High Fidelity

HOME MUSIC SYSTEMS

Their Selection, Assembly and Installation

WILLIAM R. WELLMAN

HIGH FIDELITY HOME MUSIC SYSTEMS

THEIR SELECTION, ASSEMBLY, AND INSTALLATION

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Second Edition

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Preface to the Second Edition

THE SIGNIFICANT ADVANCES that have been made since the first edition of this book appeared have made a thorough revision necessary.

The very wide acceptance of stereophonic phonographs has greatly extended the market for high fidelity music systems. Accordingly, a chapter on stereophonic high fidelity systems has been added.

Chapter 6, covering tape recorders has been entirely rewritten and enlarged to conform to the developments that have taken place in the past few years. Although tape is as yet incorporated in relatively few factory-assembled units, it is in much wider use among those who purchase their components separately and enjoys wide popularity as an independent source of entertainment among others.

In addition, new material has been added on loudspeakers, loudspeaker enclosures and phonograph cartridges and the section on amplifier construction has been brought up to date.

As in the first edition, the objective of the book is twofold: 1. to assist the reader in making a wise selection of components, and 2. to suggest ways of arriving at a sensibly priced music system without a serious compromise in quality.

W. R. W.

Contents

1 An Introduction to High Fidelity

HIGH FIDELITY (sometimes called wide range) home music systems have been available to the general public only since the end of World War II. The movement toward realism in music was evidently started by a few engineers and technicians who were also music lovers. As might be expected, the examples of excellent equipment built by these people elicited the favorable comments of all who listened and, of course, the circle then widened. Parts for the equipment were available, for they had already been designed and developed for the broadcast industry, but the parts were quite expensive at first, and production was on a relatively small scale. Soon parts manufacturers expanded their facilities. However, it is doubtful whether the movement would have reached its present proportions without the impetus of the high fidelity, long-playing record and the advent of frequency modulation broadcasting.

Since high quality music systems are still in the minority as compared to ordinary radios and phonographs, the term "high fidelity" is often misused and misunderstood. Any discussion of the subject might, therefore, open with the question, "What is a high fidelity music system?" It is likely that if you were to put this question to the uninitiated enthusiast he would answer that it is a system capable of reproducing the higher tones in music. While this is true, the definition is inadequate. The average person is unaware that in a wide range system the bass, or lower portion of the music range, is also extended.

Figure 1-1 is a chart showing the ranges of various musical instruments, expressed in vibrations per second, and excluding overtones. Notice that the bass viol, for instance, has a range of vibrations extending from slightly higher than 40 per second to about 240 per second, while the piccolo range extends from about 500 vibrations per second to more than 4600 per second. At the top of the chart you will find two broader black lines; the uppermost of these two represents the portion of the musical range reproduced by a good high fidelity system. The lower heavy line shows the range covered by an ordinary radio or phonograph.

At first glance it might appear that an ordinary radio or phonograph would satisfactorily reproduce most of the sounds created by musical instruments, but a little thought will reveal that this is far from true. Notice that such a radio or phonograph reaches the lower limit of its range at about 100 vibrations per second; hence many of the sounds produced by the bass viol, bass tuba, and kettle drums, to say nothing of the pipe organ, are lost. Now let us explore the upper part of the range. The typical radio has a coverage extending up to about 6000 vibrations per second. Let us determine

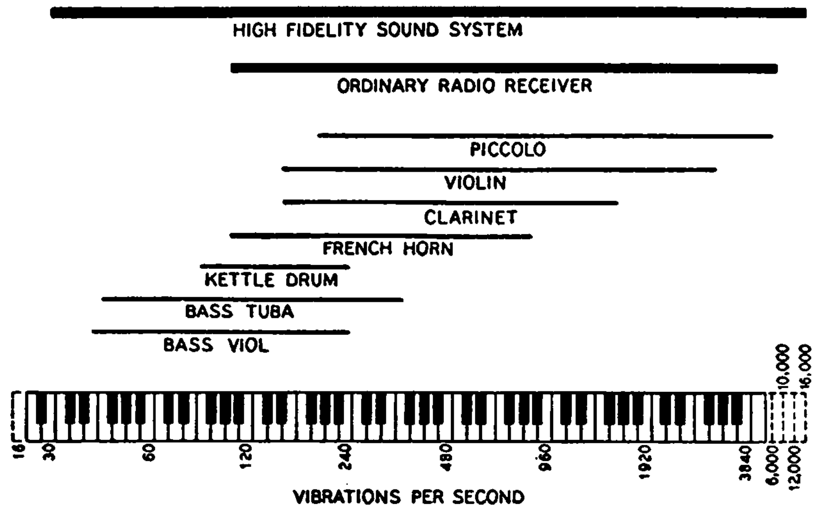


Fig. 1-1. Range of sound vibrations.

whether such equipment will prove to be adequate for the higher notes. The right-hand end of the chart, extending beyond the piano keyboard, is shown in dotted lines. This area begins just below 6000 vibrations per second and extends up to 16,000 per second, which is the limit of average hearing. In this region lie the higher harmonics, or overtones, of various instruments; these overtones enable us to distinguish between, say, a flute and a piccolo, even though both instruments are playing the same note. Obviously, these overtones are completely lost in the average radio or phonograph.

Up to this point we have learned that a good high fidelity music system not only reproduces more of the higher notes, including the overtones, but also has an extended bass response. By contrast, an ordinary phonograph does not give much more than an illusion of real music. True bass is not present; we imagine we hear the deep tones of the bass viol, when actually we do not. We perceive the rhythm, some harmonics of the original note are heard, and that is all. Furthermore, the high fidelity system gives us "definition"; this is the ability to distinguish between various instruments and is present only when the higher harmonics are reproduced.

However, we have not yet given a full description of the capabilities of a wide range system. There are other factors, such as dynamic range, lack of distortion, and the rather elusive quality known as "presence." Dynamic range refers to the ability of the equipment to reproduce the full range of loudness in music from the softest passage to the full volume of a symphony orchestra without harshness, fuzziness, or other distortion. All components in a first-class high fidelity system are selected for their ability to reproduce sounds with an absolute minimum of distortion of any kind (and there are several forms of distortion, as you will discover later). "Presence" is rather difficult to define, but may be suggested by the word "realism." The music must sound, as nearly as possible, like the original, including the loudness. Sometimes this statement is interpreted to mean that the music should be as loud as it would be in a concert hall, but this is not quite true. Actually, you should have the illusion that the music has the same loudness as if you were in the concert hall or opera house.

We shall now take a moment to distinguish between a true high fidelity system and what may be called "commercial" high fidelity. The table model phonograph, often advertised as a high fidelity unit, may well give satisfactory reproduction of the high end of the musical range and is usually far superior in that respect to the average radio set or phonograph; but because of the limited size of cabinet and the small speaker such so-called high fidelity equipment more often than not has poor bass response.

Throughout this book you must bear in mind that true high fidelity is usually quite expensive. A system that will reproduce, without distortion, all sounds from 30 to 15,000 vibrations per second may be beyond your budget, but it is possible to buy equipment at reasonable prices which will reproduce the range from perhaps 50 to 10,000 or 12,000 vibrations per second. You will have to decide whether the extension in range and an absolute minimum of distortion warrant the increase in price. Your decision can only be made on the basis of a comparative listening test and you may then find that the wider range of the more costly system is not worth while in your particular case. For all except the most discriminating, a system with a range of tones from 50 to 12,000 should be fairly satisfactory and will afford an amazing improvement over an ordinary radio or phonograph. A system of this kind need cost no more than $150 to $200, if cabinet work is not included.

In this chapter we shall discuss the requirements to be met by the individual components of a wide range system. However, before we can do this it will be necessary to review some of the basic principles of sound and its behavior.

Sound

Sound may be defined as those vibrations that affect the organ of hearing. Although sound may be transmitted through solids, liquids, or gases, for our purposes we need discuss only the transmission through gases—more specifically, through air.

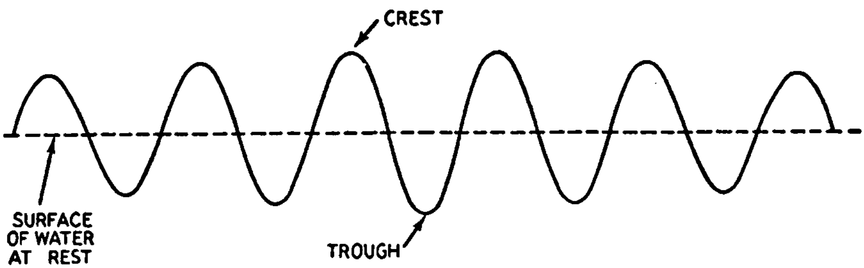


Fig. 1-2. Waves in water.

To understand how sound is propagated through air, we may turn to a familiar analogy: the origin of waves in water. Imagine the smooth surface of a pool of water. If a stone is dropped into the pool, ripples spread out in ever-widening circles from the source of the disturbance. Looking down upon the water's surface, we see a series of concentric rings, but if we were able to direct our view horizontally across the surface we would find that at some points the water is "piled up" in a ridge. This effect has been labeled "crest" in the drawing, Figure 1-2. Following each crest there is a point at which the water has been depressed below its normal surface level (indicated by the dotted line); these points are the "troughs."

Sound waves are generated and propagated in much the same way, except that they are not limited to travel in a horizontal plane as are the waves in water. The transmission of sound waves through air from point to point consists of vibratory, or to-and-fro, movements of air particles. These particles are alternately compressed (packed more closely together) and rarefied (separated more widely) and the interval between compression and rarefaction depends upon the rate of vibration at the source.

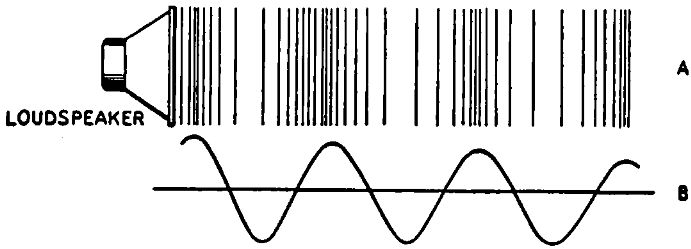


Fig. 1-3. Propagation of sound waves.

Refer to Figure 1-3A, which represents sound waves emanating from a loudspeaker. (Note: Although straight lines are used here, the waves, if visible, would actually be curved since they radiate in all directions. Furthermore, we have deliberately neglected the fact that waves are radiated from the rear as well as from the front of the loudspeaker.) You will observe that the lines are closer together at some points than at others. Such close spacing indicates maximum compression or packing of air particles; the widest spacing of lines indicates maximum rarefaction of air particles. If we were able to measure the air pressure at various distances from the speaker and then chart the results, we would have a curve such as shown in Figure 1-3B. You will immediately notice that this looks remarkably like the illustration of waves in water, Figure 1-2. Notice that, whenever a maximum compression of air occurs, the wave rises to a crest; it falls to a trough when a point of maximum rarefaction occurs. There is also a zero point, at which time the air is at normal pressure, neither rarefied nor compressed. With these facts in mind, we may proceed to a discussion of some of the characteristics of sound waves.

Speed of Propagation. The speed of sound is variable and depends upon the medium through which it travels. If we assume that the waves are passing through air at normal pressure (sea level) and at a temperature of zero degrees centigrade, the speed is then 1088 feet per second. As the conducting medium becomes denser, the speed increases.

Intensity. Referring to Figure 1-4, we see that the intensity (sometimes called amplitude) is measured by the height of the wave at the crest. This illustration shows two waves having different intensities or amplitudes. For measuring and comparing sound intensities, a unit called the decibel has been devised. The decibel (abbreviated db) represents the change in pressure that is just discernible to the air.

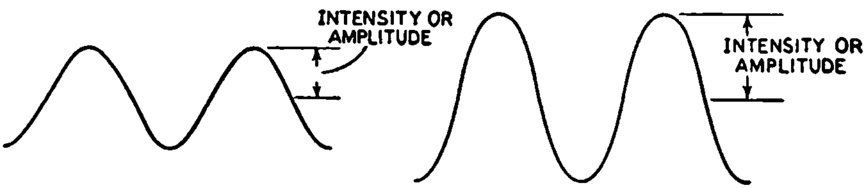


Fig. 1-4. Illustrating intensity of sound waves.

Frequency. Frequency is a measure of the number of vibrations occurring in a given time interval, usually a second. Figure 1-5 is a representation of two waves having different frequencies, although their intensities in this case are the same. Notice that the wave at the left of the figure completes just half as many vibrations in a given time as does the one at the right. The wave at the right, therefore, has twice the frequency of the one at the left.

Frequency is always expressed in so many "cycles" per second; a cycle is precisely the same thing as a complete vibration. The lowest frequency perceptible to the average ear is about 30 cycles per second; comparatively few individuals can hear sounds above 15,000 cycles per second. The frequency of middle C is 256 cycles per second; expressed in another way, when middle C is struck on the piano, the string makes 256 complete vibrations in one second.

The pitch, or frequency, produced by a vibrating object is dependent upon several things, the first of which is physical size. It is hardly necessary to add that when a large object is set into motion it will produce fewer vibrations than a smaller one and this will result in a lower pitcher tone. Any object has its own "period of vibration"; this is the particular frequency to which it will most readily respond, although it can be made to vibrate at other frequencies. The natural period of vibration is sometimes referred to as the resonant frequency of the object; in other words, the object re-

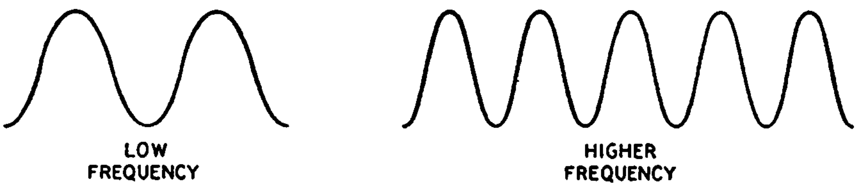


Fig. 1-5. Illustrating frequency.

sponds best to that particular frequency. We may expect that at its resonant frequency an object will emit a comparatively large amount of sound energy; a loudspeaker cone, for example, will radiate more sound at its resonant frequency than at other frequencies, although it is evident that it can be made to vibrate at those other frequencies.

Many objects, when in vibratory motion, will radiate frequencies other than the resonant frequency. These frequencies are referred

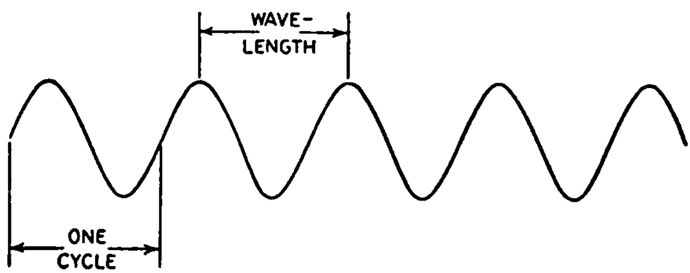


Fig. 1-6. Wavelength.

to as harmonics and bear a definite relationship to the primary, or fundamental, tone. An object vibrating at a fundamental frequency of 240 cycles per second may also radiate frequencies of 480 (second harmonic), 720 (third harmonic), and so on. As we have already learned, harmonics are important in adding definition to recorded music.

Another commonly used term is wavelength, and this is directly related to frequency. As shown in Figure 1-6, wavelength is the distance from the start of one vibration to the end of that same vibration. If we assume that sound waves travel outward from a vibrating body as do the waves in water, then it is obvious that when several vibrations have been completed they will extend into space as shown in the illustration. Suppose the vibrating body continues in motion for precisely one second, and during that interval it completes exactly 110 separate vibrations. If the vibration takes place in air at normal temperature and pressure, the sound waves will travel a distance of approximately 1100 feet (the precise distance is 1088 feet) during that one-second interval. If sound waves were visible, we would see 110 waves strung out end to end over a distance of 1100 feet. And, if sound waves were visible they would

Fig. 1-7. Distortion.

also be measurable in the ordinary way, and we would find each complete sound wave to be 10 feet long. In other words, the length of a wave may be found by dividing the distance traveled in a second (1100 feet) by the frequency (110). If the frequency happened to be 1000 cycles per second, then the wavelength would be 1100 divided by 1000 or 1.1 feet. Although frequency is the more important, wavelength is sometimes used in calculations pertaining to sound reproducing equipment; this is especially true in the case of the loudspeaker and its enclosure.

Incidentally, it is important to understand the relationship between the electrical waves existing in sound reproducing equipment and their mechanical counterparts, sound waves. The amplifier handles electrical waves and, although they are not audible as are sound waves, they possess similar characteristics in regard to frequency, wavelength, intensity, and so forth. The amplifier in a quality sound system must be capable of reproducing electrical waves having frequencies between 30 and 15,000 per second. The loudspeaker converts these electrical waves into their mechanical equivalents.

Wave Form. Not all sound waves have the form shown in Figure 1-7A. For instance, some waves may appear as shown in Figure 1-7B. However, if the original form of the wave is as shown in Figure 1-7A and the music system reproduces it as in Figure 1-7B, then one of the several forms of distortion has taken place and the reproduced sound is no longer a faithful replica of the original.

Music System Components and Their Specifications

The components of a complete home music system are illustrated pictorially in Figure 1-8. You will note that there are three sources of program material: radio tuner, phonograph record player, and tape recorder. Each of these is a device for converting impulses (of

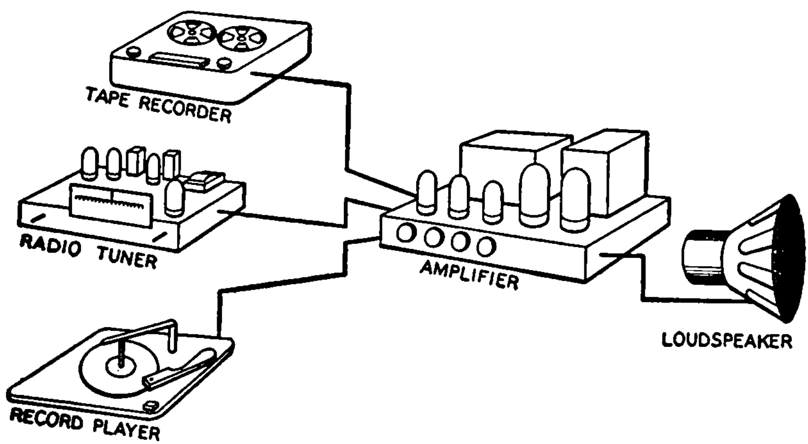


Fig. 1-8. Components of a home music system.

different types) into electrical waves. The tuner utilizes radio waves transmitted through space. In the tape recorder the impulses are stored in the form of magnetic energy in the tape. The record player converts the mechanical energy stored in the record grooves into electrical waves. In all three, the electrical waves reach the amplifier and are there built up, or strengthened, and then passed on to the loudspeaker where they are converted into air vibrations.

Not all high fidelity enthusiasts include the tape recorder in the list of components, principally because of the high cost of a quality unit. A few listeners also eliminate the tuner and depend solely upon the record player as a source of program material. Although the tuner is not prohibitively expensive, it does not always provide the type of music wanted at the time.

The amplifier is provided with a jack, or jacks, for the record player, tuner, and tape recorder, so that the program source may be selected at will. (Note: In many instances, there will be a separate, small amplifier—the preamplifier—connected between the program source and the main amplifier, but in this discussion we shall consider the preamplifier to be a part of the main amplifier.)

The program source delivers to the amplifier impulses that are—or should be—exact electrical duplicates of the original sounds. The intensity of these impulses is very small, and it is the function of the amplifier to build them up to the point where they will actuate the loudspeaker. If we have a good amplifier, it will introduce little or no discernible change in the character of the impulses. A good loudspeaker will convert the electrical waves into sound waves that are as nearly like the originals as is possible.

Before beginning a discussion on individual components, it is best to consider some factors that determine good quality in music. Listeners sometimes ask, "How may I know whether my system actually delivers quality that is as good as I believe it to be?"

It is true that, in some types of music, discordant or harsh sounds have been deliberately introduced for a specific effect. However, if we neglect such instances, good-quality reproduced music with a minimum of distortion is "easy to listen to." If the music is not tiring and does not grate upon your nerves, even after five or six hours of listening, then you may assume that you have made at least an approach to good quality.

As we have already pointed out, high fidelity does not mean good high frequency response alone; in fact, it is possible to have a large proportion of high notes, but with a great deal of distortion and, obviously, this does not mean that you have high fidelity. Very often, a listener's first hearing of high fidelity will be deceptive, for the system may appear to lack highs, and perhaps lows too; it is only after a prolonged period of listening that their presence becomes evident. Again, some record manufacturers have attempted to create an impression of wide frequency response by overaccentuation of highs. At first listening, such products may sound unusually brilliant, but after a few plays the harshness and distortion become only too apparent.

One ready-made test for good quality is the ease with which one instrument can be distinguished from others in a group; if you can readily make such identification, then the higher harmonics are being reproduced. At the same time, there must be no harshness or fuzziness of the highs. Finally, listen to the bass notes carefully. True, clean bass is readily detected when listening to such instruments as the double bass; if bass is not a monotone, but appears to move up and down the scale, if there is no suggestion of boominess or hangover, then you will have reason to be satisfied with the performance of your system.

The Loudspeaker

In this discussion, the loudspeaker enclosure is considered to be a part of the loudspeaker system. It would be almost impossible for the layman to select a speaker on the basis of manufacturers' performance figures or charts alone. Although such figures and charts do give some indication of performance, the most reliable method of judging, at least for the average purchaser, is to listen and compare.

Information offered by the manufacturer always makes some mention of the frequency response of the product. However, the unqualified statement that the speaker has a response of from 50 to 15,000 cycles is far from enlightening and, at times, may be deceptive. Although the unit might reproduce all frequencies between those limits, we still have no information as to how well the various frequencies are reproduced. As an example, consider the two charts, Figures 1-9A and 1-9B. You need not approach such

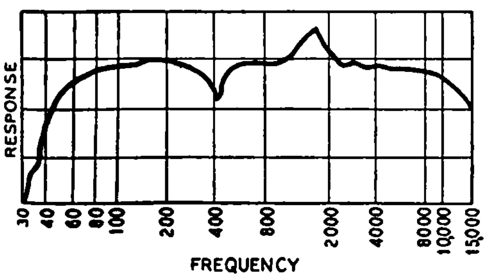


Fig. 1-9A

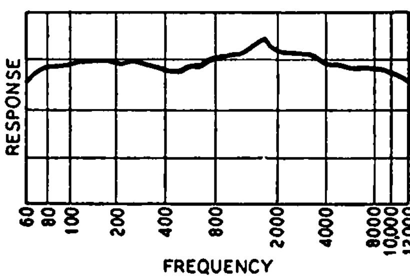


Fig. 1-9B

charts with a feeling of dread, for they do not differ greatly from the stock market charts published in daily newspapers and are no more difficult to interpret. Vertical distances represent sound intensities (instead of stock prices) while the horizontal distances are used to represent frequencies (instead of months or years). Figure 1-9A is the response curve of a speaker that we shall refer to as *A*; it shows how well speaker *A* responds to, or reproduces, all tones. Notice that the unit does respond to all frequencies between 30 and 15,000 cycles. But, on close inspection you will see that the heavy line dips sharply below 100 cycles and again above 10,000. In other words, speaker *A* will not reproduce tones above and below those points as well as it will others in the middle of the range. Furthermore, there is a noticeable dip in the curve at about 400 cycles and a sharp peak near 1500 cycles. A tone at about 400 cycles will therefore be less intense than others, while frequencies near 1500 cycles will be overaccentuated.

Now consider the response curve of speaker *B*, shown in Figure 1-9B. As charted here, this unit has a response that extends from 60 cycles to only 12,000 cycles, but there are not the sharp dips at the upper and lower ends of the curve, as in speaker *A*. Furthermore, dips or peaks in the curve are relatively minor. It is more than probable that speaker *B* would afford better reproduction than speaker *A*, even though the range of *B* is more limited. Please note that the charts shown here do not represent performances of actual speakers and are intended only for illustration.

From the foregoing you probably have gathered that when you are considering the purchase of a loudspeaker it would be unwise to depend upon a mere statement of the response limits. If you are considering a speaker in the upper price range, it is quite likely that the maker can supply more detailed information on his product than the dealer has at his command. If such information cannot be obtained upon request, then the speaker should be bought only after extensive comparison (under identical conditions) with other units in the same price classification.

The ideal speaker would have an absolutely flat response characteristic, with neither peaks nor dips, but such excellence is not to be expected of the average unit. Fortunately, it is possible to compensate for some speaker deficiencies. For instance, a sharp drop at the upper end of the curve can usually be corrected by installing a second speaker having a rising characteristic at that point. Similarly, an exceptionally good speaker enclosure will often compensate for the poor bass performance of a speaker in an enclosure of only average quality.

The performance of even the best speaker can be affected adversely by its location and, to some degree, by room acoustics. This is discussed more fully in the chapter on loudspeakers. Suffice it to say here that the radiation of high frequencies from any speaker is most intense at a point directly in front of the unit. In other listening positions, the highs drop off sharply.

In an average room, the loudspeaker is required to deliver acoustic power equal to about 0.5 watt. Assuming that the speaker has an efficiency of 5 per cent (this is the average for speakers in the $50 to $75 class), then the amplifier must deliver 10 watts of electrical power to the speaker. However, if the room is acoustically "dead" (particularly free from reverberation or echoes because of the presence of a great deal of sound-absorbing material) or if the speaker has a much lower efficiency, the electrical power will have to be increased. It is also fairly obvious that an increase in room noise level will make a power increase necessary.

Finally, the loudspeaker must be built to handle the power delivered to it with a minimum of distortion.

The Amplifier

An amplifier suitable for use in a wide range system must have the following characteristics: (a) a flat frequency response from 30 to 15,000 cycles, (b) a minimum of distortion, (c) ability to deliver at least 10 watts of power to the speaker, and (d) reasonably low hum and noise level. In addition, it is desirable that the unit be equipped with bass and treble boost circuits. There are other desirable features, and these will be taken up in the chapter on amplifiers. Suppose we now consider the requirements just mentioned.

For reasons already outlined, the amplifier must be designed to deliver at least 10 watts to the speaker; if it cannot do this, it will be overloaded at peak levels and serious distortion will result. It is important to remember that a 20-watt amplifier provides a 100 per cent safety factor, but a unit should not be chosen on this basis alone. When only a single loudspeaker is to be used, a 10- to 15-watt amplifier will almost always be adequate, but the power output should be increased by approximately 5 watts for each auxiliary speaker.

Merely providing ample power for driving the speaker or speakers will not afford realism unless the signals applied to the speaker are essentially the same as those applied to the amplifier input; in other words, the amplifier must increase the signal level with a minimum of distortion. First of all, it is important that all frequencies be amplified to the same degree, otherwise the tonal balance will be destroyed. It is difficult to find an amplifier having an absolutely flat response curve, and a deviation of plus or minus one db is considered acceptable. Harmonic distortion and intermodulation distortion (these terms will be more fully explained in the chapter on amplifiers) occur in all amplifiers, but the levels should not be above 5 per cent for harmonic distortion and 10 per cent for intermodulation distortion. Hum and noise can be considered forms of distortion and should therefore be kept as low as possible.

Finally, when we have an amplifier and loudspeaker capable of delivering sounds that are nearly perfect replicas of the originals, we discover that the human ear itself is imperfect. It is well known that the hearing of individuals differs greatly; for instance, the ability to hear the higher pitched tones decreases with age. Even the average ear does not hear all frequencies with the same loudness; the curve of hearing shows a quite noticeable drop at both ends of the frequency range. Another undesirable effect that must be overcome is the normal room and street noise present in practically all but the most remote locations. The answer to this is not simply to increase loudness, for background noise tends to mask the high and low frequencies to a greater degree than it does the mid-range.

The effects noted in the preceding paragraph can sometimes be offset by the use of adequate bass and treble controls. As a rule, such controls provide both boost (increase in intensity) and cut (decrease in intensity) at both ends of the frequency range. Some idea of the operation of the controls can be gained by studying Figure 1-10. The curve, *A*, in this illustration shows the frequency response of the amplifier with the controls set for "flat" response; in other words, there is then neither a boost nor a cut at any point in the range. The flat position will usually be at the mid-point on the control dial. With both controls advanced to the maximum boost position, the response curve of the amplifier is as shown at *B*; notice that both the highs and the lows have been emphasized.

Incidentally, such an adjustment might be desirable when the amplifier is operated at unusually low volume levels, for at such low levels there is a drop in base and treble output. With both controls set for the maximum cut position, the curve is as shown at C. Other adjustments whereby a cut in bass and a boost in treble (or vice versa) are also possible.

It is evident from this discussion that intelligent use of bass and treble controls will effectively compensate for differences in hearing, for the effects of room noise, and for the effects introduced by unusually low volume levels. It is also possible to use them in compensating for deficiencies in the program source, such as an inade-

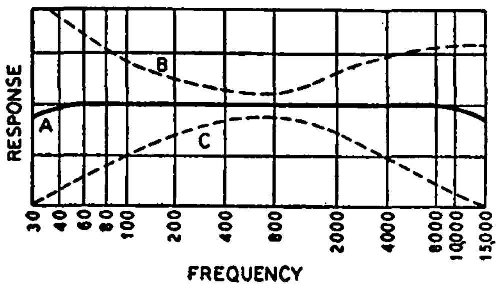


Fig. 1-10

quate recording or a broadcast of poor quality. We do not mean that any type of tone compensation will actually restore to music what has already been lost, but merely say that it may improve the over-all effect. Of course, the use of tone controls can be overdone. First of all, it is quite possible that overaccentuation of any part of the range will introduce distortion. Then, too, operating an amplifier with the bass boost set too high may increase the hum level. Excessive treble boost will certainly increase the level of any surface noise present in a recording or of any noise incidental to A-M broadcasting.

Preamplifiers

In many instances it is necessary to employ a preamplifier between the program source and the main amplifier, for example when a record player equipped with a magnetic type of cartridge is the program source. The function of the preamplifier is to compensate for the lower voltage output of such a device.

The preamplifier operates in much the same way as the main amplifier. In some respects, however, it must meet requirements that are more rigid. Since the preamplifier is placed ahead of the main amplifier, any hum, noise, or distortion resulting from its operation will be further increased by the main amplifier, and these have to be guarded against.

In some instances, the preamplifier will be physically a part of the main amplifier, constructed on the same metal base, or chassis. In other types, the preamplifier is an entirely separate unit; the latter arrangement is to be preferred. Almost all preamplifiers include some type of equalization for phonograph record characteristics; the need for such equalization will be explained in the chapter on preamplifiers. In some instances, the preamplifier unit contains the bass and treble boost controls, so that all controls are then in a central location. In a few designs the preamplifier is arranged so that it can be located at a considerable distance from the other components in the music system, thus providing a convenient remote-control arrangement.

The Record Player

The record player is a device for converting mechanical energy (in the record grooves) into electrical energy to be applied to the amplifier. The first requirement is that the device deliver a signal of such intensity that the amplifier can be driven to full power capacity, if so desired. We naturally expect that the range of frequencies be wide enough to include all tones that are audible, and most modern units will satisfy this requirement. However, neither the record player nor the record has yet attained perfection, and it is not possible to get a uniform or flat frequency response. For this reason, various kinds of compensations must be employed.

One very important requirement is that the record revolve at precisely the correct speed with as little variation as is possible. This can be achieved only through unusually excellent design, and this is reflected in the selling price. Most listeners prefer the manually operated type of player to the automatic changer, but in some instances the cost of a first-class manual player is prohibitive.

Summary

Up to this point we have discussed, in rather general terms, the components of the high fidelity home music installation. The desirable and undesirable features of each component will be discussed individually and at greater length in the chapters to follow. This material should appeal to those who must buy their components "ready made." If you are in this situation and you are located near a large radio supply house or high fidelity show room, you will be able either to select your components individually or to buy complete packaged "ensembles." The latter are available at prices ranging from $125 to $600 or more, without cabinet or speaker enclosure.

If you have some skill in the handling of tools, you will be interested in the instructions to be given for building your own cabinets and speaker enclosures, thereby saving a substantial part of the cost of an installation. For those who have had some experience in building radio or electronic equipment, there will be information on the construction of amplifiers, preamplifiers, and tuners. Finally, we shall offer some suggestions for remodeling existing radio-phonograph combinations with the objective of improving quality.

In this chapter it has been necessary to employ terms and phrases that might be unfamiliar to some readers. Although an effort has been made to give explanations where they seemed necessary, it is possible that some have been overlooked. If you have experienced any difficulty, please turn to the glossary of terms at the end of this chapter.

GLOSSARY OF TERMS

ALTERNATING CURRENT. A current which reverses its direction of flow at regular intervals. A frequency of 60 cycles per second is common in power and lighting circuits in the United States. Abbreviated: a.c.

AMPLIFIER. An electronic device used to increase the intensity of a voltage (or power). It consists of one or more vacuum tubes, together with associated circuit components.

ANTENNA. A wire or other electrical conductor used to collect radio waves (in reception) or to radiate waves (in transmission). Sometimes referred to as the aerial.

ATTENUATION. A reduction in the intensity of a signal.

AUDIO FREQUENCY. An alternating current that may be converted, by means of a loudspeaker or equivalent device, into sound waves. The range of audio frequencies is from 30 to 15,000 cycles, or vibrations, per second.

BAFFLE. A device used to increase the length of the path that sound waves travel from front to rear (or vice versa) in a loudspeaker. The simplest baffle consists of a board with the loudspeaker mounted over a hole, usually in the center of the board. The word "baffle" is also used in referring to various types of loudspeaker enclosures.

BASS. Notes at the low frequency end of the audible range.

BIAS. A voltage applied to the grid of a tube to make it negative with respect to the cathode.

BIAS CURRENT. In tape recording, a current, usually a high frequency alternating current, applied to the tape while recording in order to raise the recorded level and to reduce distortion.

CARRIER WAVE. A radio frequency wave emitted by a transmitter. When used to convey speech or music, the carrier is modulated by the audio frequency waves.

CARTRIDGE, PICKUP. A device used to translate mechanical movement of the pickup stylus into electrical energy. The two principal types are the crystal and the magnetic, or variable reluctance, cartridges.

COMPLIANCE. A measure of the ease with which a pickup stylus may be deflected from its rest position.

CROSSOVER NETWORK. An electrical circuit (usually a combination of coil and capacitor) for passing low notes to a low frequency speaker and high notes to a high frequency speaker. Also called: dividing network, frequency divider.

CYCLE. One complete reversal in direction of an electric current, or a complete vibration in a sound wave.

DECIBEL. A measure of change in power, voltage, or loudness. Abbreviated: db. Positive and negative values are used; a negative value refers to a decrease, while a positive value means an increase.

DIRECT CURRENT. A current that flows in one direction only. It generally refers to one that also remains at a constant value. A pulsating direct current is one that changes intensity but not direction of flow.

DISCRIMINATOR. A detector used in FM receivers. It converts frequency variations into amplitude variations.

DISTORTION. Any component of a signal that was not present in the original.

EQUALIZER. In tape and disc recording some departure from flat response is necessary. Usually the treble is emphasized and the bass is de-emphasized. An equalizer corrects such deviations when the recording is played back.

FEEDBACK. A method of applying a portion of the output energy of an amplifier to the input circuit so that distortion is reduced and the output impedance of the amplifier is lowered.

FIDELITY. Faithfulness of reproduction, as compared to the original.

FLUTTER. Speed variations, as in a phono turntable or tape transport, occurring at a relatively rapid rate.

FREQUENCY. Acoustically, the rate of vibration of an object. Electrically, the rate of reversal of direction of flow of a current or voltage. Expressed in cycles per second.

FREQUENCY RESPONSE. A measure of how well an amplifier or other device responds to all frequencies within a given range, with a minimum of deviation from a constant level.

FREQUENCY MODULATION. A system of radio transmission in which the frequency of the carrier wave is varied by the changes in the speech or music.

FUNDAMENTAL. Acoustically, the rate of vibration to which an object will respond. If a wave consists of several components, the fundamental is the lowest frequency.

GAIN. The ratio between the level of the output signal and the level of the input signal.

HANGOVER. A type of distortion occurring at low frequencies and caused by the inability of the loudspeaker diaphragm to stop vibrating as soon as the signal ceases.

HARMONIC. A multiple of the fundamental frequency. Thus, the second harmonic of 1500 cycles is 3000 cycles, the third harmonic is 4500 cycles, and so forth.

HEAD, TAPE HEAD. The electromagnet in a tape recorder which performs the function of record, playback, or erase, or, in some recorders, all three functions.

IMPEDANCE. The total opposition offered by a circuit to the passage of alternating current.

KILOCYCLE. One thousand cycles per second. Thus, 15,000 cycles per second is equal to 15 kilocycles per second. Abbreviated: kc.

LIMITER. A circuit used in FM receivers to remove amplitude variations from the frequency-modulated signal.

MAGNETIC PICKUP. One type of device for converting mechanical motion of the pickup stylus into electrical energy. The magnetic pickup consists of an armature, bearing the stylus. The armature is mounted between the poles of a magnet. Movement of the stylus results in a changing magnetic field which induces a variable current in a pair of coils wound on the magnet.

MEGACYCLE. One million cycles per second. For example, 20,000,000 cycles per second is equivalent to 20 megacycles per second.

MODULATION. A process in which one wave is caused to vary in accordance with the changes in another wave.

OVERLOADING. A form of distortion caused by applying to the amplifier (or loudspeaker) a signal that is greater than it can handle.

POWER AMPLIFIER. The last, or final, stage in an amplifier; it supplies power to a load, such as a loudspeaker.

POWER OUTPUT. A measure of the amount of power delivered to a loudspeaker by the amplifier.

RADIO FREQUENCY. A frequency higher than audio frequencies, but lower than heat or light waves.

RECTIFIER. A device, often a vacuum tube, for converting alternating current to direct current.

RUMBLE. A low-pitched noise originating in a record player or tape recorder, usually as a result of motor vibration transmitted to the turntable or tape drive mechanism.

SELECTIVITY. The ability of a radio receiver to reject undesired signals.

SENSITIVITY. The characteristic of a radio receiver or tuner which determines the minimum signal required to produce a desired value of output signal. Expressed in microvolts; a sensitivity of 5 microvolts is higher than one of 10 microvolts.

STYLUS. The "needle" in a phonograph pickup. The stylus vibrates as it rides along the record groove and conveys these vibrations to the pickup cartridge. Plural: styli.

TANGENT. A phonograph pickup is said to be tangent when its long axis is tangent to the record groove.

TRACKING. This refers to the proper seating of the stylus in the record groove so that distortion and groove wear are at a minimum.

TRANSFORMER. An electrical device consisting of two or more coils of wire wound close to each other. The coils may or may not be wound on an iron core, depending upon the frequencies to be handled. Iron cores are used at lower frequencies, as in power transformers and output transformers.

TRANSPORT, TAPE TRANSPORT. The mechanical portion of a tape recorder which performs the function of carrying the tape past the record, playback, and erase heads.

TREBLE. Notes at the high frequency end of the musical range.

TWEETER. A loudspeaker for reproducing the treble, or high frequency, notes.

WOOFER. A loudspeaker used for reproducing the bass, or low frequency, notes.

WOW. A change in musical pitch resulting from speed variations in a record player turntable or in a tape recorder mechanism.

2 Loudspeakers

Most books on high fidelity begin (once the preliminaries have been disposed of) with a discussion of the various components in this order: record player or radio tuner, preamplifiers, amplifier, loudspeaker, and loudspeaker enclosure. This arrangement is perfectly natural because it follows the progression of the music from its source to its conversion into sound waves. We shall depart, however, from it for several reasons, the first of which is that it is not too well adapted for use in a "how-to-do-it" type of book. The second reason, and not at all secondary, is that the loudspeaker is the most important single component of a quality music system. In fact, a mediocre system will often show a remarkable improvement when a good speaker is substituted for a poor one. Conversely, it is possible to ruin the fidelity of an otherwise excellent system by the use of an inferior speaker.

Speakers suitable for use with wide range music systems are available at prices ranging from about twenty dollars to as much as several hundred dollars. Regardless of price, almost all present-day speakers produced by reputable manufacturers are well designed and constructed and all will afford results commensurate with the selling price. That is to say, there is little to choose from between one manufacturer's fifty dollar speaker and another brand in the same price bracket.

While it is to be expected that a one hundred dollar speaker will afford definitely better reproduction than a twenty-five dollar one, it is also true that speaker *A*, at one hundred dollars, will not give you reproduction three hundred percent better than that of speaker *B*, at twenty-five dollars. However, the steep upward trend in speaker prices as compared to the less sharp increase in quality obtained is cheerfully borne by many extremely critical listeners.

Loudspeaker Fundamentals

Practically all modern loudspeakers are of the permanent magnet dynamic type. The dynamic speaker is one whose diaphragm is energized by a moving coil attached to it. The designation "permanent magnet" refers to the fact that the speaker's magnetic field is supplied by a permanently magnetized and highly magnetic alloy, in place of the iron-cored coil used in earlier dynamic speakers.

The permanent magnet dynamic speaker consists of a very light coil of wire cemented to a cone-shaped diaphragm. The coil is free to move in the field of a strong permanent magnet. Electrical impulses from the amplifier are applied to the coil, referred to as

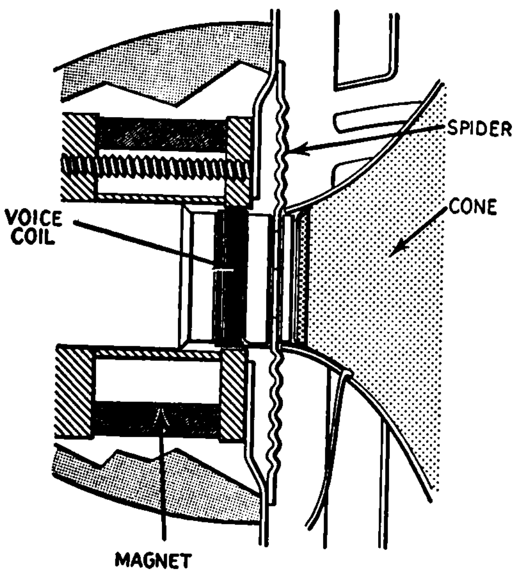


Fig. 2-1. Permanent magnet dynamic speaker.

the voice coil. Because the impulses from the amplifier are constantly changing in direction and intensity, a changing magnetic field is set up in the voice coil. This changing magnetic field reacts with the steady magnetic field produced by the permanent magnet, with the result that the voice coil is alternately attracted and repelled by the permanent magnet. Each time the voice coil is repelled, it moves forward and the cone moves with it. This forward movement compresses air in front of the cone and produces sound waves. When the voice coil is attracted by the permanent magnet it moves backward, and so does the cone, compressing air at the rear of the cone. In this way, the electrical impulses are converted into sound waves. The rapidity of vibration of the voice coil and cone depends upon the pitch or frequency of the signal from the amplifier. The distance that the cone moves is dependent upon the strength of the signal.

Figure 2-1 is a cross-sectional view of a popular make of high fidelity loudspeaker. The voice coil is wound on a cylindrical form; its diameter depends upon the make and application of the speaker. The cone, made of a special paper, is attached to the outer end of the voice coil form. The periphery of the cone is cemented to and supported by the metal frame of the speaker. Centered within the voice coil form is the permanent magnet. The gap between the inner surface of the voice coil form and the outer surface of the cylindrical magnet must be extremely narrow in order to provide a dense magnetic field. To concentrate the field further, the yoke provides a return magnetic circuit; in other words, an almost unbroken magnetic circuit exists from one end, or pole, of the magnet to the other. The only break is the narrow, ring-shaped gap in which the voice coil is centered. To prevent the voice coil from shifting and possibly rubbing on either the magnet core or the yoke, it must be adequately supported and centered, and the support used must be quite flexible if the voice coil is to be free to move. This is accomplished by the suspension, or "spider" as it is often called; this is generally a thin disc of resin-impregnated cloth and it often has annular corrugations to improve flexibility. With the spider cemented in place along its circumference, the voice coil is free to move but only in the direction of the magnet's long axis, never sidewise.

The Magnet

The strength of the permanent magnet determines, to a large degree, the efficiency of the speaker as well as the quality of reproduction. It is obvious that the magnet should supply as intense a field as possible, for then the voice coil movements will be large for a given signal. Furthermore, this will increase the power handling capacity of the speaker as well as the faithfulness of reproduction. Ordinary steels may be used, but pound for pound are far less effective than the newer alloys. A few ounces of Alnico—an alloy consisting chiefly of iron, aluminum, nickel, and cobalt—makes a far more powerful magnet than one using several pounds of steel.

Magnets used in high fidelity speakers range in weight from perhaps six ounces to as much as several pounds and, as a rule, the weight of the magnetic material has some bearing on the performance of the speaker, provided that the material has been used efficiently. Manufacturers' catalogs usually specify the weight of the Alnico magnet, and this will give you some indication (although not an infallible one) of the quality of the speaker.

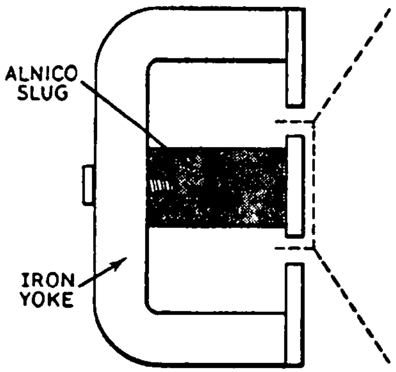


Fig. 2-2A. Slug-type magnet.

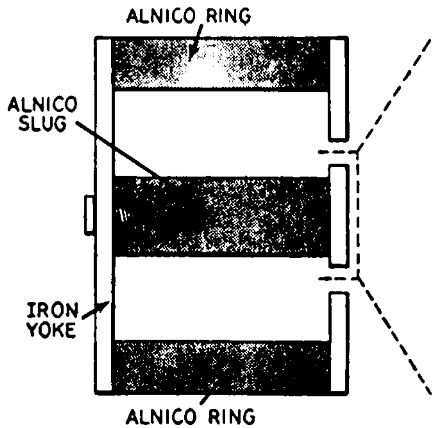


Fig. 2-2B. Ring-type magnet.

A majority of speakers use the Alnico V alloy and the material is generally in the form of a cylindrical slug, as shown in Figure 2-2A. Notice that a U-shaped piece of iron (the yoke) completes the magnetic circuit, except for the voice coil gap described above. Some speakers utilize Alnico in the form of a hollow cylinder, as illustrated in Figure 2-2B. The principle is exactly the same, except that since both the center slug and the ring are of Alnico almost the entire magnetic circuit is of this material. At first glance it would seem that this type of construction is superior because a greater weight of Alnico is used. This would be true, except for the fact that in some instances another, less powerful alloy—Alnico II—is used, and a large amount is therefore necessary. In judging a speaker magnet, then, you must take into account the type of Alnico used as well as the weight of the magnet.

The Voice Coil

Any tendency of the voice coil to warp or otherwise alter its contour may result in contact between the voice coil form and the magnet core or between the coil and the yoke. When this occurs a rasping noise will develop that renders the speaker useless until it has been taken apart and the voice coil, cone, and suspension assembly replaced. The cost of such work may be considerable and, for a low-priced unit, may warrant replacing the entire speaker. Many speakers now use aluminum-base voice coils that are not subject to warping; this is a feature to look for when making a selection. Some voice coils use flat instead of round wire, wound on edge. This permits the use of a greater number of turns and increases the efficiency of the speaker. Incidentally, it is possible for one or more turns on a voice coil to loosen with the result that the loose wire vibrates independently and sets up a rattle. There is no way of correcting this, except to replace the cone assembly, as mentioned above. Since speaker manufacturers' specifications generally describe the voice coil construction, you will do well to read them carefully. Although many voice coils are wound with wire having enamel insulation, silk-insulated wire seems to be better, for the silk affords a better grip for the cement or lacquer used to hold the turns in place.

In examining the various makes, you will find considerable variation in voice coil diameters. The usual range is from one to two inches, although in a few instances the diameter is greater. A larger voice coil diameter permits the use of a shallower cone and this improves the action of the speaker on low notes.

Speaker Power Rating

As mentioned in the first chapter, an amplifier should be capable of delivering at least 10 watts to the loudspeaker. This means, obviously, that the speaker must be able to handle at least that amount of power without distortion or rattling. You will probably have no difficulty selecting a speaker having ample power handling capacity, for even the cheaper wide-range units are rated at 10 watts; many are rated at 25 to 30 watts. However, if you have a choice between two speakers and all other considerations are equal, by all means select the one having the higher power rating.

It is well to bear in mind that, when two or more speakers are connected to a single amplifier, the power will be divided between them, although not necessarily evenly. Thus, when a total of 10 watts is applied to two identical speakers, each need be capable of handling only 5 watts. Again, this depends upon whether or not each speaker is to deliver the full range of frequencies. In multiple speaker installations where one speaker is used only for the lower frequencies and the other for the high notes, the low frequency speaker must be able to handle a larger share of the power.

Voice Coil Impedance

This is a point that is often misunderstood and neglected by listeners who have little technical knowledge. The impedance of a voice coil is dependent upon a number of factors, including the diameter of the coil, the number of turns, and the wire diameter. You need not concern yourself with the technicalities of impedance, but you should bear in mind that the impedance at the output terminals of the amplifier should match that of the speaker voice coil as closely as possible.

The voice coil impedance of a speaker will usually be stamped on the speaker frame or on the shipping container. If not found in either of these places it can be obtained from the manufacturers' literature or from an instruction sheet which you may have received with the unit. The typical commercially made amplifier is arranged so that speakers having different voice coil impedances may be connected to it. In such cases there will be more than two output terminals. One is likely to be marked "C" or "Common." One speaker voice coil terminal is to be connected to this point, regardless of voice coil impedance. (See Figure 2-3.) The remaining amplifier terminals will probably be marked "4," "8," and "15" (in a few instances the designation "16" is used instead of "15"). If your speaker has an 8-ohm voice coil, its remaining terminal must be connected to the amplifier terminal marked "8"; a 15-ohm speaker would be wired to the one marked "15." If you intend to use two or more speakers operating from one amplifier, then the total impedance will no longer be that of a single unit. There are two methods of connecting multiple speaker systems; series and parallel. The series arrangement is shown in Figure 2-4A. If two or more speakers are wired in this way, the total impedance will be the sum

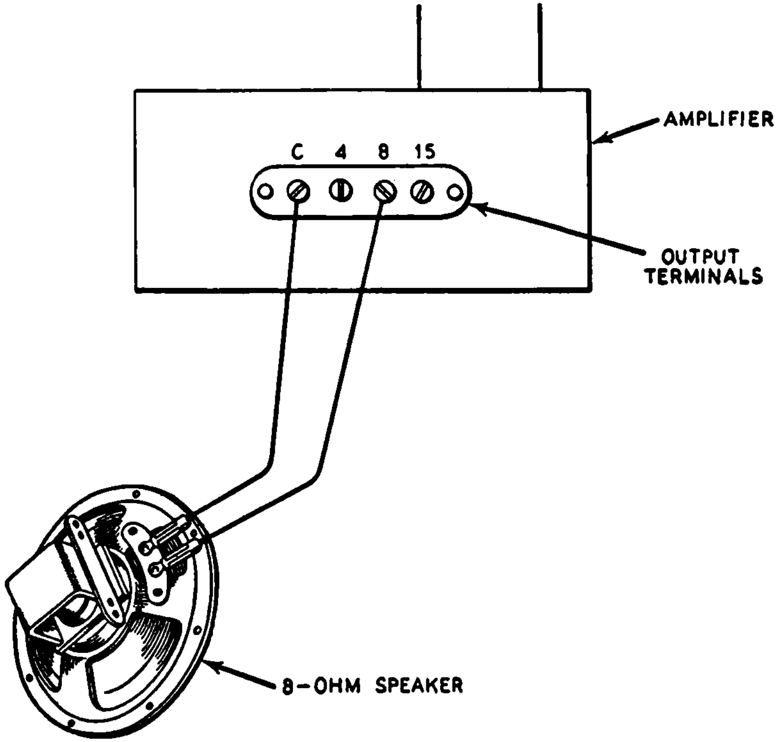


Fig. 2-3. Connecting loudspeaker to amplifier.

of the individual impedances. Figure 2-4B shows the parallel method; when two identical speakers are so wired, the total impedance will be half that of one unit.

Speaker Location

The location of your speaker system will probably be determined by such factors as convenience and room layout, but if you are willing to make such considerations secondary to good listening you will be interested in the ideal location. Experiment has shown that this location is in a corner of the room. An even more desirable location is in a corner, with the speaker close to either the floor or the ceiling. With a speaker placed in a corner, the two adjacent walls act as the sides of a horn and aid materially in radiating the sound, especially the lower tones. When, in addition, the speaker is placed close to the floor or ceiling, we have added another side to the horn.

This explains why corner-type speaker cabinets are in current favor. Even though you do not use a triangular cabinet, some experimentation may be profitable. Moving an ordinary square or rectangular speaker cabinet so that it occupies a position at the junction of two walls will usually improve the bass.

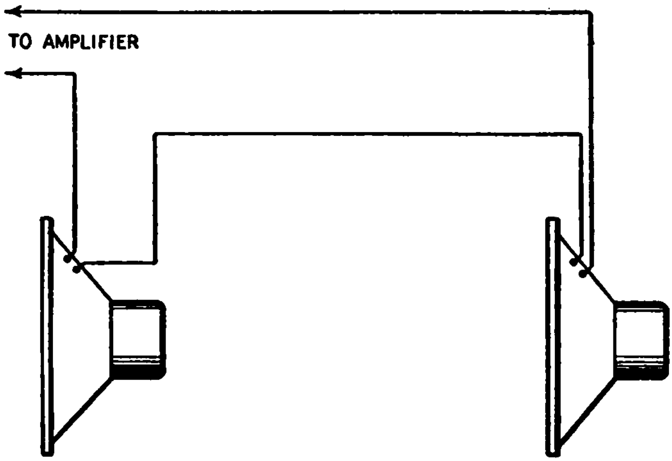


Fig. 2-4A. Loudspeakers connected in series.

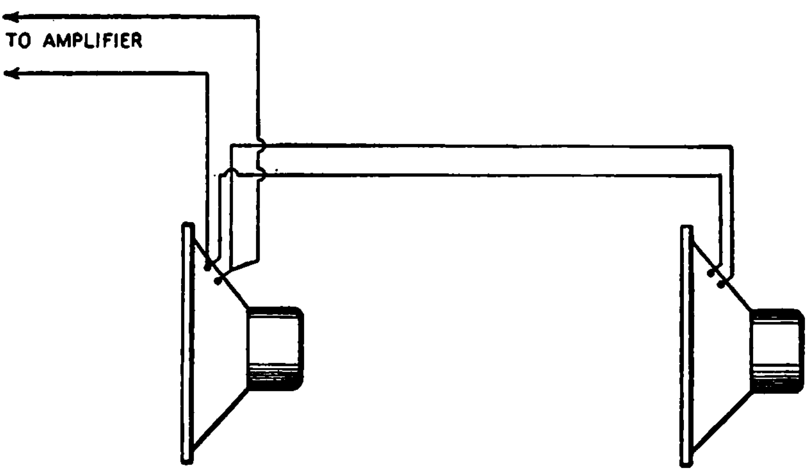


Fig. 2-4B. Loudspeakers connected in parallel.

The distribution of the higher frequencies falls off rapidly as the angle between the axis of the speaker and the listener widens. In other words, you will hear more of the higher frequencies if you are seated directly in front of the speaker than you will at any position to one side. It is important to note that this applies to the vertical relationship between loudspeaker and listener as well as the horizontal; in many installations the loudspeaker is placed too low for best results. You will soon discover that the high frequencies drop off much more rapidly than the lows as you get farther from the loudspeaker, but, of course, there are limits to the distance between speaker and listener, especially if the system is operated at a relatively high volume level. If you are an extremely critical listener, try turning your head sidewise toward the loudspeaker and note the pronounced increase in highs. Cupping your hand over your ear will have a similar effect.

Selecting a High Fidelity Loudspeaker

It is quite unlikely that you will be able to hear the loudspeaker in your own home before purchasing it. This is unfortunate, for it is well known that room acoustics have a great deal to do with the quality of reproduction. Furthermore, it is usually unwise to select a speaker on the basis of a short listening period. However, if you do not intend to build your own components such as speaker enclosure and amplifier but will use commercially made equipment, there is at least a partial solution to the problem. Most large radio distributors, high fidelity showrooms, and even some large record dealers now have facilities for conducting what may be called the "A-B" test. Display rooms of this type usually have a long shelf on which are placed record players of different manufacture; a second shelf holds amplifier equipment, and finally there will be a row of loudspeakers in enclosures. By a switching arrangement, any combination of record player, amplifier, and speaker may be heard. When you have chosen your amplifier and player, have the salesman demonstrate that player and amplifier connected to the various speakers you are considering. Do not use snap judgment, but insist upon a prolonged listening test; if necessary, return for a second or third hearing.

