HI-FI STEREO INSTALLATION SIMPLIFIED

Derek Cameron

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This comprehensive book is designed to meet the growing demand for simplification of stereo installation procedures and achievement of optimum sound reproduction.

The author begins with an overview of basic systems options and an explanation of the principles of acoustics. Control of acoustic characteristics for listening areas is discussed, with descriptions of speaker enclosures and their placement in various acoustic environments.

Basic installation procedures are detailed, with attention to interconnection requirements. The reader will particularly appreciate the timely inclusion of precautions in installation procedures.

In Chapter 3, Mr. Cameron discusses in detail interconnections and specifications for commercial audio equipment from selected manufacturers. An excellent cross-section of 20 audio manufacturers has been compiled to illustrate practical installation situations. Photographs, diagrams, and basic schematics have been provided to facilitate reader understanding of the material discussed.

Practical perspective is provided by inclusion of specifications for typical high-quality audio equipment and illustration of required interconnections. The coverage ranges from basic installations to elaborate systems.

(continued on back flap)

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Contents

Preface

The thrust of this comprehensive text is toward simplification of hi-fi stereo installation procedures and achievement of optimum sound reproduction. To this end, an overview of the basic system options is provided and the principles of acoustics are explained. Control of acoustic characteristics in various listening areas is discussed and the choice of speaker enclosures with their placement in various environments is described and illustrated. Basic installation procedures are detailed, with coverage of interconnection principles and requirements, including consideration of impedance and current-flow characteristics. Common installation errors are pointed out and basic system adjustments for optimum performance are noted.

To provide practical perspective for the reader, interconnections and specifications for typical high-quality audio equipment are provided. Although many meritorious units were necessarily omitted for lack of space, twenty representative manufacturers have been selected to illustrate a cross-section of practical installation situations. The often-neglected topic of antenna installation is covered in some detail, and the reader is shown how to cope with reception problems in fringe areas. Control of audio-system interference is also explained, with attention to details of shielding and filtering. Because of the marked trend to installation of residential hi-fi wiring systems, approved techniques have been detailed and illustrated. For the benefit of the more technically minded reader, the author has included a chapter on audio component checks and tests. Both quick tests and basic instrument applications are described.

It has been assumed that the reader is not primarily concerned with technical matters but that he is seeking a "common-sense" guide to hi-fi installation procedures. To this end, no details that may concern the installer have been omitted (at least intentionally). It is anticipated that this handbook will be of value to junior college and vocational students as well as to home-study audiophiles. Hobbyists and experimenters will also find numerous practical pointers in this book. The author wishes to take this opportunity to thank the many manufacturers who have made the book possible by supplying important photographic and technical material, as credited throughout the text. He also wishes to acknowledge his debt to his fellow instructors and associates who have made many valuable suggestions and constructive criticisms.

DEREK CAMERON

Chapter 1

Survey of Hi-Fi Stereo Systems

1-1 GENERAL CONSIDERATIONS

A good high-fidelity/stereophonic-quadraphonic installation requires careful planning. First, room acoustics should be closely analyzed. Second, the optimum speaker system may be selected. Third, appropriate audio components, such as preamplifier, power amplifier, tuner, tape deck, cassette deck, and turntable, may be chosen. Fourth, an optimum FM/AM antenna installation should be planned. Some installations will include various accessories, such as frequency equalizers and color organs. After the installation has been completed, various adjustments may be required; for example, the bass and treble levels for the speakers may be optimized, and equalizers should be set to compensate for the particular acoustical characteristics of the room.

Industry authorities generally agree that high-fidelity reproduction requires a frequency response that is uniform within ± 1 decibel (dB) from 20 hertz (Hz) to at least 20 kilohertz (kHz), with a harmonic distortion value of less than 1 percent at any frequency within this range. A block diagram of a typical high-fidelity system is shown in Fig. 1-1. *Component systems,* such as those illustrated in Fig. 1-2, are very popular and are often

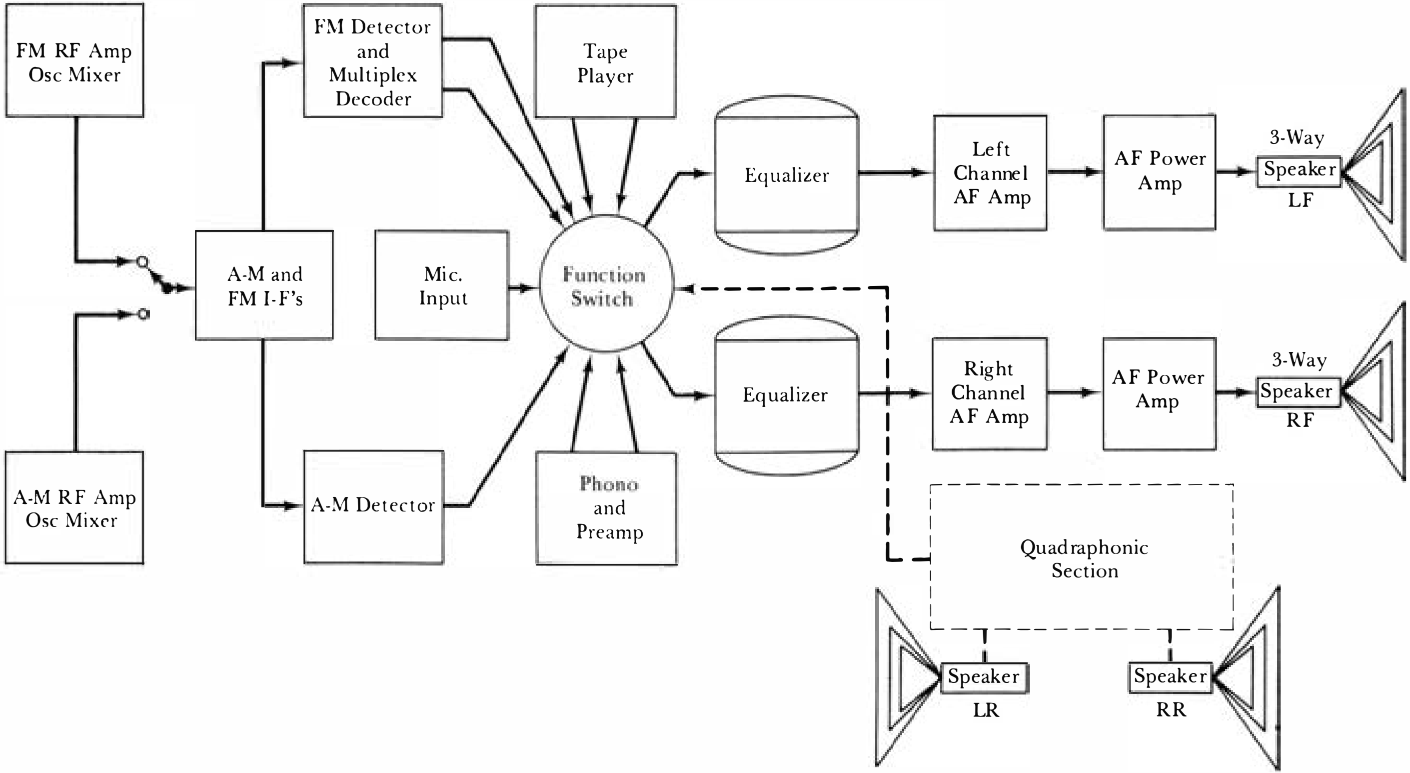
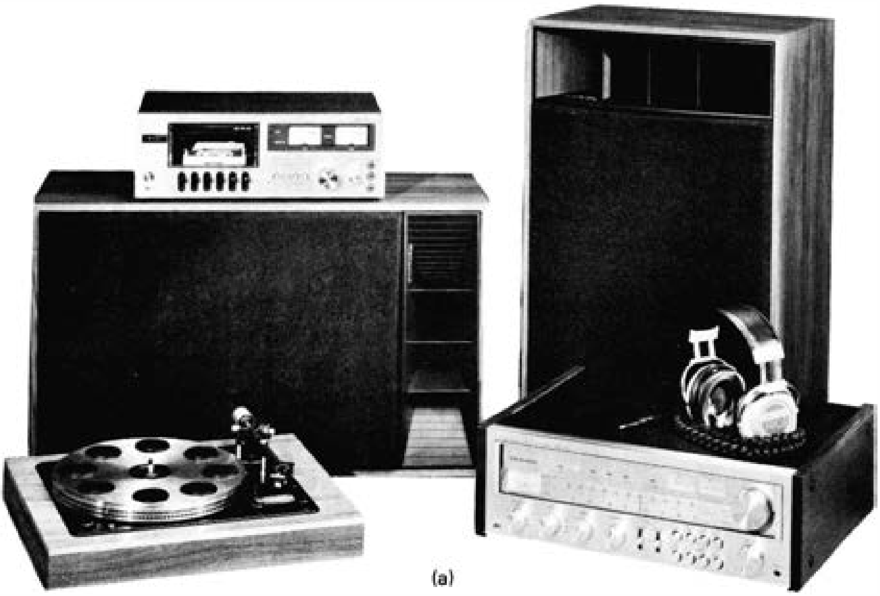


Figure 1-1. Block diagram of FM/AM stereo multiplex high-fidelity system.

