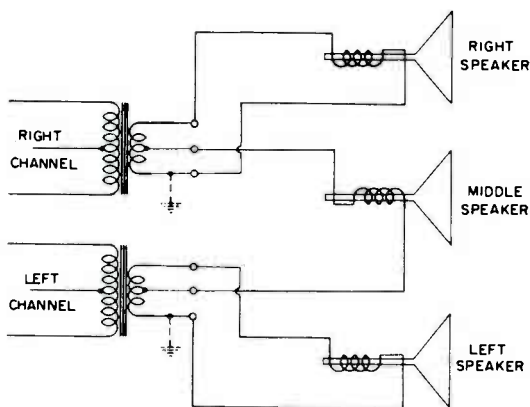
Fig. 8-21. One method of connecting a middle speaker in a stereo system.

277
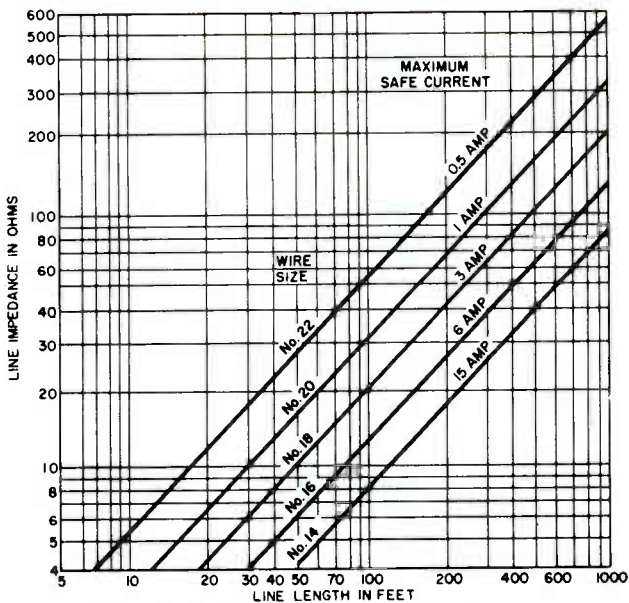
Fig. 8-22. Wire sizes for speaker lines.

be fed through the amplifier. By listening to the speakers, it is possible to determine whether or not the low frequencies are being cancelled. If the low frequencies are absent when the speakers are facing each other, then the phasing is correct if the speakers are be mounted so that they face in the same direction. When the test is made, if the low frequencies can be heard, then the low frequencies are not being cancelled, and the phasing is correct if the speakers are to be mounted facing away from each other. If the phasing is incorrect in either of the two cases just described, all that is necessary is that the connections to one of the voice coils be reversed.

The phasing of speakers may also be checked by using the circuit shown in Fig. 8-24. The apparatus consists of a pair of headphones connected to the input of an amplifier through two long cords and a double-pole, double-throw switch. The amplifier is equipped with an output indicator. The sound system whose speakers are to be checked for correct phasing should be turned on and a constant tone fed into its input. The
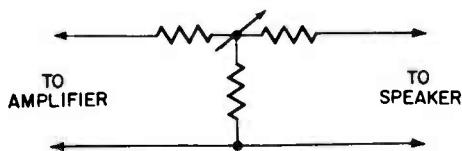
Fig. 8-23. A variable T-pad speaker control.

double-pole, double-throw switch should be marked to show "in phase" and "out of phase." This may be done by holding both phones in front of one speaker and noting the position of the switch which gives the greatest indication on the output meter. This position should be marked "in phase," and the other position should be marked "out of phase." The equipment may now be used to check the phasing of two speakers. One phone is held in front of each speaker. If the greatest output is indicated when the switch is in the "in phase" position, then the speakers are phased properly.
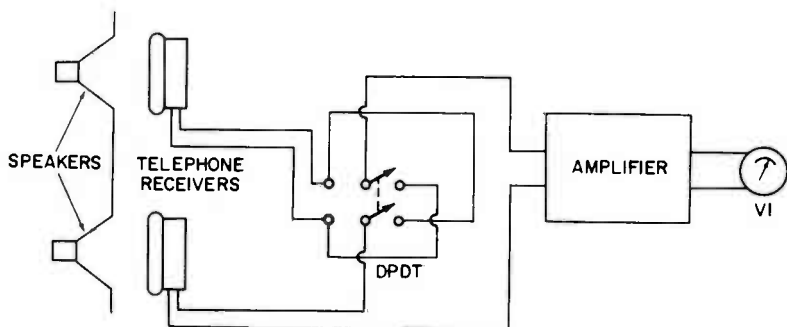
Fig. 8-24. Setup to determine speaker phasing.