Since the wide range music system must cover a broader range of tones than the average mass-produced radio or phonograph, the loudspeaker system must be capable of radiating the full range of frequencies. Designing a single speaker to do this has challenged engineers and, while it is possible to build a single speaker to cover the full range from 50 to 15,000 cycles per second, it admittedly is difficult. However, there are many single speakers available that extend the range to perhaps 12,000 or 13,000 cycles per second. We shall discuss these speakers later on.

Almost everyone is aware that small objects vibrate faster than larger ones. In a common household door chime, for example, you have probably observed that the smaller tube or bar produces a higher note than the larger one. The same thing applies to loudspeakers; the larger the cone, the lower its range of tones and, of course, a large cone is essential if we are to realize the magnificent bass response that true high fidelity systems are capable of. The use of a large cone, while necessary for good bass reproduction, may often result in poor highs, and it is usually more desirable and efficient to design a single speaker for a limited range of frequencies. This explains why the majority of high fidelity enthusiasts prefer to use two speakers, a large one for the low notes and a smaller one for the highs. But a single speaker with fairly good high frequency response is far less expensive and you may want to select this type, especially if you are not extremely critical and if cost is an important factor. There are several types of wide range speakers, and we shall discuss their features in the order of increasing cost.

Extended Range Speakers. As mentioned before there are available several makes of speaker that afford a range of frequencies considerably wider than expected of units in commercial radios or phonographs. They are often described in manufacturers' literature as "extended range" units and, although they do not afford true high fidelity in the strict application of the term, they do make possible at least an approach to high fidelity at very moderate cost. To give you some idea of their capabilities, we may say that it is rare to find a speaker in an ordinary radio receiver that will reproduce notes higher than 6000 cycles per second. Incidentally, such speakers, as a class, generally have a limited bass response, extending no lower than about 100 cycles per second, although this is often due to the limited volume of the enclosing cabinet. By contrast, even the cheapest extended range unit is capable of covering 50 to 10,000 cycles when used in a suitable enclosure. In fact, some types will cover 45 to 15,000 cycles. Earlier in this chapter we pointed out that the listener should compare the rapid rise in speaker prices with the increase in frequency coverage. Of course, the fact that the more expensive speakers do reproduce the full range with far less distortion than the cheaper types must also be taken into account. Whether the freedom from distortion and increased frequency coverage are worth the price is something you will have to decide after extended listening tests.

Extended range speakers use a variety of methods for increasing the response, as well as for better radiation of the highs. In one model, an aluminum alloy dome is fastened over the front of the voice coil. On low notes, the entire cone vibrates, but at high frequencies only the dome is in action. In another make, a multicellular horn, placed at the front of the unit, gives better dispersion of the highs over a wide angle. A spiral horn is used in another type, and a fourth uses a miniature subcone in addition to the main one. No one of these methods offers a marked advantage over the others and the final selection will depend upon critical comparison.

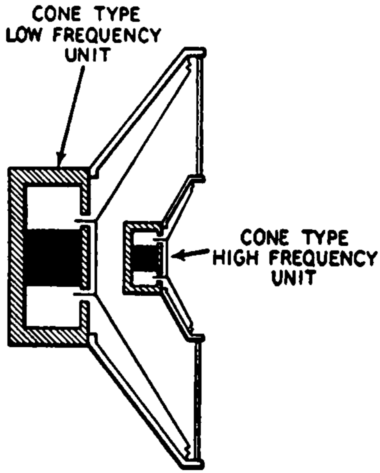


Fig. 2-5. Coaxial speaker.

Priced at under twenty-five dollars, the average speaker in this class is well designed and constructed and when used in an adequate enclosure will give good results. Typical of this group are the General Electric models 1201D and 1203D (the latter uses a smaller magnet than the first type), the University model 6200, and the RCA model SL-12.

Duplex or Coaxial Speakers. Speakers in this class follow one of two distinct design patterns. In one design, the unit consists of two separate cone speakers, one small and the other large. The smaller speaker is mounted at the front of the larger, as shown in the cross-sectional view of Figure 2-5. In the other design, a horn-type unit is substituted for the small high frequency cone speaker. This is illustrated in Figure 2-6, and you will note that the neck of the horn passes through the center of the larger speaker's magnet. The basic principle is about the same in both versions: the small unit is designed to handle the highs and the large cone produces the lows.

It is evident that the duplex speaker offers the advantage of two separate units mounted on a single frame, and there is thus a saving in space and added convenience in mounting. One problem encountered in the design of such units is the proper diversion of the high and low tones to the two speakers. This is accomplished by means of electrical filters known as frequency dividing networks,

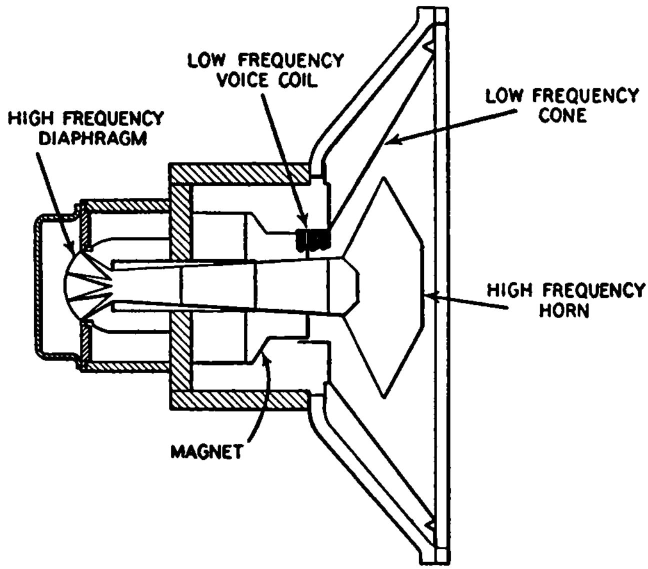


Fig. 2-6. Duplex speaker with horn-type high frequency unit.

which will be discussed in greater detail when we take up the matter of multiple speaker installations. In some dual speakers the dividing network is mounted directly on the speaker frame, but in other types it is a separate unit and is then placed at the bottom of the speaker enclosure.

Prices of speakers in this category show a rather wide variation. As a rule, units employing a cone-type high frequency unit are less expensive than those equipped with a horn-type driver. The price range is from about twenty-five dollars to as high as two hundred. In most cases the price includes the dividing network, but in a few instances it does not. Some experts are of the opinion that coaxial speakers in the very low price range offer little improvement over a good extended range unit. Examples of duplex speakers in the lower price class are: General Electric model A1-400, Jensen models K-210, and K-310A. The first two mentioned are 12-inch speakers, while the third is a 15-inch. Near the upper end of the price range we find such superlative units as the Altec-Lansing model 604C.

Multiple Speaker Installations. Although the duplex speaker offers a convenient method of obtaining wide range response in a minimum of space, there is much to be said for the use of two or more separate, high quality units. It is quite possible that a system using several entirely separate units provides better realism because the sound then does not emanate from a single point source. Of course, it is true that for vocal music a single source is the more desirable, but this does not apply to orchestral music. In fact, some listeners use several rather widely spaced speakers to achieve just this effect.

You will find it a comparatively simple matter to purchase separate speakers and assemble them into a complete system. Any such combination will, of course, include a large cone-type unit for the low frequencies, and you will have a choice of either a horn-type unit or one or more small cone speakers for the high end of the range. We should mention that the first-mentioned combination will almost always be the more expensive.

In any such multiple installation the low frequency speaker is referred to as the "woofer" and the high frequency unit as the "tweeter." In some factory-assembled combinations the principle is extended to include one or two low frequency drivers, a speaker for the mid-range, and a tweeter for the extreme highs.

Concerning the frequency dividing network, you will find it possible to buy high and low frequency units of the same make, together with a dividing network expressly designed for the combination. Or, if cost is of prime consideration, you may want to make up your own network of simple design, following the instructions to be given later.

Among the manufacturers who offer excellent two-way combinations and dividing networks for use with such combinations are: Altec-Lansing, Electro-Voice, University, and Jensen. To avoid confusion, we must mention that you will often find the frequency dividing network referred to as a crossover network.

If you wish to assemble your own two-way system based upon the use of cone-type speakers for both woofer and tweeter you will need some information concerning the basic principles of such combinations. Some experimenters select a large speaker, known to have good bass response for the woofer and one or more smaller units having good high frequency response for the tweeter section, and connect them together with no attempt to provide diversion of the highs and lows to the appropriate speakers. Such arrangements can be made to work fairly well and certainly are far less expensive than more elaborate two-way systems. One objection, however, is that care must be taken not to overload the smaller speaker or speakers since the lower notes are not being diverted from them. In selecting the components of such a combination, the woofer should be as large as possible, at least 12 and preferably 15 inches in diameter. A high quality 5-inch speaker will serve as the tweeter. The method of wiring the two units will depend upon their individual impedances, and the parallel method is to be preferred. The reason for this preference is that if one speaker fails the system will still remain in operation, although the over-all frequency response will be seriously affected. If parallel wiring is inconvenient on account of the total impedance, then the speakers may be connected in series. For illustrations of both series and parallel circuits, refer to Figures 2-4A and 2-4B.

The next item to be considered is the proper "phasing" of two or more speakers. If this were to be disregarded, the cone of one speaker might move outward at the same instant that the cone of the second moved inward. This would not only result in partial cancellation of the air waves set up but would also have the effect of shifting the apparent source of the sound.

Phasing of Loudspeakers

Loudspeakers must be phased individually, before they are connected in the circuit with other units; consequently, whether they are connected in series or parallel has no bearing on the phasing operation. If, for some reason, the two speakers are arranged at opposite sides or ends of a room, then phasing becomes of lesser importance, but if they are to be mounted in a single housing or enclosure so that they point in the same direction you must be careful to connect them so that when one of the cones moves forward, the other does likewise.

The simplest way to phase two speakers is to apply a direct current to each of the voice coils in turn, so that the respective cones are caused to move backward or forward. Of course, it is important that the speakers be located so that the motion of the cones can be observed while the work is being done.

To do the job, you will need an ordinary dry cell; this may be of the flashlight variety or the larger type commonly used for doorbells and the like. The center terminal of any dry cell is the positive connection; the terminal located at the circumference of the metal container is negative. (This negative connection is the bottom of the container if a flashlight cell is used.)

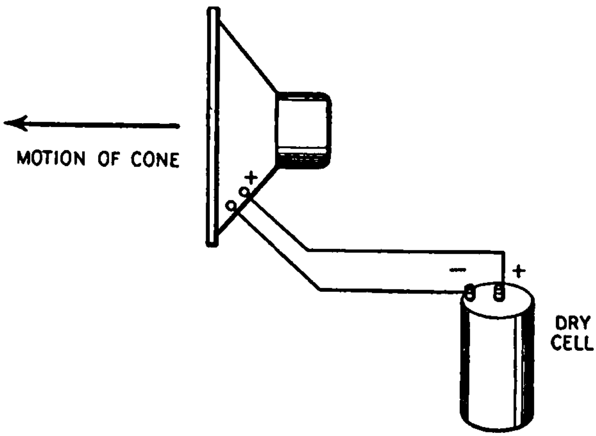


Fig. 2-7. How to phase a speaker.

Connect the terminals of the dry cell to the speaker voice coil terminals as shown in Figure 2-7. As you close the circuit you will hear a distinct click and the loudspeaker cone will move either backward or forward. Observe the direction of motion carefully; if the cone happened to move backward, reverse the connections to the dry cell so that it now moves forward. Now determine which of the voice coil terminals is connected to the positive pole of the battery, and mark this voice coil terminal with a "+." Check the remaining speaker or speakers in exactly the same way, marking the appropriate terminals with "+" designations. If the speakers are to be connected in parallel, wire all of the "+" terminals together. For a series connection, wire the "+" terminal of one speaker to the unmarked terminal of the next, and so on.

All of the foregoing applies to the phasing of cone-type speakers only. If a horn-type tweeter is used, the procedure is different. With the woofer connected to the amplifier, play a record or tune in a radio program and adjust for average room volume. Now connect the tweeter wires in the circuit, then try reversing their connections. You will notice that the volume appears to be greater with the tweeter connected one way. When the maximum volume has been obtained, the speakers are properly phased.

An alternative method of phasing speakers may be used and is useful in cases where the speakers are already enclosed in cabinets and the movement of the cones cannot readily be observed. Turn on the system and place the cabinets a few inches apart, facing each other. Now try reversing the wires to one speaker. When you find a connection that apparently increases the total sound output, the speakers are properly phased. If the output drops off when the wires are reversed, the speakers are incorrectly phased.

Frequency Dividing Networks

The simplest type of divider network consists merely of a condenser connected between the high frequency speaker and the amplifier. A condenser passes high frequencies more easily than lower ones, hence the lows are diverted from the tweeter. The basic

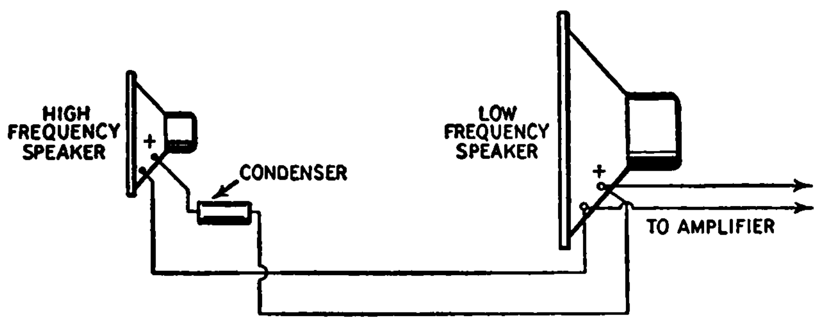


Fig. 2-8. Simple condenser-type frequency divider.

circuit is illustrated in Figure 2-8. The size of the condenser is dependent upon two things: the voice coil impedance of the speaker and the desired "crossover frequency." Crossover frequencies used commercially range from 400 to 2000 cycles. If, for example, a 1000-cycle crossover frequency has been selected, all frequencies below that point will be reduced in intensity, insofar as the tweeter is concerned. You will note that no provision is made in this simple, low-cost arrangement for diverting the high frequencies from the woofer. Although commercial systems do make such provisions, the dividing network then becomes more elaborate and more costly. We shall take up such methods later on.

There are formulas for determining the value of the condenser, but to save you the trouble of making the calculations a list of condenser sizes for various crossover frequencies and for several different voice coil impedances is given below.

Voice Coil Impedance (ohms)	*Crossover Frequency (cycles per second)*	*Condenser Capacitance (microfarads)*
4	1,000	40
4	1,500	25
4	2,000	20
8	1,000	20
8	1,500	12
8	2,000	10
16	1,000	10
16	1,500	6
16	2,000	5

It will be noticed that at lower crossover frequencies the condenser capacitance becomes greater; also, as the speaker voice coil impedance becomes higher the necessary condenser capacitance is smaller.

Certain audio engineering textbooks have pointed out that the electrolytic type of condenser should not be used in a crossover network. Nevertheless many high fidelity enthusiasts have used them and report good results. In a few cases they have been used in factory-built networks. There is no question but that the best type of condenser to use is oil (with paper as a second choice) if best results are desired. Unlike a condenser with paper or mica dielectric, an electrolytic condenser has considerable leakage current and it would seem that this might cause distortion.

On the other hand, oil and paper condensers have the decided disadvantage that for a given value of capacitance they are far larger than an equal value of electrolytic condenser, and are likely to cost more. For these reasons you may want to try electrolytics but bear in mind that the ordinary type cannot be used. Ordinary electrolytic condensers are intended for use in direct current circuits only and the voltages present in the output of the amplifier are alternating in character. Consequently, a "non-polarized" condenser must be used. Electrolytic condensers of this type are to be found in the motor starting circuits used in refrigerators and in other appliances.

You may make your own non-polarized condenser by connecting together two electrolytic condensers, each having twice the capacitance recommended in the table above. Connect the two condensers as shown in Figure 2-9. Be sure that you connect either two negative terminals or two positive terminals together. Electrolytic condensers intended for use in radio receivers and available at all large radio supply stores are quite satisfactory for the purpose. All such condensers have their terminals (or connecting wires) clearly marked, either with "plus" and "minus" signs or with the words "positive" and "negative." Every electrolytic condenser is also marked with its maximum voltage rating. As the amplifier output voltages rarely exceed 25 or 30 volts, any voltage rating in excess of this will be ample. There is no harm in using a condenser of higher voltage rating than necessary.

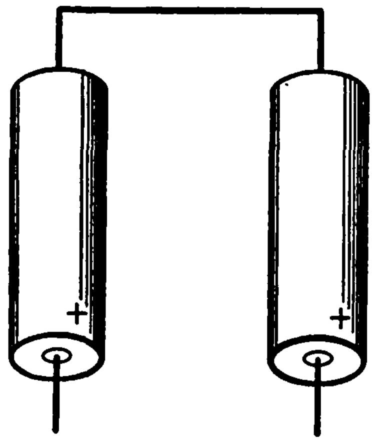


Fig. 2-9. How to connect two electrolytic condensers to make a non-polarized condenser.

You may want to try making up a more effective type of divider network for your speaker system; the work is not too difficult and the results will be noticeably better than with the simple condenser-type filter just described. As you will note from the diagram, Figure 2-10, it consists of a condenser and an inductance. An electrolytic condenser may be used as in the simple divider just described. The inductance may be home-made.

Briefly described, the filter functions as follows: a condenser passes higher frequencies much more easily than it does lower ones. Consequently, the high notes are permitted to pass through the condenser to the tweeter, but the lower notes are diverted away from it. On the other hand, the inductance, or choke coil, passes the lower tones more readily; these are allowed to pass to the woofer but the higher notes are blocked off. The advantage of such an arrangement will be evident at once. Since the high notes are kept out of the woofer, there are no longer two sources of high frequency energy as before.

The first step in making this type of divider is to select the crossover frequency—the point at which the sound energy divides. The

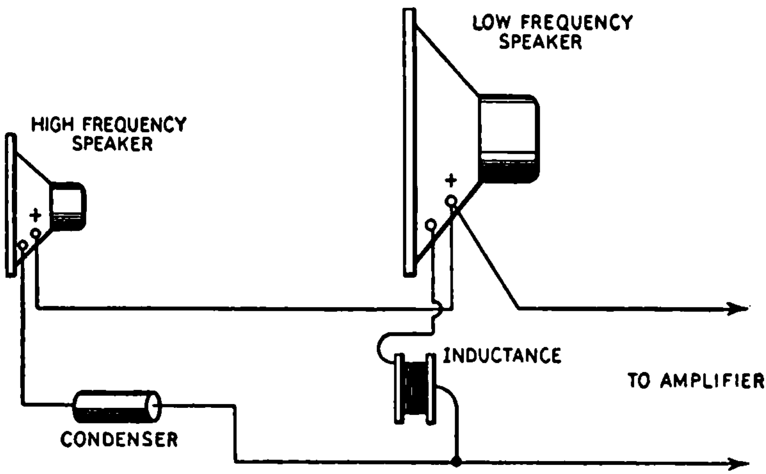


Fig. 2-10. Condenser-inductance type of frequency divider.

selection of a crossover frequency will depend on several things, including the design of the speakers at hand and, as stated before, speaker manufacturers employ crossover points ranging from 400 to 2000 cycles. You will probably find that a crossover at 1000 cycles is quite effective in most cases. Remember that a divider network does not sharply cut off all frequencies above and below a certain point.

The crossover frequency selected determines the values of the condenser and the inductance. If, for example, you are using an 8-ohm tweeter, the condensers must have a capacitance of 20 microfarads. Two 40-microfarad, 150-volt condensers (these are commonly stocked sizes) are then connected back-to-back as described and illustrated above. Should your tweeter happen to have a 16-

ohm voice coil, then two 20-microfarad units must be connected together to make up a 10-microfarad condenser.

You are now ready to construct the inductance. This consists of a number of turns of wire on a cylindrical winding form; the exact number of turns will depend upon the voice coil impedance of the woofer and the desired crossover frequency. If, for example, you have selected a 1000-cycle crossover point, you will need a coil having 160 turns for a 4-ohm speaker and 225 turns for an 8-ohm

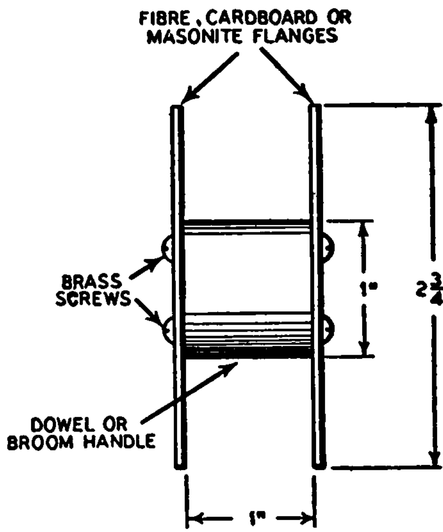


Fig. 2-11. Winding form.

speaker. For a 16-ohm speaker, 310 turns must be used, but the winding form must then be larger than for the first two types.

Figure 2-11 shows the winding form. For 4- and 8-ohm speakers use a one-inch length of broom handle or one-inch diameter wood dowel for the spool. For a 16-ohm speaker the diameter is the same, but the length must be increased to 1½ inches. The two flanges are made of heavy cardboard, fiber or thin masonite; for 4- or 8-ohm speakers they should be 2¾ inches in diameter, but for a 16- ohm speaker this dimension must be increased to 3½ inches.

For the winding you will need some No. 17 enamel-insulated magnet wire; this may be purchased from any large electrical supply house. It is quite likely that the wire will be sold in quantities of one pound; you will need a maximum of ¾ pound for any of the coils mentioned.

Assemble the form by fastening the flanges to the spool with brass woodscrews; do not use iron screws or nails, since they are likely to create distortion. Begin winding by punching or drilling a small hole in one of the flanges, close to the spool. Pass the end of the wire through this hole from the inside. In winding the wire, be sure that each turn is as close as possible to the one next to it, and that there are no kinks or bends in the wire. Each layer must lie perfectly smooth on the one preceding it. When you have wound on the correct number of turns, punch or drill another small hole in one of the flanges, close to the end of the winding. Pass the end of the wire through this hole and the inductance is completed.

It should be noted that the sizes given for condenser and inductance are not extremely critical; even though the values do not work out exactly as specified, the only difference will be a shift in the actual crossover frequency which is relatively unimportant in a unit of this kind.

Before leaving this topic, your attention is directed to one point. While the ordinary electrical crossover will do a satisfactory job in most cases, it does introduce a serious loss of power, which becomes even more serious in view of the inefficiency of most good high fidelity speakers; in many cases speaker efficiency is no more than 5 percent. If a crossover introduces a loss of no more than one decibel, then the effective power of a 25-watt amplifier is reduced to 20 watts.

In addition, losses due to this (and other) causes frequently have effects that are still more serious. For instance to compensate for losses it may be necessary to operate the amplifier or preamplifier at a higher level, and then such deficiencies as hum, turntable rumble, tape noise, and the like will become much more prominent.

Electronic Crossovers

The losses introduced by the usual electrical crossover have led to the development of an electronic crossover. While the usual electrical crossover is introduced between the amplifier and the speakers, this unit is inserted at the input of the basic or power amplifier, as shown in Figure 2-12. You will note that two basic amplifiers are needed, one for the highs and midrange, and the second for the lows. This results in a relatively expensive installation but the manufacturer (Heathkit) claims that this cost is offset by a number of advantages. Most important among these is the feasibility of adjusting the system over a wide range to suit an endless variety of speaker combinations. This is made possible by a high pass filter employed in combination with the high-frequency amplifier and a low pass filter placed ahead of the low-frequency amplifier. In each case the cut-off point of the filter is adjustable by means of a switch to 100, 200, 400, 700, 1200, 2000, or 3500 cycles. A

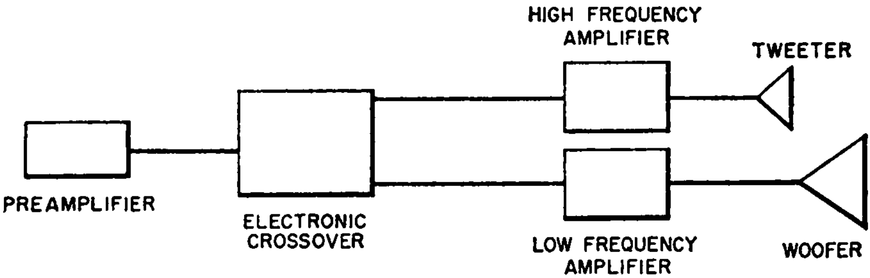


Fig. 2-12. Block diagram of electronic crossover.

second advantage is that a change in speakers may be accomplished without any change in the crossover other than a readjustment of the switches.

Other Crossover Systems

In addition to the simple electrical crossovers already described, other systems have been devised that offer one or more advantages. In one instance, crossover is accomplished, not between amplifier and speaker, but ahead of the output transformer. The advantage of this method is that smaller capacitances may be used.

In still another method, the frequency range is divided between two amplifiers, as in the electronic crossover, but with the difference that the crossover unit here is a conventional electrical type, rather than electronic. Although this system still requires filters, they operate more efficiently since they are placed ahead of the amplifiers and therefore do not have to handle any appreciable power. Part of the cost of this system is offset by the fact that in both amplifiers cheaper output transformers may be used since the transformers are required to cover only a limited portion of the frequency range. In a modification of this system, a single amplifier is used having dual output stages, with the frequency range divided between them. In this case, the filters are placed ahead of the output stages.

Amplifier-to-Speaker Wiring

In many home music systems the loudspeaker is an enclosure separated from the rest of the equipment. The distance between speaker and amplifier will depend upon the size of the room and its layout. Questions then arise concerning the maximum allowable distance between the two and the most suitable type of wire to use.

There will be a natural tendency to use an excessively small diameter wire, since it is easier to conceal. Resist this tendency, otherwise you will introduce losses that may have a serious effect on the performance of your equipment.

The maximum length of the speaker line and the size wire to be used will depend upon voice coil impedance. For the average 8-ohm speaker, ordinary lamp cord may be used for a run up to about 25 feet without introducing serious losses, and this should take care of most rooms.

As the voice coil impedance becomes higher, a smaller wire size may be used for a given distance. For example, for the same run of 25 feet, No. 22 wire may be used if the voice coil impedance is 16 ohms, but for a 4-ohm speaker the wire size must be increased to No. 16. The safest plan is to use a larger size than is actually needed. Of course, the wire should be stranded, rather than solid, to afford maximum flexibility.

3 Loudspeaker Enclosures

THE SPEAKER ENCLOSURE is more than just a convenient housing for the speaker; its primary function is to enhance the bass response and a well-designed enclosure can contribute a great deal to the quality of your home music system. On the other hand, a poor enclosure can impair the quality of even the best system.

All speakers require some kind of enclosure, or "baffle." As explained earlier, the movement of the cone or diaphragm alternately compresses and rarefies air at the front and rear of the speaker. When the cone moves forward, air is compressed at the front and rarefied at the rear; on the other hand, when the cone moves backward, just the reverse is true. It will be seen, then, that the cone movement sets up air waves at front and rear.

The basic function of a speaker is to set air waves in motion; the greater the air disturbance, the more efficient will the system be. Now, if we were to suspend a speaker in free air with no baffle or enclosure, we would find that the air pressure created at one face of the cone would cause air to move toward the opposite face, where the air has been made thinner. If the air path is relatively short, the front and back air waves tend to cancel, with the result that little air is actually set in motion. This is true because the front and back waves, at any given instant, are equal and opposite. The relatively slight air disturbance would be particularly noticeable at the lower frequencies with the result that the speaker system would be deficient in low tones. We see then, that if the bass response is to be good, some method must be used to prevent interaction between the front and back waves that are set up. This is accomplished by the baffle, or enclosure.

We shall now proceed to a discussion of the various enclosures; following this we shall describe the construction of several different types. If you decide to build one of these, the type you select will naturally depend upon your preferences, the adaptability of the particular unit to the decorative scheme of your room, and your mechanical ability. Should you elect to purchase a ready-made enclosure, examine the construction carefully. It is well recognized that any speaker enclosure should be built of heavy materials and, in most instances, should incorporate interior bracing. The latter is not solely for the purpose of strengthening the enclosure, but to prevent its walls from vibrating. A good enclosure will be lined with some type of sound-deadening material. The kind of wood used and the exterior finish will, of course, be reflected in the selling price. Factory-made enclosures sell for $40 to $175, and it is obvious that you can save much of this expense if you are handy with tools. The painted enclosure to be described in this chapter should cost $10 or $12 for lumber, plus a small amount for accessories. If you prefer finishes such as mahogany, walnut, or blond, the lumber cost probably will be doubled.

The Flat Baffle

This is the simplest type of baffle and is illustrated in Figure 3-1. It consists merely of a large board with an opening at the center to accommodate the loudspeaker. The baffle operates on the principle that the front air waves set up by the speaker are forced to travel a longer path in order to reach the rear of the cone, and therefore there is less likelihood of cancellation of the two waves. The area of the baffle determines the effectiveness of the speaker at low frequencies, and the response limit may be lowered by using a larger baffle.

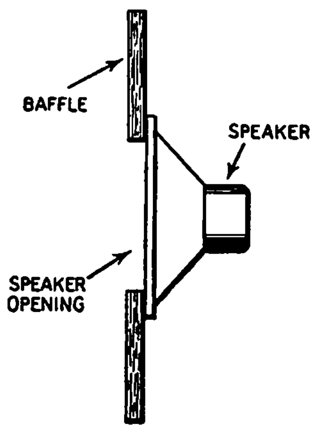


Fig. 3-1. Flat baffle.

Theoretically, the distance from the center of the cone at the front, around the edge of the baffle to the center of the cone at the rear, should be equal to one half wavelength of the lowest tone to be reproduced. The length of a sound wave may be determined by dividing the speed of sound (approximately 1100 feet per second) by the frequency. For a 60-cycle note the wavelength would be about 18.3 feet. A half wavelength would then be 9.15 feet. This would mean that the ideal flat baffle for reproducing this frequency would measure almost 4.8 feet from center to edge; a square baffle would have sides measuring 9.15 feet. Here we find the principal disadvantage of the flat baffle: its inordinate size. However, such dimensions are rarely used in actual practice. Sometimes compromises are made by using an approach to the specified dimensions or by folding the baffle, as will be described later.

The material used for any baffle must be as inert as possible to prevent it from vibrating. If the baffle vibrates, it will do so at its own resonant frequency and this will introduce an artificial peak in the response of the speaker. One way to prevent such vibration is to use heavy material. In no case should lumber thinner than ¾ inch be used. Plywood is suggested since it is readily available in large sheets. Crossbracing will also help to prevent vibration; two or more one-by-two-inch strips nailed or screwed to the rear face of the

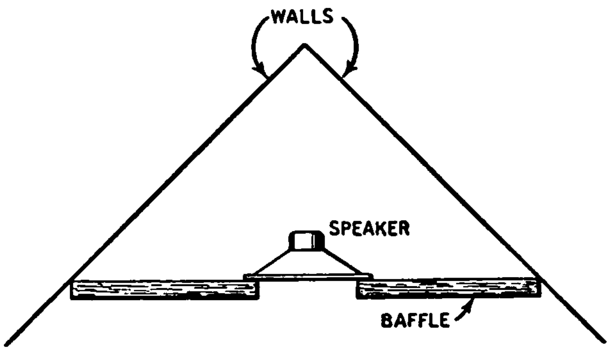


Fig. 3-2. Corner installation of flat baffle.

baffle will be quite effective. Like the enclosed baffles to be described later, the flat baffle should have its back face covered with a sound-absorbing material, such as Celotex, Kimsul, Fiberglas, or Ozite. These materials can be bought from building supply dealers and mail order houses. In most cases they come in convenient "bats" or "blankets." The sound-absorbing covering, regardless of the type used, should be one inch thick. Some of the materials listed cannot be held in place with ordinary nails; in such cases use roofing nails with washers under the nail heads. If the insulating material has a tendency to shred or come off in small particles, it is a good idea to spray it with a coat of lacquer, available in convenient pressure cans.

Flat baffles are not widely used in in home music systems, principally because they occupy more area than other types. However, it is sometimes convenient to employ such a baffle in the type of installation shown in Figure 3-2. In this case, the baffle has been placed so as to form the base of a triangle, with two walls of the room forming the remaining two sides.

The Open Box Baffle

As pointed out in the preceding section, the chief disadvantage of the simple flat baffle is the space requirement. A saving in space and, incidentally, the conversion of the baffle into a piece of furniture are effected by using the open box baffle shown in Figure 3-3. A little thought will reveal that such an enclosure is, in reality, a flat baffle that has been folded.

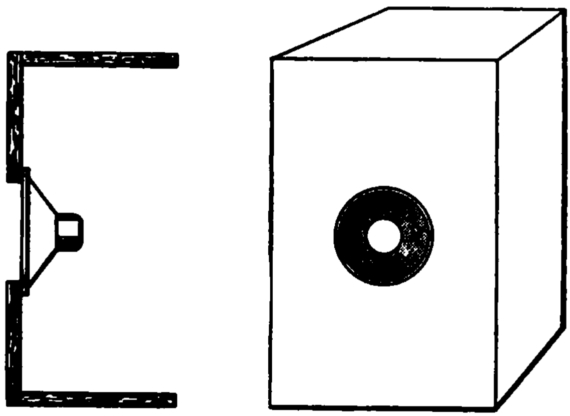


Fig. 3-3. Open-box baffle.

Earlier, we stressed the importance of using a large baffle if good low frequency response is to be expected. This applies equally well to the box type of baffle, except that now we are concerned with the volume or cubic content of a box rather than the area of a flat board. The minimum volume requirement for an open box baffle used with a standard 12-inch speaker is about 6 cubic feet. But, as mentioned in connection with flat baffles, there is no reason why you should use the minimum size if you have the space, and there is an excellent chance that the low frequency response of the speaker will be extended thereby. A square box baffle measuring 24 inches on a side and having a depth of 18 inches will enclose a total volume of exactly 6 cubic feet. If you prefer to use an enclosure having a rectangular front panel (and this is likely to be the case), then the dimensions (in inches) may be: height, 30; width, 24; and depth 14½. A rectangular front panel usually presents a more pleasing appearance. Incidentally, it is worth noting that most good furniture design is based upon application of the so-called "golden triangle." An object having a rectangular front appears to be more graceful when the longer side has a ratio to the shorter side of four to three. The diagonal of a rectangle of those proportions would then have a length of five, and the proportions of the three sides of the enclosed triangle would then be three, four, and five; hence the term, "golden triangle." For instance, in planning an enclosure you may decide upon a width of 24 inches. For this width, the height should be 32 inches. If the depth of such a box were to be 18 inches, then the volume would be 8 cubic feet, and this would be more than adequate for a 12-inch speaker. In applying the golden triangle principle to enclosure design, the proportions must be reasonably close, but not necessarily precisely so.

Like the flat baffle, all box-type baffles, whether of this variety or others to be described later, should have a lining of sound-absorbing material. It is customary to line one side, the front or back, and the top or bottom of the box, so that no two untreated walls face each other.

The Closed Box Baffle

A large completely enclosed box (having a back panel) is sometimes referred to as an infinite baffle, although the application of the term is not strictly correct. A true infinite baffle is one in which the speaker front and back air waves do not cancel; this is not approached until the volume of the enclosure is 10 cubic feet or more. Incidentally, the closest practical approach to a true infinite baffle is made by mounting the speaker in an opening cut in the wall of a room; the larger the enclosure at the rear of the speaker, the better. Installations of this kind are fairly popular among high fidelity enthusiasts. As an alternative to such an installation, the speaker may be mounted in the door of a large closet. The mechanics of such installations will be described in the section to follow.

Although most listeners agree that a closed box baffle does not compare with a wall or closet installation, insofar as the volume of enclosed air is concerned, still it is evident that it is an improvement over the box baffle with an open back. It should be reiterated that sound-absorbing material must be used. If you have an opportunity to compare the open box with the completely enclosed box (and this can readily be done by alternately removing and replacing the back panel) you will probably find that the total sound radiation drops off slightly when the box is closed. However, the bass response will undoubtedly be better with the back in place.

Acoustic Suspension Systems

The acoustic suspension system is a specially designed speaker-enclosure system offering an improvement in the linearity of the response curve. Although some special treatment of the speaker is required, most of the work centers on the enclosure itself, therefore the system is described here rather than in the chapter on loudspeakers.

Very briefly described, it consists of an enclosure completely sealed (airtight). The enclosure is filled with a sound absorbing material, and the speaker rim is sealed to the front of the baffle so that the joint is airtight. The outer edge of the speaker cone is treated so that the cone is very compliant.

A system of this kind, using two woofers (with the tweeter in a separate enclosure) can be built around relatively inexpensive speakers. The most important requirement is that the speakers have fairly good bass response. A kit for softening the rim of the cone may be obtained from any large electronics supply house and will be found listed in their catalogs. Follow the directions accompanying the softening kit and be careful not to puncture the speaker cone while treating it.

For two 12-inch speakers, the enclosure should be about 28 inches long, 15 inches high and 10 inches deep. The material is ¾-inch thick plywood. Follow the instructions on enclosure construction given later in this chapter, but two points must be observed. All joints must be carefully planed so that the parts fit together perfectly. Use screws for fastening the panels together, but before fastening apply a generous quantity of glue to all joints. Remember that they must be airtight.

When the top, bottom, front, and end panels have been assembled, you are ready to make the cutouts in the front panel to accommodate the speakers. In all other enclosures described in this chapter, the speaker was installed from within the cabinet; that is to say, the rim of the speaker was fastened to the inside surface of the front panel. The arrangement here is quite different; the speakers are to be installed from outside. This means that the inside of the speaker rim rests against the outside surface of the front panel. To get this just right you will have to measure the inside diameter of the cone ring exactly. If you are not quite sure of your measurement, make the opening a trifle small and enlarge it later with a rasp.

The speakers are held in place with the usual screws or bolts, but before the final installation is made, a sealing compound must be placed between the inside of the cone ring and the panel. There are many compounds suitable for this job; some are made by manufacturers of roofing products. Since some products may not be available nationally, it is suggested that you consult your local hardware or paint dealer. The compound should be one that will remain plastic or semi-soft and yet will not shrink.

The next step is to cover the rear of the speakers with a very coarsely woven cloth (buckram will do well). Next, fill the entire interior of the cabinet with Fiberglas insulation or similar acoustic material. The rear panel is held to the enclosure with screws and sealed either with glue or the sealing compound used for the speaker rims. Before installing the rear panel drill a small hole for passage of the wires. After the back is in place and sealed, this hole should be filled with sealing compound.

Wall Installations

Should you decide to install your speaker system in a wall, a few hints concerning the procedure will be helpful. The first step is to locate the wall studs; these are the vertical wooden members providing the framework for the partition and normally they are located 16 inches apart, center to center. They usually can be located by tapping on the wall. A hollow sound will indicate that you are tapping on plaster or wallboard only, a more solid sound will tell you that you have located a stud. When you have found two adjacent studs that are suitably located, mark on the wall a 16-inch square at the desired height above the floor. Remember that, for best results, the speaker should be placed at about ear level of a seated listener.

Next, using a hammer and cold chisel, cut away the plaster inside the outline already drawn on the wall. This will expose part of each of the two studs, as shown in Figure 3-4. You now will have to install two cross braces, made of 2 by 4 inch lumber (see Figure 3-5). The cross braces should be cut to such a length that they fit tightly between the studs; if you have to drive them in place, so much the better. Further secure the cross braces by toe-nailing them to the studs. Before you proceed with the actual installation

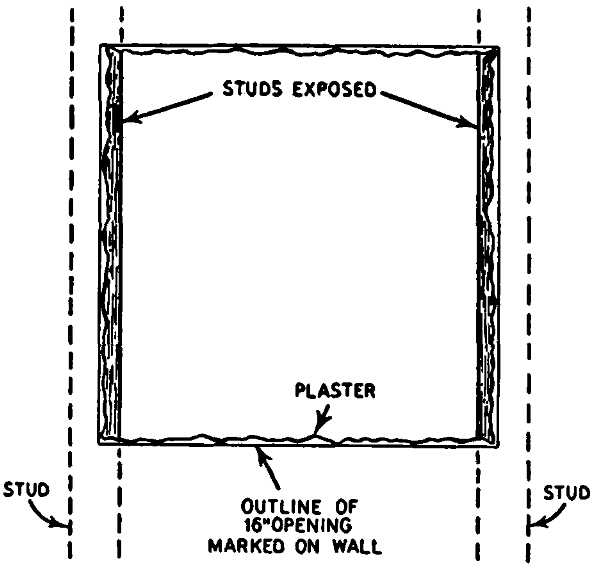


Fig. 3-4. Removal of plaster for wall installation of speaker.

of the speaker, it probably will be necessary to cut away some of the plaster on the opposite side of the partition to admit the speaker magnet and housing. The size of this opening will, of course, depend upon the construction of your particular unit.

You now have a square opening to receive the speaker sub-baffle, but it is best to add baffle mounting strips. These can be made of 1 by 1 inch lumber. Install them so that they are flush with the edges of the studs and cross braces on the opposite side of the room. The location of the strips is shown in Figure 3-6.

Now make up a sub-baffle board of ¾-inch thick plywood. The dimensions should be such that the board will just slip into the opening and rest against the mounting strips. Locate the exact center of the sub-baffle (by drawing diagonals) and, from this center, describe a circle the size of which will accommodate the speaker opening. For a 12-inch speaker, the opening will be 10½ inches in diameter. The sizes of the openings for smaller and larger speakers will be found in the section on bass reflex enclosures. Drill one or more one-inch holes inside the circumference of the circle,

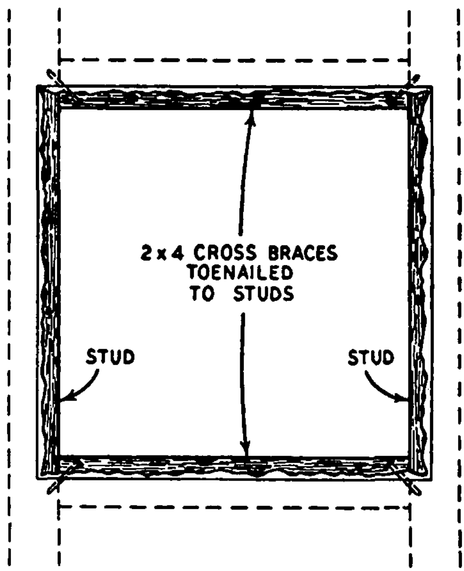


Fig. 3-5. Installation of cross braces.

and cut the opening with a compass saw or a power scroll saw, if you have it. Fasten the loudspeaker to the sub-baffle with four 10-24 machine screws and nuts.

Before mounting the speaker in the opening, the voice coil wiring may be installed, unless you prefer to have the wires emerge in the adjacent room. The wiring from the speaker down to the floor may easily be concealed behind the plaster; the remainder of the wiring may then be run along the base molding to the amplifier. Drop a plumb line from the center of the speaker opening to the floor, and mark the position of the line on the base molding. If there is no molding, make a mark on the wall instead, just above the floor. At this point, drill a hole, large enough to admit the wires, through the base molding or wall, completely through the plaster. Drill another hole of the same diameter vertically through the lower baffle mounting strip and cross brace. You will note that this hole will then be to the rear of the sub-baffle. Connect the wires to the speaker voice coil terminals and drop them through the hole in the baffle mounting strip and cross brace. Slip the baffle and speaker in place and fasten the baffle to the mounting

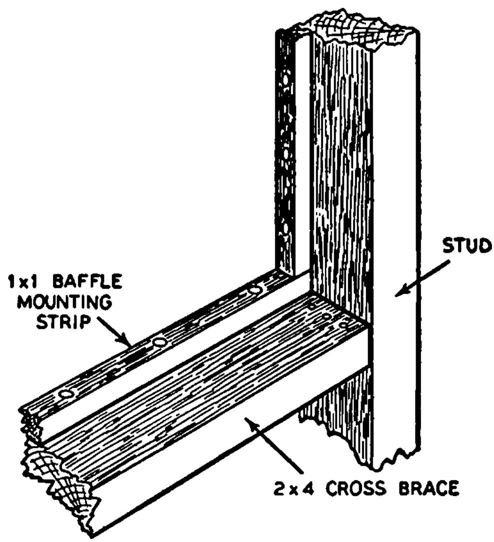


Fig. 3-6. Installation of baffle mounting strips.

strip with woodscrews. The method of fishing the wires out through the hole in the wall will depend upon the tools you have at hand. An electrician's steel snake with a small hook bent in the end is best, but any piece of stiff wire will do the job.

All that remains now is to conceal the opening in the wall and there are several methods for doing this. One simple and rather obvious solution is to hang a tapestry over the opening. Of course, it is important that the cloth of the tapestry be quite loosely woven, otherwise the passage of the higher frequencies will be impeded. Some constructors have employed small framed tapestries or shadow boxes having a rear covering of coarse cloth. As an alternative, you might want to install some kind of decorative bronze grill, backed up by a suitable grill cloth. When properly made, a wall installation will give excellent low frequency response.

If you cannot install your speaker in a wall, you might want to use a large closet as an alternative. The fact that clothing must also be kept in the closet need not be a drawback, as long as you are able to get a reasonably large volume of enclosed air. In fact, the presence of clothing might be beneficial. In some installations, the record player and amplifier are installed in the same closet. And, while on this topic, we might also mention that a closet makes a most convenient location for record player, amplifier, and tuner, even though a separate speaker enclosure is used. At least it provides a place to keep the equipment, without the expenditure of time and labor in constructing some other kind of housing. If a closet door is used for mounting the speaker, the work is far less complicated than it is for a wall installation. The problem of concealing the opening is simplified, for obviously it is easier to fasten a decorative grill to a wooden door than to a plaster wall.

Bass Reflex Enclosures

The bass reflex enclosure is a completely enclosed box with the usual lining of sound-absorbing material and the addition of an opening, referred to as a port, or vent, in the front panel, below the speaker opening, as illustrated in Figure 3-7. Properly designed and constructed, the bass reflex cabinet is capable of providing a smooth bass response that is equal, if not superior, to that of any other kind of baffle or enclosure.

Some authorities have pointed out that a bass reflex enclosure is difficult to design and, if designed for a specific type of speaker, does not give fully satisfactory results with any other speaker. It is further claimed that the design of such enclosures depends upon formulas. In spite of these objections, several writers have given instructions for the construction of reflex cabinets, and we feel that you will be interested in the information to be included here. A great deal of emphasis is given to the port or vent; apparently it is difficult to calculate the size accurately in advance. For that reason, we are suggesting a method of "tuning" or adjusting the port opening to optimum size; although we do not claim that the method is original, we do know that it works. Even though you build such an enclosure and, after the port adjustments have been made, you are dissatisfied with the results, no material and very little time will have been wasted, for it is always possible to block up the port entirely and use the cabinet as a simple box type baffle with or without a back panel. In view of this, we feel that the bass reflex enclosure is well worth trying.

In reading through available literature on high fidelity systems, you will most likely find some disagreement among authorities con-

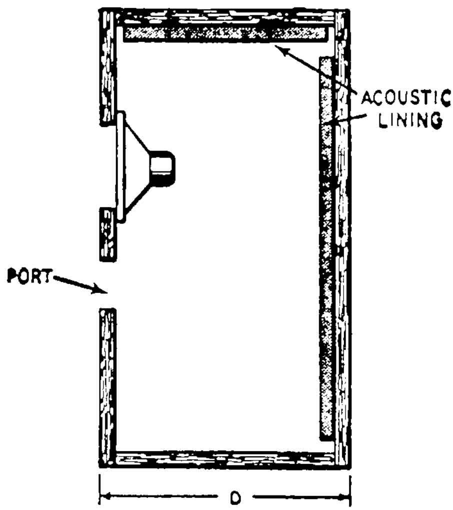


Fig. 3-7. Bass reflex enclosure, cross section.

cerning the minimum volume of the enclosure and the area of the port opening. One writer feels that a volume of 4 cubic feet is sufficient for a 12-inch speaker; another advises no less than 6 cubic feet for any size of speaker. Since, as we are now aware, good bass response depends to a large degree upon cabinet volume, we believe that the larger estimate is safer. Estimates for the area of the port opening vary from one half to one times the speaker opening. The table below should be of helpful guidance.

Also presented will be some suggestions for building a bass reflex enclosure in accordance with the specifications for pleasing appearance, mentioned earlier. Figure 3-8 shows the typical cabinet as viewed from the front. In the drawings critical dimensions have been lettered, and the letters appear at the headings of the columns of dimensions in the tabulation. All dimensions are given in inches.

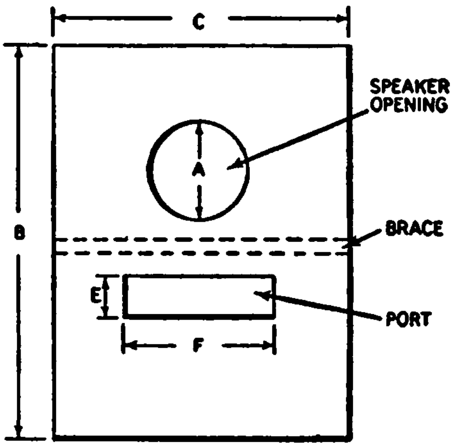


Fig. 3-8. Bass reflex enclosure, front view.

	A	B	C	D	E	F
Speaker Diameter	*Speaker Opening*	*Height*	*Width*	*Depth*	*Port Height*	*Port Width*
8	6½	23	18	10	3	9½
10	8½	28	22	13	4	13
12	10½	34	27	14	5	17
15	13¾	41	30	18	7	22

By altering these dimensions, the cabinet proportions can be made to conform either to your ideas of pleasing design or to the available space. The latter may adjoin one or more cabinets used to house the remainder of your equipment. For instance, the enclosure for a 12-inch speaker has a volume of something over 12,000 cubic inches. Within reasonable limits, you may alter the proportions to suit your requirements. If the unit is to fit into a wall installation resembling a set of bookshelves, there is no reason why the width cannot be, say, 40, the depth 14, and the height 23; these dimensions would still give a volume of about 12,000 cubic inches. Concerning the port opening, the important thing is the area rather than the actual height and width. Again, the areas given are considered to be the maximum and are intended to be modified later by partial closure of the opening by means of an adjustable flap. The approximate port areas for various speaker diameters are as follows: 8-inch, 28 square inches; 10-inch, 50 square inches; 12-inch, 86 square inches; and 15-inch, 150 square inches. Although the ratio of height to width of the port opening does not seem to be important, it will probably be convenient to use a rectangular-shaped port.

This enclosure, like the types already described, should be constructed of heavy materials to prevent vibration of the unit. Plywood, ¾-inch thick, is suggested. In addition, it is well to add cross bracing. Run one 2 by 2 inch brace across the inside of the front panel; the best location is between the lower edge of the speaker opening and the top edge of the port. This is indicated by the dotted lines in Figure 3-8. Another similar brace may be placed across the inside of the back panel, at about the center. Some constructors add a brace running from the front panel to the back panel. For maximum rigidity, fasten the braces to the panels with woodscrews and glue. If a brace from front panel to back is used, it is obvious that no glue should be used on the back end of this piece.

To adjust the port opening to final size, you will need a piece ot plywood at least 3 inches larger in each dimension than the port. This piece should have two elongated holes or slots near the edges, as shown in Figure 3-9. Two large round-head woodscrews or two 10-24 machine screws with nuts hold this port reducer to the inside of the front panel, as indicated in the drawing. By loosening the woodscrews or nuts, the reducer may be adjusted to cover as much of the port as is necessary. Adjusting the port size from the inside of the cabinet will mean that the back panel must be removed each time a new adjustment is tried, unless you care to provide in the back panel a removable door, large enough to admit your hands. Another possible method is to cover the port temporarily from the outside with a piece of plywood until the best position is found and then make the final adjustment with

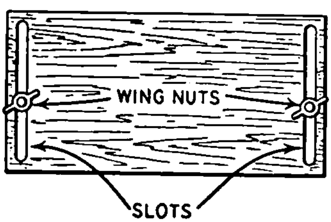


Fig. 3-9. Port reducer.

the permanent reducer from the inside. However, this method has disadvantages; often it is difficult to judge the performance unless an extended listening test is made, and you may need an assistant to hold the temporary reducer, since it is difficult to judge the response while you are quite close to the speaker. From time to time you will no doubt read of various methods of adjusting reflex enclosure ports; some of these methods require the use of measuring instruments. However, the easiest and perhaps the best method of all is critical listening. Select a program with prominent bass, and as you vary the port opening size, listen for excessive booming or

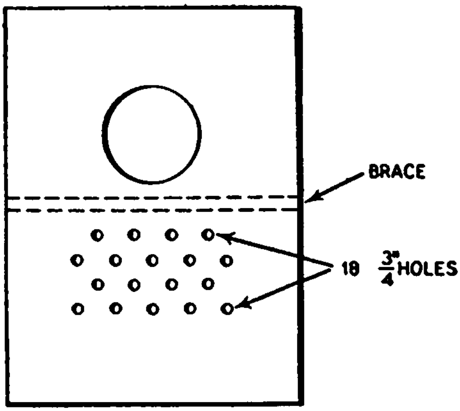


Fig. 3-10. Distributed port enclosure.

any indication of hangover. The latter may be defined as a tendency of the tone to persist after it has actually ceased. When you find what appears to be the best adjustment, listen carefully for an extended period.

An interesting variation of the bass reflex enclosure is the type recommended by the General Electric Company for use with their models A1-400, 1201D, and 1203D high fidelity loudspeakers. This type is related to the one just discussed, but in place of the single large port a "distributed" port arrangement is used. In the enclosure shown in Figure 3-10, eighteen ¾-inch diameter holes are drilled in the front panel below the speaker opening. The holes, as you will note, are arranged in four horizontal rows, two of four each and two of five each. The top row of holes is located 8 inches below the center of the speaker; the bottom row is 14 inches below the speaker center. The remaining two rows of holes are equally spaced between the top and bottom rows.

Labyrinth Enclosures

Earlier in this chapter we mentioned one function of the baffle: to prevent cancellation of the front and back air waves set up by the speaker. Since the front and back waves do not occur at precisely the same instant, they are said to be "out of phase" and thus the effect of one wave may partially nullify the effect of the other. This, as we have seen, may be prevented by the use of a large baffle, providing the air waves with a longer path. In addition to this, the speaker converts electrical impulses into air waves and, to perform this function at all efficiently, it must be provided with a sufficiently large mass of air that can be set in motion.

One rather obvious method of doing this is to couple the speaker to some kind of horn. You may have seen such horns in public address systems; motion picture sound systems also depend upon them to a considerable extent. However, the problems encountered in motion picture installations are far different from those found in the home. For one thing, there is usually no space problem; many theater installations employ enclosures having a volume of 60 cubic feet. Furthermore, it is usually desirable to have the sound

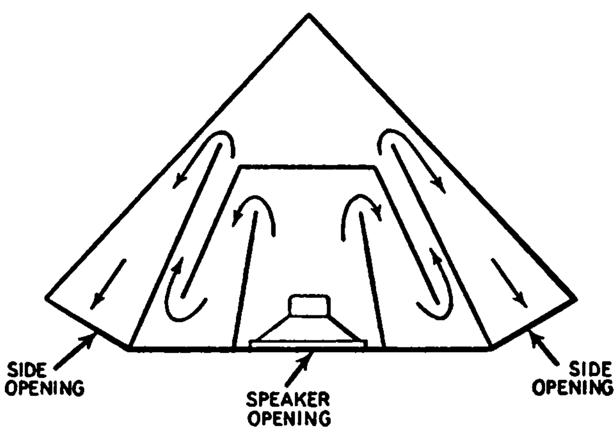


Fig. 3-11. Corner-type labyrinth enclosure (top view).

directed so that it appears to come directly from a single point, the screen.

In recent years attempts have been made to apply the horn principle to speaker enclosures for home music systems. In most installations, the horn is folded to conserve space. By the use of a series of baffles it is possible to cause the sound wave to traverse a devious, and therefore a longer, path in leaving the loudspeaker. This is the labyrinth principle, illustrated in Figure 3-11. This particular enclosure is designed for installation in a corner of the room, a design feature common in enclosures of this kind. The arrows indicate the path of the sound waves from the speaker diaphragm to the room. Notice also that the chamber is coupled to the back rather than the front of the loudspeaker.