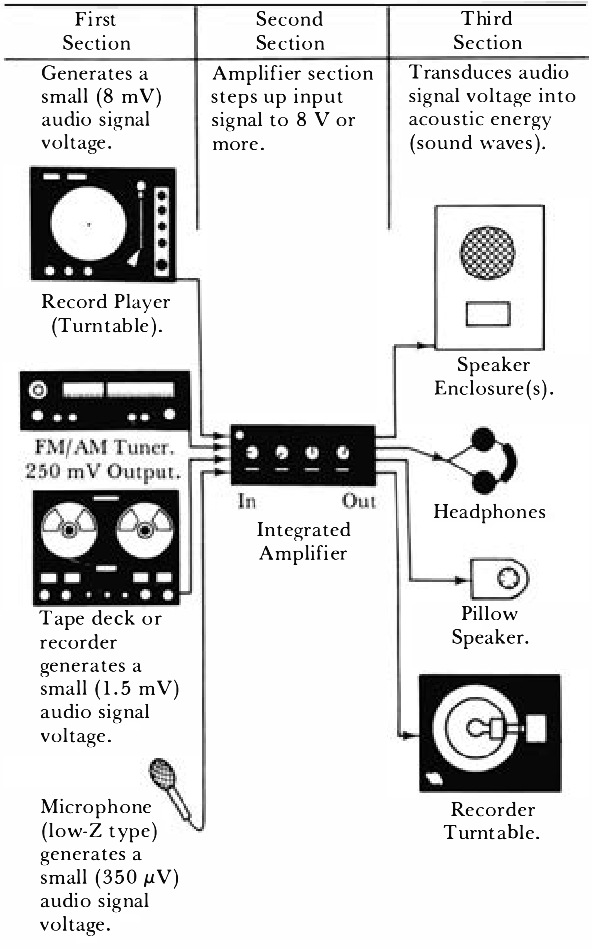


(a)

Figure 1-2. Component system: (a) appearance of a high-quality component system with an integrated receiver. (*Courtesy Radio Shack, a Tandy Corporation Co.*)

preferred by critical audio enthusiasts. An example is a chosen set of speakers, utilized with a preferred type of amplifier, plus a selected type of record player (turntable), a chosen design of FM/AM tuner, a selected reel-to-reel tape deck and/or an eight-track deck, or a preferred type of cassette deck. There is a marked trend to the inclusion of stereo frequency equalizers (Fig. 1-3) in component systems. Note that some amplifiers contain built-in frequency equalizers. A frequency equalizer is more elaborate than a conventional tone control, in that it permits the listener to increase or decrease the frequency response of a speaker through five sectors of the audio-frequency range.

Note that component systems were originally monophonic; even today mono hi-fi systems are available. This system utilizes single-channel preamplifiers and power amplifiers. When stereophonic reproduction was introduced, dual-channel preamplifiers and power amplifiers gradually replaced corresponding pairs of amplifiers. This was the beginning of combination arrangements. With the advent of stereo-FM broadcasting, stereo-multiplex adapters were added to component systems. The multiplex adapter was connected between the FM tuner and the preamplifiers. It served to decode the demodulated FM signal into left (L) and right (R) audio signals. Multiplex decoders were subsequently built into FM tuners,



(b)

Figure 1-2. (cont.) (b) Basic features of a component system.

thereby elaborating combination arrangements. Preamplifiers were combined with power amplifiers, and the combination was termed an *integrated amplifier.* Further, FM/AM tuners with stereo decoders were combined with integrated amplifiers to form integrated receivers.

High-fidelity stereo-quadraphonic systems are also designed in unitized form and enclosed in elegant furniture cabinets. A unitized hi-fi system is termed a *console,* and it contains at least two speakers. Another form of stereo-quadraphonic system is called the *compact;* it features separate

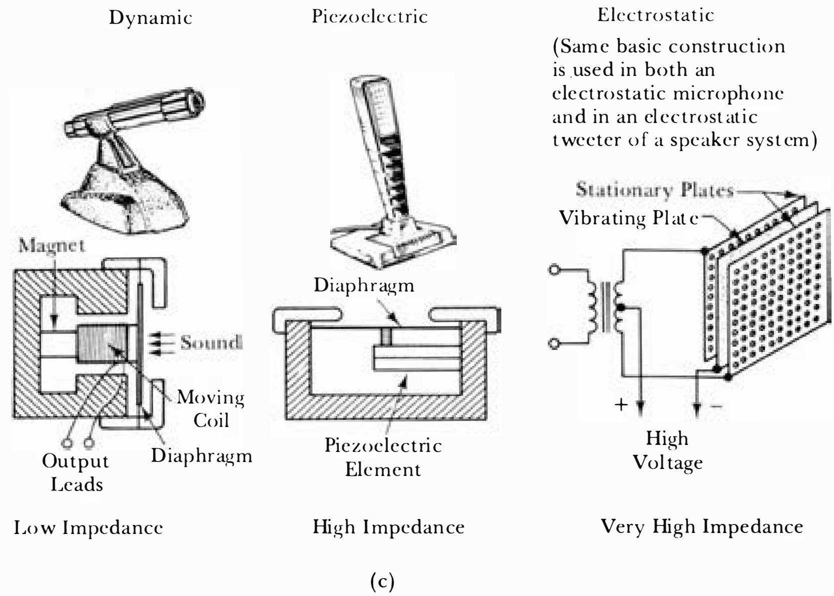


Figure 1-2. (cont.)

Figure 1-3. Appearance of a stereo frequency equalizer. (*Courtesy Dynaco*)

speakers with a record turntable and a stereo amplifier on the same base. The principal element in a compact system may include an FM or FM/AM tuner with a multiplex decoder, plus a turntable. Another design of the compact system includes a record changer mounted on top of the main unit, with a clear plastic protective cover. Audiophiles also refer to a compact arrangement as a *modular system* (see Chart 1-1). Note that a high-fidelity speaker enclosure is usually designed with several speaker units (drivers) of various sizes. The largest speaker in the group is termed a *woofer,* and it operates to reproduce the low bass tones. Most of the power in a typical audio signal is contained in the bass tones.

Chart 1-1

BASIC SYSTEM OPTIONS

All-in-One System. Console (unitized) design, wherein a single large cabinet contains a pair of speakers and all the system electronics. A typical unit comprises a record player, FM/AM tuner, stereo decoder, preamplifier, power amplifier, and two speakers.

Compact (Modular) System. Consists of a record player and stereo amplifier on the same base, with separate speaker enclosures. Some designs include an FM/AM tuner on the same base with the record player and amplifier. A few also include a tape deck. The most elaborate designs provide a four-channel amplifier for reproduction of quadraphonic discrete four-channel tapes.

Integrated Amplifier. An audio-amplifier unit that contains both a preamplifier and a power amplifier. Often provides more control refinements and more inputs (signal input facilities) than an all-in-one system.

Integrated Receiver. An integrated receiver contains an FM/AM tuner, preamplifier, and power amplifier in the same cabinet. It is used with an external record player, external tape player, external speakers, and optional earphones.

Speakers. A bookshelf type of enclosure has approximately half the efficiency of a large enclosure; or, a bookshelf enclosure will require about twice as much audio power input to develop the same amount of sound output as a large reflex and ported enclosure.

Tape Machines. A tape recorder provides both recording and playback facilities; a tape deck lacks recording facilities. A tape deck lacks a built-in power amplifier and must be used with an external amplifier and speaker system. Some tape decks provide recording facilities.