In stereo systems, phase checking and correcting are made relatively simple. In most stereo systems, a phase-reversing control is included, connected either in the speaker output circuits or earlier in the amplifier chain. However, this control does not provide for phasing of the center speaker. If a single tone or other signal is applied to the inputs of both channels at the same time, the relative outputs from the right and left speakers will indicate the phasing. If the phasing is wrong, the two signals will differ by 180 degrees and tend to cancel each other. If the phases are correct, the signals reinforce each other. Thus, the correct phasing should produce much greater volume.

It should be remembered that there is a difference between speaker phasing and system phasing. The final output phase depends on the windings of the output transformers, connections between preamplifiers and amplifiers, connections of cartridges and input devices, and other things besides speaker connections. Therefore, checking the relative phasing of the speakers themselves is not always enough.

FEEDBACK

Careful attention must be given to the factors which create feedback; otherwise, considerable trouble will be experienced. There are two forms of feedback: They are *acoustic feedback*, which is caused by coupling between the speakers and the program source units, and *electrical feed-*

back, due to capacitive couplings between the output circuits and the input circuits.

Electrical feedback can be avoided by keeping the input and output wiring and components well separated and shielded.

Mechanical or acoustic feedback is often difficult to eliminate and, in severe cases, may limit the usable output of a system. Mechanical feedback effects and microphonics may be caused by direct or resonant coupling between the speaker system and the program source equipment, or through mechanical pickup in the first-stage elements of the preamplifier or the mechanical elements of phono and mike pickups. This may be corrected by mounting the units concerned on rubber, as has been described, or by moving the units concerned out of the mechanical contact area or resonant condition area. Moving the speaker should have the same effect.

Feedback manifests itself as a continuous tone or tones and spurious noises radiated by the speakers of a sound system. Feedback can be avoided by careful attention to the location of speakers and microphonic elements of the system.

There are a few general principles of microphonics and speaker location which, if followed, give reasonable assurance that little or no trouble will be experienced with feedback other than feedback due to direct microphone pickup.

A microphonic element may be described as any element located in the hot circuits of the front end of the system. The most typical examples are tubes which are highly sensitive to any vibrations of their elements. A vibrating element in a tube in the input stage at audio frequencies may induce the same effect as a microphone in the circuit by variation of plate-to-cathode resistance, thereby introducing undesirable interference.

A microphonic element should never be located in front of a speaker. If the speaker or speakers used in a system are located in front of the microphonic elements, the possibilities of feedback will be greatly reduced.

Speakers should not be mounted close to reflecting surfaces. When there is a flat surface close to and in front of the speakers, sound will usually be reflected by the surface.

If the system is installed in a location and is intended for use in a room filled with people, feedback may occur when the room is empty, but may not occur when the room is full of people.

Cardioid and other directional microphones are often very effective in eliminating feedback. The side of the microphone which is not sensitive to sound should be pointed in the direction from which the reflected sound is coming.

Attenuation of high-frequency response when equipment is used in a room with hard walls is desirable. Under these conditions, feedback

usually occurs due to reflected sound, and, since the high frequencies reflect more efficiently than do the low frequencies, attenuation of the high frequencies usually reduces feedback effects. The high-frequency tone control is usually useful when background music is being reproduced as at dinner, etc. Under these conditions, the high frequencies should be attenuated, since they tend to make conversation difficult.

Control of low-frequency response when speech is being reproduced is desirable. Under these conditions, attenuation of the lower frequencies helps remove the "tubbiness" which occurs at times in certain types of installations.

Index

287

HI-FI STEREO HANDBOOK
THIRD EDITION

by William F. Boyce

Whether you're a neophyte on the verge of buying your first piece of equipment, or an experienced audiophile who wants to gain more knowledge about hi-fi and stereo, you'll find this book invaluable. For the beginner, it explains the difference between monophonic and stereophonic systems, the meaning of high fidelity, the different sources of program material, the various kinds of distortion, etc. The three major stages in hi-fi reproduction are thoroughly covered: the program-source equipment, preamplifier and amplifier sections, and the speaker system. Neophytes and veterans alike will appreciate the discussion on stereo, and the material on designing, selecting, and installing hi-fi systems.

This third edition includes new material on stereo needles, headphones, adapter circuits, tape-cartridge players, and multiplex operation. Several new transistor circuits have been covered, including preamplifiers, amplifiers, and stereo control circuits. Other additions contain information on recording techniques, compact hi-fi systems, and multipath-checking equipment.

ABOUT THE AUTHOR

William F. Boyce has been writing on the subjects of audio and hi-fi since 1942. In addition to authoring several books, he has supervised the preparation of thousands of technical publications. A graduate of Newark College of Engineering and of General Motors Corporation's four-year engineering training program, Mr. Boyce has been active in radio amateur science since 1932. Currently he is president of Boland & Boyce, Inc., a firm of technical writers, editors, illustrators, and publishers of military handbooks.

HOWARD W. SAMS & CO., INC.
THE BOBBS-MERRILL CO., INC.