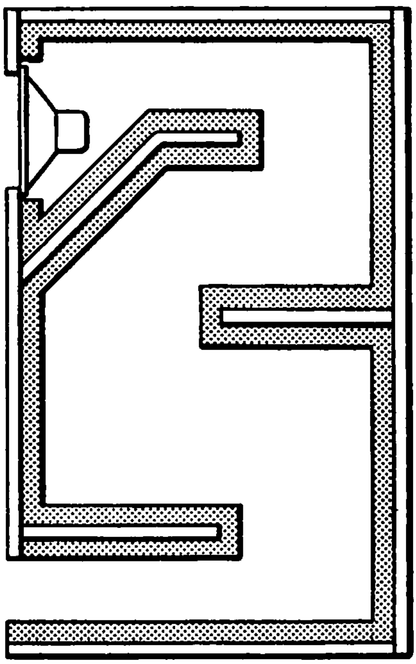


Fig. 3-12. Labyrinth enclosure.

Figure 3-12 shows a rectangular labyrinth enclosure designed for use with a 12-inch loudspeaker. All of the cabinet sides as well as the various partitions should be made of plywood, at least ¾ inch thick. Since all partitions run the full width of the cabinet, they also serve as braces. In the drawing, the plywood walls and partitions are shown unshaded, while the lining of sound-absorbing material is shaded. The acoustic material should be 1 inch thick and is applied to all inner surfaces of the cabinet. The principal difficulty you might have in constructing a cabinet of this type is making a proper joint between the inclined partition immediately in back of the speaker and the other members of the cabinet. For proper results, the joints should be beveled. Bevel joints can easily be cut on a bench saw or jointer or, if you lack these tools, by an experienced woodworker at moderate cost.

How to Build Your Own Speaker Enclosures

In this section we shall give detailed instructions for building speaker enclosures that will not only perform the desired acoustic function but, if carefully built, will also present an attractive appearance. Two basic designs have been chosen: a rectangular enclosure to harmonize with modern furniture and a rectangular design suitable for a room having traditional furnishings. In addition, suggestions will be offered, wherever possible, for altering the basic design to suit your individual preferences. However, since the features and specifications of the various enclosures (completely enclosed box, bass reflex, labyrinth, and so forth) have already been discussed, we shall confine the discussion to a description of the outside of the cabinet, leaving the selection of acoustic features to you. We shall also offer instructions for finishing the cabinet, with a choice of two general types of finish: (a) an opaque finish (paint, enamel, or colored lacquer) and (b) the more conventional transparent hardwood finish (in mahogany, walnut, and so forth) using varnish as the finishing material. In selecting the type of finish, several things must be kept in mind. The transparent finish requires much more labor, the pre-finishing preparation of the cabinet must be more carefully done, and the cost is considerably greater since plywood having a face veneer of more costly wood is required. Furthermore, if a transparent finish is to be applied, much more care will be necessary in constructing the enclosure in order to conceal the end grain of the plywood. Your selection of a finish, then, will probably depend upon your skill and the money you care to spend.

The cheapest available factory-made enclosure (built of composition board) sells for about $40. More elaborate styles run from $75 to $175. Assuming that your home-made cabinet is to be painted or lacquered, the amount of plywood needed for the average size will be perhaps 30 square feet; common fir plywood may be used for an opaque finish, and it currently sells for about twenty-five cents per square foot, or a total of $7.50. Add a few dollars for incidentals (hardware, lumber for braces, finishing materials, and grill cloth) and you will see that the total cost is not likely to be over $15. For a much more elaborate style, finished in walnut or mahogany, the expenditure is not likely to be much more than $30. Obviously, the moral here is to exercise your latent skill with tools.

We shall begin with a cabinet intended for opaque finishing, then give the constructional techniques necessary to produce the three design variations, and conclude with a discussion of the more difficult finish treatments. The finished cabinet is shown in Figure 3-13.

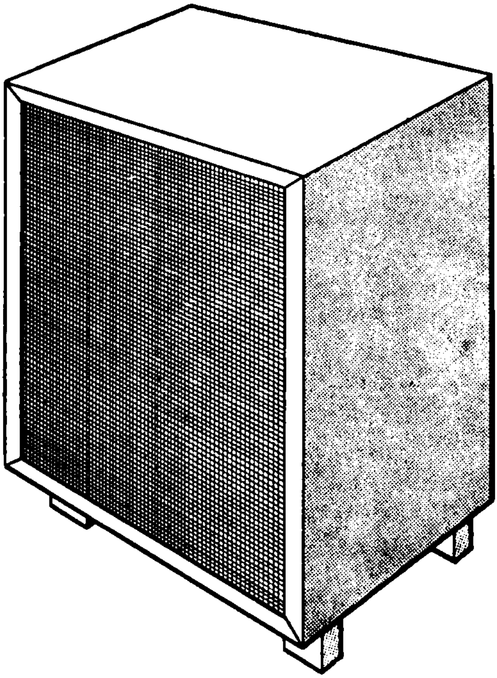


Fig. 3-13. Modern speaker enclosure.

We are proceeding upon the assumption that you are building a rectangular cabinet measuring 24 inches wide, 32 inches high, and 18 inches deep, corresponding to the proportions of the golden triangle mentioned earlier; the measurements may be scaled down or up to suit, as long as the essential proportions are preserved. The cubic content needed for any standard size of loudspeaker can be taken from previous sections of this chapter.

The enclosure should be made of ¾-inch plywood, but, if your particular needs are for an unusually small cabinet (no larger than 2 or 3 cubic feet), then ½-inch plywood may be used. Plywood commonly is sold in sheets measuring 3 by 6, 4 by 6, and 4 by 8 feet.

Most lumber dealers sell it only in whole sheets and, where only part of a sheet is bought, the cost per square foot is likely to increase alarmingly. For this reason, it is well to plan the layout of the panels on the sheet before buying to avoid purchasing more stock than you need. For example, the cabinet we are building must be cut from a 4 by 8 foot sheet. There will be some waste, but this is unavoidable because a 4 by 6 foot sheet is just a little too small. Figure 3-14 shows one way to lay out the panels so that the waste is in the form of long narrow strips; if you are building a bass reflex cabinet, part of the waste can be used for the port reducer. If you

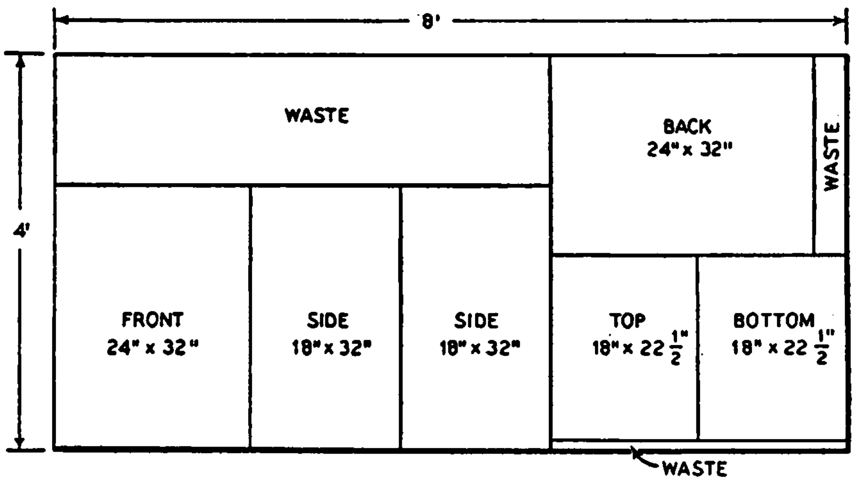


Fig. 3-14. How to lay out panels.

have difficulty in estimating the amount of lumber needed or the size of the sheet required, take a sketch of the cabinet to your lumber dealer. Most dealers will be glad to help you estimate, and many have facilities for cutting the panels accurately in the event that you are unable to do so.

As mentioned before, fir plywood is quite satisfactory for a cabinet to be finished in opaque lacquer, paint, or enamel. In fact, used plywood may be satisfactory, provided that it is clean, is not warped, and has no serious dents or scars. If the lumber dealer is not willing to cut the panels for you, use a hand saw, portable power saw, or a bench saw and do it yourself. As indicated in Figure 3-14, you will need two pieces measuring 24 by 32 inches (for the front and back), two measuring 18 by 32 inches (for the side panels), and two measuring 18 by 22½ (for the top and bottom). You will notice that the top and bottom panels are each 1½ inches short of the full 24-inch width of the enclosure; this is because the thickness of the plywood has been taken into account at the corner joints. It would be possible to make up the cabinet by cutting the top and bottom panels the full width of the cabinet, with the side panels each 1½ inches short; in this way, the joints would appear at the sides of the cabinet rather than on the top. However, there would then be four visible joints instead of two. Notice that the thickness of the front and back panels is added to the total outside depth of the cabinet, making this dimension actually 19½ inches. You need not worry about concealing the end grain in a cabinet intended for an opaque finish. However, since such grain is quite porous, it will need additional preparation before the final finish of lacquer or enamel is applied. The first step in the preparation is to sand well all end grain that will be visible.

With the panels cut to size, you are ready to begin putting the cabinet together. The method of assembly is illustrated in Figure 3-15. Before starting the assembly, prepare the four corner blocks shown in this drawing. These are cut from lumber measuring at least 1 by 1 inch in cross section and are 18 inches long. Since paint, enamel, or lacquer will readily cover small imperfections, it is possible to use large brads for holding the parts together; 1½ inch No. 14 is a suitable size. We strongly advise that you apply glue to each joint as it is assembled, either prepared glue or, better yet, one of the powdered glues such as casein, Cascamite, or Weldwood. Begin by gluing and nailing the four corner blocks to the 18-inch edges of the 18 by 22½ inch top and bottom panels. The blocks must be absolutely flush with the edges of the panels, as shown in the detail sketch, Figure 3-15A.

Now assemble the top, bottom, and side panels as shown in the center part of Figure 3-15. Use two rows of brads at each corner joint; one row should be ⅜ inch from the end of the side panel; the brads are driven through this panel into the end of the adjoining top or bottom panel. The second row of brads is 1¼ inches from the end of the side panel and they are driven through that panel into the corner block. Again, don't forget to apply glue to each joint as it is made up. As the work of assembly proceeds, check now and then with a try square to be sure you are getting truly square corner joints; it is too late to correct the situation once the glue has set.

While the glue is setting, you may cut the speaker opening and the port, if the enclosure is to be of the bass reflex type. You will note that the drawing shows a port, although this is optional. At

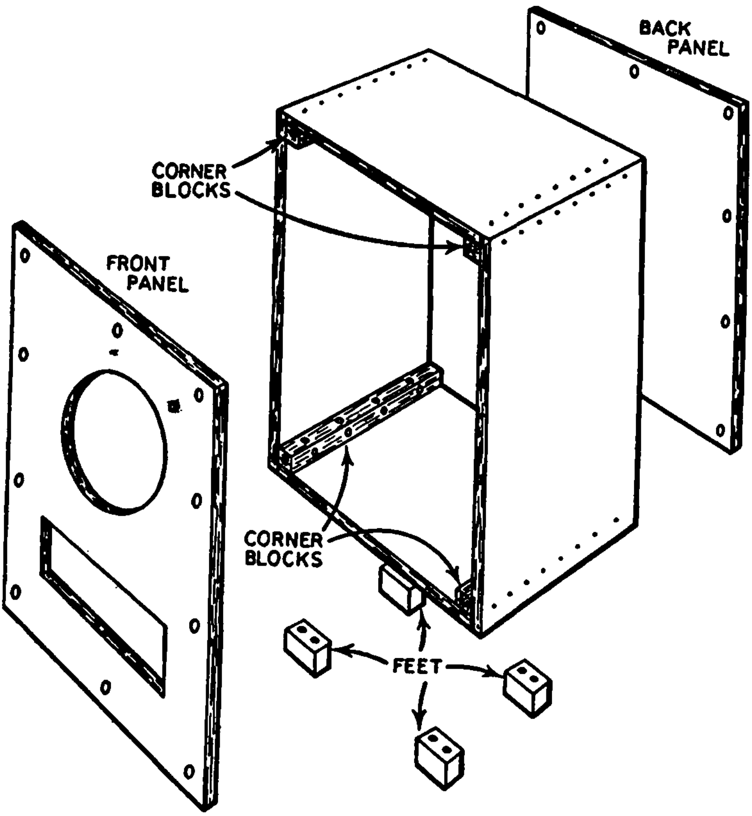


Fig. 3-15. Cabinet assembly.

the same time you may also drill the front and back panels for woodscrews. The method of laying out and cutting a speaker opening was described in the section on wall enclosures, so that there is no need to discuss the matter at this time. For holding the front and back panels in place, 1½-inch, No. 6 flat-head woodscrews are suitable. Ten screws will be needed for each panel. To lay out the positions of the screw holes, scribe lines around all edges of the front and back panels on the inside of the panels. The lines should be ⅜ inch from the edges. Marking may be done either with a marking gauge or a straightedge and pencil. A marking gauge leaves a visible scratch in the wood; that is why the layout should be done on the inner surfaces of the panels. A screw should be used at each panel corner, and these positions are marked by the intersections of the guide lines. On the shorter edge of the panel, locate a screw hole midway between the corners, then place two more on each longer side, equally spaced between the corners.

Center-punch the location of each screw, then drill with a No. 28 drill. Turn the panels over and countersink the holes to take the screw heads. Apply glue to the joint between front panel and the body of the cabinet, place the panel in position, and fasten with screws. The block feet, shown in the drawing of the finished enclosure, Figure 3-13 may now be made up. These are sections of 1¼ by 2¼ inch lumber, 3 inches long. The outer ends of the feet are arranged in a rectangle spaced 2 inches inside the bottom of the enclosure. Fasten to the bottom with glue and countersink flathead woodscrews or counterbored round-head screws. As an alternative, you may prefer to use the more decorative iron legs that are now available. The method of fastening them will depend upon the construction of the individual leg. Other possible treatments include the base shown in Figure 3-16. The base need not be solid, but can be assembled from four pieces of ¾ by 2½ inch plywood cut to the necessary lengths and nailed and glued together to form a rectangle. Fasten to the bottom of the cabinet with screws and glue. Incidentally, this is one way to use up the remainder of your sheet of plywood. If you prefer, the rectangular base, legs, or feet may be omitted entirely.

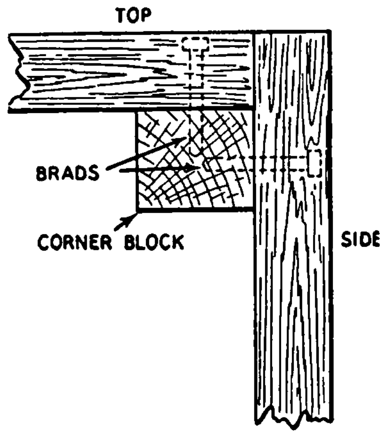


Fig. 3-15A. Detail of corner block.

It is best to prepare a mounting for the loudspeaker before the finish is applied to the cabinet. Either 10-24 machine screws and nuts or No. 10 woodscrews may be used for this purpose. If machine screws are to be used, drill the holes for them somewhat smaller than the actual diameter of the screws. The screws will then have to be driven into the holes; this will prevent them from dropping out if it ever becomes necessary to remove the speaker. If the screws do drop out after the grill cloth is in place, you will be in for a lot of unnecessary work. It is usually best to finish the cabinet

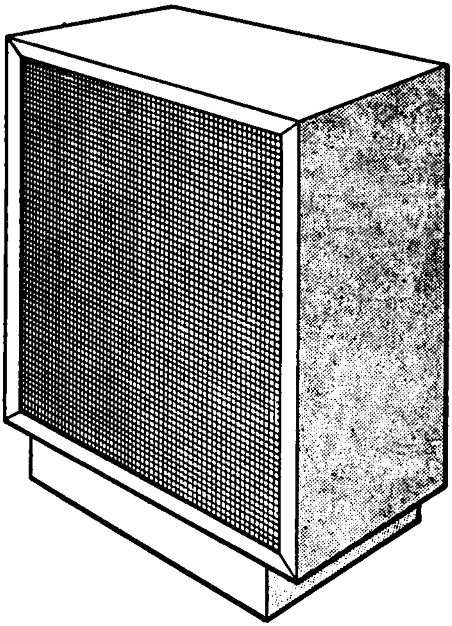


Fig. 3-16. Alternative base treatment.

and the back panel separately, before the back panel is fastened in place.

To complete the cabinet you will need the following, besides the finishing materials: a piece of grill cloth large enough to cover the entire front panel, and about 10 feet of triangular cross-section molding an inch to an inch and a half wide. For the grill cloth, monk's cloth is quite effective, although other fabrics may be used, including woven plastic, provided the material is loosely woven. Closely woven cloths will not allow the high frequencies to pass readily. The molding may be bought from a lumber dealer or may be cut on a bench saw. It should have the approximate cross section shown in Figure 3-17.

If you decide to make the molding yourself, obtain some white pine or poplar lumber 1¼ or 1½ inches square and of suitable length. Adjust the saw table (or arbor) to about 25 degrees and set the ripping fence quite close to the blade. In this operation it is best to use an auxiliary fence fastened to the regular metal ripping fence. If the molding is unobtainable from a dealer and you lack the facilities for cutting it yourself, it can be made by a local cabinetmaker. It should be planed before use, unless it is

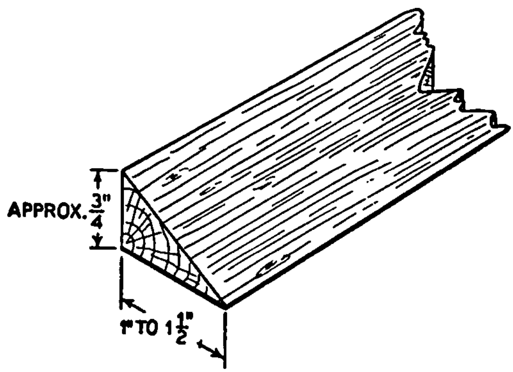


Fig. 3-17. Detail of molding.

ripped to shape with a hollow-ground or planer type saw blade. In that event, sanding only will be needed.

Carefully measure and cut the four pieces of molding needed, remembering that the edges of the molding are to be flush with the edges of the front panel. Now miter all ends to 45 degrees. This can most easily be done on the bench saw, but if you do not have one, mark the miter angles with a 45-degree drawing triangle or a miter gauge, then cut with a fine-toothed backsaw. Extra care devoted to this part of the work will help to produce a professional-looking job. Several 1-inch brads, spaced along the center line of the molding, are used to hold it to the cabinet. It is best to drill holes for the brads, slightly smaller than the actual diameter. The drilling of holes is done before any finish is applied. Any brad holes or other small imperfections in the cabinet may now be filled with plastic wood. When this has hardened, you are ready to begin finishing.

Inspect the cabinet for any signs of roughness. If any are found, sand as required. Either white or orange shellac may be used for the undercoats. Two coats are usually necessary on the cabinet and molding. The end grain will need three or four coats. Thin the shellac with alcohol for the initial coat. When dry apply a second, heavier coat. Apply as many coats as needed to the end grain, until all signs of porosity have disappeared and the surface presents a uniform glossy appearance. The final shellac coat must be quite smooth, otherwise the finish coats will not be so. Any ripples or other unevenness appearing in the shellac finish may be reduced by light sanding with 6/0 garnet paper.

When the final shellac coat is satisfactory, you are ready to apply the finish coats. Lacquer is best applied with a spray gun, but fair work can be done with a brush if you observe one or two precautions. The lacquer should be quite thin for all coats. (As a rule, only two coats will be necessary.) Do not attempt to brush the lacquer out, but instead flow it on the surface with a rather heavily loaded brush. Once it is on the surface, let it alone. Enamel may also be used, but ordinary enamels have a very high gloss that may not be desirable. Of course, the gloss can be reduced by rubbing down each coat with very fine steel wool and, finally, applying paste wax. However, this is a lot of work. Many paint stores now stock dull enamels that nearly duplicate the effect of a hand-rubbed finish; one such preparation is Sapolin "New Mode" satin finish. A half-pint to a pint will be needed, depending upon the number of coats you apply. For an enclosure of the size specified, a half-pint will be enough for one coat. Apply only one coat to the molding; the second coat is to be applied after the molding has been nailed in place. Apply finish to the edges of the back panel, and on the inner surface for a distance of an inch or two beyond the edges.

When the final coat of lacquer or enamel is dry, install the loudspeaker. The grill cloth may then be tacked in place. If you use monk's cloth, it will have to be stretched slightly during application to keep it from sagging and, at the same time, you must be careful to keep the weave parallel with the edges of the cabinet. The process of stretching will be easier if the cloth is a bit larger than the front panel. Do not attempt to hold this kind of cloth in place with tacks alone. Instead, cut strips of thin cardboard to run the full length of the panel edges. The strips should be a trifle narrower than the width of the molding. The strips are to be laid over the grill cloth and tacked in place; this will prevent the cloth from creeping. Another suggestion: do not drive the tacks all the way in until the cloth has been stretched to your satisfaction. Finally, attach the molding to the cabinet with brads and set the brad heads slightly below the surfaces. Then fill the brad holes with plastic wood and, when this is dry, carefully apply a final coat of lacquer or enamel to the molding.

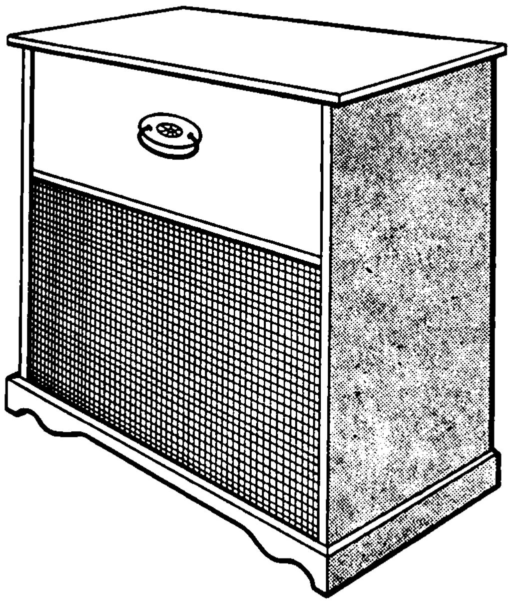


Fig. 3-18

Finally, here are a few suggestions regarding the choice of color. The color you use will depend partly upon your preferences and the location of the enclosure. In some instances, it may be painted to match the walls; in others, a contrasting color may be more suitable. However, a cabinet of the style we have just described looks rather well and blends with most surroundings when painted black.

Since this type of enclosure may be thought of as modern in design, you may consider the use of light finish. While possible, applying a light finish is attended by many difficulties, chiefly the problem of exposed end grain. As the grain at the edges of plywood is much more porous than on the surfaces, it will absorb more of the finishing material and therefore will appear different in color. An expert finisher might be able to avoid this, but the average amateur probably cannot. It is feasible to construct the cabinet so that a minimum of end grain is exposed, but you will need some skill in cutting special types of joints. If, after all the difficulties have been pointed out, you still wish to try your hand at applying a finish of this kind, we earnestly suggest that you bear in mind

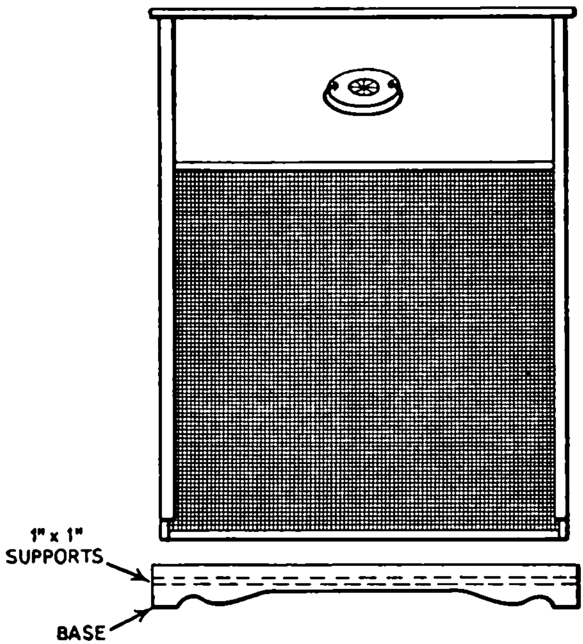


Fig. 3-19. Detail of cabinet construction.

our intention to give directions for constructing a low-cost cabinet. Do not buy a quantity of expensive plywood unless you feel that you have sufficient skill to do a creditable job.

Another style of speaker enclosure is illustrated in Figure 3-18. This is offered for those who do not care for modern furniture. It is intended for a walnut or mahogany finish and requires much more care and skill in construction and finishing. The general construction, however, is about the same, except for minor changes. The enclosure rests on a rectangular base made up of four strips of

¾-inch plywood. For a 24 by 32 inch enclosure, the strips should be about 4 inches wide. The band-sawed (or scroll-sawed) effect may be applied to the sides and front of the base or to the front alone, as you choose. The two sketches, Figures 3-19 and 3-20 illustrate several changes in the method of assembly that are made necessary by the type of finish to be used.

In Figure 3-19, you will notice that the cabinet proper and the base unit have been separated slightly so that you may see the method of assembly. As you will observe, the enclosure proper fits down inside the base unit. Fastened to the inside of the front and rear base strips there are rails to support the cabinet. (The construction of the base will be more fully described later.) This method of construction conceals the end grain. Instead of locating the cabinet top between the two side panels, as before, it is now made so that it overhangs the front and sides by about $\frac{3}{16}$ inch. When the overhanging edges are finished with colored shellac (this process to be described later), the effect of a narrow molding around the top is obtained and the edge grain can not be seen.

In the first project, the front panel was screwed to the edges of the main part of the cabinet; this method cannot be used here, for there would then be edge grain visible. Instead, the front panel is set within the frame formed by the sides, top and bottom so that its surface is flush with all edges as shown in Figure 3-20. Then, a molding is applied to the edges of the front panel. In addition, the design includes a simulated drawer; this effect is created by using molding to frame the area and adding a purely decorative drawer pull. Incidentally, in this type of construction you must be sure that the part of the front panel area to be covered by grill cloth is large enough to accommodate both the speaker and port openings, if the bass reflex principle is to be used.

The plywood selected for the project should be ¾-inch thick and must have a face veneer of suitable cabinet wood. Mahogany or walnut is suggested for an attractive finish. Current prices are in the neighborhood of 75 cents per square foot. Gum may also be used and generally can be finished to resemble either walnut or mahogany with fair results; the price per square foot is about 45 cents. Prices of plywood (and other lumber) may vary from one area to another. Be sure that the amount purchased is sufficient for all panels, as well as for the four base strips. You will need two strips measuring 4 by 26 inches and two measuring 4 by 19 inches.

Ten feet of molding will be required, about one inch wide and of the plain half-round variety or more elaborate, if you choose. In addition, you will need: sixty 1½-inch No. 10 flat-head screws, ten 1½-inch No. 6 flat-head woodscrews, about 20 feet of 1 by 1 inch square lumber for corner blocks, grill cloth at least 24 inches square, and a drawer pull.

Begin by preparing the corner blocks for the top, bottom, and sides. Cut four pieces of the 1 by 1 inch stock, each 17 inches

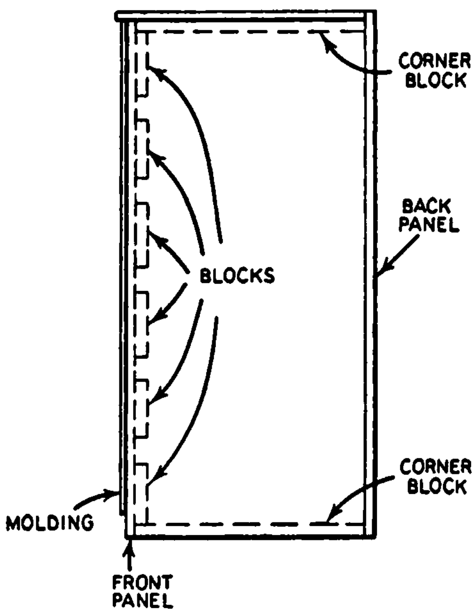


Fig. 3-20. Detail of cabinet construction.

long. As no brads are permissible in this type of cabinet, the corner joints are made by driving screws through the corner blocks, from the inside, into the top, bottom, and side panels. This is illustrated in the detail sketch, Figure 3-21A. In the joint at the top right corner of the enclosure, for example, there are four screws passing horizontally through the corner block into the right wall of the cabinet, and four more passing vertically through the block into the cabinet top. All other corner joints are made in the same way. Drill and countersink holes to take the 1½-inch No. 10 screws.

Be sure that the holes are fairly well spaced and that vertical and horizontal screw holes do not intersect.

The next step is to cut the six panels. Here are the sizes: sides (2) 18 by 31 inches; bottom 18 by 22½ inches; top 18 3/16 by 24⅜ inches; front 22½ by 30¼ inches; back 24 by 31¾ inches. The finished sizes of all panels must be quite close to the dimensions given, otherwise the parts will not fit properly. Furthermore, it is very important that all pieces be absolutely square. You will note

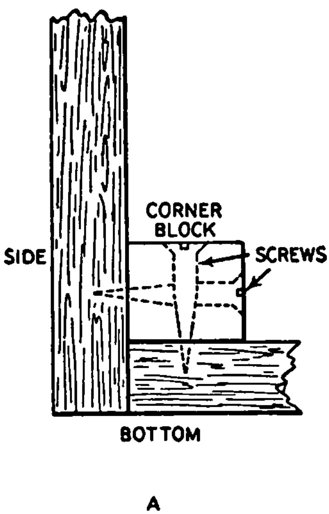


Fig. 3-21A. Assembly of corner joints.

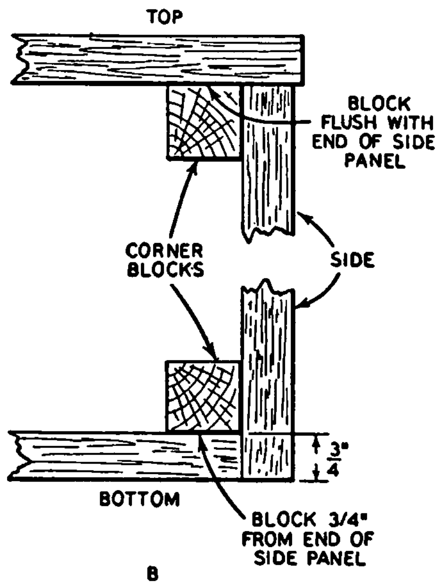


Fig. 3-21B. Placement of corner blocks.

that the size of the top panel allows for a 3/16 inch overhang at front and sides, but not at the back.

The corner blocks may now be glued and screwed to the side panels. As there is a possibility of excess glue oozing out onto the face veneer, we suggest that you use a non-staining powder glue, such as Cascamite or Weldwood. In assembly, use just enough glue to make a good joint, and if any excess runs out on a veneered surface, wipe it off immediately with a damp cloth. In assembling the corner blocks, refer to Figure 3-21B. You will notice that the blocks used at the upper corners are flush with the upper ends of the side panels, while those used at the bottom are set in exactly ¾ inch from the lower ends of the panels. Furthermore, the front ends of all four blocks are set in a distance of an inch from the front edges of the panels; in other words, the back ends of the blocks are flush with the back edges of the cabinet. This is to permit locating the front panel flush with the edges of the other panels.

You are now ready to add the top and bottom panels. Apply glue to the lower edges of the bottom corner blocks and to the inner surfaces of the side panels, below the blocks. Place the bottom panel in position between the sides, making sure that its front and rear edges are aligned with those of the side panels. Insert the screws in the corner blocks and draw up tightly. Draw a pencil line on the inner surface of the top panel parallel with the edges and 3/16 of an inch distant from them. Apply glue inside this line in a band about an inch wide. Place the top on a pad of newspaper on the floor or on a low bench. Now turn the partially assembled cabinet upside down and place it in position on the top. When you are sure that all edges are aligned with the pencil marks, insert the screws and draw up tightly. As the cabinet parts come together, notice whether glue runs out of the joints; if so, wipe it off at once. Check all corners with a try square.

The front panel sets inside the frame of the cabinet and rests against corner blocks, as shown in Figure 3-20. The blocks are glued and screwed to the cabinet walls, while other screws hold the front panel to the blocks. The corner blocks may consist of continuous strips running around the cabinet ¾ inch from the front edges, or you may use a series of smaller blocks as shown in the drawing. In any event, ten screws should be used for fastening the front panel. Space the front panel screws in this way: one near each corner, one in the center of each short side, and two equally spaced along the long sides between the corners. The first step is to draw a guide line for positioning the blocks on the inside of the cabinet. The guide line should be ¾ inch from all front edges and must be drawn quite accurately, otherwise the front panel will not be perfectly flush. When the blocks have been drilled and countersunk for the screws, fasten them in place, being sure to position them accurately. Apply glue to the front surfaces of the blocks, slip the front panel in place, turn the cabinet over, and screw in place. This completes the assembly of the main portion of the enclosure.

When the glue has dried, inspect all outer surfaces for raised grain, which may occur wherever a damp cloth was used to remove excess glue. Sand all such places lightly with 3/0 sandpaper.

Start construction of the base by cutting two 4 by 26 inch and two 4 by 19 inch pieces of ¾-inch plywood. If the base is to be built so that end grain is concealed, miter joints must be used at the corners. Details of base construction are shown in Figure 3-21C. The corner joints are fastened by means of corner blocks, screws, and glue. Notice that, although the corner base strips are 4 inches

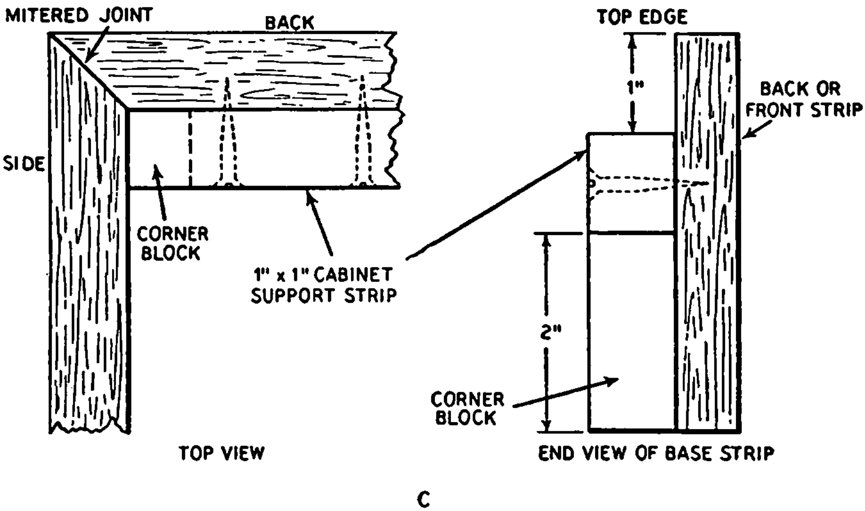


Fig. 3-21C. Detail of base construction.

wide, the corner blocks are only 2 inches long; this is to permit the use of 1 by 1 inch strips for supporting the cabinet.

When the cabinet and base have been assembled, you are ready to apply the finish; the process is briefly described below. All finishing instructions apply to the base, cabinet, and molding. When all are ready for final assembly, apply the grill cloth as described in connection with the modern cabinet. Before finishing has begun, you should prepare the four lengths of molding required. The side strips are applied to the cabinet so that their lower ends are one inch above the bottom of the enclosure. The horizontal piece, used to form the lower edge of the simulated drawer, is then added. Note that the ends of this piece are not mitered, but are cut square.

If you attempt to do any finishing of cabinet or molding after assembly, the work will be more difficult. The enclosure is completed by adding the drawer pull and finishing the edges of the top panel with colored shellac.

Cabinet Finishing

The application of a varnish finish requires only a little more skill than does painting. However, considerably more labor and patience are necessary for success. Complete instructions for applying this type of finish are beyond the scope of this book, but we are including a schedule of finishing operations. Additional information may be obtained from any standard work on cabinet finishing.

1. Final sanding of cabinet, molding, and base should be done with 3/0 garnet paper. After sanding inspect carefully for scratches or blemishes. Dust all surfaces well.
2. Either oil or water stain may be used. Oil stain is easier to apply and requires no mixing, but water stain is more penetrating. Water stain raises the grain of the wood, but this can be overcome by first sponging the entire surface lightly with clear water. Allow the surfaces to dry, then sand lightly to remove fuzz. Stain may now be applied without raising the grain.
3. Allow the stain to dry 24 hours, and apply a coat of wood filler. This is obtainable in colors or natural; the type to use depends upon the effect you want. For a conventional mahogany or walnut finish use a filler that matches, approximately, the color of the stained wood. Remove excess filler by wiping across the grain with burlap, coarse cloth, or similar rough material. At all points where end grain will show, apply two or more coats of filler.
4. When filler is dry (24 hours), apply a thin coat of white shellac. When the shellac is dry, rub down lightly with very fine steel wool or 6/0 garnet finishing paper moistened on the back. Rub only until all gloss has been removed and the surface presents a uniformly dull appearance with no highlights or hollows.
5. Carefully remove all traces of dust from previous use of steel wool or garnet paper by wiping with a damp cloth. Dry surfaces immediately. Now apply a coat of rubbing varnish and allow it to dry at least 24 hours. When dry, sand with 6/0 finishing paper moistened with benzine. Sanded surface should be dull and smooth when the work is completed.

6. Carefully remove all traces of dust and apply a second coat of varnish. Rub down with 6/0 paper moistened with benzine, after varnish has dried. Two coats of varnish will usually give a satisfactory finish. If not, apply additional coats, rubbing each coat as before.

7. When last coat has been sanded, rub it down with a piece of felt (a standard rubbing block or a piece of an old felt hat) dipped in water and lightly sprinkled with very fine pumice powder. Rub a small area at a time. Clean off pumice occasionally to inspect the progress of your work. When the entire surface is uniformly dull and smooth, with no obvious blemishes, the rubbing is completed. This will give a satin gloss that is best brought out by the application of a coat of paste wax. If a high finish is desired, saturate a clean felt with rubbing oil and apply a small quantity of rottenstone. Rub as before.

The success of the operation will depend upon careful inspection at frequent intervals and a strong right arm. It is important that all parts (base, cabinet, and molding) be finished before final assembly. However, the molding may be given the final varnish coat after fastening it to the cabinet. After the molding has been stained, shellacked, and given the first varnish coat, fasten it to the cabinet with brads. Set the brads slightly below the surface, and fill holes with plastic wood or a mixture of glue and sawdust. If the sawdust used is the same color as the rest of the cabinet, it can easily be concealed with a dab of stain and a second coat of varnish.

When the cabinet has been finished and assembled, the end grain of the top panel may be concealed with colored shellac. This is made by dissolving alcohol-soluble stain powder in thin shellac until the desired color is obtained. Apply the shellac to the edges of the panel with an artist's brush. Use as many coats as needed.

4 Record Players

In practically all wide-range home music systems the record player is the principal source of program material. In fact, it was the development of the modern long-playing record and the resulting excellence of recorded music that led to the popularity of such systems.

All of the components of a high fidelity system should be carefully selected only after critical examination and extended listening periods; the record player is no exception to this rule.

Many high fidelity enthusiasts prefer to use a manually operated player for reasons that will become clear a little later. However, under some circumstances the use of a manual player may be inconvenient. One reason for the popularity of manual players is that the individual components (motor and turntable, pickup arm, cartridge, and stylus) may be individually selected and assembled by the owner. Another reason is that the manufacturer of a record changer attempts to make a product as compact as possible and quite often cuts corners in construction to meet competition, with the result that the average moderately priced changer is likely to use components that are lighter, not as carefully engineered, and often inferior in construction.

When the high fidelity listener selects and assembles the parts of a manual player, he does not do so to save money. In fact, a really excellent unit usually costs several times as much as a good automatic changer. However, he then has the privilege of selecting each component for its superlative qualities. This should not be taken to indicate that all changers are undesirable or poorly constructed; on the contrary, there are many fine units available. But in the final analysis, the finest changer cannot compare with the finest manual player.

Basically, a player, whether manual or automatic, consists of a small motor for driving the record turntable. There is also a cartridge, used to change the mechanical impulses stored in the record grooves into electrical impulses that can be applied to the amplifier. The variations in the record grooves are communicated to the cartridge by means of the stylus, or needle. The cartridge and stylus are supported and guided along the record grooves by means of the pickup arm. Up to this point, we have listed all of the components of a manual player. The changer, in addition to these, has a mechanism for automatically playing a sequence of records, for raising and lowering the pickup arm at the proper time and, usually, for stopping the motion of the turntable and returning the pickup arm to the rest position after the last record has been played.

Phonograph Motors

Many motor-turntable combinations for use in manual players use the hysteresis synchronous type of electric motor. It operates on much the same principle used in the common electric clock and, as we might expect, one of its features is constant speed under a variety of load conditions. Another important feature is that motors of this type are far less noisy than others. We are not now referring to noise that may be detected directly, but to electrical disturbances that may be picked up by the cartridge (particularly a cartridge of the magnetic type) and conveyed to the amplifier. We shall enlarge upon this point a bit later.

Early record changers (and some manual players) used a two-pole induction motor. With the advent of the long-playing microgroove record, it was soon discovered that this type of motor was entirely unsatisfactory. The reason is that some phonograph cartridges have a tendency to pick up hum from the motor windings as they pass over the record surface. This tendency can be minimized by increasing the number of motor poles. The weakness of the two-pole motor became more evident with the introduction of long-playing records, because this type of recording has a much lower output than the 78 rpm variety. As a consequence, the amplifier volume control must be turned up higher and any hum present then becomes more evident. The situation was further complicated by the use of the magnetic cartridge. This type has a much lower output than the crystal variety and thus requires a higher degree of amplification. And as amplification is increased, electrical disturbances become more noticeable. The result was the record player manufacturers substituted the four-pole motor for the two-pole type. As mentioned before, manufacturers of motors for quality manual players employ the synchronous principle. And almost all record changers of current production use four-pole induction motors. We may say, without qualification, that the use of one or the other is mandatory in a wide-range music system. If you can afford it, the synchronous motor is to be preferred. Under no circumstances buy a player using a two-pole motor if you intend to use it for 33 1/3 or 45 rpm records.

You can easily check the level of hum induced in a pickup cartridge by the motor. Turn the amplifier volume control up to full room volume. With the motor running, very carefully guide the pickup arm across the turntable, without allowing the stylus to touch it. Repeat this several times and note whether or not the hum increases above normal. If it does increase noticeably, the additional hum is caused by the motor and the unit should be rejected. Bear in mind that hum due to this condition cannot be eliminated without substituting another type of motor.

Another form of distortion that may or may not be caused by the motor (or its drive mechanism) is the "wow." This may be described as a rise or fall in pitch; sometimes both an increase and decrease in frequency may be noted. When this condition results from defects in the motor or its associated drive mechanism it is almost always caused by either speed variations in the motor itself or irregularities in the drive. Sometimes wows can be detected only by critical and extended listening. Incidentally, an excellent selection for checking this condition is Rimsky-Korsakoff's "Scheherazade." Near the end of the fourth movement there are sustained notes that will immediately reveal any speed changes in the system. However, you must be certain that the particular recording used is a perfect product, for this condition may also be caused by speed changes that occur during recording.

Many phonograph motors apply power to the turntable through the medium of a selective drive similar to that shown in Figure 4-1. The main motor spindle carries three bushings, each of a different diameter. One bushing is provided for each of the record speeds commonly used: 33 1/3 rpm, 45 rpm, and 78 rpm. An idler wheel with a rubber rim is held against the bushing by spring pressure. In turn, the idler wheel bears against the inner rim of the turntable. By shifting the idler wheel from one bushing to another (this is accomplished by a speed change lever) the turntable is driven at the desired speed. Irregular or incorrect speed may result from eccentric or poorly machined bushings. Also, oil or grease on the bushings or on the rubber rim will cause slipping and consequent speed variations. Oil or grease may also rapidly deteriorate the rubber.

This brings us to another important point in the selection of a motor. If the rubber idler wheel is allowed to bear against the turntable during long periods of inactivity, the constant pressure at only one point on its circumference may produce a flat spot at that point and the flat spot will almost certainly cause wows. Some manufacturers have guarded against this by providing a neutral position in the speed change mechanism; when in the neutral position the idler wheel is free. You are advised to select a motor or player having this feature.

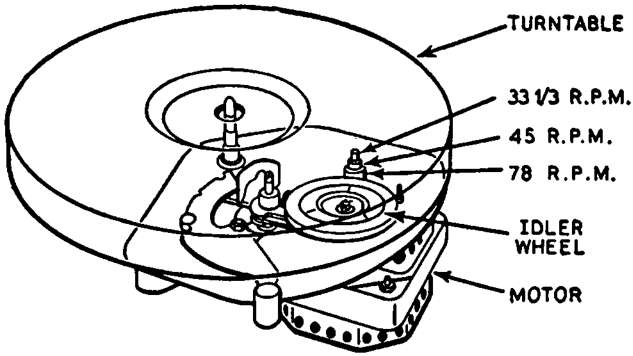


Fig. 4-1. Three-speed phonograph motor.

Most record changer motors are provided with the three standard speeds only. However, the speed of some motors intended for use in quality manual players is continuously variable. In one model the speed range is from 25 rpm to 100 rpm. And at least one mass-produced phonograph has a variable-speed record changer motor with a speed range down to about 16 rpm. This may seem to be an unnecessary feature, but it is interesting to some record collectors for the reason that, during the early 1900s, record cutting speeds were not standardized at 78 rpm as they are today. Many such collectors' items were cut at speeds ranging from 70 to 100 rpm.

A motor-turntable combination with continuously variable speed must be provided with some means for accurately adjusting the speed to the desired rate. Although some such units have built-in provision for speed regulation, you will, in most cases, have to resort to the use of a stroboscope disc. This device may also be used for checking the speed of a fixed-speed motor. It is a circular card printed with several series of dots or bars, each series related to a turntable speed. To use a stroboscope disc, place it on the turntable as you would a record and adjust the speed change lever to give the desired speed. If, for example, the turntable is to be checked at 78 rpm, you then observe the circle of dots or bars labeled "78 rpm." The observation is best made by the light of a fluorescent, argon, or neon lamp, although it is possible to use an unfrosted incandescent lamp. The motor should be allowed to run for two minutes or so before the test is made. If the motor has continuously variable speed, adjust the speed control until the proper circle of dots or bars appears to be motionless. If you are testing a motor with one or more fixed speeds, then any apparent rotation of the circle of dots or bars indicates that the actual turntable speed is either higher or lower than it should be.

Unless a motor is properly insulated from the motor board or record player main plate, its vibrations will be communicated to the board or plate and so to the turntable and record. The vibrations will show up as the familiar low-pitched "rumble" often quite audible during the softer passages in a recording. To minimize this noise, the motor is suspended by means of rubber bushings. However, if the bushings are not soft enough or if, after a long period of use, the rubber becomes dried out or is damaged, the rumble will be evident. The quickest way to check for rumble is to use one of the test records having plain grooves. Listen carefully with the amplifier adjusted for full room volume. Under these conditions, a high-grade motor should show little or no rumble.

Of course, you can expect to find some rumble in low-priced players. It can be minimized by placing a circular cushion of foam rubber between the record and the turntable. Such a pad has several other advantages. First of all, when used with an automatic changer it helps cushion the shock as the record drops from the spindle. When playing the very light 45-rpm records it almost entirely eliminates the tendency of the record to slip on the turntable. Finally, it helps combat one of the worst enemies of a record—dust. It is certain that a great deal of dust will collect on the average flocked or felt-covered turntable; in fact it may be classified as an ideal dust-catcher. The rubber pad, however, can be easily removed for frequent cleaning. Pads of this type are currently selling for from two to three dollars, but you can make your own for a fraction of this cost. A piece of ¼-inch thick foam rubber a foot square can be bought at most large department stores and at practically all shops selling upholstery materials. Cut it to the exact diameter of the turntable with a pair of shears, then cut a 5/16-inch hole in the center for the spindle. Don't worry about a slight eccentricity in the center hole as this will have no effect on reproduction.

Turntables

The turntable is considered to be and is sold as part of the motor-turntable assembly, yet it is sufficiently important to be considered by itself. Some of the desirable features of a good turntable are: (a) it must be heavy, (b) it must be machined so that its periphery is absolutely concentric with the spindle, (c) it should be cast rather than stamped, and (d) its top surface must be absolutely flat.

When in operation, the turntable acts as a flywheel and so helps maintain constant speed. We can easily understand why weight is important here. A heavy turntable will develop more flywheel effect than a lighter one, and this is a second reason why the more critical listeners prefer manual players to automatic changers. All high quality players (as well as motor-turntable assemblies sold for this purpose) have turntables machined from heavy castings (usually of aluminum alloy), whereas the average changer employs a much lighter, pressed steel part. It is true that some changers have heavier turntables than others; this should guide you in making a selection, providing that all other considerations are equal.

We need hardly emphasize that the turntable must be accurately machined (or stamped, if the pressed steel type is under consideration). Any tendency toward eccentricity, any undulation of the top surface during rotation, will certainly show up as some form of distortion. Such errors, unless they are very serious indeed, cannot be detected by visual examination, but require critical listening over an extended period.

Aside from flatness of the top surface, the turntable must be level, although this is often a matter of installation. It is hardly likely that any record changer or any factory-assembled manual player worth considering will have a turntable that is not level with reference to its bed plate or motor board. But it is possible to destroy entirely the accuracy the manufacturer has built into the player by installing it in such a way that the entire mechanism is tilted slightly. We see then that a final check must be made after installation to be sure that the player is mounted so that the turntable is in a level position. This is particularly true in units designed for spring mounting. In such cases it is often necessary to insert thin shims of cardboard under the springs to level the unit. Checking can be done with a small spirit level, available at all ten cent and hardware stores. Be sure to check the conditions of the turntable with the spirit level in two positions: at right angles to the pickup arm and parallel with the pickup arm.

Pickup Arms

It is obvious that if you decide to buy an automatic changer you will have little or no choice of pickup arm, although some manufacturers do offer one type of arm for crystal cartridges and another type for magnetic cartridges. A few manufacture plug-in type arms so that the cartridge of your choice may be inserted.

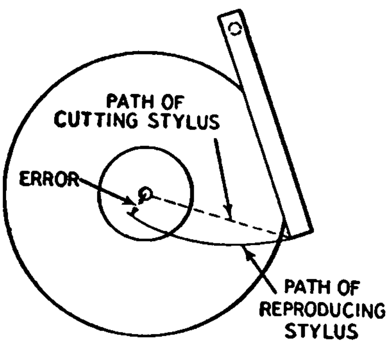


Fig. 4-2. Tracking with short pickup arm.

Discriminating listeners (who also have the means) prefer to use a manual record player because they may then select a pickup arm having certain excellent mechanical features that cannot possibly be built into arms used in low-priced changers. This does not mean that all changer arms are bad, but it is true that they cannot compare with high-quality, transcription-type arms.

The function of a pickup arm is to hold the cartridge and guide it along the record grooves. This may seem, at first thought, to be a simple matter, but if the process is not carried out accurately the result may be poor reproduction and excessive record wear. The pickup arm must follow, as closely as possible, the path of the cutting head used in making the original recording. As we shall see, no pickup arm can do this precisely, but the error can be reduced to a minimum and, as we might expect, the smaller the error of the pickup arm the higher the price.

In recording, the cutting stylus moves in a straight line from the circumference of the record to the center. The stylus used for reproduction must move in an arc because the pickup arm is pivoted at one point. The ability of the reproducing stylus to follow, as nearly as possible, the original path of the cutting stylus is called tracking. Poor tracking, or a high degree of tracking error, results in distortion and excessive wear of record groove and stylus. It is evident that an arc approaches a straight line if its radius is large enough. This means that a long pickup arm is desirable. Figure 4-2 shows why a relatively short pickup arm introduces serious tracking error. Notice the discrepancy between the path of the reproducing stylus (solid line) and the path of the cutting stylus (dotted line). Now refer to Figure 4-3. Here a longer arm has been used and it is obvious that the tracking error has been reduced.

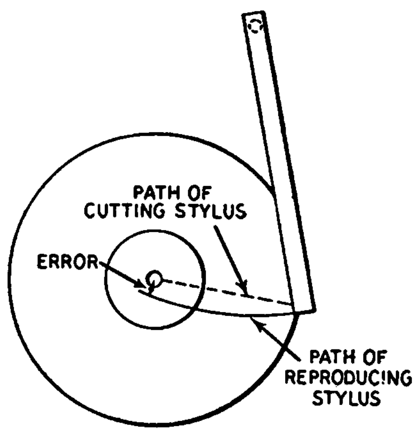


Fig. 4-3. Tracking with longer pickup arm.

The effective length of an arm is not its over-all length but the distance between pivot point and stylus point. Of course, there are limits to the length of the arm, imposed by space considerations. In the average automatic changer the radius of the arc described by the stylus point is from 7½ to 8 inches. One reason for limiting the arm length is a perfectly natural desire on the part of the manufacturer to produce a unit that is as compact as possible. By contrast, the better-grade, transcription-type arms may have a length of as much as 12 inches, with a corresponding reduction in tracking error. When we compare prices, we find that a really excellent arm, less cartridge, often costs as much as a complete, high-grade automatic changer.

Before leaving the subject of pickup arms, we must consider some other factors. Earlier pickup arms were almost invariably of the straight type, see Figure 4-3. It was soon found that such arms introduced another kind of error in the motion of the stylus, for the stylus could then be tangent to the record groove at only one point in its travel from record circumference to center. This type of error is reduced by using either the offset head, Figure 4-4A, or the curved arm, Figure 4-4B.

If your player is to be used for all record speeds, you will need either two separate cartridges or a cartridge with two styli. The

Fig. 4-4A. Offset head pickup.

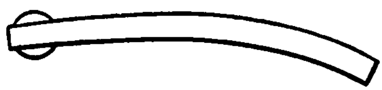


Fig. 4-4B. Curved arm pick-up.

reason for this will be explained more fully later on. Manufacturers of pickup arms (and, to some extent, manufacturers of changers) have solved the problem by using the plug-in type of construction. Two separate cartridges are required and they are easily interchangeable. Although not the only solution, many owners of high fidelity systems claim that this method is the best. This is probably true in models where the arm has a convenient stylus pressure adjustment, for then the user is assured of correct pressure for each cartridge.

Another method (this is employed by most changer manufacturers) is to use a turnover type of cartridge or a cartridge with a reversible stylus. The turnover principle is generally confined to crystal cartridges and need not be discussed here. At least one make of magnetic cartridge uses a reversible stylus. By turning a small knob on top of the pickup arm the proper stylus is brought into playing position.

To sum up the differences between manual players and automatic changers we may list the following points:

1. The high-grade manual player has a superior type of drive motor.

2. Manual players are almost always equipped with a heavy, cast, accurately machined turntable. Changer turntables are often much lighter and stampings are frequently employed.

3. Manual players usually employ longer pickup arms, almost always of excellent design and construction. This reduces stylus and record wear and affords a noticeable improvement in reproduction. Changers are limited to relatively short arms due to space considerations.

4. A really good manual player (or its components) costs far more than a good changer but is highly desirable if price is not a primary consideration.

5. The automatic record changer offers greater convenience, but this becomes of lesser importance if only long-playing records are to be used.

Cartridges and Styli

During recording, the cutting stylus describes a spiral from the circumference of the record disc to the center. This spiral consists of a groove cut into the record blank by the stylus. When the cutting head is not actuated by the amplifier and microphone, the walls of the spiral groove will be quite smooth, but when sound waves strike the microphone they are converted into electrical impulses and produce rapid side-to-side movements of the stylus, thus cutting very slight lateral variations or undulations in the walls of the spiral groove. For purposes of this discussion we may say that the degree of side-to-side motion depends upon the intensity of the sound waves while the rapidity of the motion is governed by the frequency of the sound waves.

The reproducing (playback) stylus closely follows the variations in the groove walls. In the magnetic, or variable reluctance, type of pickup cartridge, the stylus motion makes an armature located between the poles of a magnet vibrate, which sets up electrical impulses in a pair of coils wound on the magnet poles.

The crystal type of pickup cartridge depends for its operation on the piezo-electric effect, discovered by the Curies in 1880. When certain types of crystals are subjected to mechanical stress, such as bending or twisting, an electric charge is developed at the faces of the crystal. Crystals exhibiting this property include quartz, Rochelle salt and tourmaline. Early crystal pickup cartridges used Rochelle salt exclusively.

Rochelle salt crystal cartridges have never offered serious competition to variable reluctance and other magnetic cartridges. The reasons were that the magnetic cartridge offered wider response and was far more rugged. The Rochelle salt crystal was subject to breakage due to rough handling and would deteriorate under conditions of high humidity or high temperatures (over 120° F.).

Recent developments in the crystal cartridge include the ceramic, which employs barium titanate. These newer cartridges are not so delicate as older Rochelle salt cartridges and late models have a response approaching that of magnetic cartridges. Furthermore, they have a generally higher output voltage than magnetics and in some cases no preamplifier is needed.