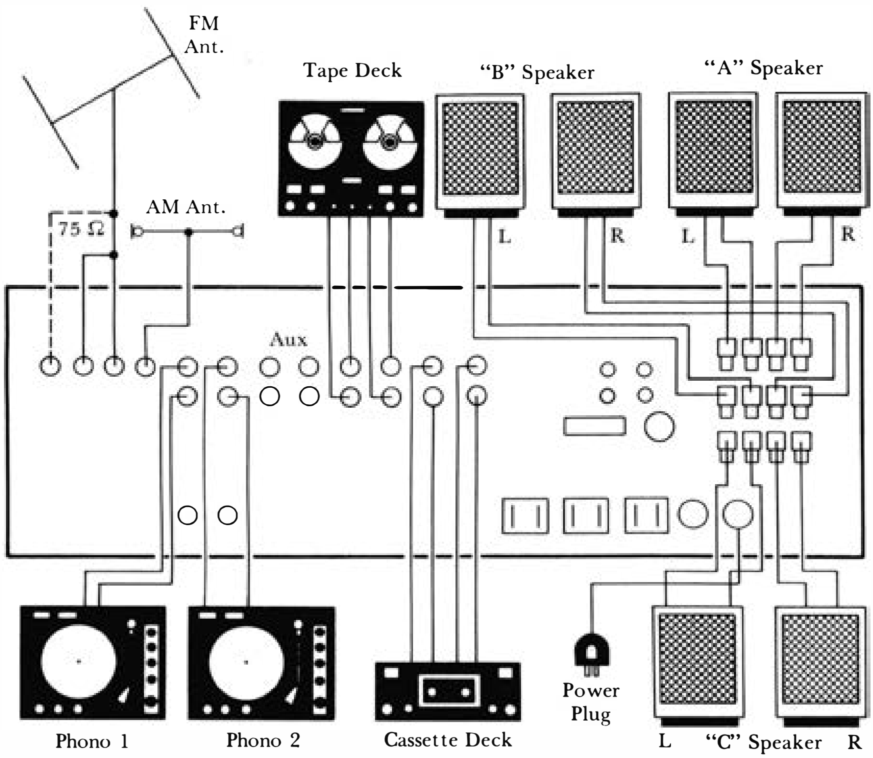
On the other hand, the smallest speaker in the group is called a *tweeter;* it serves to reproduce the high treble tones. Most speaker enclosures also include an intermediate size of speaker, termed a *midrange* or *"squawker"* unit. It functions to reproduce the intermediate tonal range between the low bass and high treble tones. Some enclosures contain a pair of midrange speakers, one of which is larger than the other. As a general rule, the size of a speaker is proportional to the amount of acoustic power that it can radiate. Speakers in an enclosure operate in association with crossover networks that direct suitable ranges of audio frequencies to each speaker.

These speakers, with their associated electrical networks in an enclosure, are called a speaker system. Level controls may be provided for the tweeter and/or midrange speaker to achieve correct tonal balance.

Stereo headphone jacks are ordinarily provided in stereo amplifiers and compact units. A pair of stereo headphones is illustrated in Fig. 1-4. Some hi-fi connoisseurs prefer headphones because of their acoustical characteristics. Other audiophiles utilize headphones for privacy. Various amplifier input facilities are provided. As an illustration, an input jack is

Figure 1-4. Pair of high-quality stereo headphones (*courtesy Radio Shack, a Tandy Corporation Co.*) and a headphone amplifier (*courtesy Heath Co.*).

ordinarily provided for an FM/AM tuner; another jack, with associated frequency compensation, is provided for a reel-to-reel tape deck; still another input jack, with an appropriate frequency-compensating network, is generally provided for a cassette player. High-fidelity amplifiers are also designed with various operating features, such as bass and treble tone controls, which may be supplemented with loudness controls. Terminals are often provided for additional speakers. A stereo balance control is usually included, and auxiliary input facilities, such as for microphones, are generally provided. Typical interconnections for a hi-fi system are shown in Fig. 1-5. Hi-fi connoisseurs who make their own tape recordings require an amplifier that provides an appropriate stereo signal for a particular tape recorder.



(a)

Figure 1-5. Typical interconnections for a hi-fi stereo system that utilizes an integrated receiver: (a) cable and lead connections. (*Courtesy Radio Shack, a Tandy Corporation Co.*)

Figure 1-5. (cont.) (b) Appearance of a high-quality integrated receiver.

A receiver for a hi-fi system is basically a tuner; it must be supplemented by an amplifier to operate a speaker system. A component system may be designed with a separate tuner and a separate stereo amplifier. All stereo tuners include a multiplex decoder section to reconstitute the L and R audio signals from the incoming encoded FM signal. A tape recorder provides both recording and playback facilities, whereas a tape player lacks recording facilities. A deck lacks a built-in power amplifier; it operates with an external amplifier and speaker system. Tape decks may or may not include recording facilities. Tape recorders are designated as monophonic, stereophonic, or quadraphonic types. Audiophiles tend to prefer reel-to-reel machines over cartridge or cassette-type machines. Eight-track-cartridge tape players, however, are popular, owing to their compactness and simplicity of operation. Most eight-track-cartridge tape machines are designed as player decks. Otherwise stated, a player deck lacks recording facilities. All eight-track tape players provide stereo reproduction, and many qualify as high-fidelity units.

1-2 BASIC ACOUSTICS

Any room produces reverberation and it will exhibit one or more resonant frequencies. Room reverberation time depends upon the amount of sound-energy absorption that occurs. Reverberation time is defined as the time that is required for a sound burst to decay 60 dB from its initial intensity. A result of reverberation is partial cancellation of particular audio frequencies at certain locations within the room. A room's resonant frequency is determined chiefly by its dimensions. If the walls, floor, and ceiling of a room are acoustically reflective and in turn produce substantial echoes, the sound wavefront requires a comparatively long time to decay, and the sound energy that started from a particular location then reaches the lis-

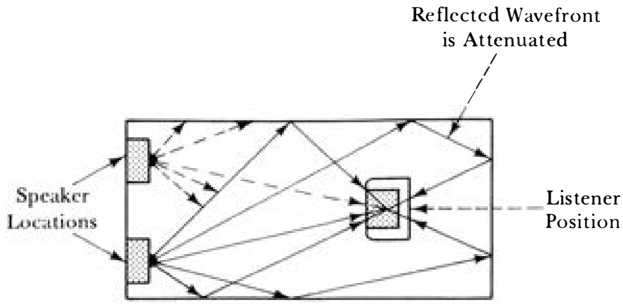


Figure 1-6. Reverberation shatters, redirects, and delays the sound wavefront.

tener from many directions, as depicted in Fig. 1-6. Note that the articulation of an audio signal is impaired by an excessively long reverberation time.

It is recognized that the human ear is tolerant of a reasonable amount of reverberation. In fact, an anechoic room that has no reverberation sounds "unnatural" to the listener. Although a sound may reach the listener from two different directions, he/she will perceive the source of the sound as in the direction of the louder wavefront, provided only that the time delay between the arrival of the two wavefronts is less than 25 milliseconds (ms). If this time delay is much greater, the listener then perceives two different sources for the sound (*Haas effect*). The velocity of sound in air is slightly greater than 1000 feet per second (ft/s). Reverberation time can be controlled by use of various kinds of materials within the listening area. Hard materials reflect a large amount of sound energy and thereby increase the reverberation time. On the other hand, soft materials absorb a large amount of sound energy and thereby decrease the reverberation time. It is instructive to note the comparative acoustic absorption coefficients listed in Fig. 1-7.

Material	*Coefficient*
Open Window	1.00
Special Acoustic Materials	0.30 to 0.9
Hair Felt	0.58
Carpets	0.15 to 0.2
Smooth Wood	0.04
Plaster	0.033
Glass	0.027
Brick	0.025

Figure 1-7. Acoustic absorption coefficients of common substances.

Furnishings in a room should include an adequate area of soft materials (carpeting, upholstered furniture, drapes, and so on) to absorb an appreciable amount of sound energy. However, there should also be provided sufficient hard materials, such as exposed walls, glass windows, or tiled surfaces, to provide adequate echoes for a "liveness" comparable to that of a concert hall. A typical acoustic treatment of a listening area is exemplified in Fig. 1-8. Observe that large areas of glass, such as picture windows, should be covered with thick drapes. Large bare walls should be partially covered with some sound-absorbent material, such as drapes. If possible, avoid covering the entire floor with thick carpeting. Thick scatter rugs may be placed at various locations as required for optimum acoustics. Upholstered furniture may be placed about the room, but it should not be pushed up against the walls. A ceiling ordinarily requires no special attention, inasmuch as the majority of reverberations take place from the walls.