The average magnetic cartridge will reproduce frequencies up to about 15,000 cycles per second. Its lower voltage output makes it necessary to use a pre-amplifier between the pickup and the main amplifier. It is not so sensitive to temperature and, although no cartridge may be considered a rugged device, the magnetic type is less likely to be damaged by rough treatment than is the crystal type.

As indicated earlier in this chapter, a record player that is to reproduce all records cut at the three standard speeds of 33 1/3 rpm, 45 rpm, and 78 rpm must have either two separate cartridges, one for 78-rpm records, the other for both 33 1/3- and 45-rpm records or a cartridge with two styli. The reason for this is that the grooves in a 78-rpm record are wider than those in the 33 1/3- or 45-rpm varieties. For the wider grooves a stylus having a tip radius of 0.003 inch is used; for LP and 45-rpm records the tip radius is 0.001 inch.

Many makes of transcription arm and a few automatic changers are equipped with the plug-in cartridge arrangement, a feature more likely to be found in the higher-priced changers. Changers in the lower-priced field usually use some type of turnover crystal cartridge or a "triple-play" magnetic cartridge.

There are at least four makes of magnetic cartridge on the market and, although all are excellent, the more popular ones at the present time appear to be those of the General Electric and the Pickering companies.

Although the stylus is the smallest component of a wide range music system, it is one of the most important, for upon its proper condition may depend the life of your record collection and the quality of reproduction.

A dozen or so years ago, the most widely used stylus consisted of a sharp steel needle; its point was rapidly worn down by the abrasive action of the record surface and the stylus had to be discarded after a single play. In an effort to increase stylus life, manufacturers applied a plating, often of chromium, and although stylus life did increase to about ten plays, the gain was made at the expense of the record. Still later came the osmium stylus and soon thereafter cartridge manufacturers introduced a stylus chuck fitted with a concealed set screw making it necessary to either remove the cartridge or use a special tool to replace the stylus. This was based upon an anticipated stylus life of about 5000 plays. It was believed then that the permanent stylus had arrived.

It is well known that there is no such thing as a permanent stylus. Currently manufactured styli are of three types: osmium (sometimes called "precious metal") styli, sapphire, and diamond. More and more attention is being focused on styli by the authors of textbooks and magazine articles and by record dealers. It is rare these days to walk into a very large record shop that has not installed a "needle clinic" where the music system owner may take his stylus for microscopic examination. And from various experts we hear estimates (often conflicting) regarding the expected life of a stylus. One source states that the osmium tip is good for no more than 15 hours of play, the sapphire for about 25 hours, and the diamond for a year. By contrast, the jacket of one make of high-quality, long-playing record advises the user to check metal or sapphire styli at least once each six months. Now, it is apparent that these two statements are in disagreement. If the sapphire tip can be counted on for only 25 hours of play, this would then be equivalent to playing both sides of about 42 long-playing 12-inch records and, in the course of six months, the owner would be allowed only 7 plays per month. It is safe to assume that either the life of the sapphire tip is often underrated or the record manufacturer mentioned (who, by the way, makes a superb product) is deliberately offering dangerous advice to the consumer. This does not seem likely.

If we analyze such statements more carefully, we shall probably find that the first source intends to convey the impression that a sapphire stylus cannot be depended upon for more than 25 hours of play; some individual specimens might fail before the end of their anticipated useful life, while still others may last far longer. At the same time, the record manufacturer probably means that a sapphire tip is safe to use for a period of six months (or more), provided that it remains in good playing condition. To be on the safe side, we advise that you be aware of the condition of your stylus at all times and this can be done only by frequent examination.

There is no question that the diamond stylus is far superior to sapphire, and no serious high fidelity enthusiast would consider using a metal stylus. When the average life of the stylus is considered, your stylus dollar goes much farther when you invest in a diamond. A few years ago, when diamond styli were much more expensive than they are now, there may have been good reasons for choosing a sapphire, but now the prices of both types are within reach of all.

Prices of both sapphire and diamond styli vary considerably, depending upon the manufacturer. In most instances the price of a replacement stylus made by a cartridge manufacturer for use with his cartridge is higher than one sold by a manufacturer who concentrates almost exclusively on styli. For instance, a diamond stylus sold by a manufacturer of a popular magnetic cartridge as a direct replacement in his cartridge is priced at $10, while an equivalent replacement made by a stylus manufacturer sells for about $3.

Some of the evidences of stylus wear are: high frequency distortion, lack of high frequencies, excessive surface noise, jumping of stylus of grooves, sticking of stylus in one groove, and "repeats." Keep in mind the fact that most of these symptoms may also be caused by other defects. However, when one or more of these effects are present and are the result of a worn stylus, the damage has already been done to one or more records. At this point you may well ask the question: "How may I detect stylus wear in its early stages?" There are two answers. The simplest, and often the best, procedure is to remove the stylus and take it to a reliable record or high fidelity equipment dealer who offers a stylus examination service. The alternative is to perform your own examination at regular intervals.

If your pickup arm is equipped with plug-in cartridges, it is a simple matter to remove a cartridge for inspection of the stylus. You should have received, with the arm, instructions for doing this in the plug-in types. In other units, the cartridge is held in place by means of several screws. As there is a wide variation among different makes, we have selected the popular General Electric cartridge as an example. Figure 4-5 is an underside view of this cartridge mounted in the arm and shows the screws as well as the connecting wires. The ends of the wires are fitted with small metal clips and these slip over small pins on the back end of the cartridge. If your cartridge is of the triple-play type, the first step is to remove the small knob at the top of the arm by pulling it straight upward. Tilt the pickup arm as far back as it will go and slip the two connection clips from the pins. Now remove the two screws and the cartridge will then drop out of the arm. The single-play type of cartridge has, of course, no stylus control knob.

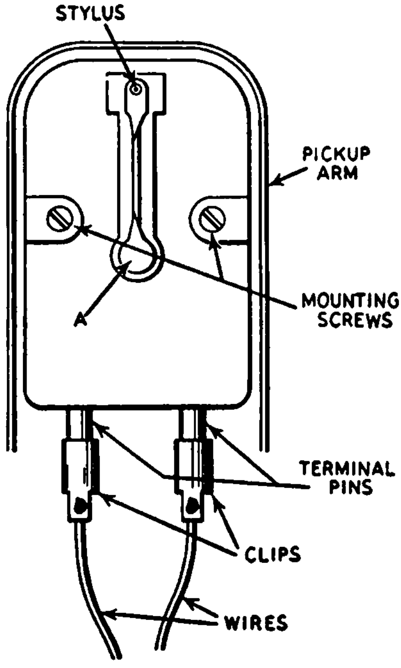
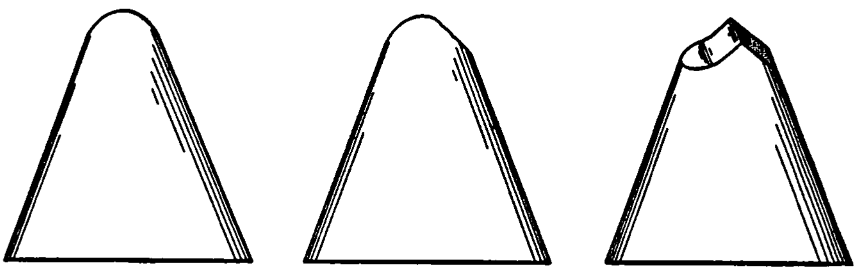


Fig. 4-5. General Electric single-play cartridge.

A 10-power jeweler's loupe may be used for examining the stylus, but careful examination must be made, for the magnification is hardly sufficient. Many optical and camera shops now sell imported pocket-type microscopes, adjustable to 20, 40, and 60 diameters. They sell for about three dollars and are ideal for this purpose, except that a little practice is needed since there is no fixed platform or stage for holding the object. The work must be done under strong light and you will find that it is necessary to hold the cartridge, the microscope, and perhaps the light in various positions to get the best results. In some cases it may be desirable to remove the stylus from the cartridge to facilitate examination. This procedure will be described later.

To judge the progress of wear, you will need some kind of guide; this is offered in the sketches, Figures 4-6A, 4-6B, and 4-6C. A new, perfect stylus is shown in Figure 4-6A. This illustration was drawn from a photograph of a perfect stylus and clearly shows the shape of the tip. Obviously, any departure from this shape constitutes wear, but it is true that very small deviations, difficult to detect even under fairly high magnification, will occur relatively early in



A B C

Fig. 4-6.

the life of the stylus and may be disregarded. When the tip has worn to the contour shown in Figure 4-6B, wear has reached an advanced stage and it is time to think seriously of a replacement. The pronounced chisel tip and shoulders seen in Figure 4-6C indicate that the stylus has definitely outlived its usefulness and is now a menace to your record collection, as well as to fidelity.

Perhaps some information on stylus replacement in the General Electric cartridge will be helpful. In all cases, the cartridge should be removed from the arm. The single-play cartridge has a small hole in the top, almost in the exact center. Straighten an ordinary paper clip and insert the end into this hole and the stylus assembly will then drop out. To insert a new stylus, lay the cartridge on a table with the stylus side up. Lay the stylus assembly in position so that the pin is entered in the hole, or socket, in the cartridge and the stylus arm centered between the magnet pole pieces. Now seat the stylus in its socket by applying pressure to the point marked "A" in the drawing, Figure 4-5. In doing this, be careful not to bend any of the stylus assembly parts.

A cross-section of the General Electric triple-play cartridge is shown in Figure 4-7. Support the stylus assembly by placing your finger or thumb at the center of the stylus arm, point "A" in the sketch. Do not apply pressure at any other point. Now compress the spring slightly and remove the retaining washer. The spring may now be removed and the stylus will drop out. To install a new stylus, slip it into position, put the spring in place, compress the spring, and replace the retaining washer.

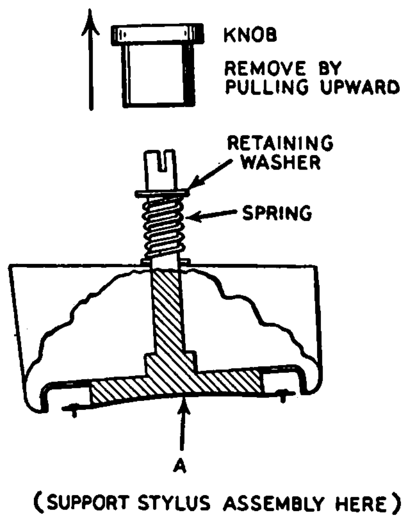


Fig. 4-7. General Electric triple-play cartridge.

The earlier General Electric triple-play variable reluctance cartridge has now been replaced by the type VR-II using "slide-in" styli. This cartridge has two distinct advantages over the type just described.

First, the frequency range has been extended so that the response is now 20 to 20,000 cycles. Second, the method of stylus replacement has been greatly improved. In the older type, both styli were permanently fixed to the stylus holder. When one stylus wore out (this was usually the 0.001, or 1-mil stylus) it was necessary to replace the entire stylus assembly and both styli, even though the 0.003 or 3-mil stylus happened to be in good playing condition because, in the average case, it received relatively little wear. Furthermore, with this type of cartridge it was necessary to remove the pickup cartridge from the pickup head or from the arm in order to replace a stylus.

The VR-II stylus assembly is constructed as shown in Figure 4-8. Notice that the individual styli are mounted on removable clips; these clips slide into a channel in the stylus holder. The slide-in tip is held in the channel by the two little flat springs or wings indicated at A, A in the drawing.

The procedure for replacing a stylus is as follows. If the pickup arm is equipped with a plug-in head, remove the head from the arm. If the arm does not have a plug-in head, then raise the arm to its maximum height. Next, depress the stylus holder knob and turn it so that the stylus assembly lies crosswise of the pickup. Grasp the slide-in tip at the points marked A, A in Figure 4-8; ap-

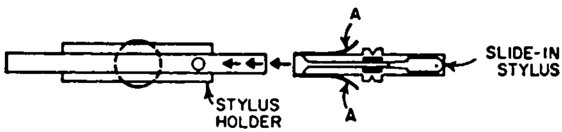


Fig. 4-8. Replacing G.E. slide-in stylus.

plying slight pressure to the springs will release the tip and permit you to slide it out of the channel in the holder. To insert a new stylus just insert the slide-in tip into the channel and slide it back as far as possible, making sure that the flat side of the slide-in tip lies flush against the bottom of the channel.

Stylus assemblies of the slide-in type are now available for the older triple-play cartridges at moderate cost. Once such a stylus assembly has been installed you will find that subsequent stylus replacement will be much easier and there will be no need to discard a perfectly good stylus because the remaining one in the assembly has worn out.

Stylus Pressure

Stylus pressure, sometimes called stylus force or tracking force is a most important factor in minimizing record wear and reducing distortion. Each cartridge manufacturer specifies a recommended pressure to be used with his product. At the present time, the average recommended force is from 6 to 8 grams for monaural cartridges and from 2 to 7 grams for stereo cartridges.

If you own a transcription or manual player, by all means follow the cartridge manufacturer's recommendations concerning stylus pressure as closely as possible. But, in a well-intentioned attempt to reduce pressure so as to minimize stylus and record wear you might easily reduce the pressure considerably beyond the recommended value with the result that on heavy modulation the stylus will tend to ride too high in the record groove and perhaps will skip grooves.

Owners of automatic changers may sometimes have to effect a compromise between the stylus pressure specified by the cartridge manufacturer and that suggested by the changer manufacturer. For example, the maker of one widely sold changer specifies 8 to 10 grams, without reference to any particular make of cartridge. The manufacturer of a very popular variable reluctance cartridge recommends 4 to 6 grams for his product. With this particular combination of cartridge and changer it was found to be impossible to eliminate groove skipping at 4 or even 6 grams. Finally, 7 grams was accepted as a compromise.

The foregoing indicates of course that you should have a stylus pressure gauge; even though your player was precisely adjusted at the factory, stylus pressure can change due to a number of conditions. Gauges are currently selling for under three dollars (in one case a dollar and a half will buy a very good imported product), and the investment will return dividends in the form of uniformly high performance and reduced wear. Gauges are of two general types: those depending upon a spring and those using the balance principle.

In the first type the stylus rests upon a flat spring; deflection of the spring is read from a scale calibrated in grams. This type is the easier to use but may develop inaccuracies on account of changes in spring tension. The balance type resembles a miniature see-saw. The stylus rests near one end of the pivoted bar. On the other side of the pivot there is a series of holes, located at varying distances from the pivot. A small weight can be shifted from one hole to another as needed in order to balance the pressure of the stylus. Each position of the weight is marked in grams, of course. The calibration of this type of gauge is not likely to change, but it is not as easy to use as the spring type.

Whether you use a manual player or a changer, the proper place to measure stylus force is as close to the surface of the turntable as possible. In some types of gauge employing the spring principle the pickup rest is as much as an inch above the actual surface of the turntable. This may not be important in the case of the better constructed, more delicately balanced transcription type arm, but may make a big difference in an automatic changer. Then too, remember that in a changer as more records drop to the turntable the height of the pickup above the turntable will increase. Ideally, the stylus pressure should remain constant regardless of pickup height, but it has been found that this is rarely true; usually the pressure decreases as the record stack builds up. If you are in the habit of loading a number of records on your changer you may have to compensate for this effect by adjusting stylus pressure to the desired value with the pickup resting on the top record of the stack, instead of at turntable level. Otherwise, there may be some skipping of grooves during the playing of the last record or two.

While on the subject of groove-skipping, you should remember that not all of this trouble is due to incorrect stylus pressure. Sometimes it is caused by improper leveling of the turntable. Many good transcription players have provision for leveling. Checking should be done with an ordinary spirit level, but do not forget to check in two directions; that is, front to back and left to right across the turntable.

Many manual arms and some changer arms are provided with adjustments for regulating pressure. One transcription arm has a calibrated adjustment so that the pressure may be read directly, but this is rather unusual. It is impossible to describe all of the generally used adjustments, and in practically all instances the maker supplies this information with the arm or the changer.

In concluding this discussion of record players, we find that considerable emphasis has been placed on manual players, although there is at least an equal chance that you will buy an automatic changer. We therefore take the opportunity of discussing, at least briefly, some of the features to be found in changer mechanisms, aside from pickup arms, cartridges, turntables, and motors.

The mechanism for changing records, lifting and raising the arm, and starting and stopping the drive motor is a more or less complicated arrangement of gears, pinions, levers, and springs. Some authorities claim that an over abundance of springs is evidence of poor design; be that as it may, the selection of a changer with a minimum of moving parts would certainly be sound judgment. The mechanism should be smooth acting and should handle records as gently as possible. Two different schemes are used to separate the bottom record of a stack from the others. In one method, the stack rests on a shelf, or ledge. Forward movement of a sliding plate pushes the record from the supporting shelf and allows it to drop to the turntable. In the alternate design, a latch located within the spindle structure supports the center of the stack. When actuated by the changer mechanism, the latch allows the bottom record to drop. There is usually a ballast arm that drops down over the spindle to hold the records in place. Under some circumstances the latch arrangement may cause wear at the inside of the record center hole and this may eventually result in wows.

A few changers are of the "intermix" type; this means that the stack may include records of different diameters. This feature would seem to be relatively unimportant to listeners who prefer symphonic music. Another variation is an intermix changer that will play both sides of the record, so that no manual turnover is necessary. Since many symphonies and concerti require both sides of a 12-inch long-playing record, this scheme has considerable merit. However, the cost of such a machine is several times that of a conventional changer of good quality.

The average unit will handle a stack comprising ten 12-inch or twelve 10-inch records. Almost all modern changers are equipped with an automatic motor shut-off that operates after the last record has been played. At the same time the pickup is automatically returned to the rest position. In a very few types, the last record is repeated until the changer is turned off. Not all types have a mute switch which effectively eliminates clicks and thuds that would otherwise be audible during the change cycle. One de luxe model is equipped to repeat part or all of any record selected and also has an adjustable pause feature for use between the playing of sides. Almost all changers are designed for spring mounting, to reduce the possibility of mechanical vibration communicated from the loudspeaker to the changer. This is an absolute necessity if the changer is to be mounted in the same cabinet with the amplifier and speaker, a situation which ordinarily does not prevail in wide-range systems.

Selection of Records

The information to be given in this section is based upon mechanical and technical considerations, rather than musical ones. A few magazines and many newspapers regularly publish reviews of current record releases and in most cases these are quite dependable. You will find that many of these review columns discuss the technical excellence of the recording as well as the performance.

Reduced to its simplest terms, the process of record manufacture is essentially a stamping operation. A plastic material (the composition varies with the type of record) is subjected to heat and pressure in a set of dies; the dies, of course, bear slight ridges that are the reverse of the grooves in the finished record. In this way, many copies of the original recording may be made. Also, it is evident that any flaws present in the original will be reproduced in the copies. There is nothing that the record buyer can do about this. However, individual copies will often turn up having flaws that were not present in the original. These are either the result of improper processing or careless handling during or after manufacture. In the paragraphs to follow we shall discuss some of these defects and how to recognize them.

Everyone is familiar with the older 78-rpm shellac records. These records are termed "shellac" because that was the principal ingredient used. If you have handled many of these, you are well aware that they are easily broken. During the years of World War II the condition was made much worse by the use of reprocessed shellac which rendered the record much more brittle. Columbia helped matters by using a laminated construction, but even this was not proof against old-fashioned record changers having knife-type selector blades. One other characteristic of shellac pressings was a high level of surface noise as compared to 33 1/3- and 45-rpm discs. In addition to these limitations, the shellac record warped easily and this often resulted in wows. While the fidelity of most such recordings is not nearly as good as you may expect from current LPs, some very excellent shellac pressings have been made. It is likely that a large percentage of 78-rpm production is at present devoted to popular music. As the longer compositions required the use of as many as twelve 12-inch record sides, the use of the 78-rpm disc in the field of serious music has dwindled.

The 33 1/3—rpm long-playing record has the outstanding advantage of uniformly excellent fidelity, except for the earliest discs and in transcriptions from older 78-rpm recordings. Transcriptions not only affect fidelity but give a high level of surface noise, which is at a minimum in most new LP recordings. In late LPs there is an almost complete absence of wows. Best of all, the LP, made of plastic, is non-breakable; it must of course be handled with some care, partly because of the narrower and therefore more delicate grooves.

The 45-rpm disc is capable of fidelity equal to that of the LP, but because of its limited length of play is not suited to symphonies, concerti, and the like. The surface noise, as in the LP, is quite low. Although the 45 offers an advantage over the 78 in the matter of storage space, this disc has one or two disadvantages that should be mentioned. Because of the large center hole, it cannot be played on the conventional changer unless a special large spindle is installed or the record is fitted with an adapter. Adapters are available in several types, including metal and plastic. One or two record manufacturers have turned out 45s with a removable center insert, that has a hole of the right size for a standard player or changer spindle. The insert, or spider, can be left in place for playing on a standard spindle or removed for use on a large spindle. Owing to its light weight the 45 has a tendency to slip on the turntable under the weight of some pickups. This is especially true when there are other records under the one in play, and the condition is further aggravated by the use of center inserts that are thicker than the record.

If you live in a large city you will sooner or later come in contact with the "supermarket" method of merchandising records. Establishments of this kind are increasing in number and enjoy the patronage of hordes of buyers for several good reasons: the buyer may select his purchases in a leisurely way, without the attention of clerks; stores of this type are likely to carry an unusually complete stock; and, finally, the very large sales volume and increased operating efficiency are reflected in savings to the consumer. The self-service idea is slowly being adopted by department and other stores but on a smaller scale.

Some years ago it was common to find a row of listening booths in a record shop; this feature is rapidly disappearing, especially in larger stores. In fact, many dealers no longer permit the purchaser to listen to the records before buying. The reasons for this policy are not difficult to comprehend. With a vast increase in sales it would be almost impossible to provide the necessary listening facilities. Then, too, there is the risk of damage when a purchaser attempts to play an LP on an unfamiliar machine. However, you will generally find that if the dealer does not permit the plaving of records in the shop he has a rather liberal replacement and exchange policy.

Although we have mentioned that many low-priced LPs are not in the same class as standard labels, this is not always true. In fact, some makers have been able to turn out an excellent product at low cost and, as a rule, that has been made possible by making the original recordings in Europe. The only way to become familiar with such record bargains is to buy them in a shop that will exchange them for others if they prove to be unsatisfactory.

Whether you select a record from the shelves yourself or it is handed to you by a clerk, you have the privilege of examining it to be sure it is in perfect condition. If you are permitted to select records directly from the stock shelves, you will usually find a half dozen or more copies of each selection to choose from. The average dealer probably will not object to critical examination of the several copies, but in doing this you must be careful to handle the records as if they were your own. Whenever you find a copy that is marred or scratched, remember that the damage was, perhaps, caused by another customer. In your inspection of records, look for these defects:

1. Scratched, worn, frayed, or soiled jackets. This is evidence of careless handling or previous use.
2. Warped records. A record that is warped to the degree that the defect can readily be seen by sighting along the edge should be rejected without further consideration.
3. Condition of label. Scratches or scars on the label may point to previous use. Examine particularly the edge of the label around the inside of the spindle hole. One or more plays on some types of changers will cause a slight wear or fraying of the label at this point.
4. Condition of playing surface. The surface of a record in new and perfect condition should present a uniform shiny appearance. Look for dull blotches; these may be indications of imperfect pressing. In most cases, scratches that are quite apparent will produce clicks or other noises during play. Many new, unplayed records will show very fine scratches in a more or less broad band extending from the outer edge of the record toward the center for a distance of an inch or more. Usually, such fine scratches will not affect reproduction, but they are evidence of careless handling. Sometimes small "blisters" or "pimples" occur on the surface and these are manufacturing defects. They will almost certainly result in clicks during reproduction.

Some record manufacturers are distributing their product in envelopes sealed at the factory. In such cases it is, of course, fairly certain that the record you buy is new and has never been played.

However observe one precaution here. Lately, some dealers, particularly discount houses, have adopted the practice of enclosing all records in their own sealed plastic envelopes. In such cases it is possible that records previously played and returned for exchange or credit have been sealed in new envelopes.

Care of Records

Regardless of type, records should not be stored in an excessively warm location; failure to observe this may result in warping. There are two schools of thought on the matter of storage. Some insist that records should be stored flat, while others prefer to store them on edge. It would seem that warping cannot occur if records are stored absolutely flat. This means that the shelf on which they are placed must be quite flat and smooth. Furthermore, they must not be piled haphazardly, but should be stacked with the edges aligned. It is not a good idea to place records of different sizes in one pile. On the other hand, storing records on edge can be a safe procedure under certain circumstances; otherwise they would not be stored in this manner in the dealer's shop. If you have a shelf two feet long and the space is only half filled with records, those unsupported at one end of the stack will stand at an angle and can warp, particularly if they are in a warm location. The solution to this problem is to use storage shelves or racks that are liberally provided with vertical partitions. This seems an appropriate place to call attention to the proper planning of storage space. Twenty 78-rpm albums, or the equivalent in LPs, weigh a great deal more than you would expect. If the shelf is unduly long, is lightly constructed, and has no support except at the ends, it is likely to sag and may eventually collapse. Be sure that your storage shelves are built of heavy lumber and have adequate support between the ends.

Care should be used when removing a record from its jacket. Never remove it forcibly; instead, bow the jacket slightly by applying slight pressure to the edges. Failure to observe this rule may not only scratch the record but will certainly cause the surface to take on an excessive static charge. As you may have noticed, static is particularly annoying on vinylite discs and will cause dust, dirt, and lint to adhere to the surface. This is impossible to remove by dusting. If dust is not removed some of it will be ground into the grooves by the stylus. Static can be minimized by bowing the jacket, as described, to reduce friction; then, if you find that the record surface is dusty, wipe it with a very soft cloth, moistened with water. A linen cloth is to be preferred. When record surfaces become noticeably dirty or fingermarked, they may be cleaned with one of the special preparations made for that purpose or simply by washing with soap and water.

When placing records on the turntable, be sure that the center hole is properly aligned over the spindle, otherwise abrasion may eventually result in a hole that is out of round or oversized. If you have an automatic changer, it is usually best to stack all the records to be played before placing them on the spindle. This is essential if the records are supported by a spindle latch. Be sure that the stack is properly aligned first, then you can be sure that all discs will slip over the spindle freely without damage.

To play a record manually, lower the pickup arm gently so that the stylus contacts the ungrooved margin of the record. Then carefully guide the arm inward until the stylus rests in the outermost groove. It is not considered a good idea to start the play at any point other than the beginning of the record.

On certain types of magnetic pickup, in spite of careful cleaning of records, dust will eventually collect between the magnet poles and on the stylus arm. The cartridge parts may be cleaned by brushing gently with a small artist's brush. Give particular attention to the gap between the magnet poles, for it is possible for small iron particles to collect here and in time this will affect reproduction.

5 Radio Tuners

SOME TYPE of tuner will be your second source of program material, unless you prefer to limit your listening to records. A tuner, of course, has the advantage that it provides a continuous source of material, but it also has the disadvantage that the type of music wanted is not always available. Much of the material originating in AM and FM broadcast stations (this is particularly true for classical music) consists of recordings, and generally there will be no need for you to make equalization or volume adjustments, once the program has been properly tuned in, for this has usually been expertly done at the station.

In order that we may begin with a fairly accurate definition, we may say that a tuner usually consists of a radio receiver, minus the audio amplifier and loudspeaker. The tuner may have its own power supply or, in some systems, may take its power from the amplifier. Some tuners are equipped with a preamplifier and, in a few instances, with bass and treble controls. While such additions sometimes may be desirable, there is unnecessary and expensive duplication if you have already selected an amplifier so equipped. Many tuners are provided with an electron-ray (or "magic eye") tuning indicator. This device insures accurate tuning and is a distinct advantage in an AM tuner and may be considered an absolute necessity in an FM tuner.

There are two basic types of tuner: AM (or amplitude modulation) and FM (or frequency modulation). One type cannot be substituted for the other, because each is dependent for its operation on a specific system of broadcasting. In addition, there is also the combination FM-AM tuner; many well-designed combination tuners are, in reality, two separate tuners built upon the same base, or chassis.

The AM tuner resembles, in design and construction, an ordinary radio receiver except, as mentioned before, the audio amplifier and speaker are lacking. It receives stations in what we have come to refer to as the "broadcast band"; such stations operate on frequencies lying between 550 kilocycles (550,000 cycles) and 1600 kilocycles (1,600,000 cycles). Dial calibrations, however, are usually condensed to read 55 to 160, in order to save space.

In order to provide space for the maximum number of stations within the available range, all stations are required by Federal law to maintain operating frequencies within very narrow limits of their assignments. The operating frequency of a station is the "carrier" frequency, but it must not be supposed that each station radiates only a single frequency. Actually, each broadcast station occupies a band of frequencies, and the width of this band will depend upon the audible frequency to be transmitted. Thus, if a 1000-cycle note is to be transmitted by a station whose carrier frequency happened to be 1500 kilocycles, the band would extend from 1499 kilocycles to 1501 kilocycles, or a total width of 2000 cycles. The maximum band width is usually about 10 kilocycles and, although such limits are necessary, they seriously affect the quality of transmission, as we shall see.

We now come to an interpretation of the terms AM and FM. In either case, program material originates in the studio in the form of sound waves. The microphone converts the sound waves into electrical waves. Suitable amplifiers then increase the intensity of the electrical waves in the same way your home music system amplifier does. Bear in mind that up to this point the electrical impulses fall within frequency limits that we are already familiar with—30 to 15,000 per second.

The function of the broadcast transmitter is to send the impulses representing speech or music over long distances. While it might be possible to send the amplified impulses directly into space, the distances covered would be so short as to render this impracticable. If distances of 20, 50, or 100 miles are to be spanned, some other method must be used. This is accomplished by the use of a carrier wave. Although audible frequencies will not cover great distances, radio frequencies—those lying beyond the range of audibility—will do this with ease.

At the radio station the desired carrier frequency is generated by an electronic device called an oscillator. After passing through another series of amplifiers the carrier wave is ready to be modulated by, or mixed with, the audio frequencies. In Figure 5-1 the carrier wave is shown at the upper left of the illustration; the audio frequency appears immediately below it. However, this representation of the two waves is not accurate technically since they cannot be shown in true frequency proportions. However, the diagram is suggestive of the modulation process. During the process of modulation, the two waves combine and then appear as shown at the right of the drawing. This is the modulated carrier and it too has the ability to travel over long distances, carrying with it the desired intelligence in the form of music or speech. You will notice that the modulated wave has an envelope or outer contour that has essentially the same shape as the original audible signal. Within this

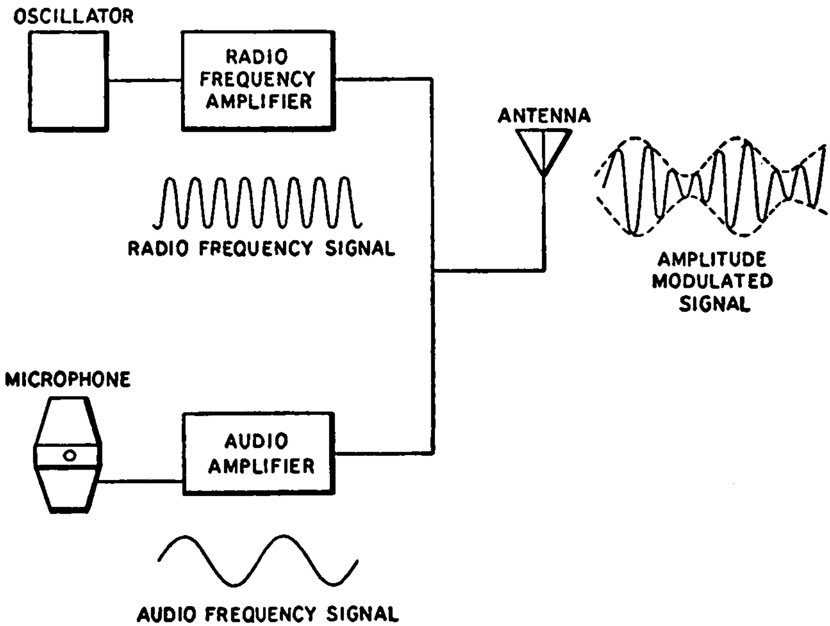


Fig. 5-1. Amplitude modulation.

envelope you will see the pulsations of the carrier wave. Note, especially, that the height, or amplitude, of the carrier has been caused to vary in accordance with the variations in the audio signal. The carrier has been modulated by altering its amplitude, hence the term amplitude modulation.

To pursue this description a bit further, we find that the primary function of the tuner is to separate the carrier from the audio signal and discard the carrier, since it has now served its purpose. This process can be performed by the use of a single tube—the detector —or, in fact, without employing any tubes at all by substituting a crystal detector. But a simple tuner consisting of just one tube or the crystal detector would be far from satisfactory; it would lack two highly important qualities: sensitivity and selectivity.

Sensitivity may be defined as the ability to pick up weak signals; it may be increased by adding one or more radio frequency amplifier tubes ahead of the detector. Selectivity is the ability to receive one signal to the exclusion of all others; it is gained by adding tuning elements—each consisting of a coil and a condenser—to the

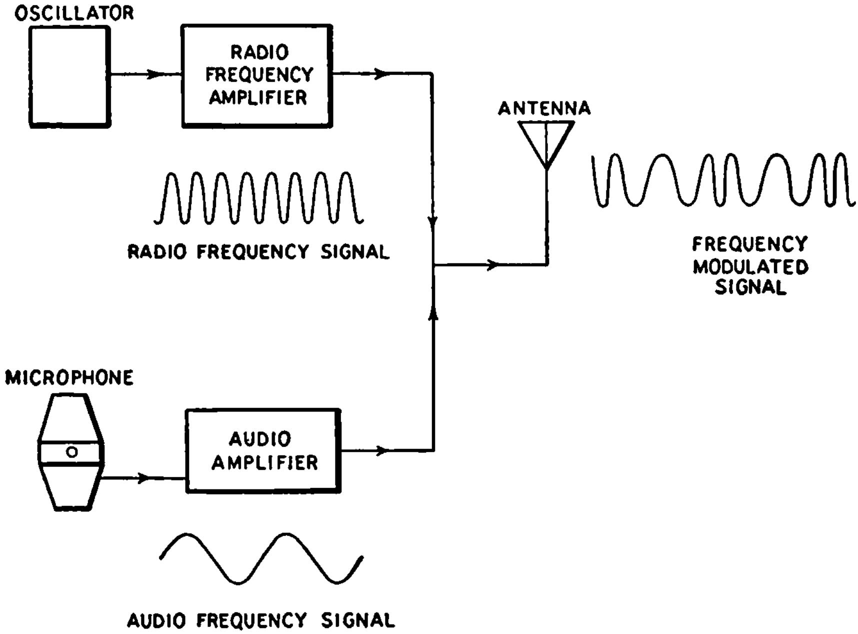


Fig. 5-2. Frequency modulation.

tuner. As the usual amplifier "stage" consists of tube together with the associated tuning elements, we find that, generally speaking, the addition of a radio frequency amplifier stage improves selectivity and sensitivity at the same time.

In 1936 Major Edwin H. Armstrong devised a method of radio transmission called the frequency modulation, or FM, system. You will recall that in the amplitude modulation system the audio pulsations caused the amplitude of the carrier wave to vary. In the FM system, the audio signal causes the frequency of the carrier wave to vary instead. Figure 5-2 is a simplified representation of the method.

As before, the two original signals—audio and carrier—are shown at the left of the sketch with the carrier wave at the top. At the right side we see the result: the frequency of the carrier has been caused to vary in time with the pulsations of the audio signal; in other words, the carrier wave has been frequency modulated.

Advantages of FM

Earlier, it was pointed out that the band width transmitted by an AM station is necessarily limited to prevent overlapping of adjacent stations. This limitation prevents the average AM transmitter from sending out audio frequencies much higher than about 5000 cycles per second. This makes high fidelity broadcasting impossible on AM, but fortunately this condition does not exist in the FM system.

In FM transmission it is quite possible to transmit the entire range of audible frequencies up to 15000 cycles per second and higher. Of course, this means that each FM station must be allocated a band much wider than is possible in AM. As the usual broadcast band is entirely too narrow to accommodate even a relatively few FM stations, such stations have been assigned operating frequencies in a part of the radio spectrum that is not already overcrowded and is wide enough to accommodate many transmitters. At the present time, FM stations operate in the region between 88 megacycles (88,000,000 cycles) and 108 megacycles (108,000,000 cycles).

In addition to the improved frequency coverage, FM has other advantages over AM. One of these is improved dynamic range. As we already know, this is the difference in sound intensity between the softest possible passage and the loudest. We may add that the full dynamic range is not reproduced in AM broadcasting. During the softer passages, the operator monitors the transmission so as to raise the level; on very loud passages the level is purposely reduced to prevent overmodulation and distortion. This is not necessary in FM broadcasting and, as a result, we get an increased volume range with better realism. Finally, FM reception is remarkably free from the static we have come to associate with AM programs. Although, strictly speaking, the term "static" applies to disturbances created by atmospheric electricity, it is commonly applied to man-made electrical disturbances as well as natural ones. Whether man-made or natural, static consists of amplitude modulated impulses, and as the FM tuner is designed to receive FM pulses only, static is almost non-existent.

While we cannot over-emphasize the fact that true high fidelity reproduction is impossible in AM reception, many AM tuners are sold either individually or in combination with an FM tuner. It would appear, then, that in spite of its shortcomings the AM tuner does have a sphere of utility. There may be areas not receiving adequate service from FM stations and others where the sole coverage is AM. Perhaps the area has one FM station, operating only part of the time, and one or more AM stations operating full time. Under such circumstances, the listener may be forced to depend upon AM as the principal source of broadcast entertainment. In such extremes, conditions can be improved by proper adjustment of amplifier bass and treble controls, to create an impression of good frequency response. Remember that the effect will be purely illusory, for you cannot add to the music something that was not present in the original broadcast. Furthermore, you cannot improve upon the limited dynamic range of the AM receiver and, under many circumstances, little or nothing can be done to eliminate static.

In the material to follow we shall discuss the characteristics of tuners with a view to guide you in making a wise selection. We shall then offer instructions for converting a small AM radio of the table variety for use as an AM tuner.

The construction of a good FM tuner is considered to be beyond the ability of all except those who have had considerable electronic experience, unless the parts are purchased in kit form with a pre-punched and pre-drilled chassis. Readers who have had considerable experience in electronics are advised to refer to the many excellent constructional articles that appear from time to time in the leading radio periodicals.

If you have had no experience or only limited experience in this field, and wish to build your own FM tuner, then it is suggested that you buy one of the many kits currently available. Building a tuner from such a kit is considered to be feasible for the novice provided that he knows how to solder and can use hand tools. The difference between building a tuner from a kit and building it from individually bought components is that in the former case all of the engineering problems connected with layout, placement of parts and routing of wiring have been solved for you by the manufacturer; all you have to do is follow the instructions. If you were to attempt to build a comparable tuner from individually selected components, even a slight error of judgment in the placement of a component might spell the difference between success and failure. Excellent FM kits are currently available at prices ranging from $25 upward, including all components and tubes and with instruction booklets that are so complete that failure is almost impossible.

Characteristics of Tuners

Sensitivity. The degree of sensitivity you will need depends upon your location, the number of stations serving your area, and their power and distance from you. In metropolitan areas served by a number of powerful stations, extreme sensitivity is usually not necessary and, indeed, may be a handicap. The reason for this statement will be evident when you understand that, although two stations in a given area are never assigned the same carrier frequency, it often happens that two widely separated stations operate on the same or adjacent channels. It is quite possible for a tuner having extreme sensitivity to receive both the local station and the distant one at the same time. While the program of the distant station may not always be clearly heard, an annoying condition called "monkey chatter" will result. Furthermore, the possibility of such interference increases as the tuner selectivity is reduced and, as we already know, an AM tuner used in a wide range system should accept as broad a band as possible, for improved fidelity.

As you might suspect, these remarks apply particularly to AM broadcasts. FM transmissions have a limited range and generally do not travel farther than the "line of sight" distance; this is about 75 miles at the most and remains fairly constant. AM broadcasts, on the other hand, travel relatively short distances—25 or 30 miles—during daylight hours, but cover much greater distances at night. For this reason, AM transmitters in some locations are required to reduce the amount of power radiated at night in order to prevent overlapping. We may conclude, then, that the listener in a metropolitan area will probably be better satisfied with a tuner having only fair sensitivity. Those who live in rural areas or who are interested in listening to distant stations require tuners with a higher degree of sensitivity.

Another factor having an important bearing on sensitivity is noise. As the sensitivity increases the level of background noise is likely to increase too, but of course this does not apply to FM reception. In fact, it is desirable that the FM tuner have as high a degree of sensitivity as possible. Sensitivity of any tuner is expressed in microvolts; a sensitivity of 10 microvolts is higher than a sensitivity of 25 microvolts, for instance.

Selectivity. As mentioned earlier, selectivity is the ability of a tuner to receive one station to the exclusion of others operating on adjacent channels. While, under some circumstances, a high degree of selectivity may be necessary, it does have drawbacks. In an AM tuner, excessive selectivity may prevent the passage of the entire transmitted band of frequencies, which, as we have found, is not too wide at best. Since the nature of AM broadcasting limits the frequency response to about 5000 cycles, it is evident that high selectivity may make a bad situation worse.

AM tuners are of two types: the superheterodyne and the tuned radio frequency type. The superheterodyne circuit is more widely used and is more likely to have high selectivity. In addition to this, the tuned radio frequency circuit is less likely to limit the dynamic range. FM tuners always employ the superheterodyne principle, but this has no bearing on the performance of such tuners since they are designed to accept a very wide band of frequencies.

The selectivity of a tuner is indicated by the width of the band it will accept at any dial setting. The average AM superheterodyne had a band width of 10 kilocycles or less; a well-designed tuned radio frequency circuit may have a band width as great as 12 kilocycles. FM tuners must accept a band at least 150 kilocycles wide and most good tuners are rated at from 150 to 175 kilocycles.

Noise. Noise in this discussion does not refer to natural or man-made static; these are characteristic of AM reception. We are now speaking of hissing noises that may be present when the tuner is adjusted to a point midway between two stations or when it is tuned to an exceptionally weak signal. Such noise can be minimized by good design and can be detected only by a critical check of the unit.

Drift. Drift generally results from a change in the value of one of the components in the tuner. It presents no problem at all in an AM tuner because of the relatively low frequencies employed, but may be quite serious in an FM tuner, which operates at high frequencies. At such high frequencies a very small change in the value of a part, such as a condenser, may represent a large deviation from the assigned frequency. The most effective method of overcoming drift is the use of an automatic frequency control (AFC) circuit. Should a change in frequency occur, the AFC circuit automatically compensates for it. Automatic frequency control has one disadvantage that should be noted. If you are attempting to tune in a weak signal that is adjacent (in frequency) to a stronger one, the circuit may reject the weaker signal and "lock-in" on the stronger one. This may be overcome by providing a switch for disconnecting the AFC circuit when conditions make this necessary.

Tuning indicator. The majority of FM tuners are equipped with an electron-ray indicator tube for greater accuracy in tuning. This device may be regarded as an absolute necessity in an FM tuner. However, it is used only rarely in AM tuners.

Power supply. As mentioned before, some tuners derive operating voltages from a power supply that is an integral part of the tuner, while others depend upon power taken from the amplifier. The latter type is, as you might expect, less expensive. Although the necessary power usually can be taken from the amplifier, it is well to determine in advance whether the amplifier has the necessary reserve capacity; some of the more elaborate tuners have as many as ten tubes and might impose a serious overload on the power supply.

FM reception systems. Although all FM tuners operate on the superheterodyne principle, there are at least two distinct circuit variations. One variation uses what is known as a discriminator type of detector; this type also has one or two tubes functioning as limiters. A limiter prevents amplitude modulated pulsations from reaching the detector. Because this type requires at least one additional tube and a number of extra parts, it is usually higher in price. The second type employs a ratio detector, and in this kind of circuit no limiter tubes are necessary.

Converting an AM Receiver for Use as a Tuner

In this section we shall offer instructions for converting a suitable AM receiver for use as a tuner. Almost any receiver in good operating condition may be so used, but of course sets of recent manufacture are likely to give best results.

Home radio receivers are of three varieties: 1. a.c. operated (those using a power transformer for supplying all of the voltages for the various tubes); 2. a.c.-d.c. operated (those using no power transformer and designed to be operated from either alternating or direct current supply); 3. three-way or "three-power" portables (those designed to operate from alternating current supply, direct current supply or from batteries). Any of the three will serve as tuners but the a.c. type is most suitable. The reason for this is that when an a.c.-d.c. set or a three-way portable is connected to a high-fidelity amplifier (which is almost always operated from a power transformer) some hum may be encountered that might be difficult to remove. The only drawback is that few sets built in recent years have been of the a.c. variety; furthermore, the chassis of an a.c. receiver is likely to be comparatively large. If this is no objection and you have an a.c. set available, by all means convert it.

Bear in mind that no matter how well designed the AM receiver is and no matter how skillfully you make the conversion, performance can never compare with that of an FM tuner for three reasons. First, the higher frequencies are not transmitted on AM because stations are limited to a band width of 10 kilocycles (with the exception of a few stations at the high frequency end of the broadcast range). Second, on AM transmission the dynamic range is limited. Third, AM receivers are generally not designed to accept a range of audible frequencies wide enough for true high fidelity. We offer the instructions here only because, as mentioned before, you might be in an area where good FM reception is impossible or you might require an AM tuner for those programs that are not available on FM or you may want to use the AM tuner as part of a binaural system.

As mentioned earlier, the selectivity of an AM tuner has a great deal to do with the fidelity obtained: the set you select for conversion should not have extreme selectivity or the available frequency band will be still further narrowed. As a rule, the performance of a set while in good operating condition will offer a clue as to its suitability. If the sound is muffled and the highs seem to be absent, the set probably has excessive selectivity. In that case, a simple adjustment called "stagger tuning" may help to reduce selectivity somewhat, but will also reduce sensitivity. Instructions for making this adjustment will be given later.

We shall assume that the set you are converting is of the a.c.-d.c.

variety. The only materials needed (aside from tools) are: a length of shielded cable and an RCA-type (also called a RETMA) phonograph plug, plus some wire and solder. The cable and plug are to be used for connecting the tuner to your amplifier and should be made up in advance, using the instructions to be found in Chapter 9.

A.c.-d.c. sets have employed two general systems of wiring; these are referred to in the trade as "chassis ground" and "floating ground." Simply explained, the first terms mean that the metal chassis is used as one of the conductors in the circuit while the second term applies when it is not so used. While either type of set may be used, the "floating ground" type is preferable and the majority of modern sets use this system. Handling the exposed chassis of a "chassis ground" set might introduce some shock hazard and interconnecting it with an amplifier or other device introduces the possibility of a short circuit under some conditions.

After removing the set from the cabinet you will have to determine whether the "floating ground" or "chassis ground" system is used. First, determine how many of the wires or parts in the set are actually connected directly to the metal chassis. If you find a considerable number so connected, it is likely that the set uses the "chassis ground" system. As a further check, locate the electrolytic filter condenser; this will usually be a tubular container with either a metal or a cardboard case. If one of the wires leading out of this condenser is connected directly to the metal chassis the set probably uses the "chassis ground" system. As one more check, examine the on-off switch. This will usually be located on the back of the volume control. If one of the wires from the switch is connected to the chassis and if one wire from the electrolytic condenser is also connected to the chassis, you definitely have a set of the "chassis ground" type. In the other type of set, the points mentioned will be connected together but will not be connected to the chassis.

The next step is to prepare the radio receiver. Remove it from its cabinet and turn it bottom up on a workbench or table. First, locate the volume control, which may easily be identified. The control will have three terminals, not including the switch terminals which are located on the back of the control. With the volume control shaft pointing toward you and the radio set turned upside down, locate the terminal farthest to the left. This is shown in the pictorial diagram, Figure 5-3. Trace the wire or wires connected to this terminal. You will find that one wire eventually leads through a hole in the metal chassis to the upper side of the chassis. If you explore a bit further, you will find that three more wires enter this same hole and lead up into a square metal can mounted on top of the chassis. This is an intermediate frequency transformer. In some sets, there may be a resistor connected between the volume control and the intermediate frequency transformer, but this is of no consequence. The important thing is to locate the wire leading out of the transformer. As stated before there will be four wires; the one you are interested in will, in almost every case, be colored black.

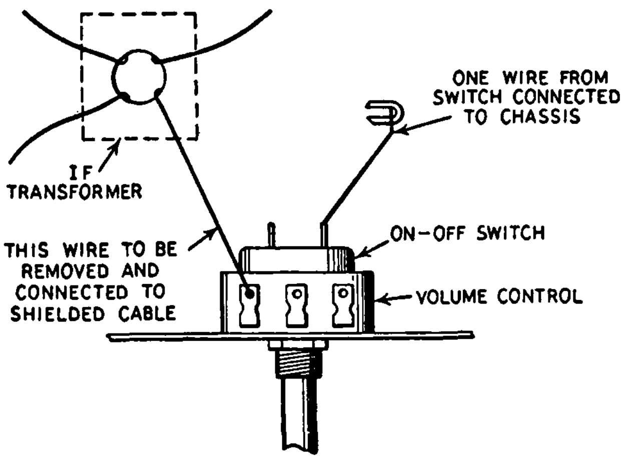


Fig. 5-3

The others, by the way, will be blue, green, and red. Disconnect the black wire from the volume control. Now connect the end of the inner wire in the shielded cable to the black wire and solder the connection. Wrap some friction tape or adhesive tape around the joint.

Examine the back of the volume control, and you will find two terminals located there; these are the on-off switch terminals. One end of the line cord (power cord) will be connected to one of the switch terminals. Trace the wire or wires connected to the remaining terminal to the switch; in most sets a wire will lead from the switch terminal directly to the metal chassis. If that is the case, then the outer shield of the cable must be connected to the chassis also;

you will remember that a wire was connected to the shield, just for this purpose. If the connection is made to the chassis be sure that it is soldered well. If you find that the switch terminal is not connected to chassis, but to some other point, then connect the cable shield directly to the switch terminal, but be sure that you do not make the connection to the terminal bearing the line cord wire. The installation of the shielded cable is shown in Figure 5-4.

The shielded wire must now either be insulated or fastened to the chassis in some way so that the metal shield cannot come in contact with any of the parts in the set. The method used will depend

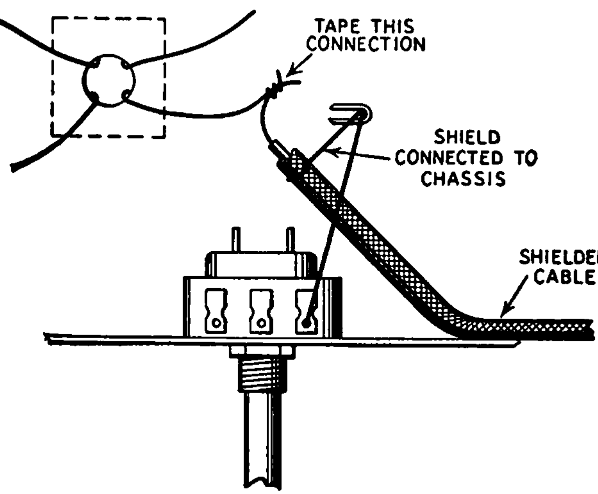


Fig. 5-4

upon your skill and familiarity with radio equipment. Perhaps the easiest way is to wrap adhesive tape over the shield.

Earlier, we mentioned that it is possible to reduce the selectivity, or broaden the tuning, of a superheterodyne in cases where it seems to be necessary. You are cautioned, however, that in no case will you get true high fidelity performance, whether the operation is performed or not.

In connecting the shielded wire to your set, you noted an intermediate frequency transformer; this is usually referred to as an i.f. transformer. Your set has two of these, and from the top of the set they appear as the Figure 5-5. You will notice that there are small screws at the top of the transformers; these are tuning adjustments, and are four in number. (Note: in a very few sets, each transformer will have one adjustment at the top and the other at the bottom. The lower adjustment is accessible by turning the set upside down.) In the drawing, the four adjustments have been labeled A, B, C, and D.

The first step is to make sure the adjustments have been properly made to begin with. Plug the tuner into the amplifier, insert the tuner line cord in an electrical outlet, and turn all equipment on. When the set has warmed up, tune in a program at any point on the dial. Reduce volume until the music can just be heard with

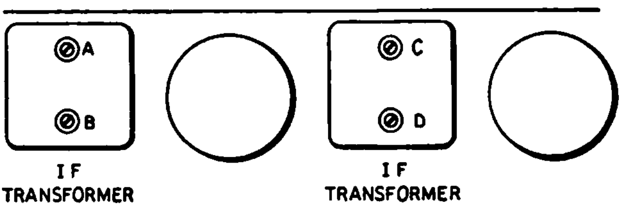


Fig. 5-5

your ear close to the loudspeaker. With a small screwdriver, turn adjustment A very slowly back and forth. Do this carefully, since only a fraction of a turn will make a decided difference. Adjust the screw for the loudest response. If the volume increases noticeably, reduce volume again until the program is barely audible. Adjust screws B, C, and D, in the same way.

Now turn screw A to the right, exactly one eighth of a turn. If the signal disappears entirely at any time during the adjustment to follow, readjust the volume level as needed. Proceed to screw B; turn it exactly one eighth of a turn to the left. Complete the adjustments by turning screw C one eighth of a turn to the right, and D to the left by the same amount. This completes the job. If you find that the selectivity has not been broadened sufficiently, you may repeat the operation, turning each screw an additional one eighth turn.

It is quite likely that the set you have converted is of the transformerless, or a.c.-d.c., type. Any such device, when connected to any amplifier is likely to introduce hum. This can be corrected by reversing the polarity of the tuner and amplifier line plugs. Merely reversing one plug may not always reduce the hum, however. Try reversing both plugs one at a time until the hum is at a minimum.

Antennas

An antenna for an AM tuner presents no particular problem. If you have built or intend to build the tuner described above, an indoor antenna ten to twenty feet long will give good results in the average metropolitan location. In suburban or rural areas, it may be necessary to install a larger, outdoor antenna; this may be a single wire about thirty feet long and it should be located as high as possible above ground.

If you decide to convert an AM set for use as a tuner, the set will, in almost all cases, be equipped with a built-in loop antenna. The average superheterodyne receiver will perform very well on an antenna of this kind if the desired stations are within a reasonable distance. Incidentally, it is worth noting that the majority of loop antennas found on small sets have a connection for adding an external antenna if conditions warrant it.

Antennas for FM reception are subject to entirely different conditions; this is because the higher frequency carrier waves travel in straight lines and reception is limited to the line-of-sight distance. This means that the effective range can often be extended by elevating either the transmitting or the receiving antenna. This is why most FM transmitting antennas are located on tall structures.

You should make an effort to install an outdoor antenna, if this is at all possible. Of course, there will be situations in which this is not feasible for one reason or another. The mere fact that the antenna is located out of doors does not insure improved reception; the point is that in an outdoor installation the advantage of antenna height is usually gained.

It is possible to build your own outdoor antenna, but the cost of a factory-made product is so little that this is hardly worth while. However, if it is necessary to install an indoor antenna, we suggest that you make your own, for very little labor and material are involved and the completed job will probably work as well as, if not better than, most commercial antennas.

You will need about fifteen feet of twin lead: this is the flat, tape-like, two-wire line used for connecting a TV or FM antenna to the set. It may be purchased at any large radio store for a few cents a foot. The length given is based upon the assumption that the antenna will be no more than 10 feet from the tuner; if this distance is to be greater, more twin lead will be needed.

Cut a piece of twin lead 58½ inches long. At each end of the piece, remove the insulation from the two wires for a distance of one inch. The plastic insulation will strip off quite easily if you will first heat it with a match. Twist together the two wires at the ends of the length of twin lead, as shown in Figure 5-6. This will give an effective length of 56½ inches. Now locate the exact center of the twin lead; this can be done by doubling it. At the center cut

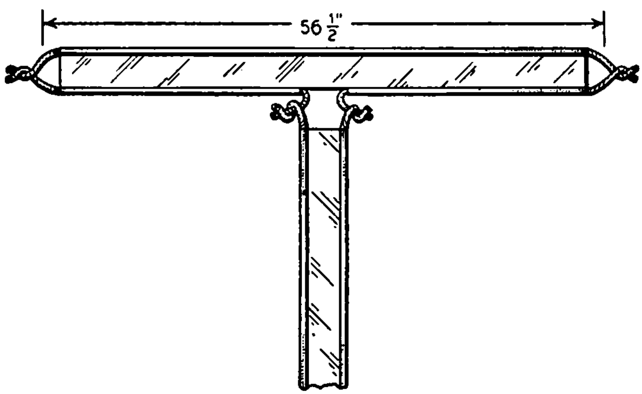


Fig. 5-6. FM antenna.

through one of the wires. Strip the insulation from the two ends thus formed.