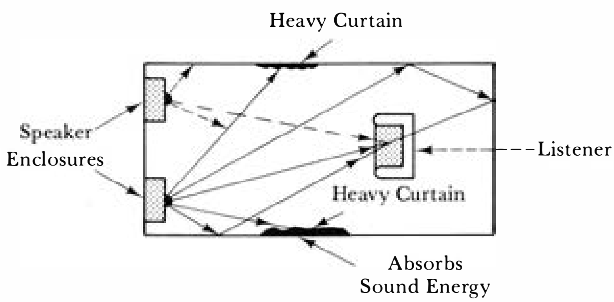


Figure 1-8. Reduction of reverberation time with heavy curtains.

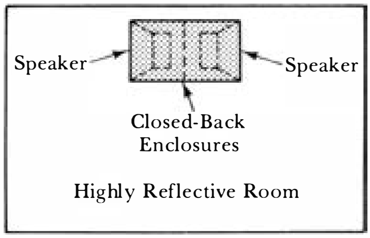


Figure 1-9. Closed-back enclosures installed in a highly reflective room.

Some types of room structure, exemplified by rumpus rooms, are ordinarily highly reflective. In turn, it is preferable to utilize closed-back speaker enclosures and to place them back to back, as depicted in Fig. 1-9. On the other hand, either open-back or closed-back speaker enclosures are suitable for installation in a "dead" room. It is desirable to place stereo speakers at floor level in the corners of a room, as pictured in Fig. 1-10. This placement has an acoustic aspect in which the walls and floor function as a semihorn structure. If the corners in a room are inaccessible at floor level, the next best procedure is to mount the speaker enclosures in the corners at ceiling level. Unless very small enclosures are necessary, there is no significant advantage in locating the enclosures at ear level. An exception may be noted, however, in the case of a speaker system that has highly directive high-frequency (tweeter) output. In this situation, high audio frequencies may not be satisfactorily reproduced unless the enclosure is installed at ear level.

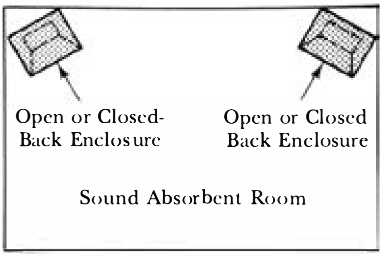


Figure 1-10. Open- or closed-back speaker enclosures installed in a highly sound-absorbent room.

Bookshelf-type speaker enclosures require the least installation space and are the least efficient form of enclosure. Note that minimum bass-frequency output occurs when the enclosure is located at the center of a wall. This is a basic consideration when a bookshelf-type speaker is installed, because this design of enclosure has comparatively poor bass reproduction. Its low-frequency reproduction can be improved by positioning the enclosure close to the side walls of the room, as exemplified in Fig. 1-11. As noted previously for larger types of speaker enclosures, a bookshelf-type enclosure will have improved bass reproduction if it is placed in a corner of the room at floor level. Sometimes, it is found that this positioning impairs high-frequency audio reproduction. This possibility depends upon the dispersion

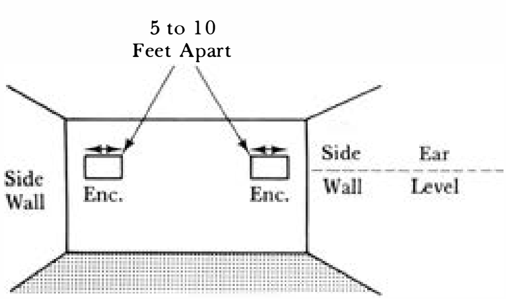


Figure 1-11. Bass output is improved by installing the enclosures nearer the side walls.

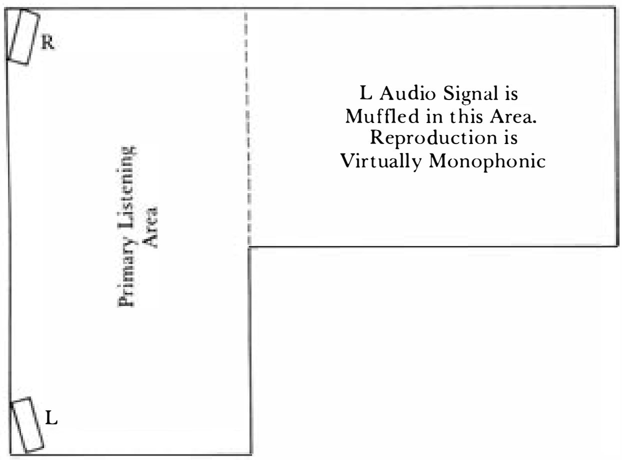


Figure 1-12. This room shape cannot be covered completely by a pair of enclosures.

characteristics of the tweeter. If an enclosure is installed at ceiling height, optimum reproduction across the entire audio-frequency spectrum is usually obtained with the enclosure mounted "upside down." This orientation places the woofer unit nearer the corner than otherwise. It is sometimes found that treble response can be improved with an enclosure lying on its side instead of standing upright. In the absence of prior experience, it is advisable for the installer to experiment with various enclosure orientations.

Room shapes vary, and this factor can affect both the enclosure placement and the enclosure complement. For example, a common "problem" listening area is the L-shaped arrangement exemplified in Fig. 1-12. Because of partial isolation from the left speaker, the entire room cannot be covered satisfactorily by the outputs from a pair of stereo speakers alone. In other words, if the R and L speakers are installed as shown in the diagram, the L signal reproduction will be impaired over approximately 50 percent of the room area. One solution to this problem is to simply designate the poorly covered area as "off limits" for stereo listening, as exemplified in Fig. 1-13. This solution has a substantial disadvantage in that the potential listening area is substantially restricted. Improved coverage can be obtained by placing the enclosures as shown in Fig. 1-14. Again, however, this installation has a significant disadvantage in that the excessive separation of the L and R enclosures produces a "hole in the middle" sound pattern.

This foregoing disadvantage can be largely overcome by increasing the enclosure complement with the supplementary midrange speaker depicted in Fig. 1-15. Note that a preamplifier may have an "L + R" output

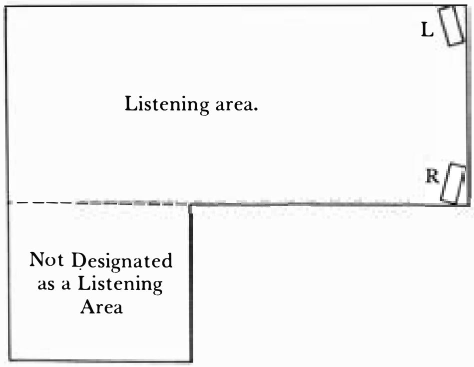


Figure 1-13. Simple but incomplete solution to the problem of the L-shaped room.

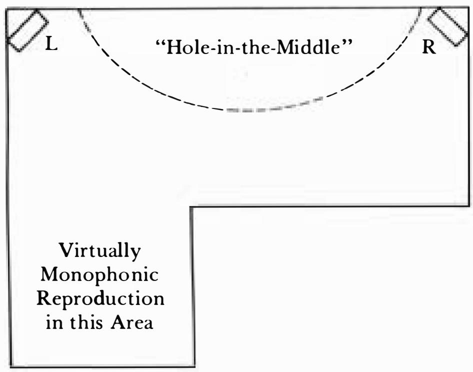


Figure 1-14. This speaker placement develops a sound pattern with a "hole in the middle."

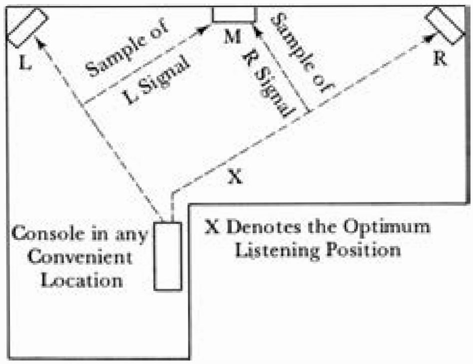


Figure 1-15. Monophonic speakers can be employed to eliminate the "hole-in-the-middle" effect.

connector for this purpose. This speaker reproduces a monophonic signal obtained by mixing equal portions of the L and R signals. A separate level control is provided for the mono speaker, so that its output can be adjusted as required for the most satisfactory sound reproduction. The optimum listening position in this arrangement is approximately at point X. A listener to the left of this point will hear more L signal than R signal. Conversely, a listener to the right of this point will hear more R signal than L signal. Similar restriction on the optimum listening position will be found in almost any stereo enclosure installation.