The remaining length of line is to be used for connecting the antenna you just made to the tuner. Strip the insulation from the two wires and connect them to the free ends of the antenna, as shown in the drawing. The antenna may then be tacked or taped along a wall or molding or supported in any other way that is convenient, as long as the antenna proper is in a horizontal position and faces in the direction of the station. If reception is not as good as desired, try turning the antenna slightly; very often just a slight shift in direction will make a noticeable difference in signal strength.

The indoor antenna just described will work quite well if your tuner has average sensitivity, unless you are in or beyond the so-called "fringe" receiving area. If you are located in such an area and do not care to install an outdoor antenna for the FM tuner but you do have an outdoor television antenna, this will generally work quite well for FM. There are two ways of using a television antenna.

If it is necessary that the television receiver and the FM tuner be used at the same time, you will have to resort to the use of antenna couplers. These devices permit simultaneous operation of two receivers from a single antenna, and are available at all electronics parts distributors. They are priced from $3 to $5 but provide the

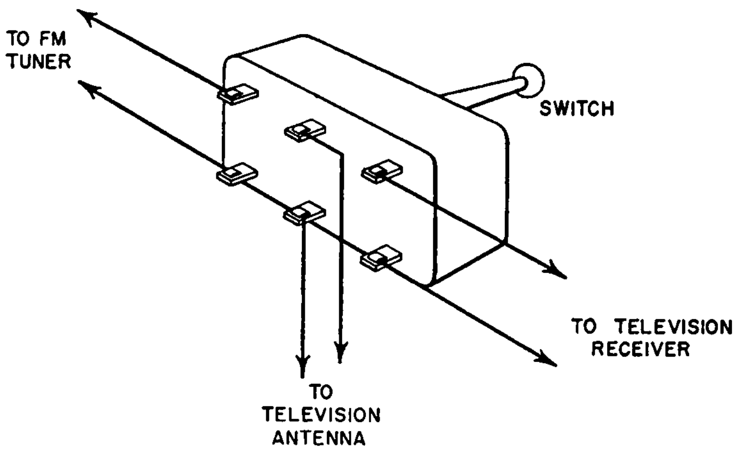


Fig. 5-7. Transfer switch for TV receiver and FM tuner.

only satisfactory method of using one antenna for two sets when both must be used at one time.

In cases where the television receiver and the FM tuner are not to be used simultaneously, a simple low-cost solution is the installation of a transfer switch. This is a double-pole double-throw switch of the toggle variety (although a low-capacitance switch such as a telephone anti-capacity switch might be better). The switch is installed as shown in Figure 5-7, and for convenience should be located either at the television set or at the FM tuner. With the switch in one position the television set is connected to the antenna; with the switch in the opposite position, the FM tuner may be used. All wiring between television set, switch, FM tuner and antenna should be made of the usual television twin lead.

6 Tape Recorders

THE MAGNETIC TAPE RECORDER is a third source of program material and can be a most valuable addition to your home music system if carefully chosen and properly used; in fact, its potentialities are probably greater than those of either the record player or the tuner. This chapter will be limited to consideration of monaural tape recorders. Stereophonic versions will be discussed in Chapter 10.

Compared to a record player, a tape recorder has many advantages, and among these are the following:

a. Recordings of excellent quality may easily be made, even by those with little or no experience in the art.

b. Tape recordings have a longer life than disc recordings. Tape is not nearly as subject to deterioration through wear as disc recordings. Many authorities claim that a tape recording is good for thousands of plays.

c. Recordings may be edited; undesirable passages may be cut out or erased and new portions substituted for them.

d. An uninterrupted playback time of up to an hour and a half per tape side is possible, compared to a maximum of 30 minutes per LP record side.

e. Tape recordings are not as subject to damage from mechanical causes as are disc recordings (as for example, accidentally dropping the pickup on a record). If a tape breaks, a splice can be made in a matter of minutes and, if properly made the effect is hardly perceptible.

f. Mechanical imperfections such as frequently occur in disc records (scratches, etc.) are not present in tapes. Since the recording is impressed upon the tape by magnetic means only those imperfections that are created magnetically will show up in the playback.

g. A tape recorder has a very wide range of applications. In addition to serving as a unit in the home music system it may also be used to supply narration and background music for home movies and color slides, as a means of instruction in learning to play a musical instrument, for language instruction, for recording conferences, meetings and family gatherings, for recording the voices of friends and relations (particularly children), and in many other ways.

h. Unwanted or unsatisfactory recordings may be completely erased and the tape then reused. This process may be repeated as often as desired.

Magnetic tape and wire recording devices are based upon principles discovered by Vladimir Poulsen, noted radio pioneer and inventor. His telegraphone recorded speech magnetically on a steel wire. In an early form, the wire moved past the poles of a strong magnet. Since the voice currents are of a pulsating nature, the degree of magnetization in the wire depended upon their intensity. By running the wire through a suitable pickup device the sounds could be reproduced at will. Furthermore, the speech record could be erased if desired, leaving the wire ready for a new recording.

During World War II the principle of magnetic recording on a stainless steel ribbon was widely used for recording conferences and for training purposes. Shortly thereafter the wire recorder, using stainless steel wire, made its appearance and before long was being sold for entertainment purposes. However, the wire recorder for reproduction of music has almost passed out of existence. Such recorders are now used only as dictating machines.

The next development in magnetic recording devices was the use of a tape or ribbon having a plastic or paper base coated with a finely divided magnetic material such as iron oxide. Fidelity proved to be excellent, and today professional tape recorders are used in a variety of fields. One important application is the making of original recordings in the phonograph record industry. The paper-base tape, however, is now largely obsolete; most modern tapes have a base of cellulose acetate or Mylar.

Basic Principles of Tape Recording

The recording process. When a tape recording is being made, the magnetic tape passes at constant speed in front of the poles of a powerful electromagnet. The electromagnet consists of a core of magnetic material wound with many turns of wire and is called the recording head. Signal currents corresponding to voice or music pass through the coils of the electromagnet; since the signal currents are quite small, the tape must pass very close to, in fact must be in actual contact with, the poles of the electromagnet. Figure 6-1 is a simplified representation of the recording process.

As the tape passes before the poles of the recording head, the magnetic field of the head exerts an effect upon the magnetic material on the tape. The magnetic material is made up of groups of atoms each of which is said to form a "domain" which has magnetic properties. Each domain may be compared to a tiny bar magnet. As the tape moves past the magnet poles of the head it is subjected to a varying magnetic field because the current through the winding of the head is rapidly changing. This causes the do-

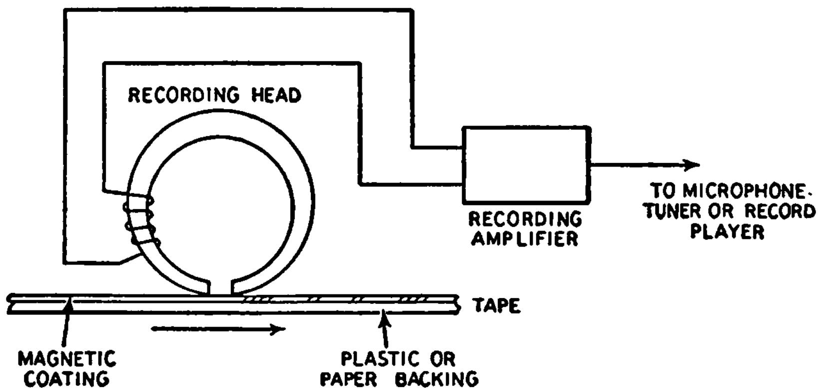


Fig. 6-1. Tape recorder—recording function.

mains to orient themselves, or arrange themselves so that they lie in one direction or the other.

Figure 6-1A shows how the domains are arranged or oriented after a section of the tape has passed the recording head. We are imagining, of course, that the domains are visible. Note that at the extreme right of the sketch all of the domains have been oriented so that they point away from you; in the next group to the left they are oriented so that they point toward you. We are aware that in an alternating voltage (such as that applied to the recording head) there are positive and negative peaks. Each bar in our sketch represents the orientation of a group of domains under the influence of either a positive or a negative voltage peak; a positive peak causes them to point one way, while a negative peak causes them to point in the opposite direction. Between the groups shown in the drawing there will be other groups less completely oriented, the degree of orientation depending upon the strength or intensity of the voltage at that particular instant. Also, the spacing between groups having maximum positive and maximum negative orientation will depend upon the frequency of the applied voltage.

We see then, that during the process of recording on tape magnetic impulses are stored up in the magnetic coating of the tape. These impulses are in the form of more or less fully oriented do-

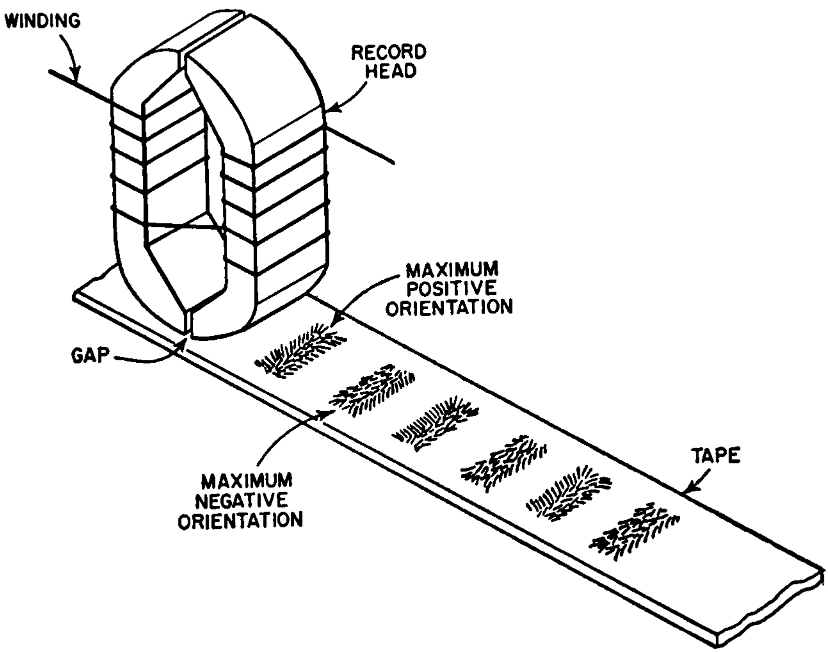


Fig. 6-1A. Tape recorder—basic principle of magnetic recording.

mains or groups of magnetic particles. The degree of orientation depends upon signal intensity and the spacing between fully oriented groups depends upon signal frequency.

The playback process. Figure 6-2 is a simplified illustration of the playback process. The pre-recorded tape passes in front of the poles of an electromagnet (the playback head) at the identical speed used in the recording process. A basic principle of electricity tells us that whenever a changing magnetic field cuts across a conductor a voltage is set up, or induced, in the conductor. The magnetic field has been stored up in the tape in the form of magnetized particles and when the tape passes the poles of the recording head the motion of the tape causes the magnetic field to vary, thus developing a voltage in the coil windings of the playback head. The strength of this voltage depends upon the degree of orientation of the domains and therefore also upon the intensity of the original signal. The frequency of the signal developed in the playback head winding will depend upon the spacing between magnetic groups having a maximum degree of orientation. This, in turn, depends upon the frequency of the recorded signal.

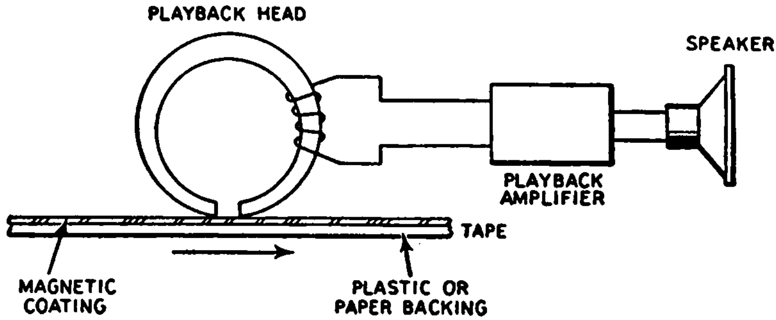


Fig. 6-2. Tape recorder—playback function.

The erase process. Erasure is performed by subjecting the prerecorded tape to the alternating magnetic field set up by an electromagnet called the erase head. This process is shown, in simplified form, in Figure 6-3. The erase head is supplied with an alternating voltage at a constant frequency; the magnetic field thus created destroys the magnetic pattern stored up in the tape by causing the domains to lose their orientation and point in random directions.

The alternating magnetic field in the erase head is developed by a high-frequency alternating current supplied by an oscillator tube; the frequency in general use is between 30 and 100 kilocycles (30,000 to 100,000 cycles). The high-frequency current is also used for "bias" which is necessary to the recording process and which will be explained in the section to follow.

Bias. If signals of ordinary magnitude were to be applied alone to the recording head, severe distortion would result. This is because the magnetization of the recording material is not directly proportional to the magnetizing current in the record head winding. Expressed in more technical language, the characteristic of the magnetic material is non-linear.

To overcome this condition, a very simple trick is used. A high-frequency current, usually between 30,000 and 100,000 cycles, is applied to the record head winding along with the signal voltage. This shifts the magnetization of the tape material to a different portion of its characteristic, so that distortion is eliminated and output is increased. Usually, bias current is supplied by the same oscillator that generates the high-frequency erase current. Some tape record-

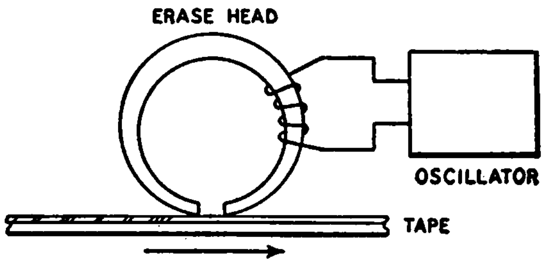


Fig. 6-3. Tape recorder—erase function.

ers are designed so that the amount of bias current can be varied to suit different recording conditions; this is a very useful and desirable feature, but is found only in more expensive equipment.

Tape Recorder Components

The several components comprising a tape recorder are: a. the tape transport mechanism; b. record, playback, and erase heads; c. bias-erase oscillator; d. preamplifier, or preamplifiers; e. the power amplifier (not always used); f. the loudspeaker (not always used); g. the power supply; h. level indicator or indicators. To assist you in choosing a recorder, each of these units will be discussed at some length; following this will be some information on making a wise selection.

The tape transport mechanism. The function of the tape transport mechanism is to carry the tape at constant speed in front of the poles of the record, playback, and erase heads.

Figure 6-4 shows, in simplified form, the type of tape transport mechanism used in the majority of tape recorders with slight variations. In this particular type the tape travels from right to left during recording and playback; in some machines the direction of travel is reversed.

As the tape leaves the supply or feed reel, it first passes over an idler then across the faces of the erase, record, and playback heads, in that order. If the machine is adjusted for the playback mode, the erase and record heads are then out of the circuit and previously recorded material is reproduced. In the record mode, all previously recorded material is first erased by the erase head, after which the

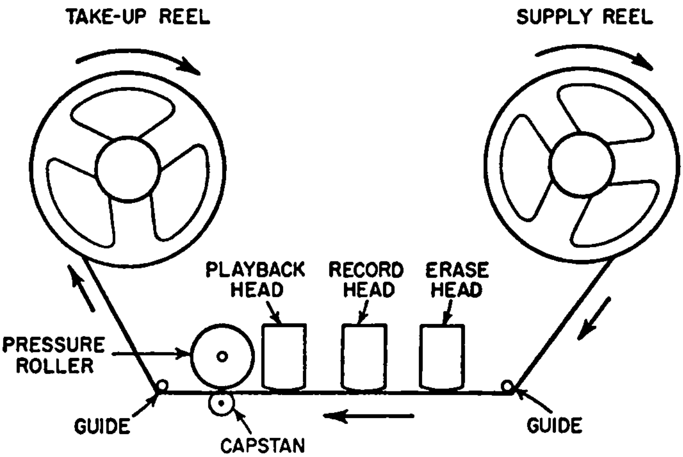


Fig. 6-4. Tape transport mechanism.

tape passes the record head and new material is impressed upon it.

After leaving the playback head the tape passes between the capstan and its associated pressure roller. These maintain constant tension on the tape, insuring a constant speed. The tape then passes over one or more guides and on to the take-up reel.

Since only very small voltages are developed in the heads, the tape must be kept in contact with the magnet poles during both playback and recording. This is accomplished in one of two ways. In more expensive machines a system of guides is used; in popular-priced recorders a set of felt pressure pads holds the tape against the poles during recording and playback.

The primary function of the tape transport mechanism is to carry the tape past the heads at a speed that is accurate and constant. Any variation in tape speed during recording or playback will result in a rise or fall in the pitch of the recorded sounds. If such changes occur only a few times per second, the result is called "wow"; variations occurring at a more rapid rate are referred to as "flutter." These effects may also occur in disc recordings due to speed variations of the turntable during either recording or playback.

It is obvious that in any tape transport mechanism three things must be accomplished: a. The feed, or supply, reel must revolve; b. The take-up reel must revolve; c. The capstan must revolve. The number of different methods used to accomplish these three functions is almost as great as the number of different makes of tape recorders, so that it is impossible to describe tape mechanisms except in general terms. In low-priced recorders a single motor is usually employed to drive the capstan and the reels. In most cases the motor drives the capstan directly and power is transmitted to the reels through a system of belts or, in some cases, through a system of rubber drive wheels. Although very good results have been obtained through the use of one motor, more expensive equipment frequently uses three motors: one drives the capstan, a second drives the feed reel and the third drives the take-up reel.

All recorders are equipped with means for rapidly winding or rewinding a reel of tape. In most cases a 1200-foot reel can be wound or rewound in about a minute and a half. There are also brakes for rapidly stopping the motion of the reels; these are applied to the undersides of the reel hubs.

Various control systems are used to select the desired function: "Record," "Playback," "Wind," or "Rewind." They differ somewhat in ease of operation, but no one system appears to offer distinct advantages over the others. One popular recorder uses a control similar to a miniature version of an automobile gear shift lever. In addition, there is a push button that must be depressed to place the recorder in the record mode. Different forms of rotary selectors have been used. In one model, "Wind," "Rewind," or "Playback" may be selected merely by turning the control; in order to record, the control must also be depressed. At the present time, manufacturers seem to favor push-button controls with a separate button for each function.

A wide variety of extra features are in use; some of these are almost indispensable. For example, an automatic counter enables the operator to determine quickly the footage that has been run and permits him to locate a desired point on a tape. Another useful feature is an automatic device that stops the mechanism in the event that the tape breaks or runs out.

Record, playback and erase heads. A typical tape head, Figure 6-5, consists of a core made of a magnetic material on which is wound a coil of wire. The magnetic material must be of a kind

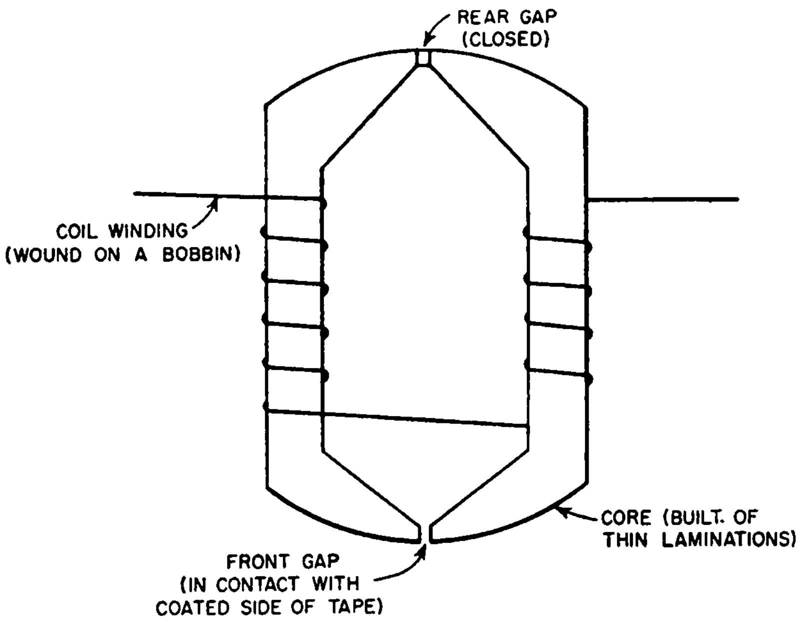


Fig. 6-5. Construction of a tape head.

that is easily magnetized and demagnetized. For this reason, well-designed heads use a special alloy known as Mumetal. While solid cores can be used, better results are obtained by constructing them of thin sheets, or laminations.

Better tape recorders use a separate head for each of the three functions of record, playback, and erase; this is the most desirable arrangement. However in attempting to design a recorder to sell at a low price, engineers are forced to cut corners, and this has led to the development of the two-head recorder or, more properly expressed, to the dual-purpose head. In this type of design, one head performs the dual functions of record and playback and there is a second head for the erase function. Some excellent home recorders using this principle are available, but the arrangement does have certain disadvantages. If a head is to be designed solely for playback, the number of turns on the winding can be increased so that the available output voltage is ample under all conditions. But a design in which the head must also serve for recording requires that the coil winding consists of fewer turns, thus decreasing the output voltage. Of course, the output voltage can be built up by using a higher gain amplifier, but then other sacrifices must be made. Furthermore, with a three-head recorder the operator can listen to the recording as it is made. If it is not satisfactory the required corrections can be made before the recording has progressed very far. Of course, monitoring can be done even in a recorder having a dual-purpose head but not in the same fashion. In such cases, the operator listens to the incoming signal from the program source, not to the actual signal from the tape.

As you will see from Figure 6-5 the typical laminated head has a core made of two C-shaped sections. When these sections are brought together a narrow gap remains at the front of the head. There is also a gap at the rear, but this is usually closed up. The width of the gap determines the upper limit of frequency response of the head. In most modern recorders the gap is no more than 0.0005 inch wide and in professional-type equipment it is often as little as 0.00015 inch.

The length of the gap determines how much of the full width of the tape is occupied by the recording. The tape is 0.25 inch wide, and this full width may or may not be occupied by the recording, depending upon the design of the recorder. In professional equipment and in some older home recorders the head gap is long enough to span the full width of the tape, as in Figure 6-6A. This is called a full-track head. With some loss in output, a recording

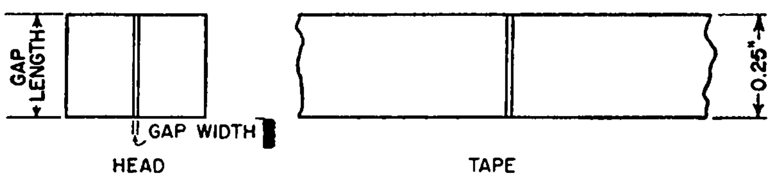


Fig. 6-6A. Full-track recording.

can be made that uses slightly less than half the full width of the tape. When this principle is utilized, the head gap appears as in Figure 6-6B. It is then possible to make another recording on the remaining width of the tape, thus doubling the playing time for a given length of tape. This is called two-track recording (or sometimes half-track recording).

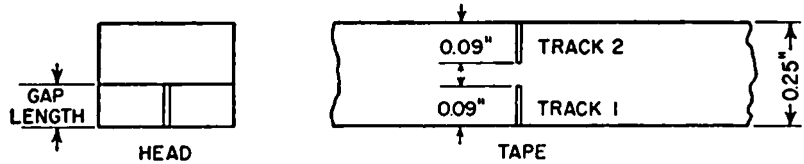


Fig. 6-6B. Half-track recording.

The tape recorder playback head delivers a very small voltage output, usually only a few thousandths of a volt. As the head does not have a uniform frequency response, this response must be corrected or made as nearly uniform as possible. This can be done by the use of equalizer circuits, which usually introduce severe losses. The result is that a high-gain preamplifier must be used to build up the weak signals before they are applied to the power amplifier. In most designs, the preamplifier uses two dual-triode tubes, affording a total of four stages. Such a preamplifier is shown in block form in Figure 6-7. Sometimes a pentode tube is substituted for the first two triodes, as in Figure 6-8.

In the recording process a preamplifier is needed to build up the signals from the record player, tuner or other program source before they are applied to the record head. Usually the record preamplifier is similar in design to the playback preamplifier; in fact, in lower priced recorders a single preamplifier serves both func-

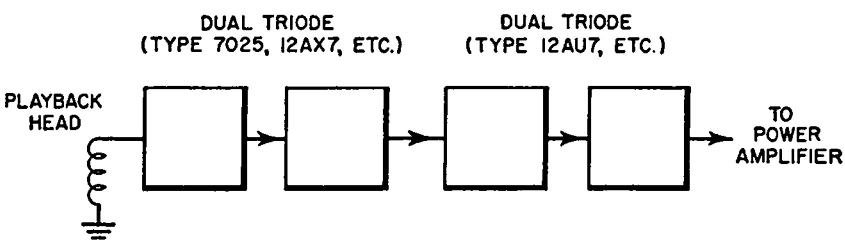


Fig. 6-7. Block diagram of playback preamplifier.

tions, as we shall see later. Notice that a power amplifier is not used between the last preamplifier stage and the record head. This is common practice in the majority of recorders which use a high-impedance type of record head. However, exceptions are found in the few recorders using low-impedance heads.

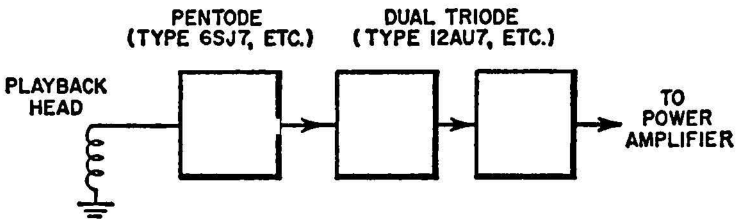


Fig. 6-8. Block diagram of alternative playback preamplifier.

Bias-erase oscillator. As the magnitude of the bias and erase signals is considerably greater than the recorded signals, a power output type of tube is used for supplying them; generally a 6V6 tube or similar type is employed. In a few recorders, two tubes in a push-pull arrangement are used.

In professional-type equipment, in tape "decks" and in a very few home recorders the bias-erase oscillator tube performs only those two functions. However, in most popular-priced recorders this tube must also perform the function of power output stage. This, plus the fact that the preamplifier performs two functions, leads to some rather complicated switching arrangements.

Figure 6-9 shows the complete line-up of tubes, in block diagram form, in a popular home recorder when used in the record mode. Notice that signals from the program source pass through the preamplifier and on to the record head, where they are joined by signals from the bias oscillator. At the same time, the bias oscillator also feeds signals to the erase head.

Now look at Figure 6-10, which shows the same recorder in the playback mode. By means of a switching arrangement the input of the preamplifier has been transferred from the program source to the playback head. Other switch contacts have disconnected the bias oscillator from the record-erase head. Still other contacts have rendered the erase head inoperative. Finally, the bias-erase oscillator has been transformed into a power output stage which feeds signals from the preamplifier to the loudspeaker.

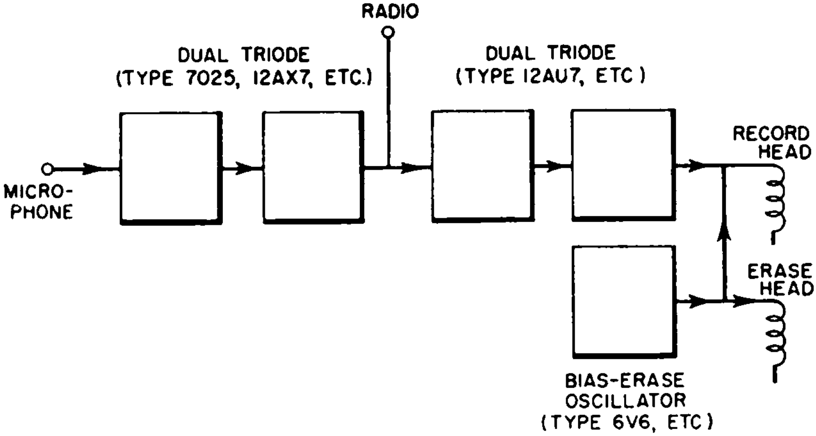


Fig. 6-9. Block diagram of tape recorder in record mode.

Recording level indicators. In any tape recorder there must be a means of determining the level of the signal applied to the record head. If such an indicator is not used there is danger of overloading during recording, with consequent distortion, or the level may be so low that noise becomes a problem. Recording level indicators are of three types: a. One or more neon lamps; b. An electron-ray indicator tube and c. A milliammeter, calibrated in volume units. The neon lamp is used on low-priced recorders and the meter-type indicator on the more expensive types. In using a neon lamp indi-

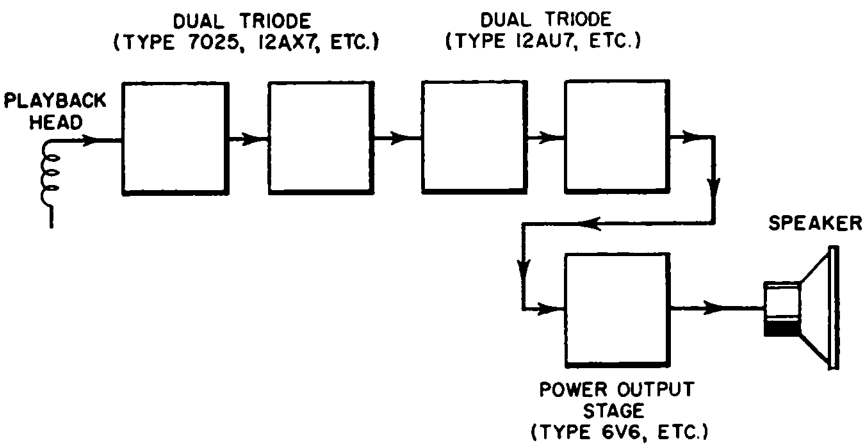


Fig. 6-10. Block diagram of tape recorder in playback mode.

cator, the proper recording level is judged by flashing of the lamp. If a single lamp is used, occasional flashes indicate a high level without overloading; when the lamp lights constantly, the recording level is too high. Sometimes two or more neon lamps are used, so that the correct level may be more accurately determined. Of the three types of indicator, the meter type is the most accurate.

Selecting a Tape Recorder

The tape recorder industry has now progressed to the point where a very wide variety of models, with an even wider array of features, is available. While this does make a choice more difficult, remember that there are only three basic types of recorder. We shall attempt to classify these (although a clear-cut classification is almost impossible) as follows: a. The so-called professional-type recorder; b. The tape "deck" and c. The home-type recorder.

The professional-type recorder almost always has a very well designed tape transport mechanism, and always uses separate heads for record, playback and erase. In most cases there is a preamplifier for playback and another for record. Many models do not include a power amplifier and loudspeaker, although a few do.

Originally, the term tape "deck" referred to a tape transport mechanism complete with heads, but without the bias-erase oscillator and preamplifiers. Lately it has come to mean also a tape transport complete with heads, bias-erase oscillator and preamplifier for playback. According to this second definition a tape deck is a unit designed for playback only of pre-recorded tapes, with no provision for recording. In the present discussion we shall accept the original definition of a tape deck.

A tape deck usually consists of only the tape transport (usually of very high quality) with the three heads. Preamplifiers are not included. The bias-erase oscillator may or may not be a part of the tape deck (currently, kit manufacturers furnish the preamplifiers and the bias-erase oscillator separately in knockdown form, to be assembled by the purchaser). The home-type recorder consists of the tape transport, usually two heads (an erase head and a combination record-playback head, a single preamplifier which is used for both record and playback, a power amplifier and loudspeaker, and a bias-erase oscillator. In most cases the power amplifier also functions as bias-erase oscillator during recording, although there are exceptions to this rule. In a few instances the power output stage is of the push-pull type and there may be provision for more than one speaker. Prices, at this time, range from a minimum of $350 upward for a recorder that just falls within the professional classification. Tape decks are lower in price because less equipment is offered. The home-type recorder starts at about $60 for the cheapest imported model and ranges up to about $250.

Your first decision will be which type to select from the three classifications. If you have sufficient knowledge of electronics so that you can assemble the required electronic components, perhaps the tape deck in kit form is the best buy for you. If you want a top-grade recorder and have no knowledge of electronics, you will have to choose from among the many professional-type machines, but remember that the $350 figure mentioned is just a starter. You can easily spend $1000 on a recorder in this class. For those with a limited budget, there remains the home recorder, especially if the unit must be easily portable; this will apply if the recorder is to be used in a number of locations. For example, if it is to be used to supply sound for home movies or slides and carried from place to place, you may eliminate the tape deck from your list; this will apply also if you intend to use the recorder in locations other than your own home. Many so-called professional-type recorders are quite bulky and not easily transported, so that if you require portability, perhaps the home recorder will be your only choice under such circumstances. Remember, that while there are many very good home recorders, none will afford the results to be expected from a good tape deck or a high quality complete recorder. Those who have had considerable electronic experience, however, may find it possible (by installing suitable jacks and switches) to bypass the preamplifier and use only the tape mechanism and heads of a home recorder. In this way, such a machine can be used as a sort of tape deck in conjunction with your high fidelity system or may be used as a complete portable recorder when this is necessary. Instructions for making these changes will be given later.

Tape Recorder Specifications

When you are ready to select a tape recorder, you will need to know some facts concerning performance ratings. Also, from the previous discussion you will have decided just which features appearing in current models are important to you. The most important things to consider are the following:

1. Tape speeds provided.
2. Frequency response.
3. Wow and flutter rating.
4. Hum and noise level.
5. Types and number of controls used.
6. Type of level indicator used.
7. Maximum diameter of tape reels the machine will accept.
8. Maximum output of power amplifier (if used).

Tape speeds. The speed of travel of the tape past the heads during recording or playback is rated in inches per second. The frequency response is directly related to the speed of the tape; faster tape travel affords a wider frequency response. However, tape speed seems to affect the lower limit of response less than the upper limit. The tape speed is not the only controlling factor in frequency response; the design of the record and playback heads has a great deal to do with it. Thus, it is possible to find a high quality recorder that will respond to 15,000 cycles per second at a tape speed of 7½ inches per second, while a low priced home recorder, operating at the same speed, will respond to only 10,000 cycles per second.

Professional-type recorders used in the broadcast and phonograph record industries often operate at a speed of 15 inches per second, but this speed is rarely used in equipment available to the layman at moderate prices, even though the equipment is advertised as of professional quality. Practically all tape decks and home recorders as well as those classified as of professional quality are equipped with two speeds: 7½ inches per second and 3¾ inches per second. Obviously, the playing time of a tape running at 3¾ inches per second is twice that of one traveling at 7½ inches per second, but the frequency response covers a narrower range. The lower speed is used principally for speech recordings; the quality is about the equal of an AM broadcast.

Portable tape recorders designed to be used either indoors or outdoors (that is to say, those using a spring-driven or battery-powered transport mechanism) generally are designed for operation at 3¾ inches per second, although a few are designed for 7½ i.p.s. also. Many also provide speeds of 1⅞ i.p.s. and in at least one model a speed of 15/16 i.p.s. is used. At these lower tape speeds the frequency response is so restricted that the recorder is of little or no value in recording or reproducing music.

Frequency response. A mere statement of the upper and lower limits of frequency response means very little; this applies to all audio equipment as well as to tape recorders. For example the statement "response 40 to 12,000 cycles" simply means that the device will respond to all frequencies between those two limits but does not tell us whether the response is uniform or not. One professional-type recorder known to be of superb quality is rated as follows "Response at 7½ inches per second plus or minus 2 db from 40 cycles to 15,000 cycles." This means that not only is the recorder capable of recording and reproducing all frequencies within the stated limits but also that at no point between the limits is there a deviation of more than 2 db from flat response. Another recorder (this one a home portable currently selling for about $150) is stated to have a frequency response at 7½ inches per second of from 40 to 12,000 cycles, with no further qualification.

These statements should not be taken to mean that all so-called professional-type recorders are excellent or that all home-type recorders are poor. To make an intelligent selection you should become familiar with the meaning of specifications, then try to find a product that most closely approaches the ideal at the lowest possible price. However, you will probably find that in home recorders you will generally have to be satisfied with a mere statement of the response limits as just pointed out. Then your choice will have to be determined, in part at least, by listening tests.

Wow and flutter ratings. As mentioned earlier in this chapter, wow and flutter are the results of speed irregularities in the tape transport mechanism. Just how much speed variation can be tolerated will be determined in part by the accuracy of the listener's hearing; nevertheless you will want to buy a product that offers a minimum of such distortion. High quality complete recorders (in the $500 and higher bracket) are rated as having wow and flutter of 0.1 percent or less. In the popular-priced home recorder field (priced from $150 to $250) you are not likely to get a machine with a rating of better than 0.3 to 0.4 percent.

Hum and noise level. The amount of hum or other extraneous noise present is usually stated in terms of signal-to-noise ratio. The amount of hum or noise may be determined by playing a specially prepared tape with signals recorded at various volume levels. When a level is found such that the hum or noise is as loud as the signal, the ratio is determined from the known level of signal. A superior recorder will have a high signal-to-noise ratio; an inferior machine will have a lower ratio. For example, a 50 db rating is better than one of 40 db.

About the best that can be achieved in a top-quality recorder is a 60 db ratio; many excellent recorders in the $500-and-up class are rated at about 50-55 db. Among low-priced home recorders the rating may be as low as 35 db; this is because the design of a tape recorder to sell at a low price is quite difficult and involves corner cutting.

Crowding of components in an attempt to produce a compact, easily portable unit can increase hum; this is especially true when the drive motor must be placed quite close to the preamplifier. And, as we have seen, the use of a single head for both playback and record leads to lower output; a higher gain preamplifier is then needed and the hum level increases proportionately.

Even in the cheapest home recorder the situation is not too bad if the machine is used as it was intended to be used: as a portable recorder. The power stage and loudspeaker do not respond to very low frequencies and normally the hum is not too obvious. Trouble arises when tapes made on the portable recorder are played back on equipment having good bass response or when an attempt is made to use the recorder in conjunction with a wide-range music system. However, if you have some knowledge of electronics and an understanding of the causes of the difficulty, certain changes can be made that will improve matters. These changes will be discussed later.

Types and number of controls. Control systems were discussed briefly earlier in this chapter and as we noted there the various systems have little or no advantage, one over the other except in convenience of operation.

Be sure that the recorder you select has a foolproof interlock so that the machine cannot accidentally be set for the record mode when a pre-recorded tape is being played; if this happens, either accidentally or deliberately, the previously recorded material will be erased. Such safeguards take many different forms; one that is extensively used consists of a push button that must be depressed in order to place the recorder in the record mode. If the button is not held down the recorder will not record and therefore cannot erase; the button will stay down only when it is first depressed and the function selector placed in the forward drive or record-playback position.

Other features that must be taken up here are the input and output connections, although they cannot be classified as controls. Most home recorders have two input jacks; one is used for recording from a microphone, the second for recording from a high level source such as a record player equipped with crystal cartridge, a tuner or TV. In the majority of cases the input jacks are designed to accommodate a standard telephone plug, although in one model the microphone jack takes a standard telephone plug and the second jack takes the RETMA phono plug.

Many recorders have provision for feeding the output of the playback preamplifier to an external power amplifier. Sometimes the jack used for this purpose (it is generally marked "AMP." or "EXT. AMP.") is arranged so that when a plug is inserted the power amplifier and speaker built into the recorder are disconnected. This is a desirable and useful arrangement, but in most cases the amplifier and speaker remain in the circuit. There is usually a provision for connecting another speaker to the recorder; when the extra speaker is added the one built into the recorder may or may not continue to function, depending upon the design of the recorder.

Type of level indicator used. The various types of level indicators were discussed under "Tape Recorder Components."

Maximum diameter of tape reels the machine will accept. Some tape decks and complete recorders of high quality will accept reels up to 10½ inches in diameter, but, the vast majority of units are designed to take reels no larger than 7 inches in diameter. This affords a playing time of 30 minutes per track or one hour per reel using standard tape. As we shall see later, there are special types of tape that give longer playing time.

Maximum output of power amplifier. Most recorders incorporating a power stage use a single output tube and have a rated power output of 3 or 4 watts. A few models make use of a push-pull circuit affording an output of 6 to 8 watts.

The Tape

As noted earlier, most modern magnetic tapes consist of a finely divided magnetic material on a plastic base; the base is generally acetate or Mylar. All tapes are ¼ inch wide and are available in various lengths and on reels of 3-inch, 4-inch, 5-inch, and 7-inch diameters. The 3-inch and 4-inch reels are used principally on battery-powered portable recorders.

The amount of tape that can be wound on a reel of given diameter is determined by the thickness of the plastic base. Thus, for the standard 1½ mil tape (the base is 0.0015 inch thick) a 7-inch reel will hold 1200 feet, a 5-inch reel 600 feet and a 3-inch reel 150 feet. Playback time for a 1200-foot length of tape is about 30 minutes per track, or an hour for the entire reel.

In order to extend playing time, tapes are made with thinner base material so that a greater length can be wound on a given sized reel. Naturally, such tapes are more expensive than the standard 1½ mil thickness. Seven-inch reels of 1 mil tape hold 1800 feet, giving a playing time of 45 minutes per track. There is also a ½ mil thick tape available on 7-inch reels holding 2400 feet, which makes possible a playing time of about one hour per track. Both of these thinner tapes generally use a Mylar base instead of the acetate base used in the standard thickness.

Another development is the special high output tape which affords a signal level about 8 to 12 db higher than that of standard tapes at a price advance of slightly more than 50 percent. It is possible that a tape of this type might help solve some of the problems encountered by those who attempt to use a two-head portable recorder in conjunction with a wide-range music system and find that the output of the dual-purpose head is somewhat less than expected.

Standard 1½ mil acetate tape in 1200 foot lengths sells at a reasonable price considering the entertainment value it affords. Certainly the cost per entertainment hour is less than that of LP records. However, as in every other field, competition has led to the rather wide distribution of bargain-price tapes, some selling at a price half that of the standard article. The quality of cut-rate tapes currently available ranges all the way from good to extremely poor. Unfortunately, there is no way to determine the quality of a tape by examining it; the only way to find out is to try it. Some such tapes have a higher than normal noise level, others give poor frequency response. It appears that most manufacturers apply some kind of lubricant to their tapes to reduce the abrasive action of the magnetic coating on the heads and guides. In some brands tested it was evident that there was a lack of lubricant because excessive static electricity was generated during fast wind or rewind. This will usually show up as a bluish glow as the tape runs over the guides and is easily seen in a darkened room. Certainly a tape in this class will eventually cause excessive wear on the heads and guides of your recorder.

How To Use Your Tape Recorder

Adding a tape recorder to your high fidelity system. The addition of a good quality standard three-head recorder to an existing music system presents no particular problem. While installation techniques will depend upon the particular make or model you buy, in all cases the manufacturer will supply complete instructions. To permit recording from all of the various program sources, a switching arrangement is needed. The switch unit will consist of a standard wafer-type rotary switch, available from all electronic supply houses, and an input jack for each of the program sources. Figure 6-11 illustrates such a switching arrangement. Remember that shielded cable must be used for all connections between the program sources and the switch and between the switch and the tape recorder. The length of cable must be no more than two or three feet, otherwise here is a possibility of hum pickup. Radio and electronic suppliers sell jumper cables in various lengths and they are inexpensive. However, if you prefer to make up your own, you may use the instructions given in Chapter 9.

If you buy a tape deck, your problem will be somewhat complicated by the fact that such units are sold without preamplifiers. Most manufacturers of decks also supply preamplifiers especially designed for use with their recorder. A few kit manufacturers sell tape decks and also supply the preamplifier in kit form. In most cases such kits can be assembled by those having little or no experience in electronics; the only skill needed is a knowledge of soldering.

The switching arrangement described above and illustrated in Figure 6-11 will not be needed if you have a modern preamplifier of the control center type. Such preamplifiers generally have input jacks for tuner, magnetic and crystal cartridges, microphone, and, frequently, one or two spare jacks for other program sources. Some have, in addition, a jack for the tape recorder input. A preamplifier of this type has a selector switch for changing from one program source to another. Figure 6-12 shows how a tape recorder may be added to a music system using a preamplifier of this type. To record, the selector switch is set to the desired program source and signals from this source are fed to the tape recorder by way of the jack marked "Tape Recorder Input." During playback, signals are fed from the tape recorder output to the preamplifier by way of one of the spare jacks, the jack marked "Tape Head" or "Tape Recorder" or the microphone jack.

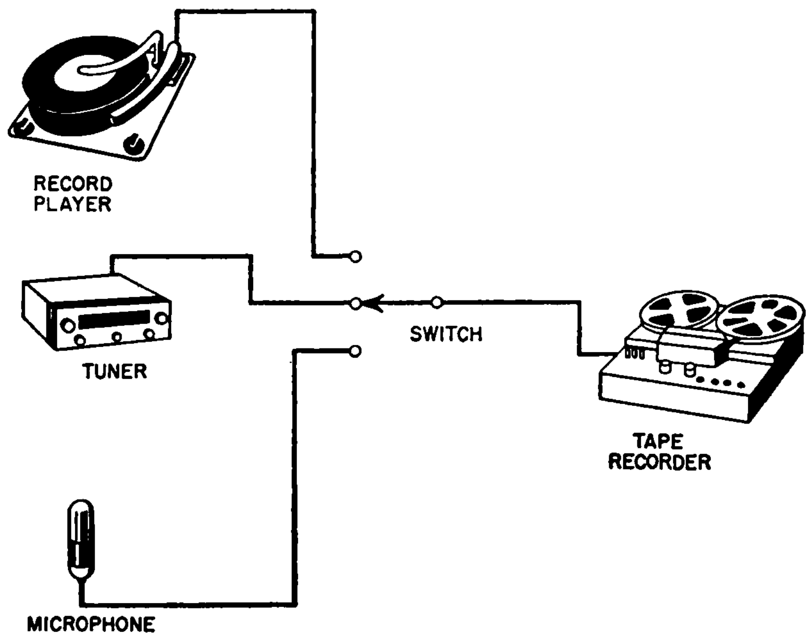


Fig. 6-11. Switch arrangement to permit use of tape recorder with several program sources.

Home recorders. As mentioned before, most popular-priced home-type recorders perform very well when used as intended: as portable, self-contained units. There is no need to give instructions for the use of such recorders for all manufacturers supply complete instructions with their products.

However, should you decide, as many owners have, to attempt to incorporate your recorder into your high fidelity system, you will almost certainly encounter trouble. Certainly, the power amplifier and speaker in the average home recorder falls far short of high fidelity standards and this knowledge may lead you to wonder whether your wide-range system could not be substituted for the amplifier and speaker built into the tape recorder. In most cases this is possible, but only by making some changes in the tape recorder, and you are not advised to attempt them unless you have had some experience in electronics, especially in reading schematic circuit diagrams.

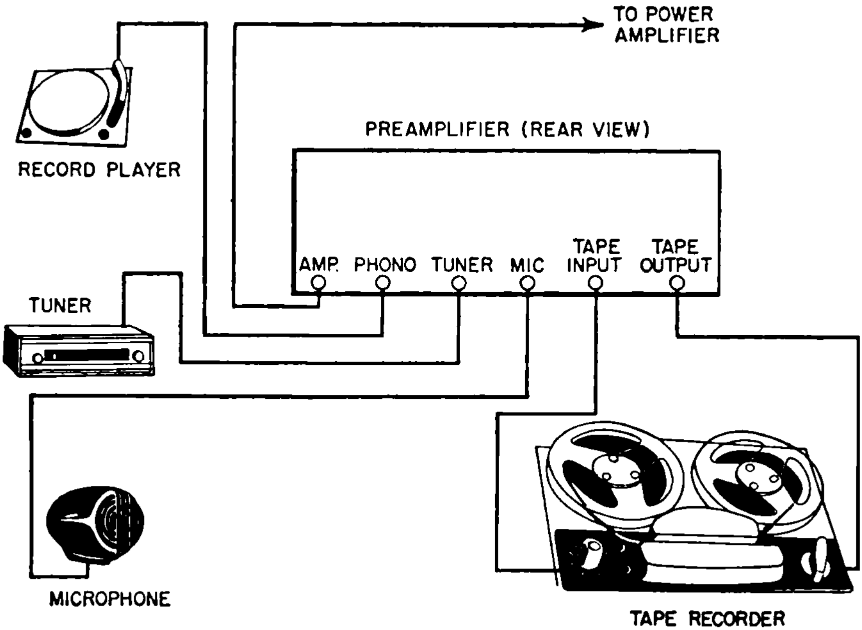


Fig. 6-12. Connections between tape recorder and preamplifier for recording and playback.

If you feel that you have had the required experience, the first step is to obtain a diagram of your recorder. This can be purchased from the manufacturer at moderate cost (usually not more than a dollar). Also, at least two firms publish technical information on all radio and television receivers, record changers, and tape recorders. This material is compiled in volumes, but some electronic distributors and dealers sell individual diagrams separately.

The next step is to study the diagram. Locate the record-playback head winding and determine the points in the circuit to which it is connected. You will find that by means of a multi-contact switch the head is connected to the grid of the first preamplifier stage during playback. When the recorder is in the record mode, the head is transferred, by the same switching arrangement, from the input of the preamplifier to the output circuit of the last preamplifier stage.

Figure 6-13 shows a portion of the typical circuit used in many home recorders. To convert your recorder so that recordings may be played back through your high fidelity system, you will have to add

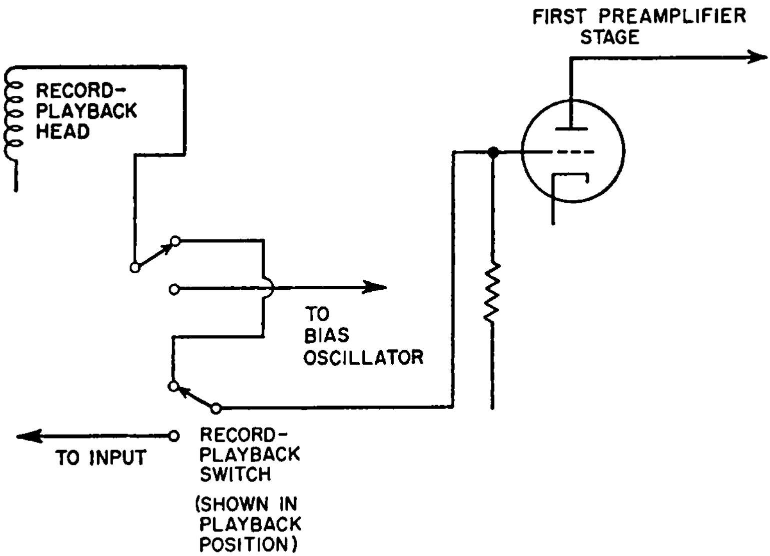


Fig. 6-13. Portion of circuit used in a home recorder.

a single-pole double-throw toggle switch and a RETMA phono jack. These may be located in any convenient place on the recorder control panel, but it is best to place them as close to the record-playback head terminals as possible so that all connections may be kept quite short. At the switch terminal, disconnect the wire leading to the record-playback head. Connect this wire to the center terminal of the single-pole double-throw switch. This, as well as all other wires involved in the conversion, must be shielded, with the shield grounded.

Using a length of shielded wire, connect one of the free terminals of the toggle switch to the RETMA jack. Now connect the remaining terminal of the toggle switch to the terminal of the record-playback switch from which you removed the head wire, again using shielded wire.

This completes the conversion of your recorder for playback. See Figure 6-14. Now, by throwing the toggle switch one way, the recorder may be used as intended, as a portable recorder. When

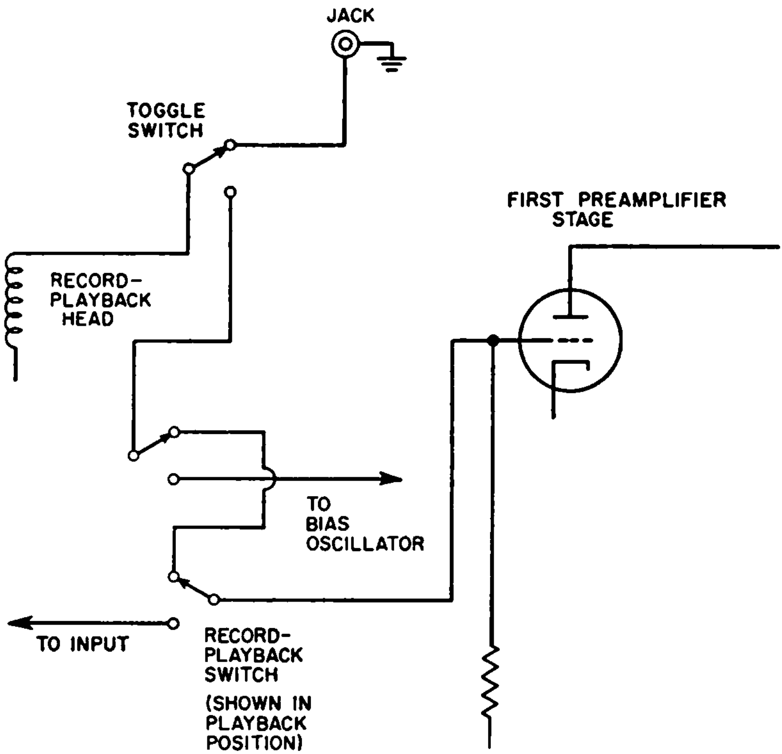


Fig. 6-14. Circuit of Fig. 6-13 after conversion.

the toggle switch is in the opposite position, a shielded cable is used to connect the RETMA jack to one of the input jacks on your high fidelity preamplifier, thus using your high fidelity system for playback purposes.

Whether you will be able to use your high fidelity preamplifier for recording, in place of the preamplifier built into your tape recorder, depends upon the design of the recorder. You will have to examine the schematic diagram carefully to determine whether it will be possible to feed signals from the program source into the record-playback head without disturbing the bias-erase oscillator. In a number of home recorders this can be done without making any electrical changes.

Figure 6-15 is a portion of the preamplifier output circuit of a popular home recorder. Note that in this model there is a jack marked "AMP." (in some other makes it is labeled "EXT. AMP."

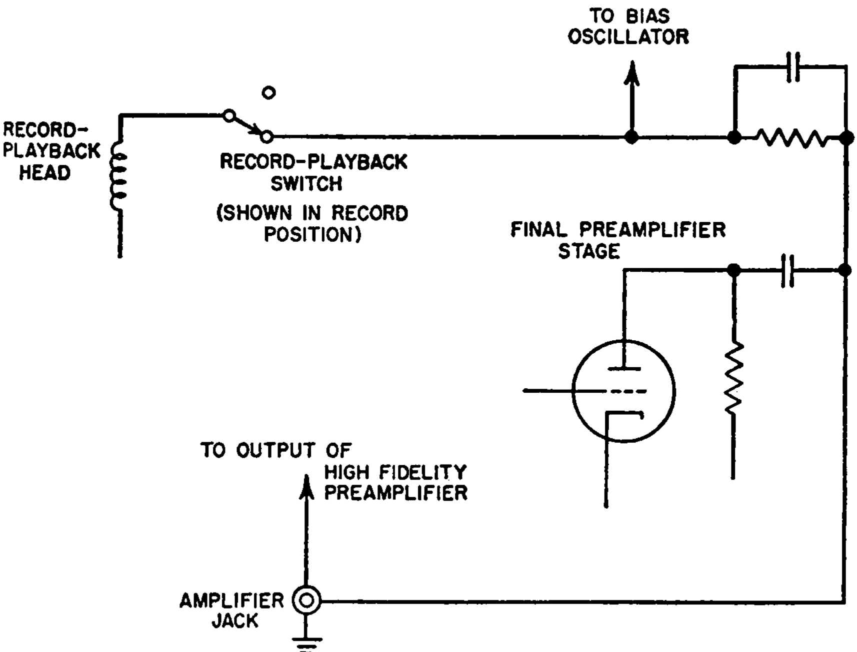


Fig. 6-15. Final stage of tape recorder.

The purpose of this jack is to provide a means of connecting the output of the tape recorder preamplifier to an external power amplifier. However, signals may usually be fed into this jack from your music system's preamplifier, thus permitting you to make recordings independently of the tape recorder's preamplifier. Usually, all that is necessary is to keep the tape recorder volume control set at zero. If hum or noise are bothersome, it might be necessary to remove the tube from the last stage of the tape recorder's preamplifier. In using this method of recording, you might find that the signal level from some sources (particularly a magnetic phonograph cartridge) is not quite high enough for good recording. This may be due to the fact that your high fidelity preamplifier does not have as much gain as the preamplifier used in the tape recorder. The difficulty can be met in several ways. One is to take the signals from the first stage in your basic amplifier. Another is to use an additional stage ahead of your preamplifier. While a stage using a tube would be undesirable because of the danger of hum, a stage using a transistor has worked out very well. Such a stage may be constructed in a small aluminum box with a RETMA plug at one end of the box and a RETMA jack at the opposite end. The stage may then be plugged into the phono input jack of the preamplifier, ahead of the pickup cartridge, when tape recordings are to be made and removed when disc recordings are to be played back.

Making simultaneous recordings of speech and music. In a number of cases it is desirable to record speech and music or instrumental music and voice simultaneously. Examples are: recording for home movies or color slides; practice in playing a musical instrument, etc. In more expensive, three-head recorders, the recordings might be made by first recording, let us say, the musical background, then running the tape through again and, with the erase head de-energized, superimposing the narration upon the music. However, this kind of recording requires considerable skill and extremely accurate timing. Furthermore, in most home recorders it is difficult, if not impossible to de-energize the erase head so that a second recording may be made on top of the first.

A much easier method of making a dual recording is to record the two signals (music and narration, or other combinations) simultaneously. For this you will need a two-channel mixer. Small mixers designed to plug into the input jack of your recorder are available at low cost (an imported model is currently selling for about $2.50).

For those who are mechanically inclined and prefer to build their own mixers, a schematic diagram is given in Figure 6-16. The entire circuit can be built into an aluminum box measuring 3½ inches by 2 inches by 1½ inches or smaller. A standard telephone plug, which will fit into the "Microphone" jack of most tape recorders is fastened to one of the small ends of the box. If the "Microphone" jack on your recorder happens to be of the RETMA type, you will have to substitute a RETMA plug for the telephone plug. On one side of the box a RETMA jack and a standard telephone jack are mounted. The overall size of the shield box will be determined by the size of the controls; with miniature controls the box may be even smaller than the dimensions given above. The knobs for these controls should have pointers or calibrated dials.

As an example of this method of making dual recordings, let us suppose that you want to provide narration for color slides and that you would like to have suitable background music in addition

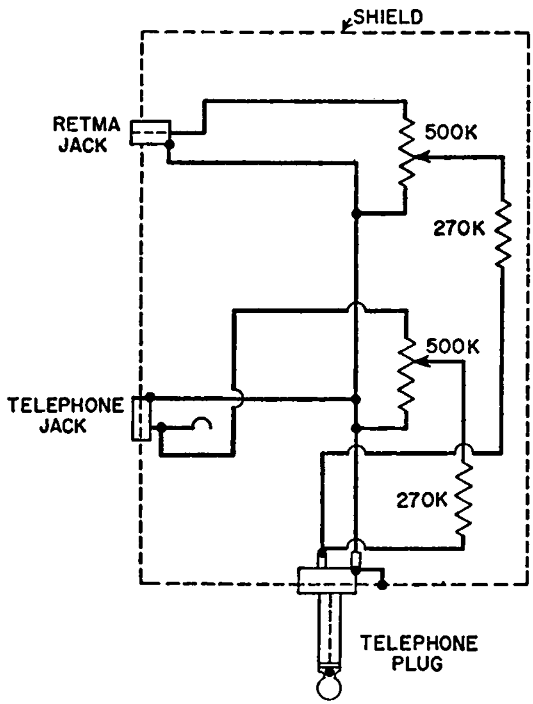


Fig. 6-16. Two-channel mixer.

to the spoken comments. It is advisable first to prepare a "script" for the narration and this should be done with the slides before you. After each slide is shown there will, of course, be an interval during which slides will be changed; this interval should be indicated on your "script". Then, when the recording is made, the proper signal notifying you of the time to change slides can be recorded also.

When all of the spoken comments have been written, you should note on the script the musical selections to be recorded at various points. We shall assume that the music is to be provided by a record player. You are now ready to make a trial recording to determine the proper volume levels. Set up the equipment as shown in Figure 6-17. As you read your prepared comments from the script, adjust the volume controls so that signals from the microphone are below the level at which the level indicator shows overloading. Then, adjust the music level control so that the music provides a pleasant background without being too obtrusive. When the comments for the first slide have been read, the signal for the

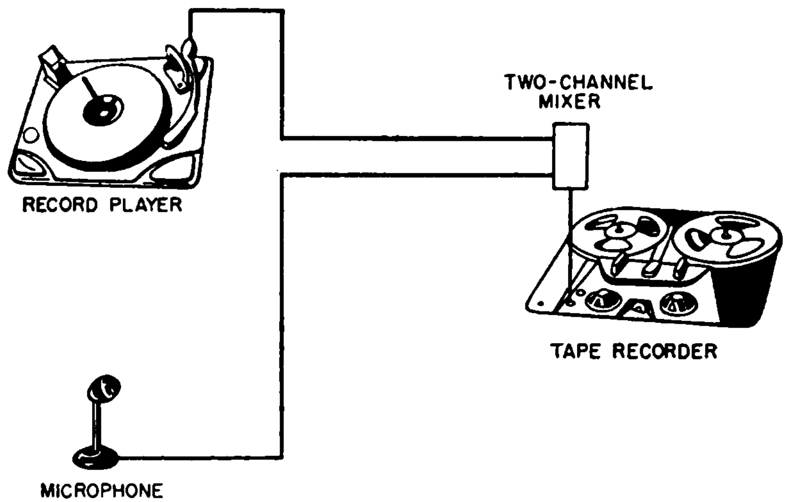


Fig. 6-17. Equipment set-up for making simultaneous recordings.

change of slides is sounded; this is done merely by having the bell, chime, buzzer or whatever device you prefer to use close enough to the microphone so that its sound will be recorded. However, a much better method is simply to turn up the music level control; an increase in the level of music signals the time to change slides. When you judge that enough time has passed to permit you to change slides, return the music volume control to its original setting and proceed with recording the comments for the second slide. This is why the two controls should have some type of pointer or indicator to permit you to return them to a predetermined level. After one or two short practice recordings you will find that you can produce tapes that are nicely synchronized with your slides and that have a professional touch.

Care of Tape Recorders

Cleaning. After several hours of use you will notice a brown deposit on the guides, capstan, capstan pressure roller and other parts of the tape transport mechanism. Loose deposits may be removed with a soft brush, such as an artist's sable or camel hair brush, but you will find that some residue will continue to adhere despite brushing. Some manufacturers recommend alcohol for removing persistent deposits; others prefer carbon tetrachloride. In either case, the fluid should be applied with a lint-free cloth. Generally there will be a removable cover over the heads, providing easy access to all parts. The following points should be cleaned: guides; capstan; capstan pressure roller; surfaces of head pole pieces and the area of the tape transport bed plate in the immediate vicinity of the heads and guides. Do not allow the cleaning fluid to come in contact with plastic parts of the tape transport. A number of special cleaners are on the market; one of these consists of a length of fabric tape impregnated with a cleaner and is run through the mechanism like an ordinary tape. Another preparation consists of two liquids: one is a cleaner, the other is a lubricant. When cleaning the pole surfaces of the record, playback, and erase heads, never allow any metal object to come in contact with these surfaces. There are two reasons for this precaution: first, the polished surfaces might be scratched, and second, the metal object might be magnetized. In fact, you should always be sure that any metal object brought into the vicinity of the heads is not magnetized.

Lubrication. The manufacturer's recommendations concerning lubrication should be closely followed. Most manufacturers stress the fact that periodic lubrication is neither required nor desirable. Either the moving parts are lubricated at the factory, in sufficient quantity for the life of the machine, or oil-less bearings are used. Applying lubricant to a part that is already adequately lubricated might cause the excess oil or grease to work into places where it would result in improper functioning or even permanent damage. An example of this is found in the rubber capstan pressure roller. A small amount of lubricant is needed on the roller bearing, but an excess might easily work out to the surface of the rubber roller causing slippage, with consequent speed variations. If a part is replaced or if periodic lubrication is specified by the manufacturer, be sure to use the correct type of lubricant. In most recorders, only three different types are used: motor oil (generally No. 20 SAE), grease, and Lubriplate. If the specifications call for oil, use only a drop, applied with a dropper, a small oil can, or a toothpick. When applying grease or Lubriplate, a very small dab will suffice.

The manufacturer of the special cleaner-lubricant described in the section on "cleaning" advises that the lubricant supplied is scientifically designed to reduce friction and eliminate wow and flutter caused by uneven tape tension. The lubricant is to be applied to the heads and guides and may also be applied to the tape. However, the recommendations of the manufacturer of the recorder or of a reputable electronics supplier should be followed in the selection of a lubricant for this purpose.

Care and storage of tape. In storing tape you will not encounter the problem found in disc recordings: the possibility of warping. Tapes may be stored either flat or on edge. However, excesses of heat, cold or humidity should be avoided. Tapes must not be stored in the vicinity of electrical equipment, motors, amplifiers, or loudspeakers. The cardboard boxes supplied with new reels of tape appear to be satisfactory as far as protecting the tape from dust is concerned, although some prefer metal storage cans similar to those used for motion picture film.

As your library of tapes grows, the problem of adequate marking or labeling of tapes will arise. Titles can of course be marked on the boxes or cans, but sooner or later reels and boxes will become mixed up. There is also the second problem of adequately marking the two tracks. Of course, the two faces of the reel are marked "Side 1" and "Side 2" and this system will work out if you are very careful always to place the empty take-up reel on its spindle with the proper face upward. Labels can be affixed to the reel used for storage, but there is still the possibility that you may wind the tape on the wrong reel. A much better plan is to mark the tape itself by splicing to each end some kind of leader tape plainly marked with the titles of the selections to follow; in this way you will always know whether you are starting track 1 or track 2.

This brings us to the matter of splicing tape. Inevitably, tapes will break from time to time; also, you might want to edit tapes by removing unwanted sections and inserting new material or by transposing sections. Bear in mind, though, that this type of editing can be done only when a single track is recorded on the tape; only rarely do the pauses in two tracks coincide, so that cutting the tape to splice one track will usually ruin the remaining track.

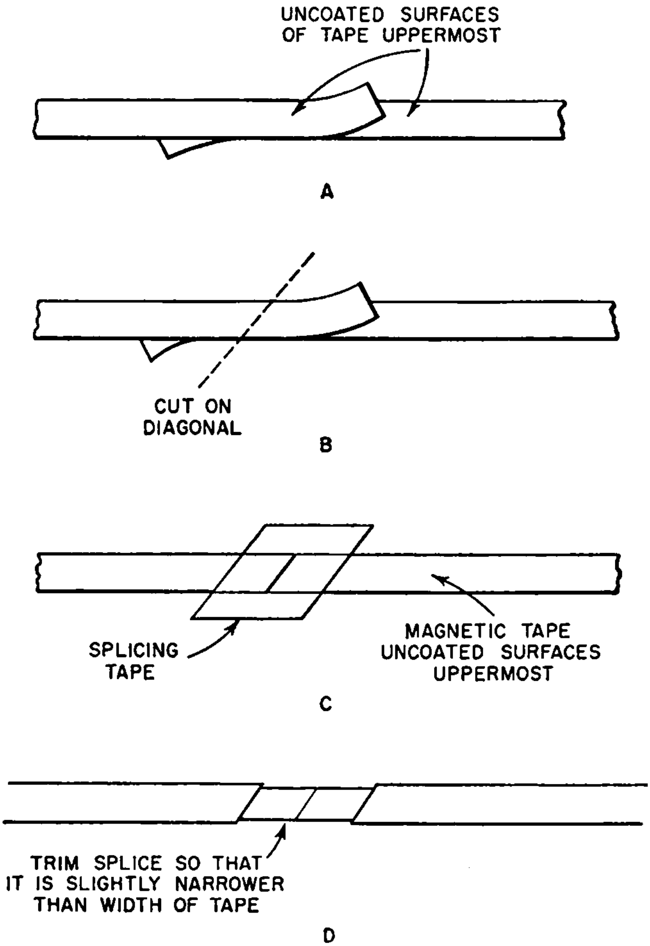


Fig. 6-18. How to splice recording tape.

To make a splice, place the ends of the two lengths one over the other as in Figure 6-18A; the overlapped portion should be as short as possible to avoid loss of too much recorded material. Now cut on a diagonal through both tapes, as indicated by the dotted line in Figure 6-18B. Lay the tapes on a flat surface with their shiny or uncoated surfaces uppermost. Bring the two cut ends together so that they fit as neatly as possible. Apply a piece of splicing tape to the joint, as in Figure 6-18C. Finally, trim away the excess splicing tape. Notice (Figure 6-18D) that in doing this the cut has been carried a trifle into the tape itself making the splice slightly narrow-waisted. This is done to avoid the possibility of a sticky splice.

When tapes are stored for long periods you might experience two difficulties. One of these is "print through" and the second is a tendency of the tape to adhere, layer to layer. Print through is caused by the magnetic impressions on one layer of tape affecting those of the adjoining layer or layers. Since a relatively long period of time is required before it occurs, it is likely that print through can be avoided, or at least minimized, by occasionally removing the tapes from storage and rewinding them. To prevent tapes from adhering layer to layer, rewind them periodically or use one of the special lubricants sold for that purpose.

7 Amplifiers

All program sources (record player, tape recorder, tuner and microphone) develop signal voltages that are quite low. A variable reluctance phonograph cartridge, for example, generally delivers an output of no more than a few thousandths of a volt. These feeble signals must be built up or amplified sufficiently so that they can actuate a loudspeaker; this is the primary function of an amplifier.

The kind of amplifier required will be determined by a number of factors including the number of loudspeakers you intend to use, the size of your listening area, the type of program source you will use and the volume level you desire.

Program sources may be divided into two classifications: low-level and high-level. These terms indicate the amount of signal voltage the source will provide. Thus we find that variable reluctance cartridges, tape recorder heads, and many microphones are classified as low-level sources because their output voltages are measured in thousandths of a volt, or millivolts. Crystal phonograph cartridges, ceramic cartridges and tuners have considerably higher output voltages (sometimes as much as one volt or more) and are therefore regarded as high-level sources.

In some cases the signal voltage delivered by a crystal or ceramic cartridge or by a tuner is sufficient so that the program source output may be fed directly to the amplifier or, as we shall refer to it from now on, the basic amplifier. However, when a low-level source is to be used, additional amplification is necessary to compensate for the lower signal level of such a source. A preamplifier is then used and it is placed between the program source and the basic amplifier. The preamplifier builds up the much weaker signals from the low-level source so that they may then be applied to the basic amplifier. In this chapter we shall consider only the basic amplifier, leaving the preamplifier for Chapter 8, although in many cases the preamplifier is physically a part of the basic amplifier if it is built on the same chassis with it. Many audiophiles lean toward the idea of an entirely separate preamplifier and there are several excellent design reasons for this preference.

In our discussion we shall subdivide the basic amplifier into several sections and discuss them separately. These subdivisions are: a. The voltage amplifier; b. The phase inverter; c. The power output stage and d. The power supply.

The Voltage Amplifier

All basic amplifiers include one or more voltage amplifiers. The function of such a stage or stages is to receive weak signals from the program source and to amplify them, thus delivering a stronger signal to the following stage. A voltage amplifier stage is so named because it receives a small voltage and delivers a larger voltage; it is not called upon to deliver appreciable power as a power output stage must.

A voltage amplifier stage (this applies as well to any other type of stage) consists of a vacuum tube together with associated components such as resistors and condensers. The vacuum tube may be either a triode (three-electrode tube) or a pentode (five-electrode tube). The difference is that the pentode affords higher gain, or amplification, per stage. However, the triode has other advantages which is why it is frequently used in spite of the pentode's higher gain. Generally, when more gain is needed than one triode is capable of giving, the designer has three choices: to use a pentode in place of the triode; to use two separate triode tubes; or to use a dual-triode tube, that is to say, a tube having two sets of triode electrodes enclosed within a single envelope. Examples of dual-triode tubes are the 12AU7, 12AX7, 7025, ECC82 and ECC83. All of these are commonly used as voltage amplifiers. The last two numbers refer to tubes of British or German manufacture and are widely used in high fidelity equipment because they afford a lower noise level.

The voltage amplifier stage will usually be located close to the input jack of the basic amplifier, unless a preamplifier is built upon the same chassis. In that case, the preamplifier tubes will be located between the input and the voltage amplifier stage or stages.

The Phase Inverter

Practically all wide-range amplifiers use a power output stage of the push-pull type. Briefly described, a push-pull stage consists of two similar tubes operating together in one stage. The power output of such a stage is slightly more than twice that of a stage using a single tube, and hum and distortion are greatly reduced. (The use of a push-pull output stage has become so usual that it would be difficult if not impossible to find a high fidelity amplifier without it. There are exceptions of course in the very low-priced so-called high fidelity phonographs and in some cheap stereo equipment, but they are not worth considering here.)

A push-pull stage must be supplied with two signal voltages that are equal in magnitude but opposite in phase relationship (while one signal is at a positive maximum the other is at a negative maximum). There are two methods of obtaining such signal voltages: the use of a special transformer between the voltage amplifier and the power amplifier, or the use of a phase inverter or phase splitter tube. The transformer method is not only expensive but has other drawbacks, so that the phase inverter is now almost universally used.

There are a number of different phase inverter circuits, some using one tube and others two tubes. In the latter type, a dual-triode tube is almost always used. Most phase inverters use triode tubes and the types commonly employed are the same as those listed under voltage amplifiers. The two-tube phase inverter is considered to be superior to the single-tube type. The phase inverter stage follows the voltage amplifier electrically and, in most designs, physically as well.

The Power Output Stage

The power output stage differs from a voltage amplifier stage in that it must supply to the loudspeaker a considerable amount of power. Like the voltage amplifier it also increases the signal intensity, but here the degree of amplification is usually less than in the voltage amplifier stage.

The power-handling capacity of the power output stage depends upon the type of tube used, the d.c. operating voltages, and the circuit design. As a rule, power output tubes are larger physically than types used for voltage amplification or phase inversion. The first reason for this is that in handling higher currents the tube electrodes (cathode and plate) must be larger. Second, because they handle considerable power the operating temperatures of such tubes run higher and the envelopes must be larger in order to dissipate the heat generated.

The tubes used in the power output stages of modern high fidelity amplifiers are almost always either pentodes or beam power tubes. The pentode, as mentioned earlier, has five electrodes, while the beam power tube has four; to simplify matters you might regard the beam power tube as a special type of pentode.

Earlier amplifiers often used triodes because of their lower distortion. They did, however, have several disadvantages, among them being lower efficiency and the need for additional driving power. The result was that triodes were abandoned in favor of pentodes and still later, beam power tubes. Earlier pentodes and beam tubes had the disadvantage of relatively high distortion, but this has been overcome in modern types such as KT-66, 5881, 6550, 6973, and 7027-A.

Another development is the ultra-linear stage, which is a compromise between triode and tetrode operation. An output transformer of special design is needed for this type of circuit and it is somewhat more expensive than the usual type. In any high fidelity output stage the most expensive single component is the output transformer, link between the output tubes and the speaker system; as a rule, it is also one of the largest and heaviest units in the amplifier. Although size and weight of the output transformer do not necessarily indicate that the amplifier is well designed you would do well to avoid amplifiers using a small, inexpensive transformer. An otherwise excellent amplifier can easily be ruined by the use of a mediocre output transformer.

The Power Supply

The power supply furnishes alternating current for the operation of the heaters of all tubes in the amplifier (and frequently in the preamplifier as well, even in cases where the preamplifier is a separate unit) and direct current for the remaining tube electrodes. The power transformer, largest and heaviest unit in the power supply, reduces the 117-volt supply to voltages suitable for the tube heaters and at the same time increases the 117-volt supply to the 350 to 450 volts required for the output tubes. In performing these functions, the power transformer normally handles 150 to 250 watts and must be capable of withstanding a higher load under abnormal conditions for at least short periods of time.

The remainder of the power supply consists of a rectifier tube for changing the alternating current to direct current, and a filter system for smoothing out the direct current so that it may be applied to the amplifier tube electrodes. The rectifier tube must be capable of supplying all of the direct current that the power out-

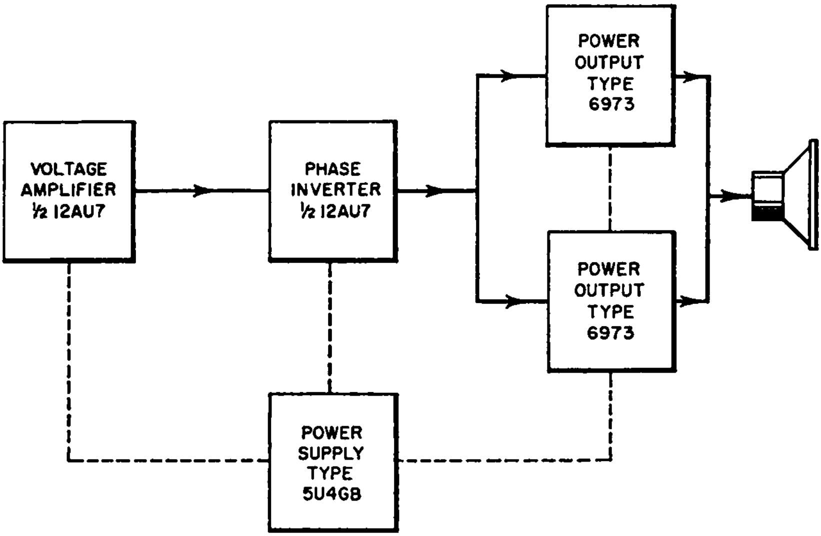


Fig. 7-1. Block diagram of basic amplifier.

put stage, phase inverter, and voltage amplifier stages (and possibly also the preamplifier stages) require, plus a margin to take care of overloads. Types in present use include 5U4G, 5R4G, 5V4G, and GZ34. A few amplifier power supplies use two rectifier tubes; some others use one rectifier tube plus a selenium rectifier.

A good power supply must meet three requirements: First, it must be capable of continuous operation over long periods without overloading or overheating. Second, the d.c. supply must be well regulated; this means that the d.c. voltages applied to the amplifier tubes must remain as nearly constant in value as possible even under widely changing load conditions. Finally, the d.c. must be well filtered; in other words, the direct current must be as nearly pure direct current as possible. In fulfilling all of the conditions mentioned, engineers have developed many excellent and ingenious designs, but, as we might expect, the closer the amplifier approaches the ideal the higher becomes the cost.

Figure 7-1 shows in block diagram form, the simplest possible basic amplifier. In a block diagram, each block or square represents a stage; that is, the tube plus all associated components. The solid

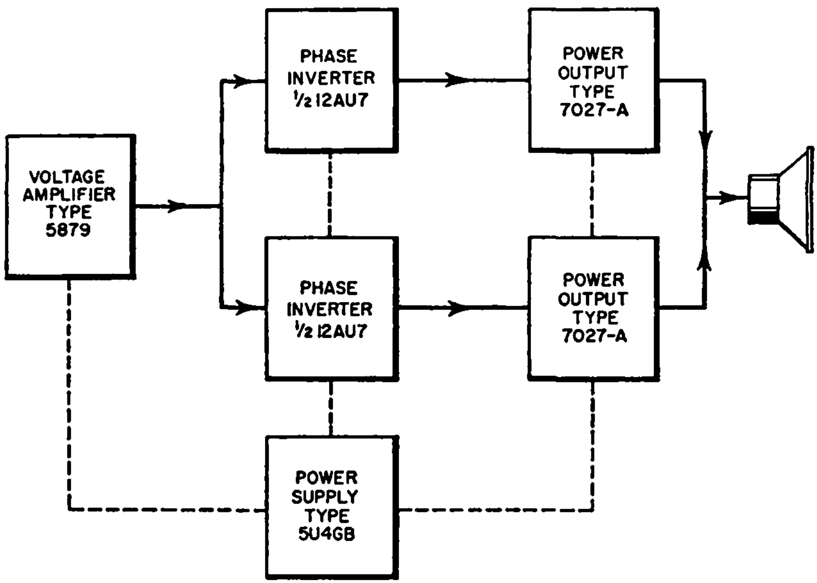


Fig. 7-2. Block diagram of basic amplifier.

lines show the route of signals from stage to stage while the dotted lines show the paths of direct current between power supply and the various amplifier stages. Note that in this amplifier one dual-triode tube performs the two functions of voltage amplifier and phase inverter. The output stage is of the push-pull type, consequently it consists of two blocks each representing a tube with associated components.

As we noted before, some phase inverters require two tubes. Present practice is to use a dual-triode tube rather than two separate triodes, for reasons of space and economy. The block diagram of a basic amplifier using a phase inverter of this type is shown in Figure 7-2. In this amplifier a pentode tube is used as voltage amplifier.

In some designs, the use of a driver stage is mandatory. If the power output stage requires more driving power than the phase inverter is capable of supplying, then a driver stage is placed between the phase inverter and the output stage. And, since the phase inverter delivers two differing signals to the output stage, the driver

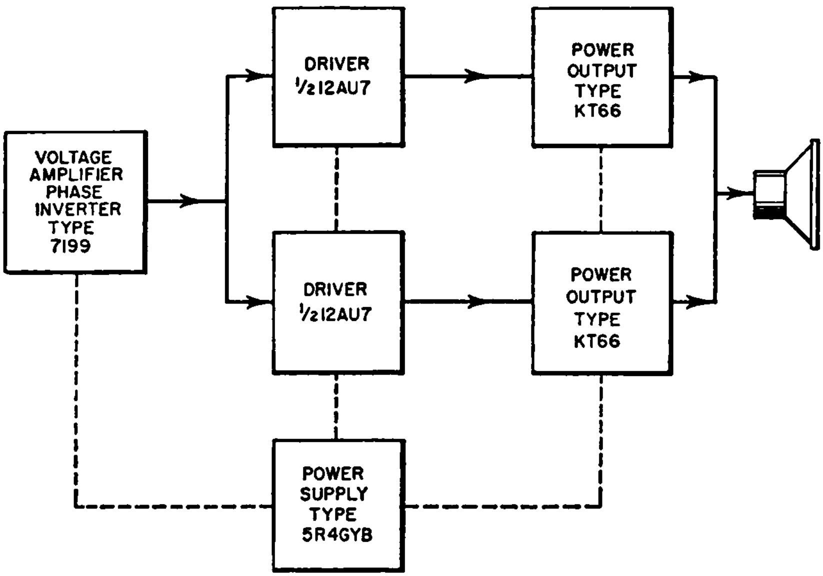


Fig. 7-3. Block diagram of basic amplifier.

stage must consist of two tubes, one for each of the two signals. This is illustrated in Figure 7-3.

Finally, some basic amplifiers require two voltage amplifier stages, one following the other as in Figure 7-4. While these four block diagrams show in an elementary way, the arrangement of stages in the majority of basic amplifiers, there is still a great variety of combinations and a number of special features. Remember too, that the tube type numbers mentioned indicate only that a tube belonging to the general classification is used. The choice of a tube type is determined by the designer's particular requirements.

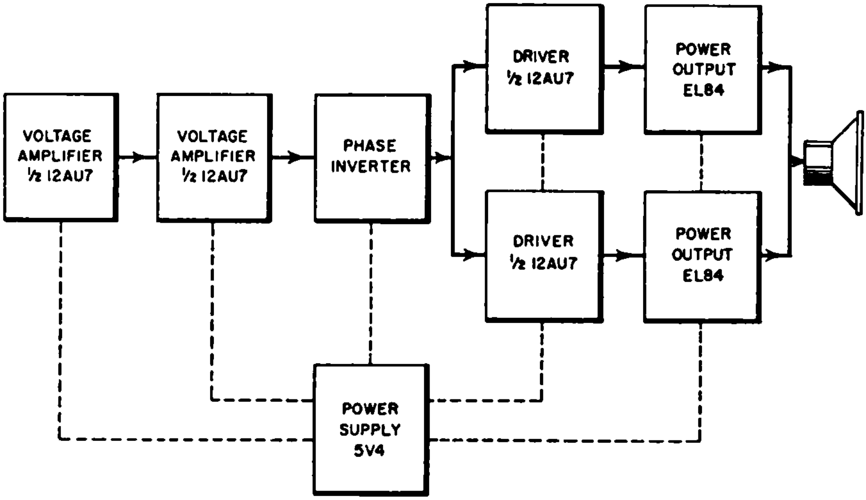


Fig. 7-4. Block diagram of basic amplifier.

Amplifier Characteristics

Frequency Response. As we noted in Chapter 1, a wide-range amplifier must be capable of reproducing frequencies as low as 30 cycles per second and as high as 15,000 cycles per second. These figures represent the minimum requirements for high fidelity. A response of 30 to 15,000 cycles is not particularly difficult to achieve; in fact many amplifiers are rated at 20 cycles to 20,000 cycles.

A mere statement of frequency response limits is not enough to give us a clear idea of the amplifier's performance. We should know also how much deviation there is from flat response. Figure 7-5A is a response curve of an amplifier and shows why the frequency response rating alone means very little. In this illustration the levels of sound intensity have been plotted vertically and the frequencies horizontally; we are thus enabled to judge the performance in terms of amplification at any frequency within the range. Notice that at both the high frequency and the low frequency ends of the range, the amplifier output drops; in other words, at the higher and lower frequencies the output of the amplifier is less than it is at other frequencies. Nevertheless, such an amplifier might still be rated as having a frequency response of 30 to 15,000 cycles.

An amplifier is said to have flat response when all frequencies within its range are amplified to the same degree. An essentially flat response curve is shown in Figure 7-5B. Notice that deviations from the ideal flat response are so minor as to be negligible. Now suppose that the curve of figure 7-5A applied to an actual amplifier. If the manufacturer advertised his product as having a response of 30 to 15,000 cycles with no further qualification, the statement would be correct but would conceal the fact that there are serious deficiencies in the extreme high and low portions of the range. On the other hand, a statement that an amplifier has a response of "30 to 15,000 cycles plus or minus 1 db" tells us a great deal more about the amplifier for we then know that at no point within the

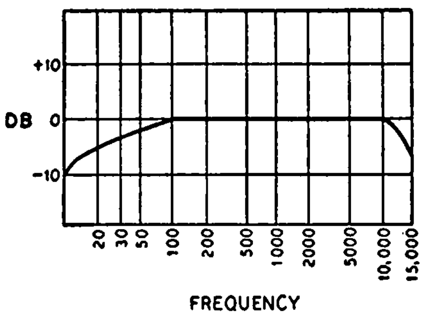


Fig. 7-5A

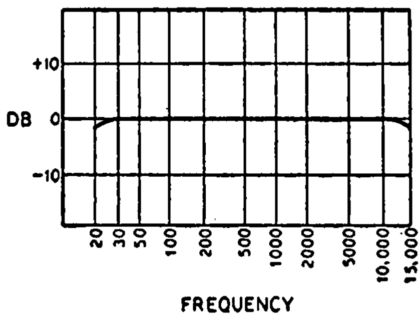


Fig. 7-5B

entire range does the output rise or fall more than 1 db from an established level. A response that is flat within 1 db is acceptable; 0.5 db or less is considered very good.

Power output. It has been demonstrated that the power needed to reproduce the louder passages of orchestral music at concert hall level in an ordinary living room is 0.4 to 0.5 watt. This represents the amount of acoustic power that the speaker must deliver. In view of this seemingly small amount of power, the reader may well wonder why it is necessary to use amplifiers rated at 30, 40 or 50 watts. The answer lies in the relatively low efficiency of wide-range speakers presently available. Bear in mind that efficiency has nothing whatever to do with frequency response or any other desirable quality of the speakers. In fact, in order to achieve good frequency response, it is usually necessary to sacrifice efficiency. The result is that the usual wide-range speaker has an efficiency of about 5 percent, which means that of the total power delivered to the speaker by the amplifier, 95 percent is wasted. Therefore, if the speaker is to deliver 0.5 watt of acoustic power, the amplifier must supply the speaker with 10 watts of electrical power. We may accept this figure as an absolute minimum for a high fidelity amplifier. If we intend to use speaker systems having multiple units (woofer-tweeter or woofer-tweeter-mid-range combinations) amplifier output power must be still further increased because the efficiency of such speaker combinations is lower than for single speakers due to losses in the crossover or frequency divider. If the room is unusually large or acoustically "dead" (containing a great deal of sound-absorbing material such as rugs, draperies, etc.) amplifier output must be increased to compensate for these conditions. In view of all of these factors we see that an output rating of 25 or 30 watts is really moderate.

Sensitivity. The sensitivity of an amplifier refers to the output level produced for a given input signal level. Sometimes the sensitivity or gain is given in decibels and may range from 40 to 70 db. More often it is expressed as the input voltage required to drive the amplifier to full output. Thus, the sensitivity of a 15-watt amplifier might be stated as 0.8 volt for 15-watt output. If you were considering the purchase of such an amplifier and intended to use it with a ceramic phonograph cartridge, it would only be necessary for you to determine the voltage output of the cartridge. Suppose it happened to be 1.0 volt. Since only 0.8 volt signal input is needed for full output, the cartridge and the amplifier would be compatible. By contrast, a variable reluctance cartridge having a voltage output of 50 millivolts would not be suitable at all since 50 millivolts is far below the required 0.8 volt; in fact such a cartridge could not be used without a preamplifier.

Distortion. We need be concerned with four types of distortion occurring in amplifiers. These are: a. Frequency distortion; b. Amplitude distortion; c. Harmonic distortion and d. Intermodulation distortion. There are other forms of distortion but usually they are not serious and are not considered in amplifier specifications.

a. Frequency distortion. This form of distortion occurs when the amplitudes, or intensities of output signals at all frequencies are not equal, even though the input signal levels are equal. Suppose that a 3,000 cycle signal at a given level is applied to the input of the amplifier and results in a certain output level. A 10,000 cycle signal of the same level replaces the 3,000 cycle signal and the output level is then much lower. This is frequency distortion and has been discussed in a somewhat different light under the heading of frequency response. When an amplifier has an absolutely flat response there is no frequency distortion.

b. Amplitude distortion. Amplitude distortion occurs when the intensity of a signal delivered by an amplifier is not directly proportional to the instantaneous amplitude of the input signal.

c. Harmonic distortion. Harmonic distortion takes place when the passage of a signal through the system generates other frequency components that were not present in the original signal. These components are multiples of the original frequency (2, 3, 4 times, etc.) and are referred to as harmonics.

Harmonic distortion in an amplifier is usually expressed as a percentage of the output power. Harmonic distortion of 1.0 percent or less at full rated power output is considered good, but many top quality amplifiers run considerably less than 1.0 percent.

When comparing the performance data of amplifiers, be sure that you read all of the figures carefully. In some cases the harmonic distortion is specified at an output less than full rated. This is an understandable tendency, for the harmonic distortion will always be greater at full output than at some lower value. For instance, one amplifier is rated at 0.2 percent harmonic distortion at 5 watts output. This is interesting information and of value to those who operate their equipment well below full output. But the rated power output of this particular amplifier happens to be 30 watts and when delivering this amount of power the harmonic distortion rises to slightly more than 1.0 percent.

d. Intermodulation distortion. When two signals of different frequencies are applied to the input of an amplifier the two should have no effect on each other and only the two original signals should appear at the output. When intermodulation distortion exists the two signals react with each other in a manner that produces new components not present in the originals. These new components are the sum and difference of the two signals and their interaction is called "beating" or "heterodyning." As in the case of harmonic distortion, intermodulation distortion is measured in a percentage of full output. One to 1.5 percent is considered good.

Selection of an amplifier should be based upon full consideration of all the published data in addition to extensive listening tests.

You should also take into consideration the compatibility of the amplifier with the program source or sources and with the preamplifier.

If you feel that you do not care to invest in a factory-built amplifier, there are two other courses open to you. The first is to buy the amplifier in kit form, the second to buy all of the components separately and build your own amplifier.

An individual who has no mechanical or electrical experience and who has no interest in this kind of work should, of course, select an amplifier from among the many factory-built units available. It is true that the cost will be higher (perhaps twice as much as for a do-it-yourself project), but there will be little or no doubt about the results, particularly if the amplifier has been carefully selected.

On the other hand, if you have had just a smattering of electronic experience, if you know how to solder and how to use hand tools such as a screwdriver and pliers and particularly if you enjoy doing such things you might want to build an amplifier. If you are the type of individual just described, you should not attempt to buy individual components for the construction of an amplifier described in a magazine or elsewhere. However, you will find that constructing an amplifier from one of the many excellent kits now available is a rewarding and profitable experience. All of the necessary engineering has been done for you and all of the work requiring a great deal of skill, such as preparing the chassis, has also been done. Furthermore, the instruction booklets accompanying these kits are so detailed that one is practically assured of success. Unfortunately, in most areas you will have to select a kit solely upon the basis of published performance data, without listening to a sample; in a few large cities electronics distributors will have sample amplifiers for demonstration.

Unless you have had a great deal of electronic experience you are not advised to build an amplifier from separately purchased components. If you feel you have had sufficient experience bear in mind that building an amplifier is not a project to be undertaken lightly, nor can you expect to save a great deal over the price of a kit unless, of course, you happen to have some of the parts already. You should have had sufficient experience so that you can rely upon your own judgment in making decisions that would have been made for you if you had decided to build from a kit. You alone will have to decide such matters as where to place a particular component and how the wiring should be located. Of course, these matters, as well as a host of others involved in construction of any electronic device could be described in detail in a book, but we would then not have a book on home music systems but a complete treatise on electronics.

Before beginning the description of an amplifier, a few apparently minor but really very important points should be considered. Although in some amplifiers ground connections are made to the nearest point on the chassis, the use of a ground bus is preferable. This bus can be made of several lengths of push-back wire, stripped of insulation and twisted together. The bus is insulated from the chassis except at one point; the single chassis ground connection should be made as close to the input jack as possible. All connections shown as ground connections in the diagrams are to be made to the ground bus. All wires, especially grid and plate leads should be as short and direct as possible. It is a good idea to keep grid wires close to the chassis; this will minimize hum pickup.

The stage-by-stage method of wiring will be used, with a separate diagram for each stage. This system of constructing electronic equipment has been thoroughly tested over a period of many years and has been found to be valuable in avoiding errors. Furthermore, if you desire, you can test each stage as it is completed. The wiring procedure in this project has been subdivided into the following steps:

a. Power supply, including line and heater wiring.
b. Output stage.
c. Phase inverter.
d. Voltage amplifier.

Circuit Description

The combination voltage amplifier-phase inverter employs a type 7199 triode-pentode tube. The pentode section functions as voltage amplifier and is direct coupled to the triode section of the tube, which performs the function of phase inverter.

Signals from the cathode and the plate of the phase inverter are applied to the grids of the output stage tubes. These are type 7027-A beam power tubes. From the 10-ohm tap of the output transformer secondary a feedback loop extends to the cathode of the voltage amplifier stage. Fixed bias for the output tubes is developed across resistor R2 (see power supply diagram, Figure 7-7).

The power supply uses a type 5U4GB rectifier. The two-section filter consists of R3, R4, C3, C5A and C5B. Voltages for tubes in the basic amplifier are taken from points "B" and "C." Point "C" may also be used for supplying voltages to a preamplifier.

Performance Data

Power output: 30 watts.
Sensitivity: 1 volt input needed for 30 watts output.
Frequency response: 15 to 30,000 cycles, plus or minus 0.5 db.
Harmonic distortion: less than 1 percent at 30 watts output.

Construction Data

The amplifier may be built upon a chassis measuring 8 by 14 by 2½ inches, or smaller. Before beginning any wiring, lay out the

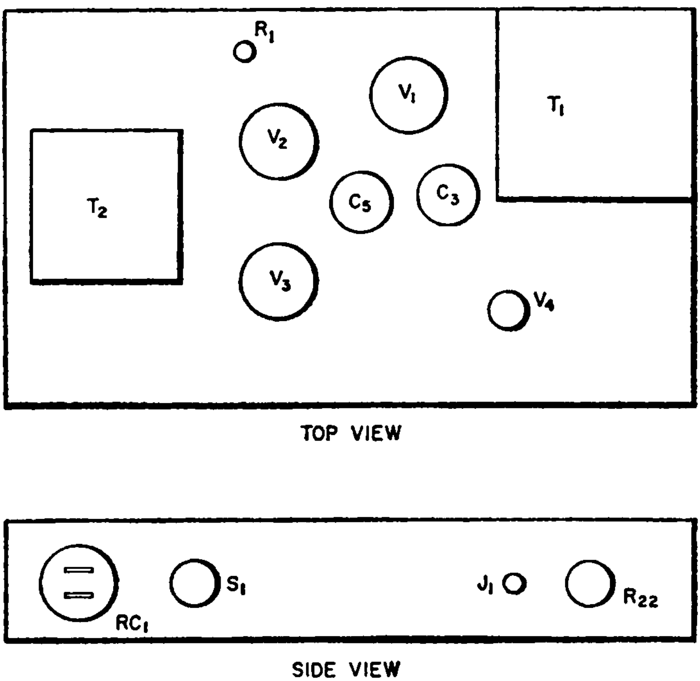


Fig. 7-6. Chassis layout.

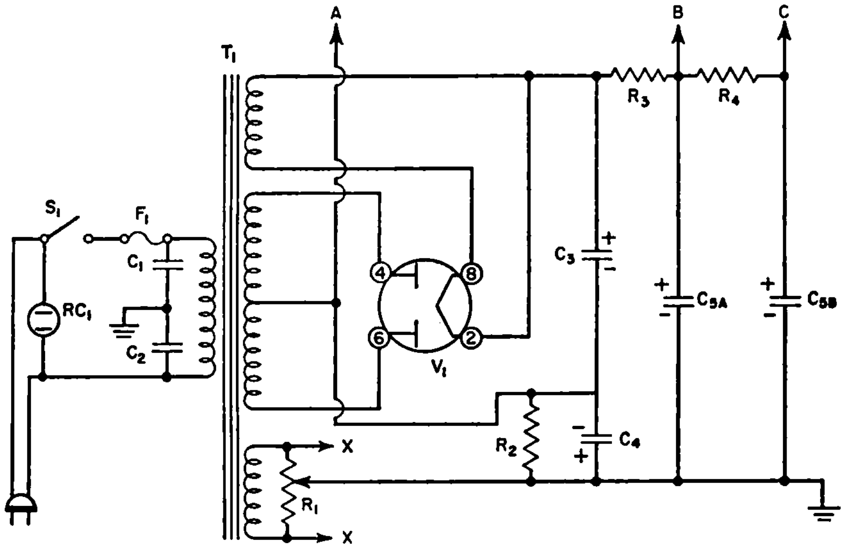


Fig. 7-7. Power supply.

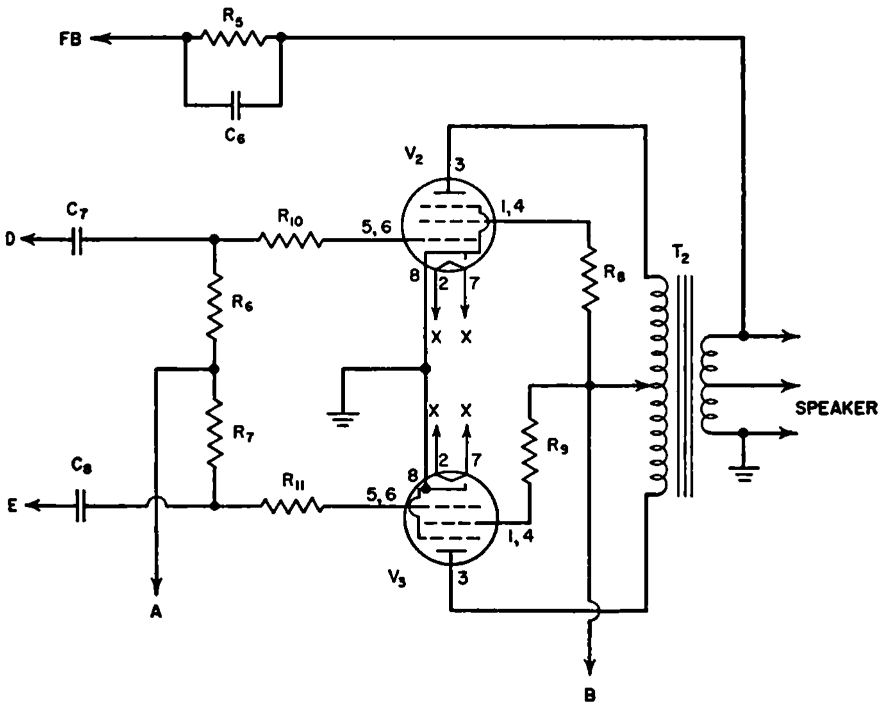


Fig. 7-8. Output stage.

chassis and drill or punch all required holes for the mounting of parts (Figure 7-6). Assemble all parts that are mounted directly on the chassis. Resistors and small condensers may be placed in position either at this time or as the wiring progresses.

Power supply. Note that the center tap of the power transformer high-voltage winding is not directly grounded, but is connected to ground through the bias resistor, R2 (see Figure 7-7). Also, observe

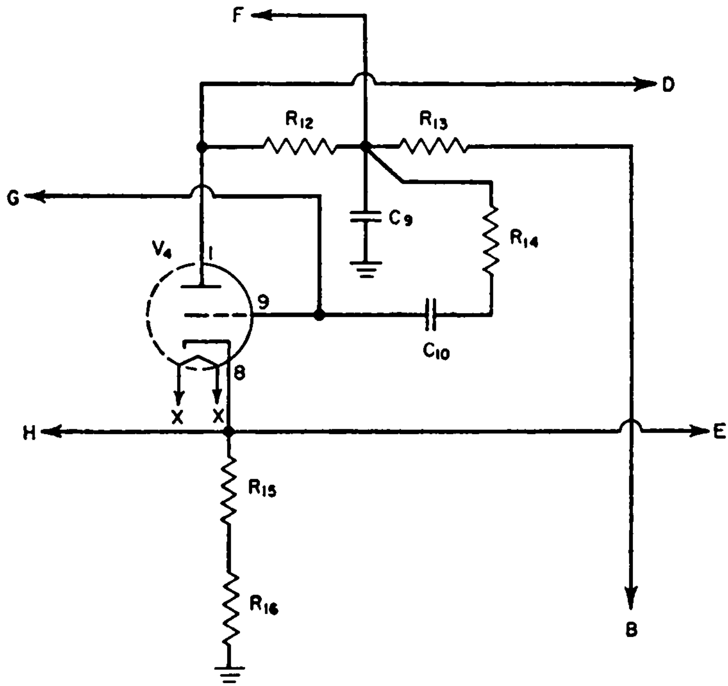


Fig. 7-9. Phase inverter.

polarities of electrolytic condensers, C3 and C4. The phono motor receptacle, RC1, is connected so that power is supplied to the motor at all times, but if desired may be connected so that power is switched off when the amplifier is off. In that case, transfer the upper connection of RC1 from the left-hand terminal of switch S1 to the right-hand terminal.

Output stage. (See Figure 7-8) The output transformer should be oriented on the chassis so that the plate wires are turned toward the tube sockets. Resistors R6, R7, R10, and R11 should be mounted on tie strips placed close to the tube sockets. Note that one lead of each of the coupling condensers C7 and C8 remains unconnected at this time; these connections (marked "D" and "E") will be completed when the phase inverter is wired. Point "B," the center-tap of the output transformer primary is to be connected to the point marked "B" on the power supply. Similarly, point "A" is connected to point "A" in the power supply (center-

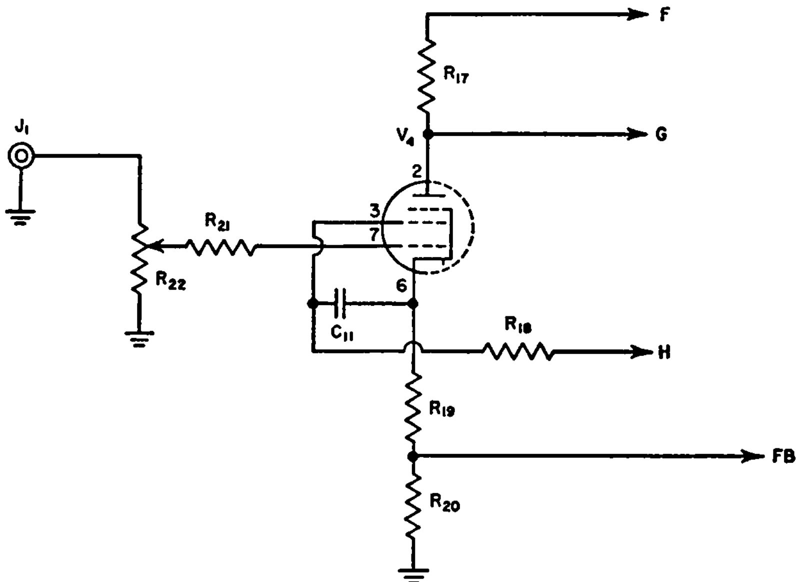


Fig. 7-10. Voltage amplifier.

tap of the power transformer high-voltage winding). Point "FB" remains unconnected until the voltage amplifier stage is wired.

Phase inverter. (See Figure 7-9) Locate all resistors and condensers on tie strips as close as possible to the tube socket terminals to which the resistors and condensers are to be connected. Note that coupling condensers C7 and C8 (see Figure 7-8) are now connected to points "D" and "E" (plate and cathode of phase inverter). Points "F," "G" and "H" remain unconnected until the next stage (the voltage amplifier) is wired.

Voltage amplifier. (See Figure 7-10) If R22 has been placed close to J1 and if both these components are close to the tube socket all input circuit wiring may be made quite short and direct.

The remaining resistors and condensers are mounted on tie strips properly oriented with respect to the proper tube socket terminals. The wiring is completed by connecting points "F," "G," and "H" to the correspondingly lettered points in the phase inverter. Point "FB" is, of course, connected to the feedback circuit consisting of R5 and C6 (see Figure 7-8).

When the amplifier has been completed, short the input jack and adjust R1 for minimum hum. If any difficulty is experienced with hum when phono motor and preamplifier are connected, try reversing the various line plugs (amplifier, phono motor and preamplifier, assuming preamplifier has a separate power supply). Be sure that all cables between phono pickup and amplifier and between preamplifier and basic amplifier are properly shielded, with the shield grounded. In some cases hum is reduced by disconnecting the ground from one end of a shielded cable; in other cases, better results are obtained with both ends of the shield grounded. If the record player has provision for grounding the motor, be sure that the ground connection is in place.

LIST OF PARTS

Capacitors

C_1, C_2 0.05 mfd. 600-volt paper tubular
C_3 40 mfd. 600-volt can type electrolytic
C_4 100 mfd. 50-volt cartridge type electrolytic
C_5A, C_5B 40 mfd.-40 mfd. 600-volt electrolytic
C_6 0.01 mfd. 600-volt paper tubular
C_7, C_8 0.25 mfd. 600-volt tubular
C_9 80 mfd. 600-volt cartridge type electrolytic
C_{10} 22 mmfd. 600 volts
C_{11} 25 mfd. 50-volt cartridge type electrolytic

F_1 3-ampere 150-volt fuse with holder
J_1 RETMA-type phono jack
R_1 100-ohm ½-watt potentiometer

Resistors

R_2 220-ohm 10-watt, 10%
R_3 50-ohm 10-watt, 10%
R_4 10K-ohm 2-watt, 10%
R_5 270-ohm ½-watt
R_6 100K-ohm ½-watt
R_7 100K-ohm ½-watt
R_8 56-ohm ½-watt
R_9 56-ohm ½-watt
R_{10} 1K-ohm ½-watt
R_{11} 1K-ohm ½-watt

R_{12} 15K-ohm 2-watt, 10%	R_{17} 220K-ohm ½-watt
R_{13} 2K-ohm 2-watt, 10%	R_{18} 180K-ohm ½-watt
R_{14} 22K-ohm ½-watt	R_{19} 820-ohm ½-watt
R_{15} 15K-ohm ½-watt, 5%	R_{20} 10-ohm ½-watt
R_{16} 1K-ohm ½-watt	R_{21} 10K-ohm ½-watt

R_{22} 500K-ohm potentiometer
RC_1 Flush-mounting receptacle
S_1 On-off switch, panel mounting type
T_1 Power transformer
Secondary 375-0-375 volts, 160 ma.
Rectifier 5.0 volts 3 amperes
Filament 6.3 volts 4 amperes
T_2 Output transformer
to match voice coil to 5,000-ohm plate-to-plate load
V_1 5U4GB
V_2, V_3 7027-A
V_4 7199
Chassis 8 by 14 by 2½ inches
Terminal strip for speaker
Knobs for on-off switch and volume control
3 octal wafer-type tube sockets
1 9-pin wafer-type tube socket
8 3-terminal solder lug strips
10 2-terminal solder lug strips
6—32 machine screws, ½-inch long
6—32 hexagon nuts
8—32 machine screws ½-inch long
8—32 hexagon nuts
Solder
Hook up wire

8 Preamplifier-Control Units

It is more than likely that a preamplifier or a preamplifier-control unit will be a part of your high fidelity home music system. In fact, if you select a record player equipped with a magnetic type of cartridge, if you plan to incorporate a tape recorder in your system, or if you wish to use certain types of microphones, a preamplifier is an absolute necessity. In addition, a modern preamplifier-control unit combination performs a number of other important functions and adds flexibility to the system. It is therefore appropriate that we consider the need for a preamplifier, the various functions of a preamplifier-control unit, the types available and their characteristics. Also, in this chapter instructions will be given for building a simple fixed type of preamplifier (one that does not include the various control functions). This unit is designed for use in connection with a variable reluctance phonograph cartridge and will perform quite satisfactorily. Building the more elaborate preamplifier-control unit is a project far beyond the skill of a novice, unless he purchases a kit. A number of excellent kits are on the market at moderate prices.

Need For a Preamplifier

The average basic amplifier (power amplifier) requires a signal input of from 1 to 2 volts to drive it to full power output. A tuner (AM or FM) will usually provide signal voltage of this magnitude, as will some crystal phonograph cartridges. If your high fidelity system is to have only one program source—a tuner (which is unlikely), it is possible that you would not need a preamplifier at all.

However, many program sources do not furnish sufficient signal to drive an amplifier to full output. Magnetic cartridges, for example, ordinarily do not deliver more than 20 to 50 millivolts (20 to 50 thousandths of a volt). A single-track tape head delivers about 10 millivolts and two-track heads usually deliver still less voltage. Since such small voltages are inadequate they must be built up or amplified before being passed to the basic amplifier; this is the first function of a preamplifier.

A simple fixed preamplifier uses one or two vacuum tubes, functioning as voltage amplifiers. When only one tube is used, it is always of the dual-unit type; that is to say, there are two tubes within a single glass or metal envelope. Two stages of amplification are thus provided by this dual-unit tube; one stage would rarely be sufficient.

Operating voltages for the preamplifier may be taken from the main amplifier, although in some types the preamplifier has its own power supply.

Equalization

In the preceding paragraphs we have mentioned "record equalization" and it is now necessary to explain what it is and why it is used.

When a recording is made, it is possible for the recording stylus to "overcut," or run into an adjacent groove. Much of the sound energy present in music lies in the lower tones. As the lower frequencies develop more energy, the excursion, or swing, of the cutting stylus tends to be greater on low notes than on high ones. It is in the lower part of the musical range that overcutting becomes a serious problem.

Overcutting is prevented, during recording, by limiting the swing of the cutting stylus on low frequencies. In other words, the motion of the stylus then is no longer directly proportional to the sound energy. The record manufacturer selects a frequency below which all tones are to be attenuated, or reduced in volume. This is called the turnover point and is usually in the neighborhood of 500 cycles. Below the turnover point, all frequencies are attenuated. However, the attenuation is increased progressively. For instance, if the attenuation is to be 6 db, it is augmented by that amount for each octave below the turnover point.

If we were to reproduce a record made in this way and made no compensation for the drop in volume level below the turnover point, the music would be noticeably deficient in bass. Fortunately there is a way to compensate or equalize for the attenuation of the bass end of the range. It can be done by introducing a relatively simple electrical filter, consisting of resistors and condensers, into the circuit.

Not all record manufacturers use the same turnover point. Originally, this point was established at 250 cycles per second, but record makers soon found that, in order to overcome the effects of surface noise, it was desirable to cut records at a higher volume level. Since this increased the energy (and the movement of the cutting stylus) throughout the entire range, it became necessary to raise the turnover point to about 500 cycles per second; some recent recordings have been cut with a 1000-cycle-per-second turnover point. If we are properly to reproduce a record, we must know the turnover point, and fortunately this information is available for most makes of records. However, if we do not have the exact information, an equalizer designed for turnover at 500 cycles per second will usually give fair results.

In addition to the attenuation of lows already mentioned, many record manufacturers pre-emphasize, or increase, the intensity of all tones above a certain frequency. This is done in order to minimize surface noise. If you have ever used the tone control on a radio receiver to reduce static, you discovered that such noises affected principally the high end of the musical range. The same thing is true in a recording and, if we increase the intensity of the highs, the signal-to-noise ratio becomes higher and the noise is less noticeable at usual volume levels.

As in the case of the turnover point, we find that all records are not the same with respect to the degree of pre-emphasis or the point at which it begins. This point varies from 1000 cycles per second to 3000 cycles per second and the rate of pre-emphasis from 2.5 db to 6 db per octave.