1-3 QUADRAPHONIC ENCLOSURE PLACEMENT

Quadraphonic sound systems include four enclosures, as pictured in Fig. 1-16. These enclosures are designated as right-front (RF), left-front (LF), right-rear (RR), and left-rear (LR). In view of variations in room acoustics and shapes, the optimum enclosure positions and orientations must usually be determined by experiment. As explained previously for stereophonic installations, each quadraphonic installation will be characterized by a primary listening area and an optimum listening position. A listener who does not occupy this comparatively limited area will experience impaired quadraphonic reproduction. In an extreme situation, the listener might be seated close to the right-front enclosure; in turn, he/she will experience monophonic reproduction for all practical purposes. Examples of enclosure positions and their orientations, with primary listening areas designated, shown in Fig. 1-17.

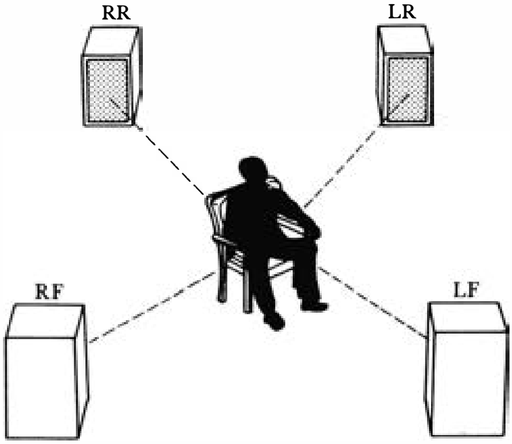


Figure 1-16. Four speakers are utilized in a quadraphonic sound system.

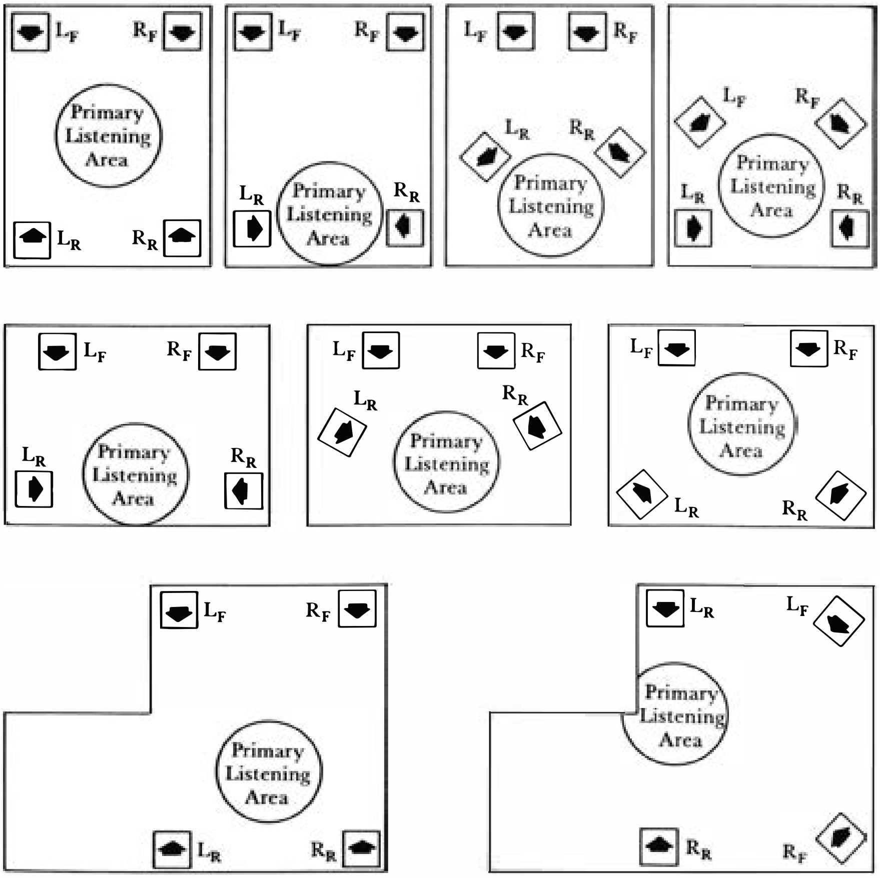


Figure 1-17. Quadraphonic speaker positions and directions for various types of rooms. (*Courtesy Marantz*)

Surround-sound is essentially the same as quadraphonic sound. In its simplest form, it is installed with *Dyna* speaker connections. Basically, this arrangement connects a pair of rear speakers across L and R stereo lines so that canceling voltages are applied to the speakers. However, the cancellation is not complete, because the L and R signals have somewhat different waveforms. In other words, the rear speakers reproduce the difference between the L and R signals. This is called a *differential* mode of surround-sound reproduction. Its theory of operation states that this stereo difference signal comprises ambient sound components that are present in the recording hall. Otherwise, stated, the Dyna signal comprises any stereo signal that does not occur in the L and R channels simultaneously. Some of these difference signals are direct program signals; others are components that are reflected by walls, floor, ceiling, and furnishings in the recording hall. Technically, these difference signals are termed *ambient sound signals*.

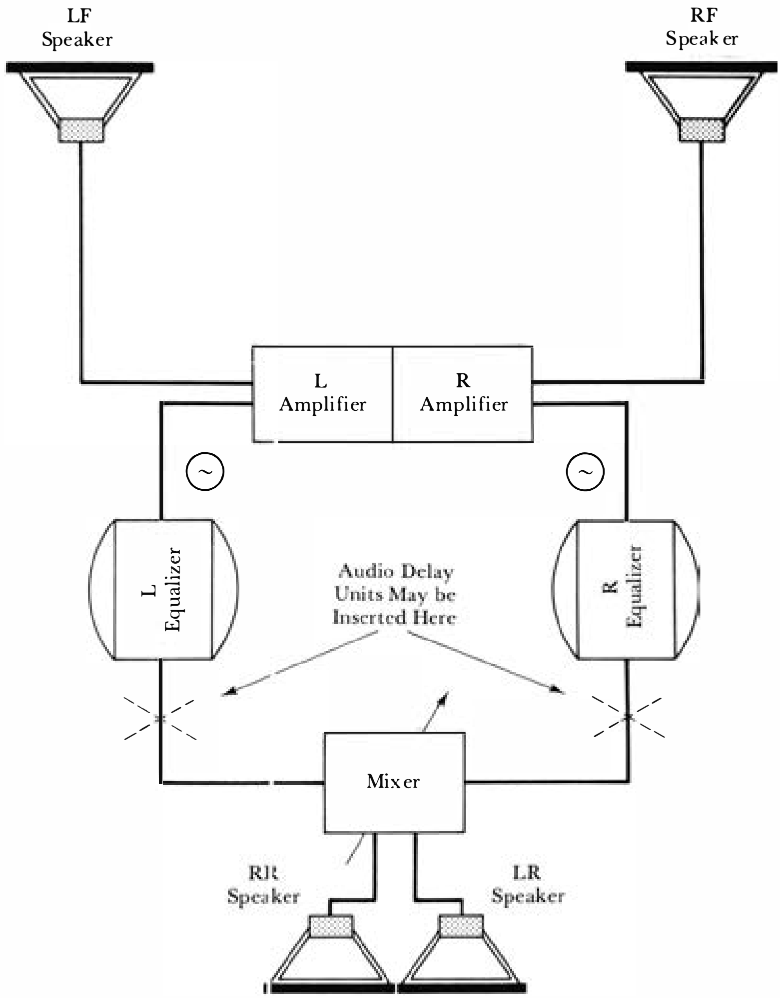


Figure 1-18. One form of synthesized quadraphonic sound arrangement.

A basic surround-sound interconnection arrangement is shown in Fig. 1-18. When Dyna-connected speakers are placed behind or to one side of the listener, he/she experiences an "enhanced-stereo" reproduction of the original sound. Note that it is a difficult technical problem to correctly reproduce inside a small room the reverberation from a large concert hall. In other words, the reverberation time of a small room is comparatively short. Apart from this problem, various stereo recordings have not been produced with surround-sound reproduction in mind. For these reasons, many audiophiles prefer to characterize the Dyna speaker connection as a quadraphonic *synthesizer* arrangement. This term denotes that the rear speakers do not necessarily reproduce true quadraphonic sound components. An elaborate synthesizer installation generally includes null, balance, and volume adjustments. In addition, audio delay may be provided in the synthesizer output lines. These functions permit the output from each rear speaker to be optimized for simulated quadraphonic sound reproduction.

1-4 MIDRANGE AND TWEETER SPEAKER CONTROLS

Previous mention was made of midrange and tweeter speaker controls. These are of basic concern to the installer. Most bookshelf-type speaker systems include these controls for adjustment of relative output levels for the bass and treble ranges. In addition to level adjustment, frequency adjustment may also be provided in the form of equalizers. An equalizer is an elaborated type of tone control that permits the listener to increase or to decrease the relative output level over selected intervals of the audio-frequency spectrum. A variation of approximately 12 dB is provided by the equalizer over each section of the frequency spectrum, as depicted in Fig. 1-19. Equalization enables the listener to compensate the speaker system for various room acoustics.

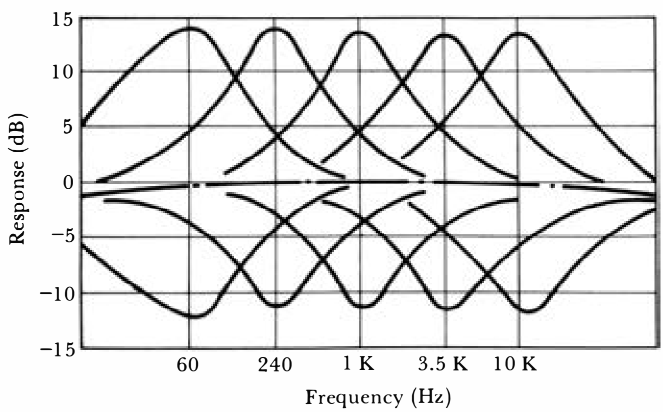


Figure 1-19. Representative equalizer frequency responses.

Balance controls are provided on preamplifiers for adjustment of relative sound output from the L and R speakers. The simplest procedure is to set the FM tuner between two stations, so that only random-noise output is heard from the speaker system. Tone controls on the preamplifier should be set to their nominally flat positions. The loudness or volume control is set for comparatively low-level output. With his/her ears approximately 1 ft away from the speaker enclosure, the listener slowly advances the midrange level control to a point such that the sound seems to be coming from one source instead of separate midrange and woofer sources. Similarly, the tweeter level control is slowly advanced to a point such that the sound seems to be coming from one source instead of separate midrange and tweeter sources. Finally, the listener moves to the primary listening area and adjusts the balance control on the preamplifier to a point where the random noise seems to have equal levels from both the L speaker and the R speaker.

1-5 SPEAKER POWER RATINGS

Speakers and amplifiers may have root-mean-square (rms) or sine-wave power ratings, or music (pulse) power ratings, or both. If a power rating is stated in watts, without any qualifications, it is understood that the rating is referenced to rms values of sine waves. This is a continuous or steady power rating; it represents the maximum power level that can be sustained indefinitely without damage to the equipment. On the other hand, a music-power rating is referenced to the peak-power value of a short pulse. This is a transient or discontinuous power rating; it represents the maximum peak-power level that can be withstood briefly without substantial distortion or damage to the equipment. A music-power rating relates to sudden surges or attacks in musical waveforms. Of course, neither an amplifier nor a speaker can be operated continuously at its rated music-power level without incurring damage to the equipment.

There is no definite relation between the rms power rating and the music-power rating of an amplifier or of a speaker. However, the music-power rating is always greater than the rms power rating. For example, an amplifier that is rated for an output of 100 watts (W) rms might be rated for 150 W of music power; another amplifier with a rating of 75 W rms might have a rating of 125 W of music power. Speakers are usually rated for rms power capability, although some are rated for music-power capability. A speaker will often have both a minimum power rating and a maximum power rating. The minimum power rating denotes the lowest power level at which satisfactory performance can be obtained in a small listening area. On the other hand, the maximum power rating denotes a power-input level above which the speaker is very likely to be damaged. Note in passing that the electrical efficiency of a cone-type speaker is quite low; efficiency values range from 1 to 10 percent, depending upon the design of the enclosure.

A speaker that is rated for a minimum power input of 6 W may be rated for a maximum power level of 100 W. Note that the power rating of a speaker system is equal to the sum of the power capabilities of the individual speakers. For example, if a two-way speaker system is utilized, the woofer might be rated at 100 W and the tweeter might be rated at 35 W. In turn, the power rating of the system would be stated as 135 W. It is evident that if a power input of 100 W were applied to the speaker system at a frequency of 10 kHz, the tweeter would be damaged. Again, if a power input of 135 W were applied to the speaker system at a frequency of 100 Hz, the woofer would probably be damaged. Therefore, various speaker manufacturers include fast-blow fuses in a speaker system, to avoid this danger of overload damage. Fuses in a speaker system may blow as a result of "flicking" a bit of lint from a phono stylus with the operator's fingernail.

If a speaker has a lower power rating than its associated amplifier, operating difficulties can be anticipated. In other words, if the listener turns up the volume control past a certain limit, the speaker system will suddenly become partially or completely "dead," owing to blown fuses. Or, if the speaker system is not fused, either mechanical and/or electrical damage will result to one or more of the speaker units. Therefore, it is good practice to select a speaker system that has somewhat greater power-input rating than the maximum power-output rating of the associated amplifier. It is important for the beginner to keep in mind that rms power ratings are consistently less than music-power ratings, and to avoid confusion between the two.

Consider the basic features and characteristics of a speaker system. A high-quality speaker enclosure (system) is illustrated in Fig. 1-20. Two of these enclosures would be used in a stereo system, and four enclosures would be used in a quadraphonic system. Installers sometimes prefer to employ a pair of less-costly enclosures for the right-rear and left-rear positions in a quadraphonic system. This is a matter of individual preference. In any case, matched pairs of enclosures should be utilized for the front positions and matched pairs for the rear positions. The illustrated speaker contains two 8-inch (in) woofers and a 3-in tweeter, and is rated for 75 W of peak program power. The frequency response of the speaker system with its crossover network is rated for a uniformity of ± 3 dB from 55 Hz to 18 KHz; its crossover frequency is 3500 Hz. A three-way (normal, high, low) L pad is provided for adjustment of treble response to match the acoustics of the listening area.

Note that a high-fidelity speaker system is sometimes interconnected with an electronic organ. In such a case, the speaker system should be derated by 50 percent, to avoid possible overload damage. This means that if the organ is rated for an output of 80 W, the speaker system should be rated for a power input of 120 W. Otherwise stated, the average power level in an organ output signal is often appreciably greater than the average power level in tape, record, and FM tuner signals. Strong bass tones are produced by large organ pipes, oboes, and so on. In turn, adequate bass reproduction requires comparatively large speaker cones. Loud bass reproduction also requires comparatively great displacement (travel) of a cone. Extensive displacement is provided by cones with "cloth roll suspension," or "soft suspension." On the other hand, the large cones are too heavy to vibrate efficiently at high audio frequencies. Hence, the woofer needs to be supplemented by a tweeter, or possibly by both a tweeter and a midrange speaker.

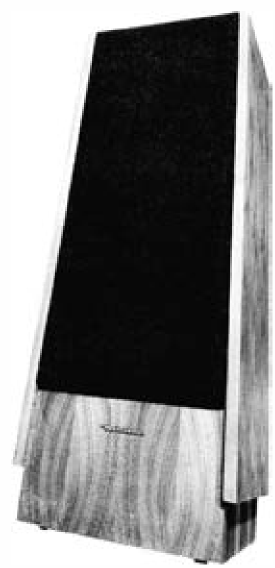


Figure 1-20. High-quality speaker enclosure for high-fidelity reproduction. (*Courtesy Radio Shack, a Tandy Corporation Co.*)

Speaker Phasing

It is essential that speakers in a system, or enclosures in a system, be connected in phase with one another. In other words, if speakers or enclosures are connected out of phase, the radiation from one unit will tend to cancel acoustic radiation from an oppositely phased unit. This cancellation is most apparent for bass tones. To operate in phase, the speaker cones must move in the same direction at the same time when a voltage is applied at the input of the speaker system. Consider the basic examples shown in Fig. 1-21. Most speakers (and speaker enclosures) have a red dot, or equivalent symbol, for checking system phase relations. In the case of two parallel-connected speakers, the red-dot terminals of the speakers are connected together. The same principle applies to parallel-connected enclosures. Or, in the case of a series connection, the red-dot terminal on one speaker (or enclosure) is connected to the black-dot terminal on the other speaker (or

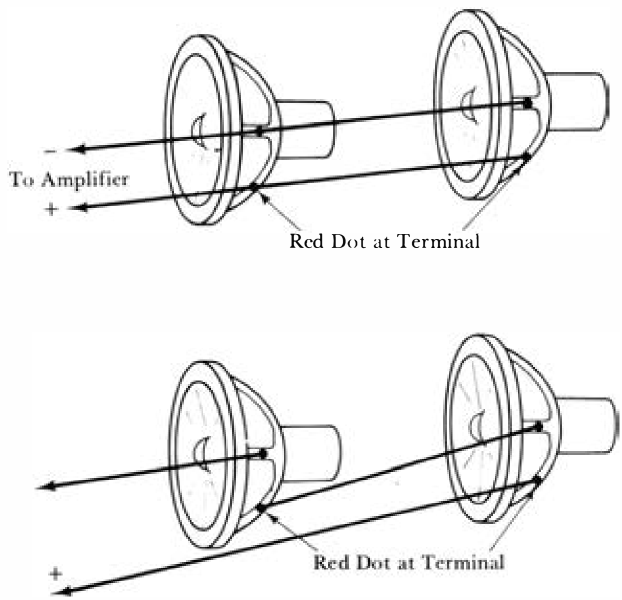


Figure 1-21. Examples of speaker phasing: (a) parallel connection; (b) series connection.

enclosure). Note that most enclosures have a terminal board mounted on the rear of the structure. However, some enclosures have a terminal board recessed into the bottom of the structure to accommodate concealed wiring.