Considerable progress has been made in the direction of standardization in the manufacturer of disc recordings. A high percentage of modern recordings are processed according to the RIAA (Record Industry Association of America) curve. However, it is quite likely that you own or will buy recordings that do not conform to this curve and different equalization circuits must be used for the several methods of recording.

All of this may seem rather complicated to you and you may be wondering whether or not this variation among recordings is not a hopeless situation. Actually, it is much simpler than it appears to be. In some cases the difference between recordings is slight and one equalizer will serve for two types.

A typical preamplifier-control unit will have either one or two knobs for equalization. These knobs control multi-position switches that cut in or out of the circuit the various equalizers. The single-knob control might have positions marked as follows: RIAA, AES, FFRR, LP. These simply refer to the method of processing the record. With the control in the RIAA position, equalization is introduced for all records processed in accordance with the RIAA curve. AES means Audio Engineering Society, FFRR is Full Frequency Range Recording (London), and LP pertains to Columbia LP.

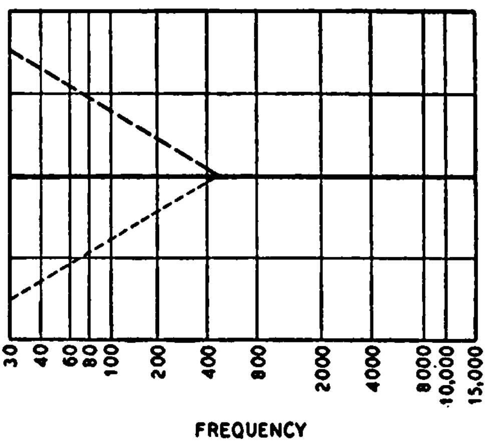


Fig. 8-1. Effect of equalization.

If the control unit has two equalization knobs, (which is likely to be the case) they will be marked "Turnover" and "Rolloff." "Turnover" of course refers to the turnover point used in recording; this was explained earlier. Therefore the turnover control provides varying degrees of compensation for the cut in bass made during recording. The rolloff control introduces suitable compensation at the opposite end of the frequency range. Here during recording, as you will recall, the treble was boosted. Therefore the control provides a varying degree of treble cut.

As a rule, the turnover control will have markings similar to those on the single knob control mentioned earlier: LP, RIAA, AES, etc.

The rolloff control usually has markings indicating the amount of treble cut in decibels. For example: 0, 8, 12, 16. In many cases the 12 db position will also be labeled RIAA, for that is the amount of treble rolloff to be used in equalizing for the RIAA recording characteristic.

The use of equalizers results in a loss in amplification, consequently more amplifier tubes must be added; that is why elaborate preamplifier control units often use three or four tubes, providing as many as six stages of amplification. You can readily understand that this results in a rather critical piece of equipment, difficult to build unless you use a pre-designed kit.

The design of preamplifiers and preamplifier-control units varies greatly with regard to the equalizers provided. At one extreme we have the single equalizer circuit used in a simple fixed preamplifier. At the other extreme, there are units providing continuously variable treble and bass equalization. And between these two extremes we find the design used in most factory-built control units and in kits: two control knobs having a number of positions, each providing equalization for a certain recording characteristic.

The effect of equalizers in compensating for the reduction in bass and increase in highs found in most recordings can be demonstrated pictorially. Look at the chart, Figure 8-1. The dotted line shows the effect of attenuation of bass notes to prevent overcutting. Notice that from 500 cycles down to 30 cycles the response declines abruptly. Now we insert in the circuit an equalizer that has a rising characteristic; in other words, as the frequency drops, the response increases. This is shown by the dashed line. The two curves are complementary and, as a result, we get a flat response characteristic as shown by the solid line. This same thing applies to pre-emphasis where the recording has a rising characteristic above a certain frequency, except that in this case we use an equalizer that applies a progressively greater amount of attenuation above a certain frequency.

Equalizers can be inserted at various points in the system. In some designs the equalizer is placed ahead of the preamplifier, often close to the phonograph cartridge, but there is a serious objection to this method. As stated before, equalization always introduces a loss; the result is that the signal is made weaker before it reaches the preamplifier. The possibility of hum and noise is increased because the ratio of signal to noise has been cut down.

To take advantage of the amplification afforded by the first preamplifier tube, most manufacturers locate the equalizers between the two preamplifier stages, as shown in Figure 8-2. Assuming that the preamplifier is separate from the main amplifier, this places the equalizer control close to the record player, where it belongs. It is perfectly logical to assume that record equalization adjustments will be made just after the record has been placed on the turntable.

Tape recorder playback heads also require equalization, and many of the more elaborate control units provide for this. Some do not, but in such cases it is generally satisfactory to use one of the positions intended for disc recording compensation. In many cases the RIAA position works fairly well.

Volume Control

Many of the simpler fixed preamplifiers are not equipped with a volume control and in such cases volume is intended to be controlled at the main amplifier. In preamplifier-control units a volume control is always included and it may be located in various places in the circuit. It is never placed at the input of the unit; that is to say, there is always at least one stage of amplification ahead of it. In some cases it is located just ahead of the last stage. There are various arguments for and against the various possible locations, many of which are not of great importance. However, one point is of enough importance to require comment. Whenever the volume control is preceded by one or more stages of amplification, there is a possibility of a very heavy signal overloading the tube or tubes ahead of the control and thus causing distortion. This is avoided by the use of level controls, a topic that we shall take up a little later.

Selector Switch

A complete high fidelity system will include several different program sources: tuner, phonograph, tape recorder, microphone and perhaps television sound. There are but two methods of changing from one program source to another. One is to provide the control unit with a single input jack and to remove and reconnect cables each time a change is to be made. This is obviously a crude and clumsy method. A modern control unit has four, five, or more input jacks and the cables leading to the several program sources remain more or less permanently connected to their respective jacks. Then, by means of a suitable selector switch, the program source is connected to the appropriate point in the preamplifier. As some program sources deliver more signal than others the jacks intended for those sources are wired to a point in the circuit such that, let us say, only two stages are used for amplification. The jacks intended for high level sources (those delivering a stronger signal) are wired to a point such that all of the amplifier stages are used.

Level Controls

In a well-designed preamplifier there will be an individual level control for each program source. Thus, the intensity of signal from that source may be controlled independently of the main volume control. This design serves two purposes. First, the sometimes widely different signal levels of the various program sources may be brought to a common level so that when the selector switch is turned through the various positions there is no serious change in speaker output. Second, by means of the level control the intensity of signal from a given source may be brought down to a point where no overloading of early stages in the preamplifier can occur.

Tone Controls

Most preamplifier-control units are equipped with bass and treble tone controls. These are almost always continuously variable controls; that is to say, they operate like the ordinary volume control except that they vary the bass and treble characteristics instead of increasing and decreasing volume. Generally the controls will have a center position mark and frequently this position will be labeled "Flat." This means that with both controls set at the center points the response is essenially flat, within the capabilities of the amplifier. Turning the bass control in one direction provides a decrease in bass and turning it in the reverse direction gives an increase in bass. This applies as well to the treble control except that here only the upper end of the frequency range is affected.

Many authorities advocate using tone controls to compensate for differences in room acoustics, but there seems to be no reason why they cannot also be used to some extent in compensating for speaker deficiencies. Furthermore, they may often be used in conjunction with the equalizer controls. For instance, if the turnover control does not give just the right equalization for a particular recording, there is no reason why you cannot compensate by small adjustments of the bass tone control. As an example, suppose that your preamplifier is not equipped with equalization for a tape head. You might select the equalizer control position that gives the best result and then make further small adjustments by means of the tone controls. Finally, tone controls often solve a problem encountered in connection with making tape recordings. If you own a home-type tape recorder and wish to substitute your music system preamplifier for the preamplifier built into the recorder you will probably find that recordings made in this manner will be deficient in bass. This deficiency can often be corrected by advancing the bass tone control during the recording process. Of course, caution should be exercised, and several trials should be made to avoid advancing the bass so far that the tape is overloaded.

Output Jacks

Some preamplifier-control units have two output jacks. One is intended to be connected to the input of the main amplifier. The second jack is often a take-off point for signal just after the first stage in the unit and is intended to be used for tape recording. This output jack is to be connected to the input jack of the tape recorder. With such an arrangement it is possible to make tape recordings from any of the several program sources while listening to the music. However, in most cases the take-off point for this jack is such that all tone controls and equalizers are bypassed and therefore ineffective.

When buying a preamplifier, you should consider the following requirements: gain, frequency response, distortion level, hum and noise level, type of equalization, and method of connecting to other components. Almost all preamplifiers show good frequency response characteristics; some manufacturers advertise their product as having a flat response up to 30,000 cycles or more. Distortion levels should be lower than the main amplifier. Remember that any hum or noise present in the preamplifier output will be increased by the main amplifier. If the preamplifier is to be separate from the main amplifier, be sure that the latter is provided with a convenient arrangement for connection of the preamplifier voltage supply wires. If the unit includes equalizer circuits, try to select a model in which there is at least one stage of amplification ahead of the equalizer.

Preamplifier Construction

An easily constructed preamplifier is illustrated in Figure 8-2; it is intended for use with magnetic pickup cartridges having a low voltage output, such as the General Electric types RPX-040, RPX-041, and RPX-050.

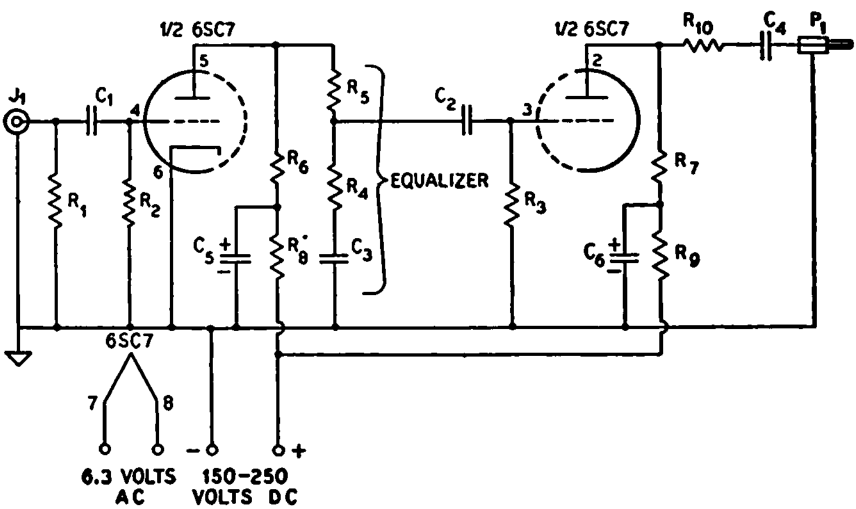


Fig. 8-2. Schematic diagram of preamplifier.

Record equalization in this unit is provided by resistors R_4 and R_5 and condenser C_3. This combination provides for a turnover point at 500 cycles per second. Although only one equalizer circuit is used, it will be fairly satisfactory for the average recording.

The high frequency response is controlled by resistor R_1; as its value is increased, the upper response limit is likewise increased. A value somewhere between 6000 and 50,000 ohms is recommended. However, for maximum high frequency response, R_1 may be omitted entirely.

The preamplifier employs one tube, a type 6SC7 dual triode. The cathode (pin 6), together with the grid and plate of one triode unit (pins 4 and 5), functions as the first amplifier stage, while the remaining electrodes comprise the second stage. Resistors R_6, R_7, R_8, and R_9, together with condensers C_5 and C_6, provide B supply filtering in addition to that used in the main amplifier. Operating voltages are taken from the main amplifier through a 4-wire cable. Two of these wires, of course, supply the heater voltage; the other two are connected, in the main amplifier, to a voltage source of 150 to 250 volts. A 24-inch length of shielded cable terminating in a plug connects the output of the preamplifier to the input jack of the main amplifier.

You will find that a chassis measuring 4 by 4 by 1½ inches will be large enough. If you have difficulty obtaining a chassis of suitable size in your area, one can easily be made from a sheet of aluminum measuring 7 by 7 inches. If you prefer, the preamplifier may be installed in a metal case, although this will not be necessary under average conditions.

LIST OF PARTS

Chassis, 4 x 4 x 1½ inches, steel or aluminum. (Substitute 7 x 7 inch sheet of 20 gauge aluminum if you make your own chassis)
Octal tube socket (Note: If preamplifier is to be installed in same cabinet as loudspeaker, the tube socket should be of the shock-mounted type.)
Shielded cable, 24 inches long
6—32 by ⅜-inch round-head machine screws
6—32 hexagon machine nuts
Tie strips
Stranded, insulated wire for voltage supply (10 feet)
Push back wire (15 feet)
Rosin core solder
J_1 standard phono jack
P_1 phono plug to match amplifier input jack

Resistors

R_1 6,000-ohm, ½-watt carbon (see text above concerning value of this resistor)
R_2 3.3-megohm, ½-watt carbon
R_3 3.3-megohm, ½-watt carbon
R_4 27,000-ohm, ½-watt carbon
R_5 200,000-ohm, ½-watt carbon
R_6 68,000-ohm, ½-watt carbon

R_7 33,000-ohm, ½-watt carbon
R_8 68,000-ohm, ½-watt carbon
R_9 33,000-ohm, ½-watt carbon
R_{10} 68,000-ohm, ½-watt carbon

Condensers

C_1 0.05 microfarad, 400-volt paper tubular
C_2 0.05 microfarad, 400-volt paper tubular
C_3 0.01 microfarad, 400-volt paper tubular
C_4 0.02 microfarad, 400-volt paper tubular
C_5, C_6 20 + 20 microfarad, 450-volt electrolytic, metal can type, vertical mounting

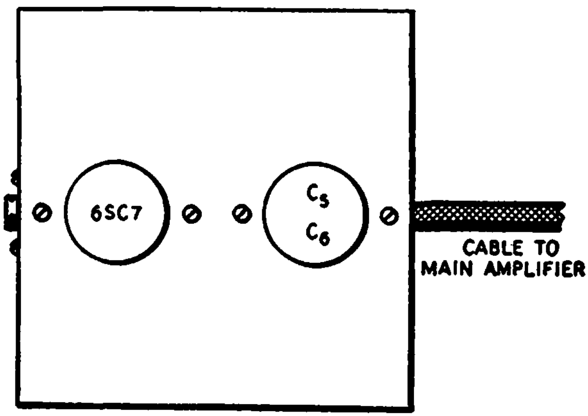


Fig. 8-3. Top view of chassis.

A suggested chassis layout is given in Figure 8-3. Layout is relatively unimportant, however, because few parts are used. You will generally find it convenient to place the input jack at one end of the chassis and the output cable at the opposite end, as shown in the illustration.

An improved version of the preamplifier just described is shown in the schematic drawing, Figure 8-4. In this unit five different equalizer circuits are provided; these are cut in or out of the circuit by means of a five-position, two-pole rotary switch.

In the first switch position, labeled A in the illustration, proper equalization is provided for records cut in accordance with NARTB standards. Switch position B provides equalization for Columbia records; position C is for all Victor 33 1/3-, 45-, and 78-rpm records; and position D is for London FFRR recordings. The last position E, gives a flat response with a 500-cycle-per-second turnover point. To provide space for the equalizer switch, it is best to use a chassis that is about 2 inches deep. If ¼-watt resistors and midget-type condensers are used in the equalizer circuits, a chassis measuring

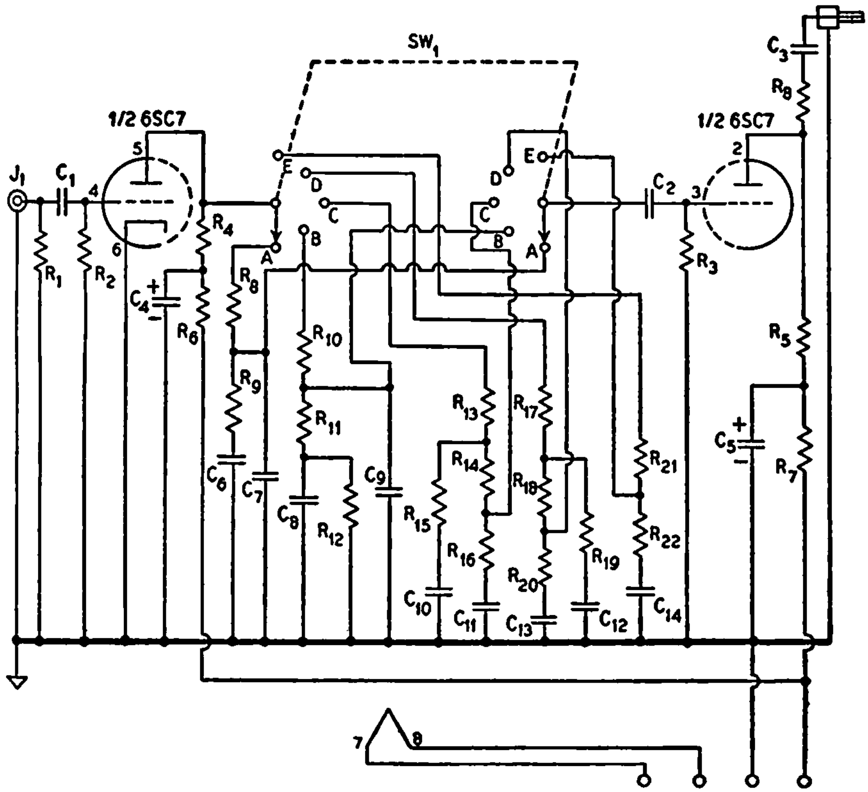


Fig. 8-4. Preamplifier with equalizer control.

4 by 4 inches should provide enough room for all parts. However, there is no objection to using a larger chassis if you prefer. A suggested layout is given in Figure 8-5, and you will note that the tube socket and filter condenser have been placed well to the rear of the chassis to provide space for the equalizer circuit components at the front.

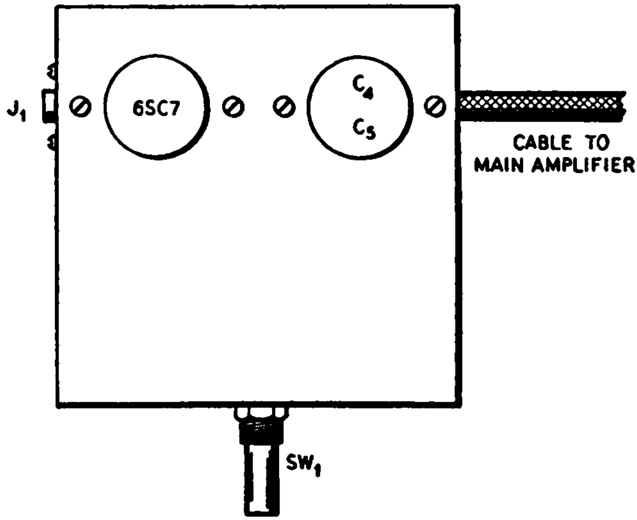


Fig. 8-5. Suggested layout for preamplifier.

LIST OF PARTS

Chassis 4 x 4 x 2 inches, steel or aluminum
Shielded cable 24 inches long
6—32 by ⅜-inch round-head machine screws
6—32 hexagon machine nuts
Tie strips
Stranded, insulated wire for voltage supply (10 feet)
Push back wire (15 feet)
Rosin core solder
J_1 standard phono jack
P_1 phono plug to match amplifier input jack
SW_1 two-pole, five-position rotary switch

Resistors

R_1 6000-ohm, ½-watt, carbon (value may be higher to increase high frequency response)
R_2 3.3-megohm, ½-watt, carbon
R_3 3.3-megohm, ½-watt, carbon
R_4 68,000-ohm, ½-watt, carbon
R_5 33,000-ohm, ½-watt, carbon
R_6 68,000-ohm, ½-watt, carbon
R_7 33,000-ohm, ½-watt, carbon
R_8 1-megohm, ¼-watt, carbon
R_9 33,000-ohm, ¼-watt, carbon
R_{10} 1-megohm, ¼-watt, carbon
R_{11} 33,000-ohm, ¼-watt, carbon
R_{12} 150,000-ohm, ¼-watt, carbon
R_{13} 1-megohm, ¼-watt, carbon
R_{14} 18,000-ohm, ¼-watt, carbon

R_{15} 33,000-ohm, ¼-watt, carbon
R_{16} 33,000-ohm, ¼-watt, carbon
R_{17} 1.3-megohm, ¼-watt, carbon
R_{18}, R_{19}, R_{20} 39,000-ohm, ¼-watt, carbon
R_{21} 800,000-ohm, ¼-watt, carbon
R_{22} 33,000-ohm, ¼-watt, carbon

Condensers (μf = *microfarad*)

C_1 0.05 μf, 400-volt paper tubular
C_2 0.05 μf, 400-volt paper tubular
C_3 0.02 μf, 400-volt paper tubular
C_4, C_5 20 + 20 μf, 450-volt electrolytic, metal can, vertical mounting
C_6 0.01 μf, 200-volt, midget paper tubular
C_7 0.003 μf, 200-volt, midget paper tubular
C_8 0.01 μf, 200-volt midget paper tubular
C_9 0.003 μf, 200-volt midget paper tubular
C_{10} 0.006 μf, 200-volt midget paper tubular
C_{11} 0.001 μf, 200-volt midget paper tubular
C_{12} 0.01 μf, 200-volt midget paper tubular
C_{13} 0.00025 μf, 200-volt midget paper tubular
C_{14} 0.01 μf, 200-volt midget paper tubular

9 Planning and Installing Your Home Music System

It is safe to say that a majority of high fidelity enthusiasts buy separate components (amplifier, speaker, and so forth) and perform at least part of the work of installation themselves. This statement is based upon a sales comparison of components against complete systems, installed in cabinets. It would seem, then, that the acquisition of a music system calls for considerable planning. The design, physical size, and other features of the various components may have some bearing upon the location of the unit and the space available for it.

If you have the inclination, time, and skill we strongly recommend that you do as much of the installation work as possible; you will not only save money but will no doubt devote much more care to the work than will the average installer. Furthermore, it is quite possible that you have tentatively decided upon a location for your equipment; perhaps you have decided that an unused credenza or set of wall shelves would be ideal. If you are really determined to make such a conversion, you will, beyond question, succeed if the plan is at all feasible. The average disinterested worker often attempts to discourage the customer by pointing out real or imagined difficulties, when actually such difficulties are not insurmountable if one cares to take the trouble to attempt the job.

Another thing that should be emphasized here is the possibility of either saving money or buying better equipment by spending less on cabinet work. Cabinets represent a very considerable part of the cost in a home music installation; you have only to glance at the advertisements in any radio or high fidelity magazine to discover that a hundred dollars does not go very far. Perhaps you don't have the skill required to build an elaborate blond or mahogany cabinet for your equipment, but comparatively little skill is required to install the components in a set of bookshelves or in a closet. And, while a simple painted or enameled enclosure certainly cannot be compared with a finely finished piece of furniture, it does serve the same purpose at a fraction of the cost. If we critically examine the mass-produced cabinets in the lower-priced fields, we discover that the careful home craftsman can usually do a better job of construction.

In this chapter we shall discuss the planning of a home music system and shall offer suggestions on buying with a view to your particular circumstances. Then we shall take up the matter of utilizing existing space for housing your components. This will also include some remarks for the benefit of those who find it economically impossible to buy all components at one time. Finally, there will be some suggestions for improving the performance of existing radio-phonograph combinations.

If you live close to one of the larger radio supply houses or high fidelity show rooms, you have an excellent opportunity to select your components at your leisure. Many of these firms are in a position to supply cabinets suitable for housing the equipment of your choice; however, we suggest that you investigate other dealers in cabinets. It may be that a firm specializing in cabinets can supply a wider variety, perhaps at lower prices. The classified telephone directory is an excellent source of information.

When you are ready to purchase equipment, you have a choice of selecting each item separately or as part of a "package" deal. Most large dealers make special offers of complete ensembles, and in most cases a small saving is realized by buying in this manner. Of course, you do not have the opportunity of selecting each component and comparing it with others in its price range. On the other hand, this method of purchasing may be advantageous for those who are not interested in the technical phases of high fidelity; such equipment groupings almost always are made up of leading makes of equipment and often the individual units are the best obtainable within the particular price range.

If you decide to select each item separately, we suggest that you begin with the loudspeaker. It is a mistake to begin shopping with the determination to spend no more than a fixed amount for a speaker. Perhaps after hearing the best, you may have to settle for something less, but at least you will then know how far short of perfection your speaker is. Visit a showroom and listen to as many speakers as possible. In making a comparison the speakers should operate from the same amplifier and other components. When you have decided upon one speaker, you will then know whether it falls within your budget. If not, listen to the second or third best and compare it with your first selection.

The next step is the selection of an amplifier (and a preamplifier, if one is needed). Ask to hear the speaker that you have already decided upon connected to a variety of amplifiers. If you have trouble making a selection, a comparative test (by switching the speaker from one amplifier to another) will help you decide.

When you have selected a loudspeaker and an amplifier, the choice of record player and other components will be merely a matter of critical listening, plus an evaluation of the features of the various components. For instance, if the amplifier you have selected is equipped with a preamplifier and tone controls, it would be unwise to select a tuner also having those features.

You may be one of those who want a high fidelity system, but find yourself unable for financial reasons to buy all the components at one time. It is possible to buy the equipment piecemeal if you are willing to make some temporary sacrifices. In such a case, we recommend that you buy the loudspeaker and enclosure first. If you are prepared to build your own enclosure, so much the better. Instructions for building various types of speaker enclosures can be found in Chapter 3.

By connecting the loudspeaker to the audio system of an ordinary radio or phonograph, you will enjoy a marked improvement in tone quality, although you will not have a true high fidelity system. If you happen to have a radio-phonograph of quite recent manufacture, there is a fair chance that it will be equipped with a record changer using a magnetic type of pickup cartridge and the results will then be even better. If the set to be used is a large one, you will have to solve the problem of finding space for both the radio and the loudspeaker enclosure, but this difficulty will be only temporary.

Later on, when you are ready to buy an amplifier, you may decide whether or not to use the record changer in the radio-phonograph combination as a permanent component of your system. This will depend upon the type and quality of the changer. Chapter 4 supplies additional information that will help you decide. Instructions for connecting a substitute loudspeaker to a radio-phonograph will be found in the final section of this chapter.

Whether you select all components at one time or buy individual items one at a time, eventually you will have to make plans for the location, housing, and installation of your system. In making these plans, bear in mind that it is not good practice to mount the loudspeaker in the same enclosure with other components. The one exception to this rule is for a closet installation.

As mentioned earlier in this book the ideal location for the loudspeaker is in a corner of the room; if you find this is impossible, then the second best position is at one end of a long, narrow room. In a room of this shape, place the speaker at one of the shorter walls. If the room is square, or nearly so, the speaker may be placed at any one of the four sides, but you will have to take into consideration the seating arrangements; obviously the loudspeaker should face toward the audience, as nearly as possible.

After locating the loudspeaker, proceed to select a place for the remaining components. Keep in mind that the speaker may be placed at a considerable distance from the amplifier, but the amplifier and all other components must be grouped closely together. Unless the tuner and preamplifier are of special design, the maximum permissible distance between either of these and the main amplifier is 2 feet. The distance between record player and amplifier should never exceed that distance.

All connections between record player and amplifier, between tuner and amplifier, or between preamplifier and main amplifier must be of shielded wire. Most units come equipped with the necessary cable, and in most instances the cable terminates in a standard plug.

All wiring used to conduct signals from program sources (tuner, tape recorder, phonograph or television receiver) to the preamplifier and from the preamplifier to the main amplifier must be of shielded cable. Any attempt to use unshielded wire or to extend the shielded cable beyond 2 feet or so will usually result in excessive hum. The only exception to this limit for length of shielded cable occurs when the final stage in the preamplifier is a cathode follower. In such cases, a length of 12 feet or more is permissible. A number of factory-built preamplifiers and a few kits use a cathode follower and then the instruction booklet or sheet accompanying the unit will advise you concerning the permissible length of cable. The wiring from speaker to main amplifier should not be shielded.

Signal input and output jacks on practically all audio units are now of the RETMA type. The companion plug for this type of jack is illustrated in Figure 9-2A. All units, whether factory-wired or in kit form will, of course be equipped with the necessary jacks. Some kits also include plugs and shielded cable, but this is usually not true of factory-wired units. While shielded cables of various types and various lengths may be purchased at moderate cost, you may prefer to make up your own. In that case, follow the directions given below.

The shielded cable consists, as shown in Figure 9-1, of an inner stranded wire with a covering of insulation. Over this insulation there is an outer shield, made of woven metallic braid. Cut the cable to the desired length allowing an extra 1 to 2 inches at each end for connections. The shield is quite flexible and can be pushed back over the insulation for a considerable distance, exposing the insulation. Push it back for 2 or 3 inches, then remove the insulation from the wire for a distance of an inch, being careful not to

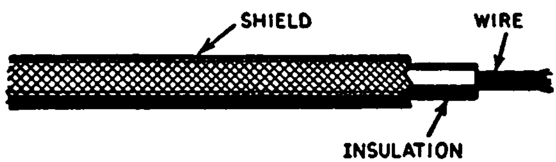


Fig. 9-1. Shielded wire.

nick any of the wire strands. The stranded wire is the principal conductor while the outer shield is the return conductor. The inner conductor is often referred to as the "high" side or "hot" side of the circuit, while the outer shield is called the "ground" side. In most cases the shield should be "grounded" or connected to the metal chassis of the units at both ends of the shield. (There are a few exceptions to this rule.)

Notice that the plug consists of a hollow center pin about 3/32 inch in diameter; the "hot" side of the cable, or inner wire, is to be connected to this pin. The other terminal of the plug is the bell-shaped shell at the rear of the plug; the shield is to be connected to this shell.

Twist the strands of the inner wire together and insert the bared end into the hollow pin, as shown in Figure 9-2B. Push the wire into the plug until you feel the insulation touch the back end of the pin. It is very important that no bare wire be exposed inside the shell of the plug, otherwise there is danger of contact between the inner wire and the shell of the plug, thus causing a short circuit and rendering the cable useless.

Now clip off the protruding wire so that no more than 1/32 inch extends beyond the end of the pin. Heat the pin with a well-tinned soldering iron and flow solder into the open end of the pin. The end may now be smoothed and rounded off with a file or sandpaper. Push the outer shield back into place until it comes over the back

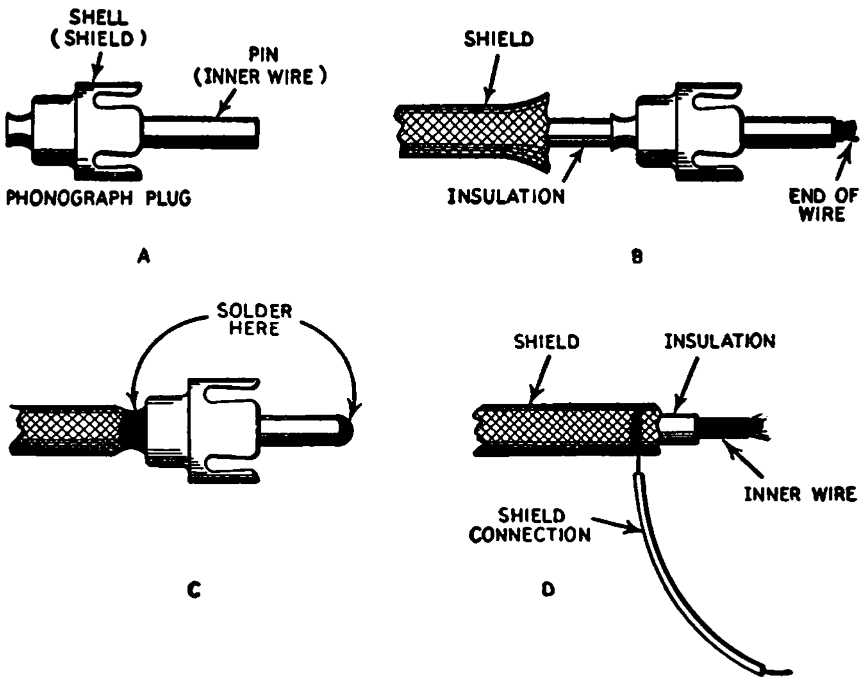


Fig. 9-2

end of the plug shell, as in Figure 9-2C. Solder the shield to the shell.

If you are making up a shielded cable for connecting the output jack of a preamplifier to the input jack of the main amplifier, you will require a plug at both ends of the cable. In that case, the remaining end of the cable is to be treated exactly as described above. In some cases (for example, to connect a record player to a preamplifier) you will need a plug at one end only; the other end of the cable will have its two conductors connected to a pair of terminals. The simplest method of preparing the cable is shown in Figure 9-2D. After exposing the insulation and inner conductor for the required distance, solder a short length of wire (stranded preferred) to the end of the shield. In doing so, be very careful not to apply too much heat, otherwise the insulation may be damaged. A much neater but slightly more difficult method of preparing the cable eliminates the need for soldering a connecting wire to the shield; the shield itself is used as the connecting wire. Push the shield back over the insulation for a distance of three inches or so. Insert a scriber, icepick or other sharp-pointed tool between two of the strands of the woven shield; be careful not to break any strands. By working carefully with the tool, you will produce an opening be-

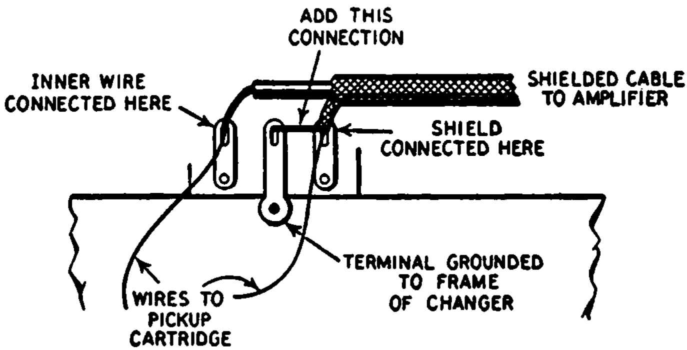


Fig. 9-3

tween the strands through which the wire with its covering of insulation may be drawn. Now twist the woven strands of the shield together, forming a "pigtail" which may be used for connecting the shield to a terminal.

Figure 9-3 shows how a shielded cable is to be connected to the terminals underneath the bed plate of a record changer. In this particular case (a popular make of changer) there are three terminals. One of these is riveted—and therefore electrically connected—to the changer frame. It is known as a ground connection. The inner wire of the cable is to be connected to one of the ungrounded terminals and the cable shield to the remaining ungrounded terminal. In many cases the equipment will operate satisfactorily and with no excessive hum with the connections made in this way. However, in a few instances it may be necessary to connect the cable shield to the changer frame; this is done by adding the short connection shown in the sketch, from the grounded terminal to the shield terminal.

There are many ways to fulfill the requirement of the loudspeaker in one enclosure and all other components, grouped together, in a separate location. One that may appeal to amateur woodworkers is shown in Figure 9-4. Twin cabinets are used and may either be painted or finished in varnish or shellac. The speaker cabinet is

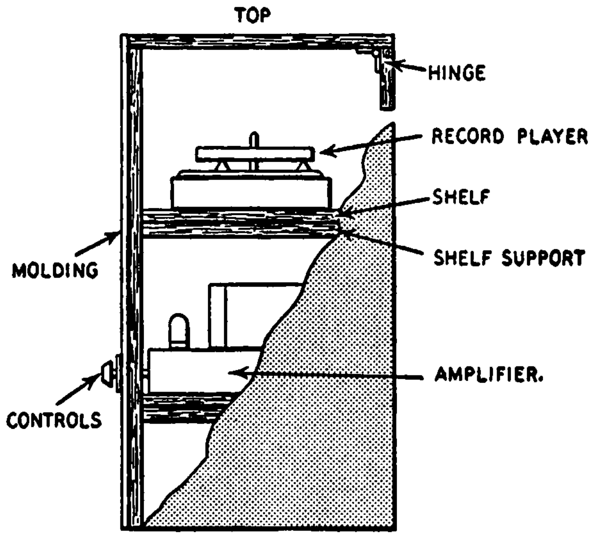


Fig. 9-4. Cut-away view of cabinet.

identical with one described in Chapter 3. The remaining cabinet houses the record changer and amplifier in this particular instance, although other arrangements are possible.

The construction of the record player-amplifier cabinet is the same as for the speaker enclosure, except for one or two details. The top of the cabinet is not fastened rigidly in place, but is hinged to provide access to the record player, which rests on a shelf as shown in Figure 9-4. The distance between the top of the cabinet and the shelf will depend upon the type of changer used, but the average changer will require from 8 to 10 inches clearance.

The amplifier may either rest on the bottom of the cabinet or on a second shelf, below the changer. The second method is illustrated in the drawing. For supporting the shelf, or shelves, use 1 by 1 inch strips running the full depth of the cabinet. The shelf rests on these strips and is fastened in place with woodscrews.

For the cabinet top you will need a piano hinge of suitable size, depending upon the width of the cabinet. It is assumed that the amplifier-record changer cabinet will have no back panel. In that case, you will have to install a support for the lid hinge; this may either be a piece of 1½ by 1½ inch lumber fastened between the

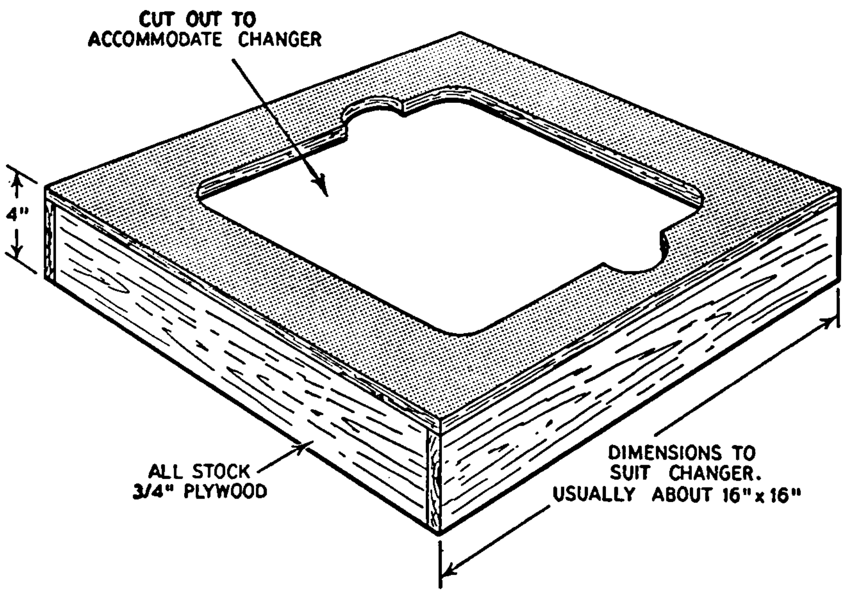


Fig. 9-5. Base for record changer.

cabinet sides, or a piece of plywood large enough to cover the entire record player compartment. The first method is preferable, as it affords more adequate ventilation, but the latter is likely to provide more rigid construction. You will need at least one, possibly two, lid supports: these can be bought from any large firm specializing in cabinet hardware, but might not be stocked by the average hardware store. Although more expensive, the automatic spring-type lid support is superior in operation to the ordinary type.

In almost every instance, the record changer will have to be mounted in a box, constructed for the purpose. The details of such a base, or box, are shown in Figure 9-5. Dimensions will depend, of course, upon the individual changer; in some cases, you will receive, with the changer, a template for cutting the top of the box.

With the amplifier mounted on a shelf or on the bottom of the cabinet, you will have to bring the controls out through the front panel. This can be done by drilling holes in appropriate places on

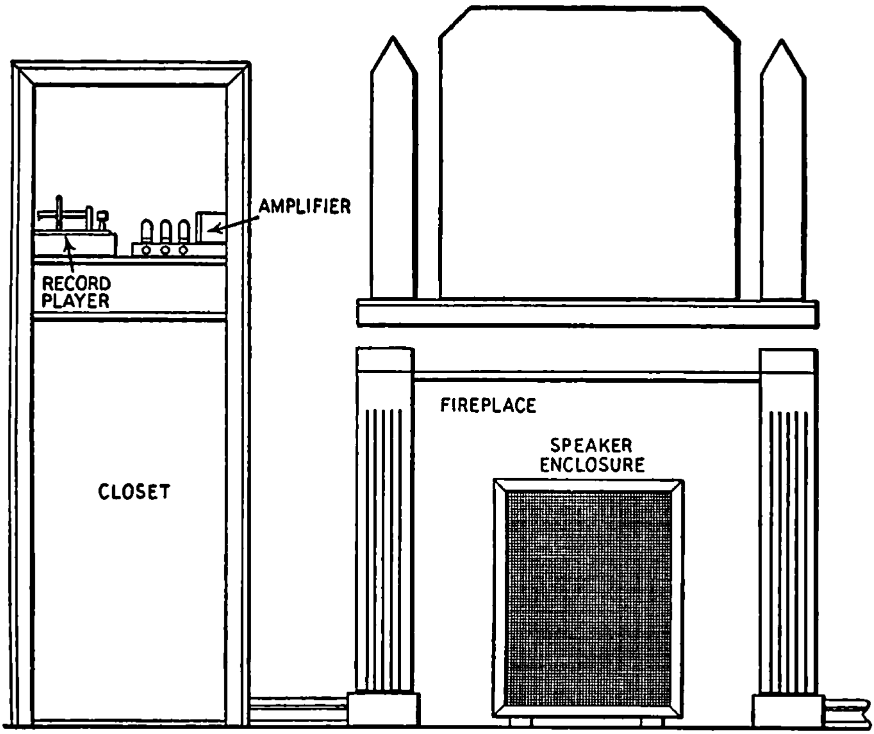


Fig. 9-6. Installation in closet and utilizing a fireplace.

the panel to accommodate the control shafts. Remove the control knobs, place the amplifier in position with the shafts protruding through the holes, then replace the knobs. You may want to mount the control dial plates on the front panel, but find that they are permanently affixed to the amplifier apron. In that event, substitute dial plates can be bought at any large radio supply store for a few cents apiece. To prevent the amplifier from shifting position, you may fasten it in place with two or more screws or bolts passing through holes in the shelf or cabinet bottom. Many amplifiers have threaded holes in the underside of the chassis for this purpose.

The amplifier can be installed so that the controls are concealed, yet accessible, by adding a hinged, drop-type door in the front of the cabinet. You will need an additional piece of plywood for the door however, since it is not possible to cut the opening without drilling one or more rather large holes for the compass saw, and of

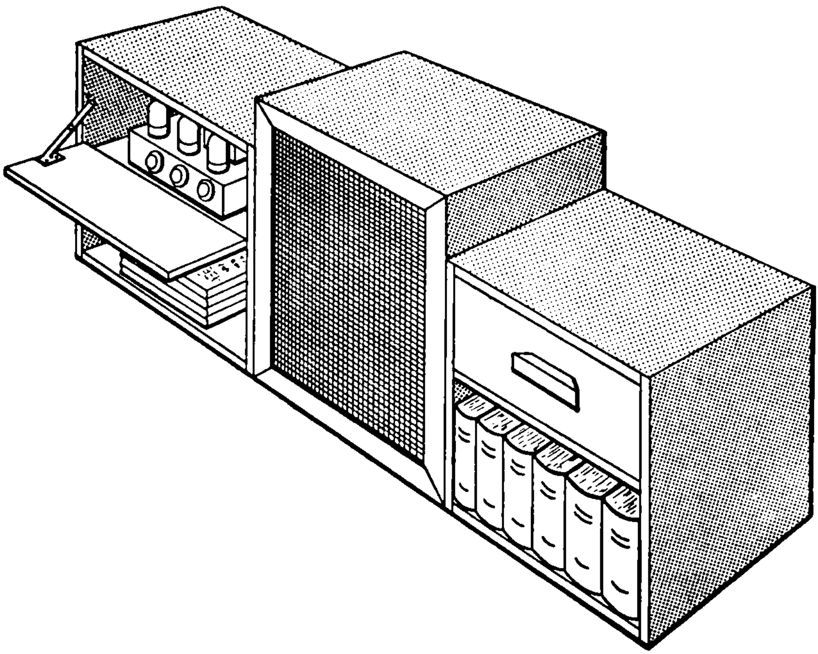


Fig. 9-7

course, this renders the cut-out piece unfit for further use. You have a choice of a piano hinge or concealed type hinges for the door. It is usually best to fasten framing strips around the edges of the door opening, on the inside of the front panel. These strips will provide additional reinforcement and will serve to support the door hinge and fastener. For the latter, use a bullet-type catch, set in the upper edges of door and door frame.

It is often possible to gain valuable suggestions from existing installations; the one shown in Figure 9-6 is a good example. The owner of this system took advantage of a fireplace, adjoining a closet. The fireplace was of the purely ornamental type but was built into the wall. As you will note in the drawing, the speaker enclosure was located in the fireplace and therefore occupies no floor space needed for other purposes. In addition, it is possible that the sloping walls of the fireplace act as the sides of a horn. The remaining components of the system (in this instance they comprise only the amplifier and record changer) were located on

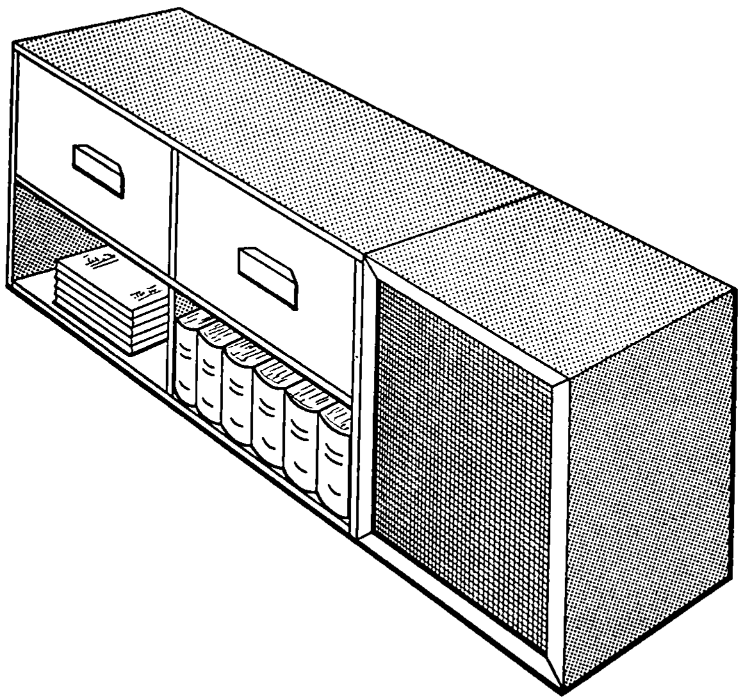


Fig. 9-8

a shelf in the adjoining closet. The only cabinet work needed was the construction of the speaker enclosure and this is a simple, painted affair of the bass reflex type. If you are contemplating an installation similar to this one, the speaker need not be adjacent to the closet; in fact, it may be located if necessary, across the room. And, if you do not happen to have an ornamental fireplace, the speaker enclosure may be placed wherever convenient.

Another actual installation is illustrated in Figure 9-7. Here, the builder uses two cabinets, resembling bookcases, one on either side of the speaker enclosure. In fact, bookshelves may be used for the purpose, provided that they are deep enough to accommodate the apparatus. The tuner, amplifier, and record changer may be installed on open shelves, or you may wish to conceal them, as shown in the illustration. Further details for installing doors of the drop type were given earlier in this chapter.

Another version of the cabinet just described is shown in Figure 9-8. In this installation both the equipment cabinets are placed on

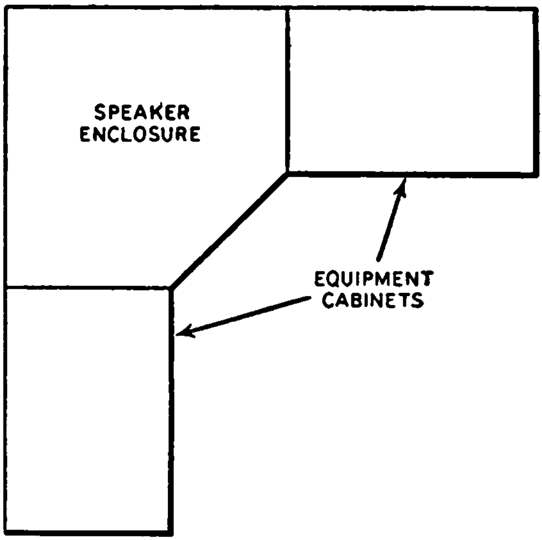


Fig. 9-9. Top view of corner installation.

the same side of the speaker enclosure, and all three units are the same height. This may be further modified for a corner installation by shaping the speaker enclosure as shown in the plan view, Figure 9-9.

In many other installations, existing furniture has been modified to serve as a housing for the equipment. Generally speaking, it is not good practice to use a credenza, step table, or the like as a speaker enclosure, unless the piece is quite large and of unusually heavy construction. Most present-day furniture is built up of panels consisting of ¼-inch or ⅜-inch plywood framed by heavier members. While this kind of construction is satisfactory when the piece is used as originally intended, it is not sturdy enough for speaker enclosures.

An acceptable compromise is often reached by housing the equipment in a suitable piece of furniture and building or buying a separate speaker enclosure. However, in making an installation of this kind, be sure that the components will receive adequate ventilation. This applies especially to the amplifier which must dissipate considerable heat.

We may summarize the points to be observed in making a custom installation as follows:

1. The components must be installed in such a way that all are conveniently accessible. This applies to both routine operation and servicing.
2. All components must receive proper ventilation.
3. The loudspeaker should not be installed in the same cabinet with other components.
4. As a rule, furniture pieces do not make good speaker enclosures.
5. The loudspeaker may be located 25 feet or more from the amplifier, but the tuner, record player, and tape recorder should be placed within 2 feet of the amplifier.
6. Shielded wire must be used for all signal wiring between program sources and amplifier and between preamplifier and main amplifier.

One more point in connection with installations: the record player, tuner, and tape recorder (also the preamplifier, if it has its own power supply) all require power from the supply line. Exceptions to this rule are the preamplifier and tuner designed to take their operating voltages from the amplifier power supply. As a result, in the average installation of tuner, amplifier, and record player, electrical outlets must be provided for three attachment plugs. Many amplifiers are equipped with a receptacle for the phonograph motor line plug, but unless this happens to be a duplex receptacle there is still no outlet for the tuner; only in rare cases will the amplifier have a duplex, or two-way, receptacle.

There are two methods of solving this problem; in either case, the use of simple wiring devices available at any hardware, electrical supply, or ten cent store will provide additional outlets.

Perhaps the simplest, but not the most satisfactory method, is to use a multiple outlet of the type intended for fastening to a base molding. The objection to this method is that the power to the record player, tuner, or other program source is not under the control of the amplifier on-off switch and it will be necessary to turn the program source on separately. Usually, it is desirable to turn on all power by means of a single switch at the amplifier.

To overcome the objection just mentioned, it will be necessary

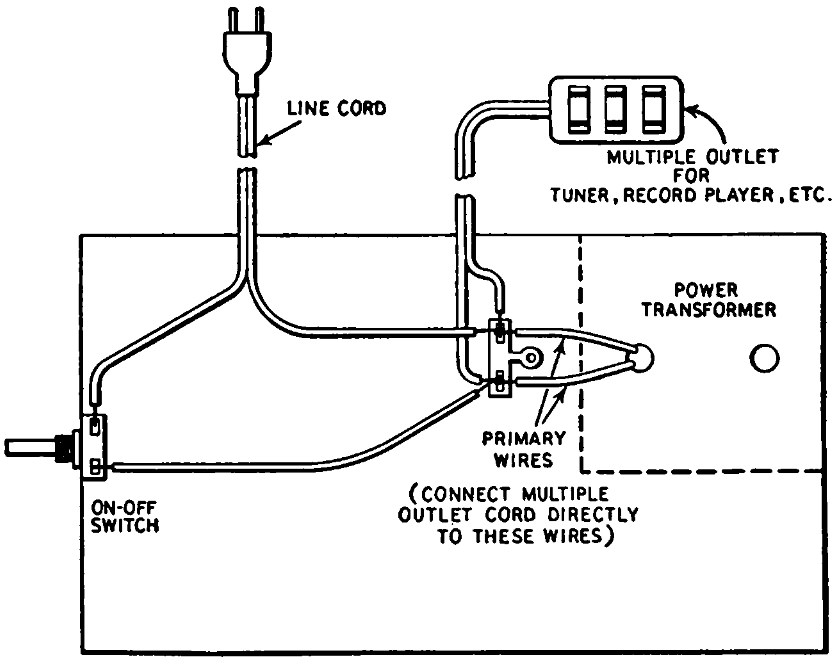


Fig. 9-10

for you to connect the auxiliary outlets to the supply line terminals in the amplifier. The connection points are shown in the diagram, Figure 9-10. Notice that the extension cord running to the auxiliary outlets must be connected to the primary terminals of the power transformer. If you connect the wires at the wrong side of the on-off switch, you will get power at the new outlets but it will not be controlled by the amplifier switch. The extension cord need be only 2 feet or so in length and can be brought out through a hole in the amplifier chassis apron. Almost any type of two- or three-way outlet may be used provided it is completely enclosed and has no exposed terminal screws.

As we have mentioned before, it is possible to make several changes in a radio-phonograph combination to improve the quality of reproduction. However, we do not advocate such changes in all cases; but, if the unit is of recent manufacture (not more than five years old), if the audio system employs a push-pull output stage, and if the radio portion is designed to receive both AM and FM broadcasts, it will usually be worth while. To determine whether or not your set has a push-pull output stage, you might examine the interior of the cabinet for a chart showing the locations and types of all tubes used. If you find that there are two identical tubes placed rather close together, you may suspect that a push-pull stage is used, but you still have no positive indication. But if both tubes are of one of the types listed here, you may be certain that there is a push-pull stage: 6K6, 6V6, 6F6, 6L6, 6AQ5, 6AK6, 7B5, 7C5, 6G6, 6Y6G.

If you find a single tube of one of the following types, you have a push-pull output stage: 6AS7G, 6AD7G (with one 6F6G).

The next step is to examine the loudspeaker to determine whether it is of the permanent magnet dynamic type or the older electrodynamic type. A permanent magnet speaker will have only two wire connections between it and the radio chassis, but an electrodynamic type will have four. Furthermore, the electrodynamic speaker will have a large coil in place of the permanent magnet.

If you have an electrodynamic speaker, a substitution is still possible but rather difficult, and you are not advised to undertake it. In most cases, the large speaker coil acts as a filter choke in the receiver power supply; to make the substitution you would not only have to replace the speaker but also have to substitute a suitable choke for the coil. Assuming then, that your speaker is of the permanent magnet type, the next step is the selection of a suitable replacement. We do not recommend that you use an exceptionally high quality speaker for this purpose, since the final result will not be high fidelity reproduction. Any single unit speaker of the extended range type will be satisfactory and will give a marked improvement over the original. Such speakers are available at prices ranging from \$12 to \$20, depending upon the size. Be sure that the voice coil impedance of the new speaker is the same as the original.

It will be more convenient if the new speaker is the same diameter as the original, for then it will not be necessary to enlarge the speaker opening. If the original speaker is fastened to the baffle with woodscrews, replacement is then merely a matter of removing the original speaker and mounting the new one in the same manner. However, in the majority of cases, the speaker will be mounted on the baffle by means of machine screws and nuts and, since the spacing of mounting holes is not the same for all speakers, you may have to remove the baffle board from the cabinet. Generally, the baffle will be fastened in the cabinet with woodscrews.

With the baffle removed, you will have to take off the grill cloth before the speaker mounting screws can be taken out and new screw holes drilled. When the grill cloth is replaced, it must be stretched fairly tightly over the front of the baffle to give a neat appearance. If you have decided to install a speaker larger than the original, the opening may be enlarged with a compass saw after the grill cloth is removed.

The method of connecting the speaker to the set depends upon circumstances. Some sets use a plug which is wired to the speaker and is inserted in a socket on the rear of the chassis; in other sets, there is a two-piece separable connector. In either case, disconnect the two wires from the speaker voice coil terminals and solder them to the terminals on the new speaker without making any changes in the plug or connector. In many late model sets there are connector clips attached to the speaker voice coil terminals; they resemble the clips used on phonograph pickup cartridges, but are larger. With such an arrangement it is only necessary to remove the clips from the original speaker and slip them over the terminals of the new one.

Quite a few late model sets use a record changer equipped with a magnetic pickup cartridge, and under such circumstances very little will be gained by replacing the changer unless you are dissatisfied with the original or it has developed some defect. If replacement must be made, it is possible that the new changer will fit the original mounting; otherwise you will have to make a new mounting board for it. There is very little difference in the dimensions of various popular record changers and it is likely that the replacement will fit into the available space in all cases. However, it is well to take measurements before buying, just to be sure; when taking such measurements don't forget to determine the over-all height of the new changer.