Consider next the principles of speaker or enclosure interconnections. The speakers or enclosures in a system may be operated in parallel, for example as shown in Fig. 1-22. In this example, each speaker has a rated input impedance of 8 ohms (Ω). Two 80-Ω speakers connected in parallel have a net input impedance of 4 Ω. Again, four 8-Ω speakers connected in parallel have a net input impedance of 2 Ω. It is instructive to consider the net input impedance of two speakers connected in parallel, one of which has a rated input impedance of 8 Ω, whereas the other speaker has a rated input impedance of 4 Ω, as depicted in Fig. 1-22(c). These two impedance values have a net value of 2.67 Ω. Note carefully that the 4-Ω speaker will draw twice as much audio current as the 80-Ω speaker. This means that if both of the speakers have the same cone diameters and the same efficiency, the 4-Ω speaker will radiate much more acoustic energy than the 8-Ω speaker. This would be an undesirable condition in most system arrangements.

Speakers or enclosures are interconnected in various ways in order to obtain an input impedance that matches the rated output impedance (load impedance) of the associated amplifier. Many amplifiers are designed to operate into a load of 8 Ω; however, others are designed for load impedances of 4 Ω, or of 16 Ω. With reference to Fig. 1-23, speakers (or enclosures)

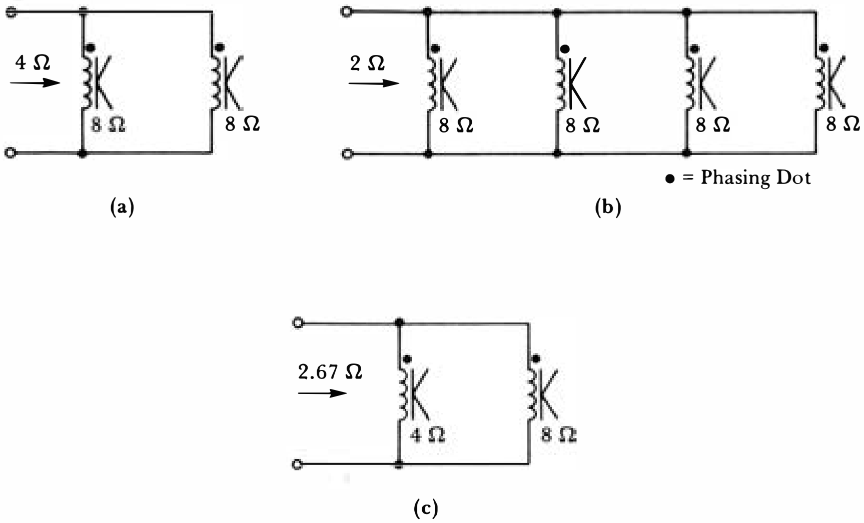


Figure 1-22. Examples of parallel-connected speakers: (a) two 8-Ω speakers in parallel; (b) four 8-Ω speakers in parallel; (c) 4-Ω speaker in parallel with an 8-Ω speaker.

may also be connected in a series arrangement. This mode of operation results in a net input impedance that is greater than the input impedance of an individual speaker. Or, the system input impedance will be equal to the sum of the impedances of the individual speakers. Note that if a low-impedance speaker (or enclosure) is connected in series with a high-impedance speaker (or enclosure), the audio voltage drop will be greater across the latter than across the former. In turn, if both speakers (or enclosures) have the same cone diameters and the same efficiencies, the high-imped-

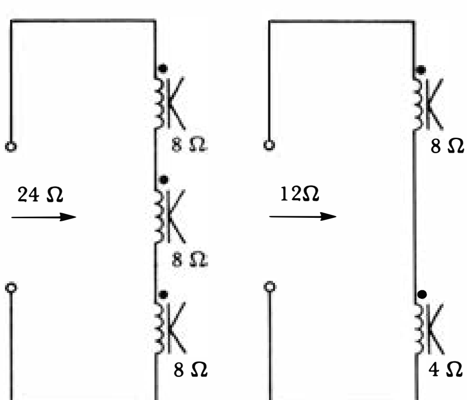


Figure 1-23. Examples of series interconnection.

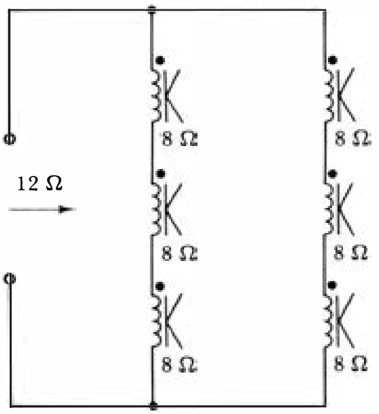


Figure 1-24. Series–parallel interconnection arrangement.

ance speaker (or enclosure) will radiate more acoustic energy than the low-impedance unit. Again, this would be an undesirable condition in most system arrangements.

Still another basic interconnection arrangement is shown in Fig. 1-24. This is a series–parallel configuration. Each of the speakers in this example has a rated input impedance of 8 Ω. Accordingly, each series string has a net input impedance of 24 Ω, and the two parallel-connected series strings have a net input impedance of 12 Ω. If all the speakers are identical, each will consume the same amount of audio power. On the other hand, if the speakers have different impedances, they will draw different amounts of audio power. The principles outlined above can be applied in this situation to determine the relative power values drawn by each speaker.

Speaker System Power Requirements

Consider a listening area with a medium or average acoustic environment. If the room has a volume of 2000 to 3000 ft^3, it can be served adequately by speakers with a power-input level of 15 W rms per channel. Acoustic-suspension speakers are assumed in this example. Next, if a room with a medium acoustic environment has a volume of 4000 ft^3, speakers rated for an audio power input of 20 W rms per channel should be utilized. Finally, a very large listening area with a medium acoustic environment and a volume of 8000 ft^3 will require speakers rated for an audio power input of 45 W per channel. Of course, larger speakers may be installed in any case, if reserve power output is desired. In a "lively" acoustic environment, the foregoing power requirements can be reduced 50 percent. On the other hand, in a "dead" acoustic environment, the foregoing power requirements should be increased by 50 percent.

Interconnections of Auxiliary Speakers

Speaker enclosures located at an appreciable distance from the main speaker enclosures are termed auxiliary speakers. Auxiliary speakers may be installed in various rooms of a residence, in a patio, or in an adjacent building. Interconnections of speakers or speaker enclosures to maintain a particular input impedance are made as explained above. In addition, it is generally desirable to include switching facilities for auxiliary speakers, as exemplified in Fig. 1-25. When auxiliary speakers are to be operated at a