If the original changer was equipped with a crystal cartridge, you may want to take advantage of the wider range of a magnetic type. Generally you will have to replace the entire changer, but in a few types it is possible to obtain a new pickup arm designed for a magnetic cartridge; this will apply only to changers of very recent manufacture. However, you are advised to write to the manufacturer for this information before investing in a new changer. The Webster model 121 changer, for example, can be converted by installing a new pickup arm at a cost of about $3. A General Electric triple-play cartridge for this unit sells for about $14 or less.

Whether you convert from a crystal cartridge to a magnetic type or replace the entire changer, a preamplifier must be added if the original cartridge is of the crystal type. There is no need to buy an elaborate, expensive preamplifier for an installation of this kind. The simple type selling for $10 or $12 will do very well. If you are interested in building your own, two types were described in Chapter 8.

You will have little difficulty in installing the preamplifier. The problem of space may develop, although this is not likely in the average large radio-phonograph combination. The preamplifier will be equipped with a jack and a cable terminating in a plug. The plug at the end of the record changer cable is to be inserted into the preamplifier jack. The plug at the end of the preamplifier cable is then inserted into the jack on the radio chassis.

You will now have to supply operating voltages to the preamplifier. These can conveniently be taken from the socket terminals of one of the output tubes in the radio. The preamplifier heater wires must be connected to the heater terminals of the socket. The preamplifier negative B supply wire may be connected directly to the radio chassis. The positive wire can be connected to any point that is at a positive potential of 150 to 250 volts; a convenient point is the screen terminal of one of the power output tubes. If you feel that you have not had enough experience to make the necessary voltage supply connections, you might consult your local radio dealer. It is quite possible that, if you explain exactly what you are trying to do and give him the type number of the power output tubes, his serviceman can make up an adapter that will eliminate the job of making any connections inside the radio receiver. Such an adapter should cost no more than two or three dollars to make. Of course, if the preamplifier you select is self-powered, no connections need be made to the radio at all.

10 Stereophonic High Fidelity Sound Systems

It is always helpful when attempting to understand a new application of science to consult the dictionary for definitions of unfamiliar words. The word "stereophonic" is relatively new to our language; it was added just a few years ago when the first stereophonic disc recordings were introduced. Therefore, you will probably not find it in your dictionary unless you have a new edition.

However, even an old edition will tell you that the prefix "stereo" comes from the Greek "stereos," meaning solid. The word "phonics" means the science of sound or acoustics. When we put the two together we get "solid sound," which, at first glance, means very little. A little more reading of the dictionary will bring to light the word "stereoscope" which is defined by the Webster's Student's Dictionary as follows: "An optical instrument with two eyeglasses producing the effect of solidity or depth by combining the images of two pictures taken from points of view a little way apart."

When we apply the above definition to stereophonic, we find that it is a method of producing realism in sound reproduction by combining two sound images from points of view a little way apart.

There are a number of ways of explaining the operation of stereo sound systems and in the course of our discussion we shall make use of two slightly different explanations.

Stereo or stereophonic sound systems may be said to represent an attempt to add realism to the reproduction of sound by providing two differing perspectives. The basic idea may be explained in this way:

Suppose we hear the sound of an airplane. We are able almost instantly to determine the direction of the sound because we are listening with two ears. One ear, let us say the right one, receives a certain impression or perspective of the sound, as in Figure 10-1. This impression is determined largely by the distance between the ear and the airplane. The left ear, however, is either slightly farther from or slightly closer to the airplane and therefore receives a slightly different impression or perspective because the sound waves, in reaching it, travel over a slightly longer or slightly shorter path.

The stereo method of reproduction adds more than mere direction to reproduced music. We may make a comparison between ordinary color slides and stereoscopic color slides. An ordinary slide gives a flat, two-dimensional impression, totally lacking in depth. On the other hand, because stereoscopic slides really consist of two separate views photographed from slightly different points of view and then combined in the viewer or on the screen, we receive a three-dimensional impression of depth; objects appear to "stand out" from the background.

When we listen to a loudspeaker reproducing music in the ordinary way we are listening to a monaural source of music with our binaural listening apparatus: our ears. The impression we receive is therefore flat and lacks depth, even though, in a good system, it may contain a wealth of detail.

In addition to the foregoing, stereo also provides an impression of direction and motion that is not present in a monaural sound system. In recent motion pictures using multi-track sound we receive a clear aural impression of direction and motion as a train or other moving object passes across the screen because the apparent source of the sound shifts from one loudspeaker to another. Some demonstration stereo recordings on tape and disc include a ping-pong game. In such recordings we may follow the motion of the ball as it issues, alternately, from two separated loudspeaker systems.

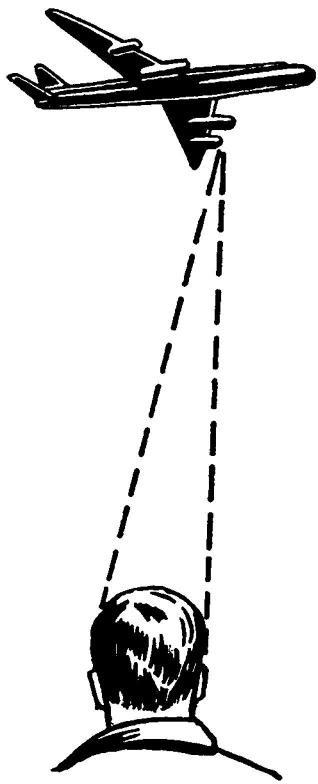


Fig. 10-1. Sound impressions on right and left ears.

From the above explanation you might conclude that stereo merely adds direction and the impression of motion to high fidelity.

While such a conclusion might be justified, it will give you only part of the story.

With modern high fidelity monaural equipment it is possible to reproduce music so close to the original that even a critical listener

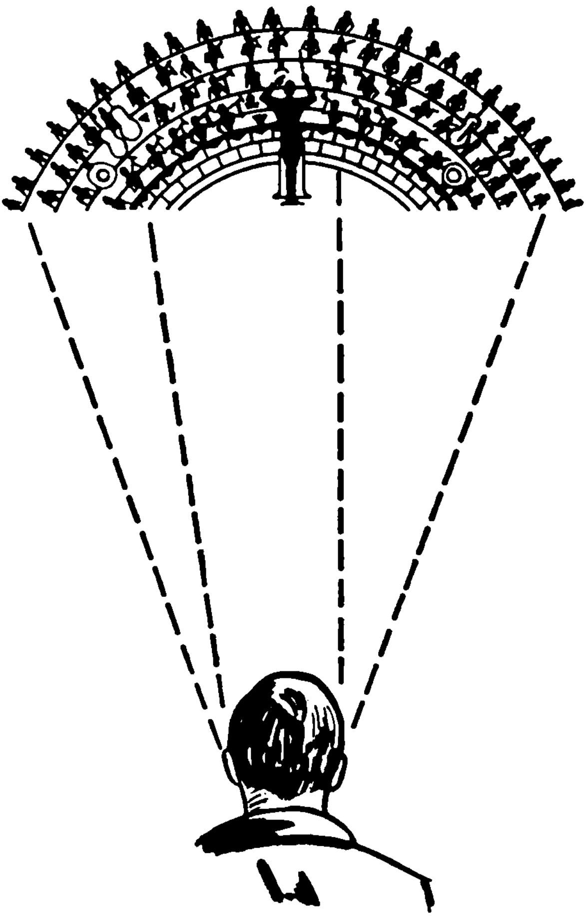


Fig. 10-2. Divergent paths of sound from a large orchestra.

would have to make comparison tests to determine which is the original and which is the reproduction, insofar as frequency range, lack of distortion and definition are concerned. However, something is lacking in realism or "presence." In other words, we still do not receive the impression that we are listening to the original soloist or group. This is not too evident in the case of a soloist, but let us see what happens when we try to reproduce orchestral music.

In a concert hall the sounds from the various instruments reach the ear by way of a number of different paths, and in the case of a large orchestra these paths may be widely divergent, as in Figure 10-2. Thus the listener receives a number of different sound impressions.

During reproduction of the music the situation is entirely different. In place of the many different and divergent sources of sound there is now a single source, the loudspeaker. During the recording process (if the music is on tape or disc) or during transmission (if the music is broadcast) all of the sounds from the widely divergent sources were combined and now issue as a single highly complex sound from a loudspeaker or loudspeaker system. Earlier we found that the ear is able to distinguish sounds partly because of location and direction. Clearly then, in ordinary high fidelity systems we are limited to identification of sounds solely because of differences in tone and we are not able to identify them because of differences in direction and location. It is this lack that good stereophonic high fidelity seeks to overcome.

Suppose we now add a second speaker or speaker system, as in Figure 10-3. Bear in mind that the second speaker receives the identical sounds as the first; in other words, the two speakers are merely connected in series or parallel to a single amplifier. If the two speakers are separated by several feet it is likely that we will succeed in spreading the sound over a wider area but we will still not have improved matters much because both speakers radiate the same impression or "point of view" of the original music. Nevertheless this "pseudo-stereo" arrangement has actually been offered as stereo. True stereo requires that each of the two (or three) speaker systems radiate slightly different impressions of the music, just as the two slightly different views in a stereoscope or in stereoscopic slides show the scene as viewed from two slightly different points of view.

At this point we should consider several misconceptions that have arisen concerning stereo. One has already been considered: the idea that stereo can be accomplished simply by adding another speaker. Another incorrect impression is that in stereo systems all of the lows issue from one speaker while all of the middle range and high notes are radiated by the second speaker. Although in some stereo systems there is a division of frequencies to a very limited degree, in the average system each of the two or more speakers radiates the full range of frequencies. Finally, some believe that all stereo systems are necessarily high fidelity systems, but this is far from true. Due to the use of inferior components employed in mass-produced stereo systems made to meet competition, some systems fall far short of high fidelity standards. In order to meet the requirements of both good stereo and good high fidelity many of the

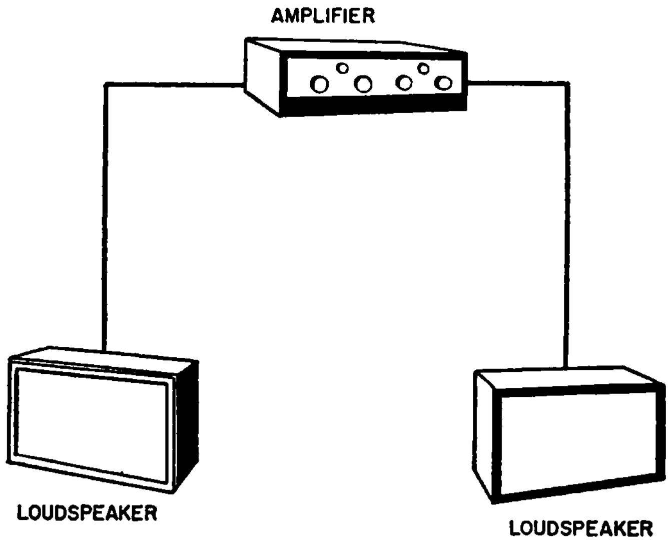


Fig. 10-3. "Pseudo-stereo" arrangement.

components must meet specifications that are far more exacting than those applying to monaural high fidelity. This is especially true of the phonograph pickup arm, the cartridge and the turntable.

How Stereo Recordings Are Made

To aid in understanding how stereophonic sound is recorded and reproduced, we shall imagine that we are about to make a disc recording of the sounds of a lively ping-pong game. Of course we know that most modern recordings are first made on tape and then transcribed to disc, but for this explanation we shall take certain liberties with the facts.

We are all aware that in monaural disc recording a cutting stylus driven by electrical impulses from an amplifier engraves lateral undulations in the surface of a disc as the stylus follows a spiral path. Later, during reproduction, the reproducing stylus follows these side-to-side movements or undulations which are then translated into electrical impulses by the phono cartridge and are then fed to the amplifier and loudspeaker. This is the lateral method of recording. It is used in all modern monaural discs and was devised by Emil Berliner in 1887. It is illustrated in Figure 10-4.

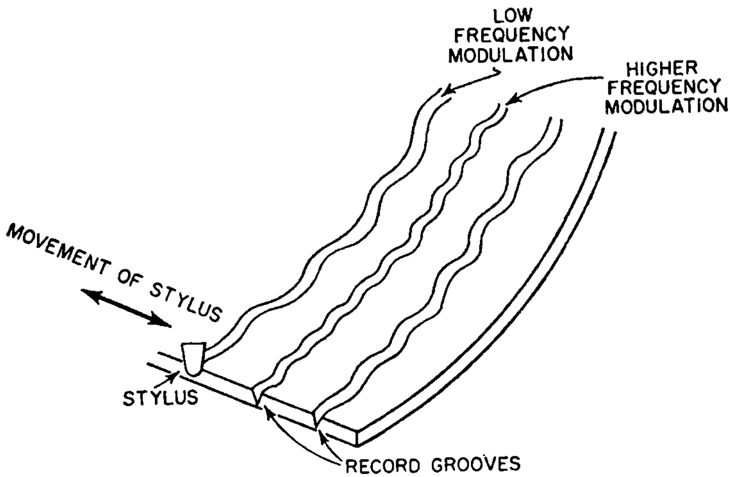


Fig. 10-4. Lateral method of recording.

Disc recordings may be made by another method, called the hill and dale system, which was used by Thomas Edison in his original phonograph. This method is shown in Figure 10-5. Note that in this system the cutting stylus, during recording, follows a spiral path as before, but instead of moving from side to side and engraving undulations or "wiggles" in the sides of the record groove it moves vertically and therefore cuts a series of depressions and elevations in the bottom of the groove, hence the name hill and dale recording.

In making our recording of the ping-pong game we shall have to imagine that we have a dual record-cutting device having a cutting stylus capable of moving both laterally and vertically. This conception, we realize, is neither accurate nor technically correct, but it will do for this discussion. Later we shall be able to consider a more complete description of its operation. Such a cutting device would engrave into the surface of the record impressions that are a combination of both lateral (side-to-side) and hill and dale (vertical) recording.

Figure 10-6 shows the arrangement of apparatus that we shall use in making the recording. Note that microphone A is placed near ping-pong player A, and microphone B is closer to player B. While both microphones will pick up all sounds of the game, microphone A will provide a more distinct sound of A's paddle against the ball than will microphone B, and vice versa. Notice also that each microphone is equipped with an amplifier and the amplifiers have

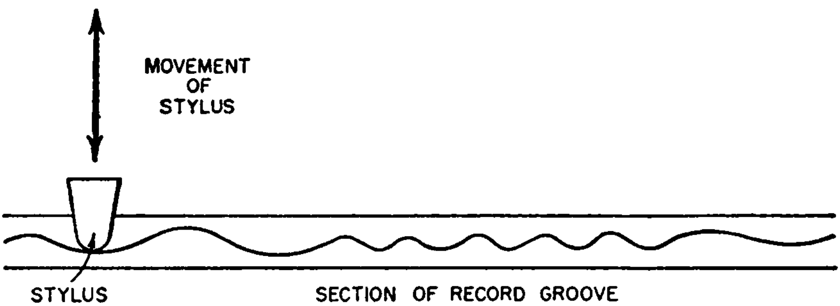


Fig. 10-5. Hill and dale method of recording.

been designated A and B. Furthermore, the output of each of the two amplifiers is fed to one of the two coils of the dual recording device. We shall refer to the combination of microphone A, amplifier A, and coil A as channel A. Similarly, channel B consists of microphone B, amplifier B, and coil B.

During the progress of the game the resulting sounds will be picked up by the two microphones, will be amplified and then applied to the two recording coils. If the two coils are arranged so that coil A imparts a horizontal motion to the cutting stylus and coil B applies a vertical motion, the result will be that a series of engravings will be cut in the surface of the disc that will be a combination of lateral and hill and dale recording.

Now let us further suppose that for playback purposes we have a stylus that is free to move both laterally and vertically. The arrangement of equipment is as shown in Figure 10-7. Again there are two channels, designated A and B. Channel A, which responds only to lateral movements of the playback stylus, employs pickup coil A, amplifier A, and speaker A. Channel B, which reproduces only the vertical movements of the stylus uses coil B, amplifier B and speaker B.

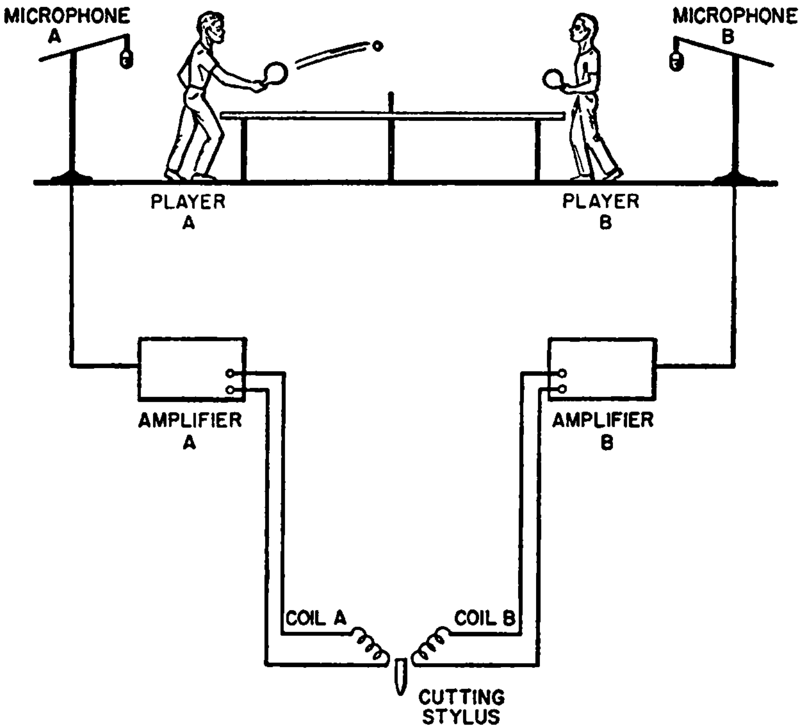


Fig. 10-6. Recording apparatus for ping-pong game.

When the recording is played back, assuming that all of the equipment is of high fidelity grade, we shall hear the sounds of the ping-pong game with all of the original fidelity as to frequency range and lack of distortion but with one important difference. While both speakers will be reproducing all of the sounds of the game, the sound of player A's paddle against the ball will be louder as it issues from speaker A than will the same sound coming from speaker B, and vice versa. In this way we may easily follow the progress of the game as the sound shifts from speaker to speaker.

We can easily imagine that when this method is applied to recordings of music, especially orchestral music, we shall have realized a new sensation of space, depth and realism in musical reproduction.

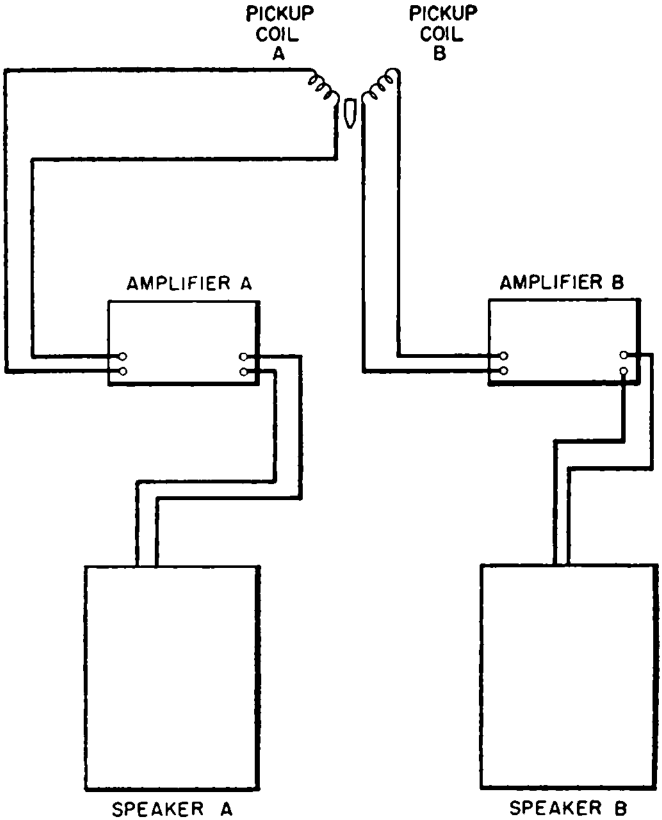


Fig. 10-7. Playback apparatus for ping-pong game.

Stereophonic Disc Recordings

Early attempts at stereophonic recording were based upon a combination of the hill and dale and lateral methods of recording just like that described in the example of the preceding section. Such a system was tried as early as thirty years ago but was not too successful. One difficulty was that under certain signal conditions the record groove became restricted or narrower. In trying to follow the narrower portions of the groove the stylus was forced to rise slightly and this rise, of course, constituted a vertical movement of the stylus. As a result a false signal was introduced, causing serious distortion.

The modern method of stereo disc recording can best be understood by first considering a simplified version of a typical stereo phonograph cartridge. We have chosen a crystal type of cartridge for the sake of simplicity, although the explanation would serve just as well for a variable reluctance type.

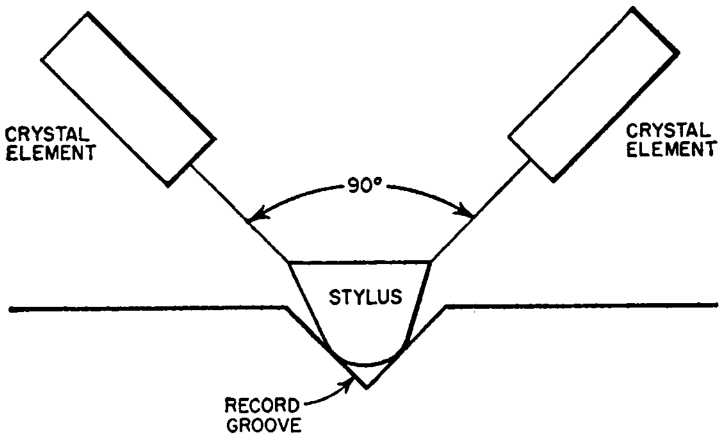


Fig. 10-8. Simplified stereo cartridge.

Figure 10-8 shows such a simplified stereo cartridge. Notice that there are two similar crystal elements located at 90 degrees to each other. Also observe that each of the two elements is placed at a 45 degree angle to the stylus.

As we noted in the preceding section, in lateral recording the modulation (engravings in the record groove produced by sounds) is contained in the side walls of the record groove. We also found that in hill and dale recording the modulation is contained in the bottom of the record groove. In a modern stereo recording the modulation belonging to one channel is contained in one wall of the record groove; the modulation pertaining to the second channel appears in the opposite wall of the groove. When such a recording is played back, the modulation in, let us say, the right wall of the groove, affects the right element of the pickup cartridge and only that element develops a signal. Similarly, the modulation contained in the left wall of the record groove affects only the left element of the pickup and only the left element produces a voltage as a result of it. In order to understand why this is so, study the diagrams of Figures 10-9 and 10-10. In Figure 10-9 only the right wall

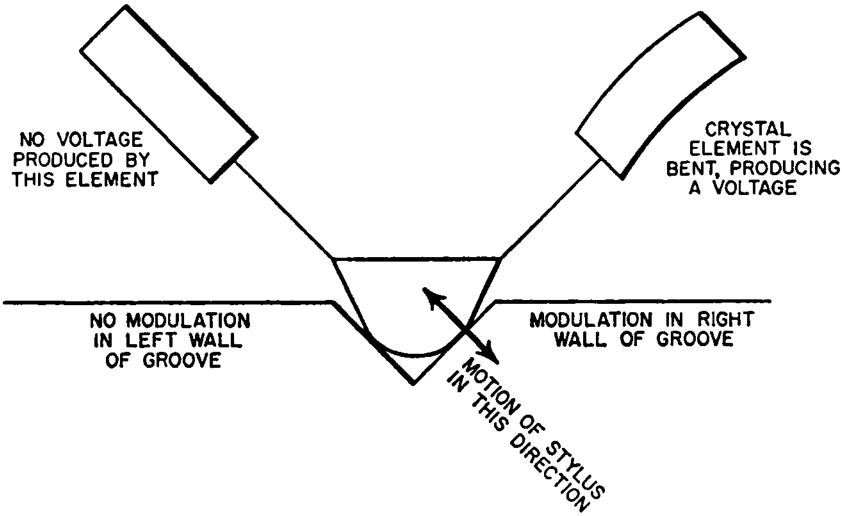


Fig. 10-9. Stylus movement when right side of record groove contains modulation.

of the record groove contains modulation; the opposite wall is smooth or unmodulated. Consequently, at that moment during the recording sound was being picked up only by channel A of the recording amplifier and therefore only the right side of the groove was affected. The modulation will result in a movement of the stylus and you will notice that this movement is in a direction at right angles to the axis of the pickup element A. Pickup element B is not affected and does not develop a voltage at this instant because the movement is parallel to the axis of element B and no voltage can be produced under such conditions.

When the left side of the record groove contains modulation there is a movement of the stylus as shown in Figure 10-10. You will observe that the movement now is in a direction at right angles or perpendicular to the axis of element B and parallel to the axis of element A. Element B then develops a voltage but A does not. The modulation in the left wall of the groove was of course produced by sound picked up by channel B of the recording amplifier. In this way a recording can be produced which contains in a single groove two sets of modulations each differing slightly from the other and each affording a slightly different perspective or point

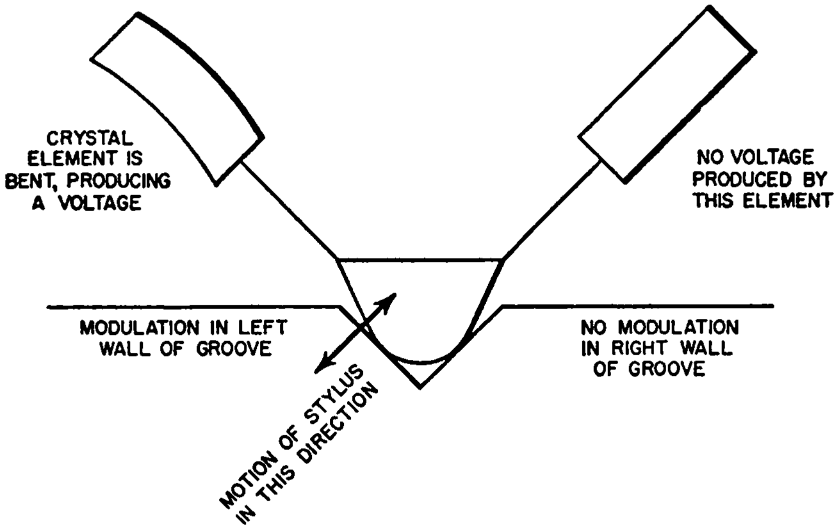


Fig. 10-10. Stylus movement when left side of record groove contains modulation.

of view of the music. Whenever both sides of the record groove are modulated both pickup elements will develop voltages; when neither side of the groove is modulated neither pickup element produces a voltage.

Basic Stereo Components

The simplest possible system for reproducing stereophonic disc recordings is shown in Figure 10-11. It consists of a record player or changer utilizing a dual-element cartridge like that described in the preceding section. If a variable reluctance type of cartridge is employed, the voltage-producing units will then be coils instead of crystal elements.

One crystal element (or coil) delivers the musical information it picks up from its side of the record groove to an amplifier; the amplifier voltages are then applied to a loudspeaker. The combination of crystal element, amplifier and speaker is referred to as a channel; we shall call this one the right channel, or channel A.

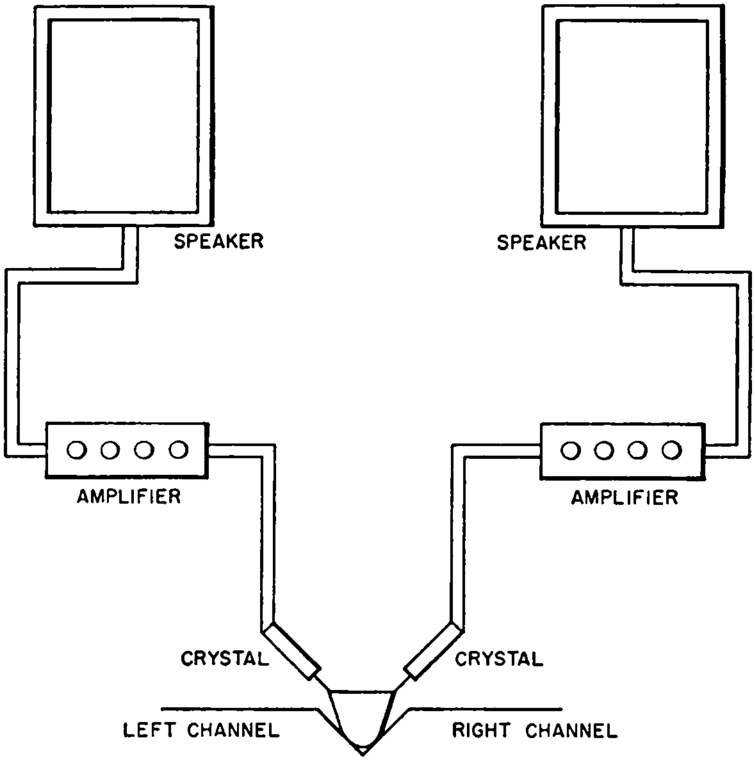


Fig. 10-11. Simple system for reproducing stereo disc recordings.

The remaining crystal element delivers its voltage to a second amplifier and the amplified voltages are applied to a second loudspeaker. This combination of crystal element, amplifier and speaker is the left channel, or channel B.

With sufficient separation between the two speakers we shall then receive the impression that the sounds of the various instruments in the orchestra reach us from the proper directions and the overall result is a sense of realism not achieved by monaural high fidelity systems.

Before leaving the subject of disc recordings, we shall consider the question of compatibility. The word compatible means capable of existing together in harmony. Applied to stereo it refers to a disc recording that may be played on either a stereo or a monaural player.

Monaural recordings may be played on stereo equipment without damage to either equipment or recording. Of course, since a monaural recording contains only one set of modulations, only one pickup element, one amplifier and one loudspeaker functions during the playing of such a recording.

Unfortunately, the reverse situation is not true in present stereo recordings, although there are developments under way that may soon make this possible. Any attempt to playback a stereo recording on a player designed solely for monaural recordings will result in damage to the recording. This may not be evident to the noncritical listener after only one playing and the danger is that the listener may not notice the damage, may assume that none has resulted and may try it again. Almost certainly the recording will be ruined after several playings.

The reason why stereo recordings cannot be played on monaural equipment is due to the construction of the pickup. A stereo stylus is free to move in two directions, as we have already seen. We might say that it has both vertical and lateral compliance. Compliance is merely a measure of how easily the stylus may be deflected from its rest position. A monaural stylus is free to move only laterally; we might say it has only lateral compliance. Consequently, when a stereo record is played on a monaural record player one side of the groove is subjected to the action of a stiff, unyielding stylus and the delicate engravings in that side wall are rapidly destroyed.

Record Players for Stereo Reproduction

A record player designed for reproduction of stereophonic recordings must meet specifications that are more exacting than those for monaural players. For example, rumble that would not be noticed in monaural reproduction may become objectionable when stereo recordings are played. Such rumble may be due to vertical motion of the turntable or the spindle. Since vertical movements produce little or no voltage output in a monaural cartridge, the rumble may not be noticed. However, the stereo cartridge must have a high vertical compliance, therefore the rumble becomes apparent.

Correct tracking is far more important than in a monaural system and, since tracking is closely related to compliance it may be impossible to obtain correct tracking in a cartridge used in a record changer pickup arm when the desired stylus pressure is used. As observed elsewhere in this book correct tracking is obtained by the use of a well-designed, well-balanced pickup arm (among other things), and it is difficult to incorporate this item in the average mass-produced record changer. For this reason, best results in stereo reproduction will be obtained by using a manual player equipped with a transcription-type pickup arm and a well-designed turntable driven by a top-quality motor.

The mere fact that a record player or automatic changer is described as a stereo player does not necessarily insure that it has been especially designed for stereo reproduction. In a few cases, the stereo model and the monaural model of a given type of changer are found to be identical in construction, except for the pickup. Also, some changers are sold as "stereo-wired" or "wired for stereo" and are supplied without cartridge, the only point of difference between the stereo-wired model and the monaural-wired model being that the pickup arm has two pairs of wires instead of a single pair. This does not mean that such players or changers are useless, but they certainly cannot afford the reproduction that a unit especially designed for stereo can give. When selecting a record player (and this applies especially to changers) compare the specifications of the stereo model with those of the equivalent monaural model. The performance data pertaining to the stereo model should be definitely superior to the monaural model if the unit has been carefully engineered for stereo service.

All current production of stereo disc recordings is recorded at 33 1/3 rpm. Therefore, stereo cartridges use a single stylus. The stereo stylus, instead of the 1 mil or 0.001 inch tip radius used in the ordinary LP stylus, has a radius of either 0.7 mil or 0.5 mil. The smaller radius stylus seems to be more common in transcription type players while the 0.7 mil stylus is more widely used in automatic changers. Most of the leading makes of stereo cartridges come equipped with a diamond stylus; a few low-priced ceramic stereo cartridges are fitted with sapphire styli.

Stereo Tape Recordings

At this time it seems unnecessary to review the basic principles of tape recorders, for the subject was covered in Chapter 6. It is only necessary to point out that early tape recorders were single track; that is to say, the entire width of the tape was used for a single recording. Later, it was found that a recording could be made on something less than half of the full width of the tape with little or no sacrifice in fidelity. This discovery led to the development of the two-track recorder. In this system a recording is made on one edge of the tape. The tape is then reversed and a second recording made on the other edge. In monaural recording, this doubles the playing time of a given length of tape. A single-track recording on a 1200-foot length of tape runs about 30 minutes; a two-track recording has a playing time of about an hour, for the same length of tape.

When stereo was introduced, it soon became evident that a two-track recorder could be used for recording or playback of stereophonic sound by using one track for the right channel and the second track for the left channel. Of course, in recording, two microphones, two recording amplifiers and two recording heads were needed. To play back such a tape, two playback heads, two playback amplifiers and two speaker systems were required. Furthermore, the playback time was limited to about 30 minutes per 1200-foot reel of tape because both tracks were in use at the same time. Most stereo recorders of the home variety were, and still are, designed for playback only of stereo recordings, but they are capable of either recording or playback of monaural music.

To extend the playing time of a given length of tape, the four-track recorder was developed. Early versions of this recorder used staggered heads, with the two heads separated an inch and a half or so. This imposed a space problem so the "in-line" heads, shown in Figure 10-12, were developed. Note that of the four recordings (one per channel in each direction) each occupies a space somewhat less than one-fourth the entire width of tape. This leaves a safety island (indicated by the heavy black lines on the tape) between each two tracks to prevent cross talk.

The drawing shows the tape passing across the heads from left to right, in order to play the first half of the recording. With the tape in this position, track #1 (belonging to the left stereo channel) passes over the upper head gap while track #3 (pertaining to the right stereo channel) passes over the lower head gap. When the tape is reversed to play the remainder of the recording, track #4 (left stereo channel) passes over the upper head gap and track #2 (right stereo channel) passes over the lower head gap. With this system a 1200-foot reel of tape gives a playing time of about an hour for stereo or two hours when used monaurally.

With the introduction of four-track stereo tape recordings, the chief disadvantage of tape, as compared to disc recordings, was eliminated. Although stereo tape recordings were far more com-

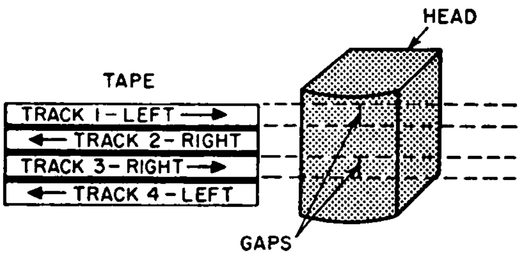


Fig. 10-12. "In-line" heads on four-track recorder.

patible than disc recordings, in that stereo tapes can be played on monaural equipment without risk of damage, two-track stereo tapes were much more expensive than disc recordings, for an equal playing time.

Stereo Broadcasts

Early efforts in broadcasting stereophonic, or binaural programs began with the use of two channels in the AM broadcast band (550 to 1600 kilocycles). By using two receivers, each feeding one unit of a pair of headphones, binaural reception was achieved. There were also a few experiments involving the use of a television channel and an AM channel.

Later, an AM transmitter operating in the broadcast band and an FM transmitter operating in the 88 to 108 megacycle range were used in a similar service. To receive such programs, which now extend to a considerable number of stations, two receivers or tuners are of course required. This system is not fully compatible, since a listener who has only one receiver does not hear the full program but only that portion picked up by one of the two microphones. Furthermore, there are differences in the characteristics of the two systems of transmission and reception. As we know, the AM system is subject to fading and electrical interference. This objection could be overcome by using two FM channels instead of one FM and one AM transmitter and in fact, such experiments have been carried out.

Many of the objections to the systems already described are overcome by means of "multiplexing." The multiplex system is fairly satisfactory and far less expensive than the others as only one tuner or receiver is needed. Briefly described, the multiplex system makes use of a "sub carrier." The main carrier is modulated by the first channel (audio). To the main carrier is added a sub carrier which is modulated by the signals of the second audio channel. If a monophonic receiving system is used, only one channel is heard, but a stereo receiver is equipped with an additional detector which is used to demodulate the sub carrier. Although the multiplex system was not in general use at the time this book was revised, many of the stereo FM tuners now being produced incorporate it. This is worth remembering if you might consider converting your system to stereo. Other than this, tuners for stereo have specifications similar to those intended for monaural reception. For information concerning FM tuners consult Chapter 5.

Stereo Amplifiers

With one unimportant exception, all stereophonic sound systems require two amplifiers and if the program source dictates the use of a preamplifier this too must be duplicated.

Modern stereo amplifiers consist of two identical amplifiers and preamplifiers built upon a single chassis. While the design and construction of each of the two units is, in general, similar to that used in monaural units, there are a few features that are different.

Specifications as to hum level, noise level and distortion are similar to those applicable to monaural amplifiers. Power output is usually rated per channel. Thus, a 50-watt stereo amplifier will deliver 50 watts when used in a monaural system or 25 watts per channel for stereo. In a few cases, the total power output is stated, rather than the output per channel. If a stereo amplifier is rated at

28 watts without further qualification, then it will be safe to assume that this means 14 watts per channel. Factory-wired amplifiers and kits are available with power output ratings of from 12 to 50 watts per channel, but the majority deliver about 20 watts per channel.

One problem that arises in stereo is that of controlling volume. Since there are, in reality, two separate music systems provision must be made for controlling the volume of each system. The simplest but not the most satisfactory method is to use two independently-operated volume controls. However, when it is necessary to increase or decrease the volume level of the entire system the settings of the two controls must be changed by the same amount, otherwise the balance will be disturbed. An improved method uses concentric control knobs, each knob operating its own volume control. Each of the two controls may then be operated independently of the other so that proper balance may be achieved. However, once balance is obtained, it is no longer necessary to work two controls in order to raise or lower volume; by means of a friction or clutch arrangement the two knobs are locked together so that turning one of them operates both volume controls simultaneously. In this way the volume level may be changed as desired without affecting balance.

Although many "packaged" stereo systems use identical speakers for the two channels, this is usually not true of systems using components selected by the owner, nor need it be true. The possibility that two different speakers might be used makes it necessary that independent tone controls be provided for the two channels. Here again, some amplifiers employ the concentric control system just described; generally the two bass tone controls are ganged together, as are the two treble controls.

While not absolutely necessary, a phase reversing switch is desirable and convenient and is found in a few models. Because stereo systems require two or more speakers, phasing can become a real problem. Of course, speakers can be phased in the usual way, as described in Chapter 2, but this may prove to be time consuming, especially if you are making a series of trials using different speakers.

The general rule in stereo work is to arrange the two channels so that the "violins are on the left"; that is to say, the sounds of the violins should issue predominantly from the left channel speaker. Some stereo disc and tape recordings do not follow this rule, so that occasionally it becomes necessary to switch channels. For this reason, the majority of currently available stereo amplifiers are equipped with a channel reversing switch.

Practically all modern stereo amplifiers are equipped with some type of balance control; the design ranges from the simplest potentiometer to a cathode-ray indicator tube. A stereo system without such a control can still be balanced but it will not be quite so convenient. A monaural disc recording may be used and the volume of the two channels adjusted until the level is the same in both speakers when the reversing switch is thrown. Amplifiers using some type of visual balance indicator, such as the cathode ray indicator or a meter, show that the system is balanced up to the speakers. They do not indicate a lack of balance due to the use of two dissimilar speakers or different acoustic conditions surrounding the speakers. Once electrical balance has been achieved by the use of the visual indicator, further adjustments will have to be made to compensate for differences in speakers or acoustics.

Stereo Speaker Systems

In the simplest stereo arrangement two similar speakers or two similar speaker systems are used, one for each channel. Figure 10-13A illustrates such an arrangement. In this case, two extended range speakers are used, each housed in its own enclosure. However, if the reproducer is to comprise a woofer and a tweeter, or a woofer, midrange speaker and a tweeter, then each of the two enclosures will house identical sets of speakers. Usually the two speakers (or speaker systems) are placed 6 to 8 feet apart. This is the arrangement used in many "packaged" stereo systems, but is not necessarily the best. Assuming that the two speakers are identical, it is quite possible that the acoustics of the room require that the spacing between speakers be something more or less than the 6 or 8 feet mentioned in order to achieve the best stereo effect. Although most amplifiers have some arrangement for balancing it is often impossible to achieve correct balance without altering the location of one or both speakers.

Another type of stereo speaker system is shown in Figure 10-13B. Here a third speaker has been added. Signals from the left and right channels are fed to the third speaker, located between the other two, thus creating a third channel. This method seeks to avoid the "hole-in-the-middle" or "ping-pong" effect noted in some two-channel systems. A few amplifiers are equipped with a control for blending signals from the left and right channels in order to set up the third channel.

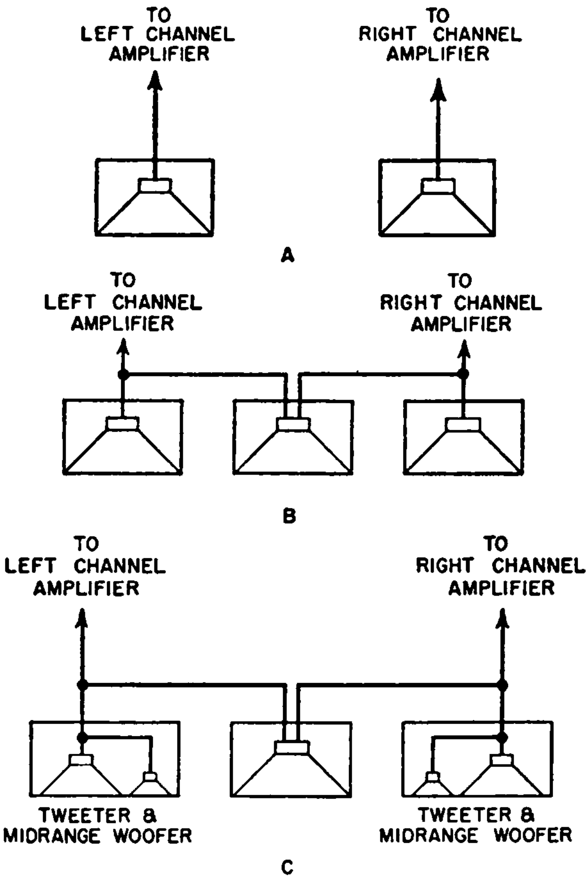


Fig. 10-13. Stereo speaker systems.

A variant of the system just described is seen in Figure 10-13C. The left channel and the right channel use speakers covering all of the musical range except the bass; that is, each speaker system consists of a mid-range speaker and a tweeter. The middle channel employs a woofer only; it receives only the lower tones from both the left and right channels. This "blended bass" system operates on the principle that the lower frequencies are non-directional and therefore only one sound source is needed for them.

Converting To Stereo

If you already have a high fidelity system and wish to convert to stereo, it may be possible to do so even though you have had little or no electronic experience. Several different types of procedure are possible and the procedure you select will depend upon your experience, the type of equipment you already have, and your financial circumstances. In brief, in order to convert to stereo you will have to do the following:

a. Substitute a stereo pickup cartridge for your present monaural cartridge. Or, if you have a record changer that does not conform to stereo standards, you may have to replace the entire changer.

b. Add a second amplifier. It may or may not be identical to your present amplifier, depending upon the results you expect.

c. Add a second speaker or speaker system with a separate enclosure. Depending upon the convenience and results you expect, this may or may not be identical to your present speaker.

d. If tape recordings are to be played and you already have a monaural recorder, it may be possible to obtain substitute stereo heads for the present monaural heads.

e. If stereo radio reception is desired, two tuners will be needed. Assuming that you already have an FM tuner, you will also need an AM tuner. On the other hand, if you now have no tuner you will need a combined AM-FM tuner.

As mentioned before, the procedure you adopt in making a conversion will depend upon a number of variables, so many that it is impossible to offer specific instructions. However, the steps listed above do require further amplification and we shall therefore consider them in order.

Record player. If you now have a good quality manual player no particular problem will arise, for it is likely that it meets standards sufficiently high so that it will be suitable for stereo. All you need do is replace the cartridge. But if you are presently using an automatic changer you should first determine whether it is suitable for stereo. You might do this by comparing the specifications for your changer with those of stereo changers listed in distributors' catalogs or manufacturers' catalogs. If your unit equals the average of a number of products you can probably use it for stereo with a fair degree of satisfaction. Your final decision will of course hinge upon just how critical you are as a listener and just what you will expect in the way of final results. Remember that almost any player or changer can be converted to stereo but an inferior unit will give poor results.

Since many record changers sold in the last few years were wired for stereo cartridges it is possible that the one you own is so wired. If so, the pickup arm will contain two pairs of wires in place of the single pair found in a monaural unit. If your player or changer is equipped with plug-in cartridge shells and is wired for stereo then both the shell and the front end of the arm will have four contacts instead of two. Some stereo cartridges are equipped with three terminals instead of four; in that case, the shield wire of each of the two pairs is to be connected to the common terminal of the cartridge. The remaining two cartridge terminals will be marked "right" and "left" or "R" and "L" or will be color coded, and the accompanying instruction sheet will tell you which of the colors pertains to each of the two channels. In any case, these channel markings must be observed when the pickup cartridge is connected to the amplifiers.

Amplifiers. If at present you have a high quality amplifier, you will be well advised to add a second amplifier that is as nearly identical to your present one as possible, for in that way you will be assured of the best possible results. In some cases amplifier manufacturers offer duplicate amplifiers together with an adaptor for making the conversion. However, if your present amplifier is of a quality less than first-class, you might still want to convert and accept the results that might be considerably short of good quality high fidelity stereo. In that case, there are a number of routes to stereo open to you. For instance you might want to buy one of the stereo conversion kits currently available. These consist of a low-power amplifier, a small speaker, and a stereo cartridge, together with all of the required accessories. Or you might prefer to build a second amplifier. If you decide to follow the latter course, it might be well to use a ceramic cartridge, for then the amplifier can be quite simple in design, and usually the preamplifier can be dispensed with. Finally, it should be noted that almost any amplifier of good or even fair quality can be used; however bear in mind that the final results can never be better than what the poorer of the two amplifiers can deliver. This limitation applies to both distortion and power output. You might, for instance, have a good quality tape recorder; most home type tape recorders can be used as an amplifier-speaker system when the "Record-Playback" control is placed in the neutral position. If your machine is designed in this way, then the problem of amplifier and speaker are solved at once. Even if you are prepared to accept results that would not be described as good high fidelity stereo, it is still best to follow at least one rule in the selection of an amplifier: it should at least have a push-pull output stage.

Speakers. Ideally, for two-channel stereo the two speakers should be identical. However, some departure from this rule is permissible. It is not a good idea to use a second speaker which has a limited frequency range as compared to the first. Furthermore, the second speaker should compare favorably with the first in so far as distortion is concerned. It is not necessary that the second speaker have the same cone diameter, power handling capacity or efficiency as the first but you must remember that the addition of a second speaker having a lower power handling capacity than the first will impose some limitation on the stereo system. If, for example, you add a 10-watt speaker to a system already using a 25-watt speaker then the total power handling capacity of the stereo system will total 20 watts, or 10 watts per channel. However, this limitation may not be serious (or even worth considering) if you customarily operate your system well below the power handling capacity of the original speaker. Differences in efficiency between the two speakers, unless quite marked, can usually be compensated for by balancing the system.

Tape recorder. Some makes of tape recorders may be converted to stereo by substituting stereo heads for the present monaural heads. However, the installation of a head in a tape recorder is not a job for a novice. In any case, you should have at hand the manufacturer's instructions for aligning the heads after they have been installed, and these instructions should be followed most carefully. You should also have one of the special tapes for checking the final result. An improperly installed head will result in poor performance in so far as signal to noise ratio is concerned. If you can obtain the necessary replacement heads for your machine but do not feel that you can do the job properly, then it would be best to have them installed by the manufacturer's service representative. In the event that the manufacturer of your unit does not offer substitute heads, then the entire recorder must be replaced.

Tuners. If you now own an FM tuner and wish to convert to stereo, the simplest solution is to adapt an AM receiver for use as a tuner. Instructions for doing this were given in Chapter 5. However, the FM tuner will be useless for receiving multiplex programs, when such broadcasts are in general use. However, it will still serve for receiving monaural FM broadcasts, and meanwhile by merely adding an AM tuner you will be able to enjoy FM-AM stereo broadcasts as long as they are transmitted. Furthermore, it is likely that adaptors for converting existing monaural FM tuners to multiplex will be offered.

Appendix A
High Fidelity Directory

Dealers

(All firms listed below conduct a mail order business and will send catalogs upon request.)

Allied Radio Corp., 100 N. Western Avenue, Chicago 80, Ill.
Hudson Radio Corp., 212 Fulton Street, New York 7, N.Y.
Lafayette Radio, 165-08 Liberty Avenue, Jamaica 33, N.Y.; 100 Sixth Avenue, New York 13, N.Y.; 110 Federal Street, Boston, Mass.; 542 East Fordham Road, Bronx 58, N.Y.; 24 Central Avenue, Newark, N.J.; 139 West 2nd Street, Plainfield, N.J.
Olson Radio Corp., 609 S. Forge Street, Akron 8, Ohio.
Radio Shack Corp., 730 Commonwealth Avenue., Boston 17, Mass.

Manufacturers

(All manufacturers listed will send catalogs or brochures upon request. In many cases these firms manufacture components other than those listed.)

Amplifier Kits

Acro Products Co., (Acrosound), 410 Shurs Lane, Philadelphia 28, Pa.
Arkay International, 88-06 Van Wyck Expressway, Jamaica, N.Y.
Allied Radio, (Knight), 100 N. Western Avenue, Chicago 80, Ill.
Dynaco, Inc., (Dynakit), 3916 Powelton Avenue, Philadelphia 4, Pa.
Eico, 33-00 Northern Boulevard, Long Island City 1, N.Y.
Grommes, Division of Precision Electronics, Inc., 9101 King Avenue, Franklin Park, Ill.
Heath Company, (Heathkit), Benton Harbor 15, Mich.

Paco Electronics, Inc., 70-31 84th Street, Glendale 27, Long Island, N.Y.

Amplifiers

Bell Sound Division, Thompson Ramo Wooldridge, Inc., Columbus 7, Ohio.
Bogen-Presto, Paramus, New Jersey.
De Wald, Division of United Scientific Laboratories, Inc., 35-19 37th Avenue, Long Island City 1, N.Y.
Dynaco, Inc., (see *Amplifier Kits*)
Eico, (see *Amplifier Kits*)
Fisher Radio Corporation, 21-25 44th Drive, Long Island City 1, N.Y.
General Electric Co., Audio Products Section, Auburn, N.Y.
Harman-Kardon, Inc., Westbury, Long Island, N.Y.
Pilot Radio Corporation, 37-02 36th Street, Long Island City 1, N.Y.
H. H. Scott, Inc., 111 Powdermill Road, Maynard, Mass.
Sherwood Electronic Laboratories, Inc., 4300 N. California Avenue, Chicago 18, Ill.
Stromberg-Carlson, Division of General Dynamics, 1477-010 N. Goodman Street, Rochester 3, N.Y.

Cabinets

Homewood Industries, Inc., 26 Court Street, Brooklyn 1, N.Y.
Cartridges, Phonograph Pickup
The Astatic Corporation, Conneaut, Ohio.
Fairchild Recording Equipment Corp., 10-40 45th Avenue, Long Island City 1, N.Y.
General Electric Co., (see *Amplifiers*)
Grado Laboratories, Inc., 4614 7th Avenue, Brooklyn 20, N.Y.
Dynaco, Inc., 617 N. 41st Street, Philadelphia 4, Pa.
Pickering & Company, Inc., Plainview, N.Y.
Shure Bros., Inc., 222 Hartrey Avenue, Evanston, Ill.
Sonotone Corp., Elmsford, N.Y.

Loudspeakers

Altec Lansing Corp., 1515 S. Manchester Avenue, Anaheim, Calif., 161 Sixth Avenue, New York 13, N.Y.

Bozak, Darien, Conn.
Electro-Voice, Inc., Buchanan, Mich.
General Electric Co., (see *Amplifiers*)
Jensen Manufacturing Co., 6601 S. Laramie Avenue, Chicago 38, Ill.
James B. Lansing Sound, Inc., 3249 Casitas Avenue, Los Angeles 39, Calif.
Neshaminy Electronic Corp., (Janszen), Neshaminy, Pa.
North American Philips Co., Inc., (Norelco), 230 Duffy Avenue, Hicksville, Long Island, N.Y.
Stephens Trusonic, Inc., 8538 Warner Drive, Culver City, Calif.
University Loudspeakers, Inc., White Plains, N.Y.

Phonograph Turntables

Fairchild Recording Equipment Corp., (see *Cartridges*)
Gray High Fidelity Division, 16 Arbor Street, Hartford 1, Conn.
Lafayette Radio (see Dealers)
Rek-O-Kut Company, Inc., 38-19 108th Street, Corona 68, N.Y.
Thorens, New Hyde Park, N.Y.

Pickup Arms

Fairchild Recording Equipment Corp. (see *Cartridges*)
General Electric Co. (see *Amplifiers*)
Gray (see *Phonograph Turntables*)
Pickering & Company (see *Cartridges*)
Rek-O-Kut Company, Inc. (see *Turntables*)

Record Changers

Audiogersh Corp., (Miracord), 514 Broadway, New York 12, N.Y.
Electronic Importers, Inc., (Musicmaster), 2128 Third Avenue, Seattle 1, Wash.
Garrard Sales Corp., Port Washington, N.Y.
Glasser-Steers Corp., 155 Oraton Street, Newark, N.J.
Stromberg-Carlson (see *Amplifiers*)
Thorens (see *Phonograph Turntables*)

Stereo Conversion Kits

Weathers Industries, 66 E. Gloucester Street, Barrington, N.J.

Styli, Phonograph

Fidelitone, Chicago 26, Ill.
Jensen Industries, Forest Park, Ill.
Recoton Corp., 52-35 Barnet Avenue, Long Island City 4, N.Y.

Tape Recorders

Crown International, Elkhart, Ind.
North American Philips Co., Inc. (see *Loudspeakers*)
Revere Camera Co., Chicago 16, Ill., Los Angeles 7, Calif.
Roberts Electronics, Inc., 1028 N. La Brea Avenue, Hollywood 38, Calif.
Superscope, Inc., (Sony), 8150 Vineland Avenue, Sun Valley, Calif.
Tandberg, 10 E. 52nd Street, New York 22, N.Y.
Telectrosonic Corp., (Telectro), 35-18 37th Street, Long Island City, N.Y.
V-M Corporation, Benton Harbor, Mich.

Tuner Kits

Arkay International (see *Amplifier Kits*)
Lafayette Radio (see Dealers)
Paco Electronics, Inc., (see *Amplifier Kits*)
H. H. Scott, Inc. (see *Amplifiers*).

Appendix B
Equalization for Popular Makes of Records

The chart below gives the correct settings for turnover point and treble rolloff for many popular makes of records. If you happen to have a one-knob equalizer, set it in accordance with the first column, then make additional adjustments for treble by operating the treble tone control.

Label	*Turnover*	*Rolloff*
Angel	NAB	AES
Atlantic	NAB	NAB
Capitol	AES	AES
Cetra-Soria	COL	NAB
Colosseum	AES	AES
Columbia	COL	NAB
Concert Hall	AES	AES
Decca	COL	NAB
Epic	COL	NAB
Haydn Society	COL	NAB
London	COL	LON
Mercury	AES	AES
MGM	NAB	AES
Oceanic	COL	NAB
RCA Victor	NAB	AES
Remington	NAB	NAB
Urania	COL	NAB
Vox	COL	NAB
Westminster	NAB	NAB

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