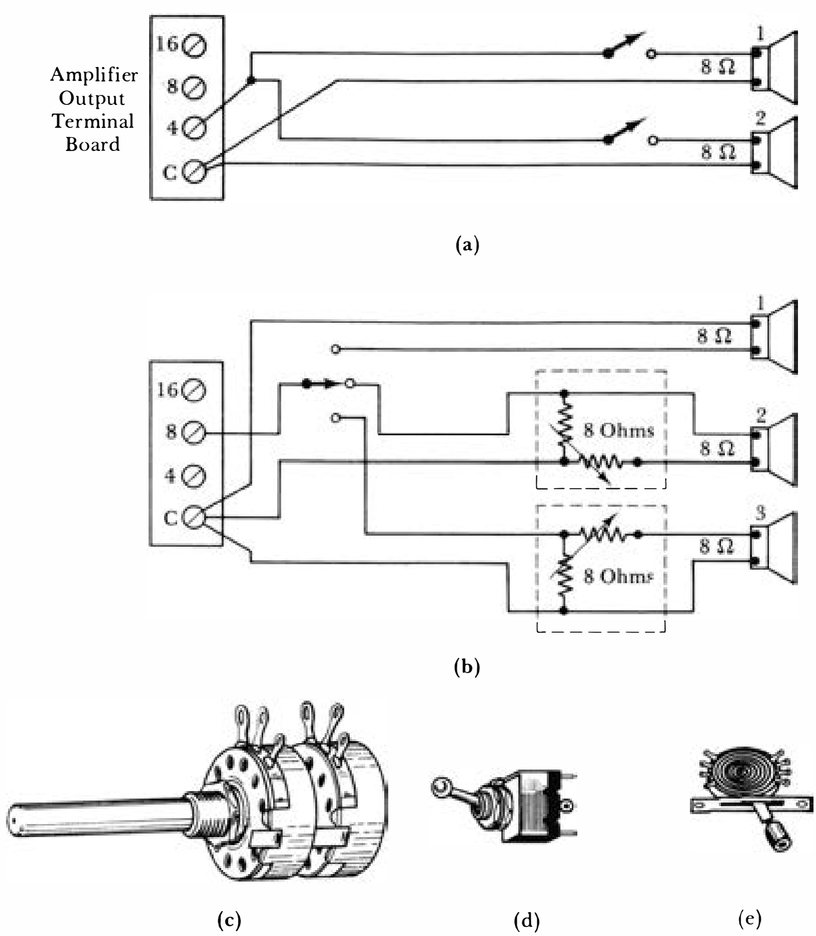


Figure 1-25. Typical auxiliary speaker circuitry: (a) two remote speakers operated individually or in parallel in a 4-Ω circuit; (b) three remote speakers operated individually in an 8-Ω circuit, L pads included in two of the speaker circuits; (c) appearance of an L pad; (d) toggle switch; (e) rotary switch.

lower level than the main speakers, L pads are included in the auxiliary circuits, as shown in the diagram. Auxiliary speaker installation procedures are detailed subsequently.

Although mismatches between the power amplifier and its speaker system are not a highly critical consideration, there is a hazard encountered in the event that a loudspeaker system presents a load impedance of less than 4 Ω to a representative power amplifier. A load value less than 4 Ω will often cause the output transistors to overheat and can result in catastrophic failure. If multiple speaker systems are configured in suitable series or series–parallel arrangements, as previously explained, a load value less than 4 Ω can be avoided, regardless of the number of speakers that are driven. Of course, an accidental short circuit in the speaker input circuit is very likely to cause the power-amplifier output transistors to burn out. Some power amplifiers are fused, and some have built-in automatic overload-protection circuitry. However, most power amplifiers are likely to be damaged if their output terminals are short-circuited.

Chapter 2

Basic Installation Procedures

2-1 GENERAL CONSIDERATIONS

Certain basic principles apply to installation procedures for any audio system. Consider the typical component stereo system depicted in Fig. 2-1. It comprises a record player, tape recorder, FM tuner with auxiliary multiplex decoder, a two-channel preamplifier, two-channel equalizer, two-channel power amplifier, and a pair of speaker enclosures. An external antenna is connected to the FM tuner. A microphone (two may be employed, if desired) is connected to the tape recorder. In this kind of system, the FM tuner, tape recorder, preamplifier, and power amplifier each has its separate power supply. Auxiliary multiplex decoders are not provided in modern audio systems. However, if you are concerned with an old-style arrangement, note that the multiplex decoder will also have a separate power supply.

Three different kinds of conductors are utilized in system interconnections. Most of the connections are made with *audio cable.* This is a shielded and stranded type of conductor; the shield braid that surrounds the central conductor is grounded to the chassis (or to the common power-supply bus) of the equipment. Thereby, undesirable pickup of hum and other stray fields is virtually eliminated. Note in passing that in spite of observance of all good practices in installation procedures, an occasional audio system will nevertheless pick up objectionable interference. This is a result of operation in a location that has abnormally high stray field intensity. Several methods are available to minimize or to eliminate objectionable interference, as explained subsequently.

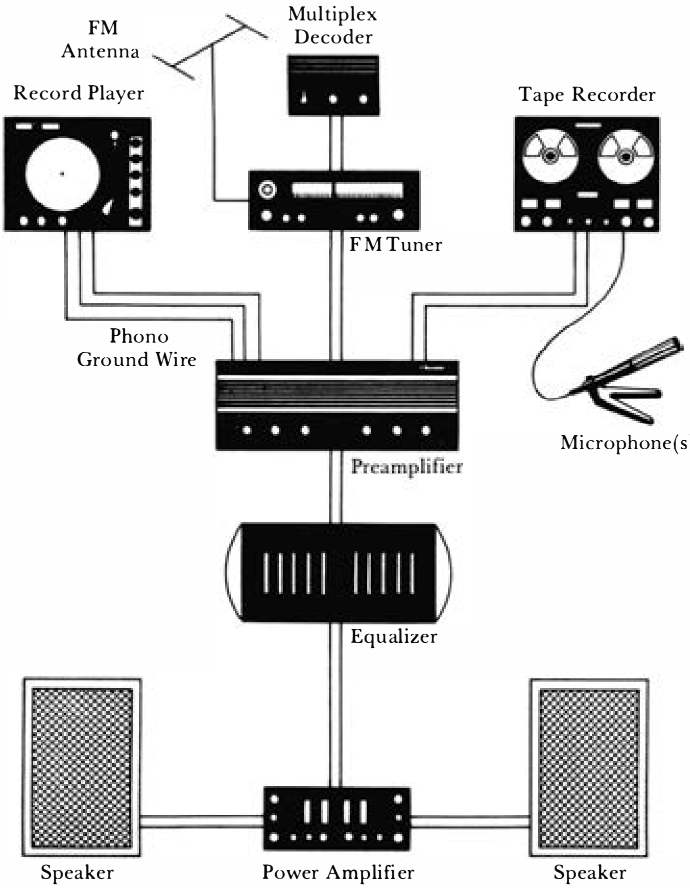


Figure 2-1. Plan of a typical component system.

An audio system can be divided in a general way into low-level circuits and high-level circuits. Thus, the output circuit from the tape recorder in Fig. 2-1 is a low-level circuit. Similarly, the output circuit from the record player is a low-level circuit. Low-level circuits are much more susceptible to hum and interference pickup than are high-level circuits. Therefore, the low-level circuits must be very well shielded, and the braid of the audio cables must be well grounded. In addition, it is general practice to connect a separate ground wire from the record player to the preamplifier, as depicted in Fig. 2-1. This is ordinarily a heavy copper wire that provides a low-impedance ground return for the phono circuit, and thereby minimizes the possibility of audible hum from this source.

By way of comparison, the outputs from the power amplifier are high-level circuits, and in the majority of situations, they are essentially immune to hum or interference pickup. In turn, unshielded conductors (such as conventional lamp cord) may be used to connect the speakers to the power amplifier. The lead-in from the FM antenna to the tuner often consists of twin lead, such as that employed in connecting a TV antenna to a television receiver. However, as will be detailed in a subsequent chapter, the FM antenna lead in may pick up objectionable static and interference in fringe areas unless shielded conductors are used. Power-supply leads from equipment components consist of conventional appliance or lamp cord. Some of these may be switched, and others may be unswitched, as will be explained.

2-2 INTERCONNECTION PRINCIPLES AND REQUIREMENTS

Various interconnection principles and requirements apply to any audio system; some of these are "common-sense" considerations, whereas others are not apparent to those who may be unfamiliar with the electrical characteristics of audio components. Basic cabling and wiring features for a stereo system are exemplified in Fig. 2-2. Connections are made from the several components to terminals on the terminal board of the main unit—in this case, an integrated amplifier. As noted previously, audio cable is used for interconnecting low-level circuits. If prepared cables are supplied with the system, the installer's task is simplified. Prepared cables are stripped of insulation at the ends of the conductors, and appropriate terminal lugs have been soldered at the ends of the conductors. In the event that you must work with stock cable, suitable lengths must be cut, and the ends of the cables must be prepared for connection, as depicted in Fig. 2-2(a) and (b).

When a conductor is connected to a screw-type terminal such as shown in Fig. 2-3(a), the end of the wire should be formed into an eyelet, as depicted in (b). In other words, the eyelet should be oriented so that it tends to close, instead of to open, as the screw is tightened. *Close attention should be directed to prevention of accidental short circuits between adjacent terminals.*

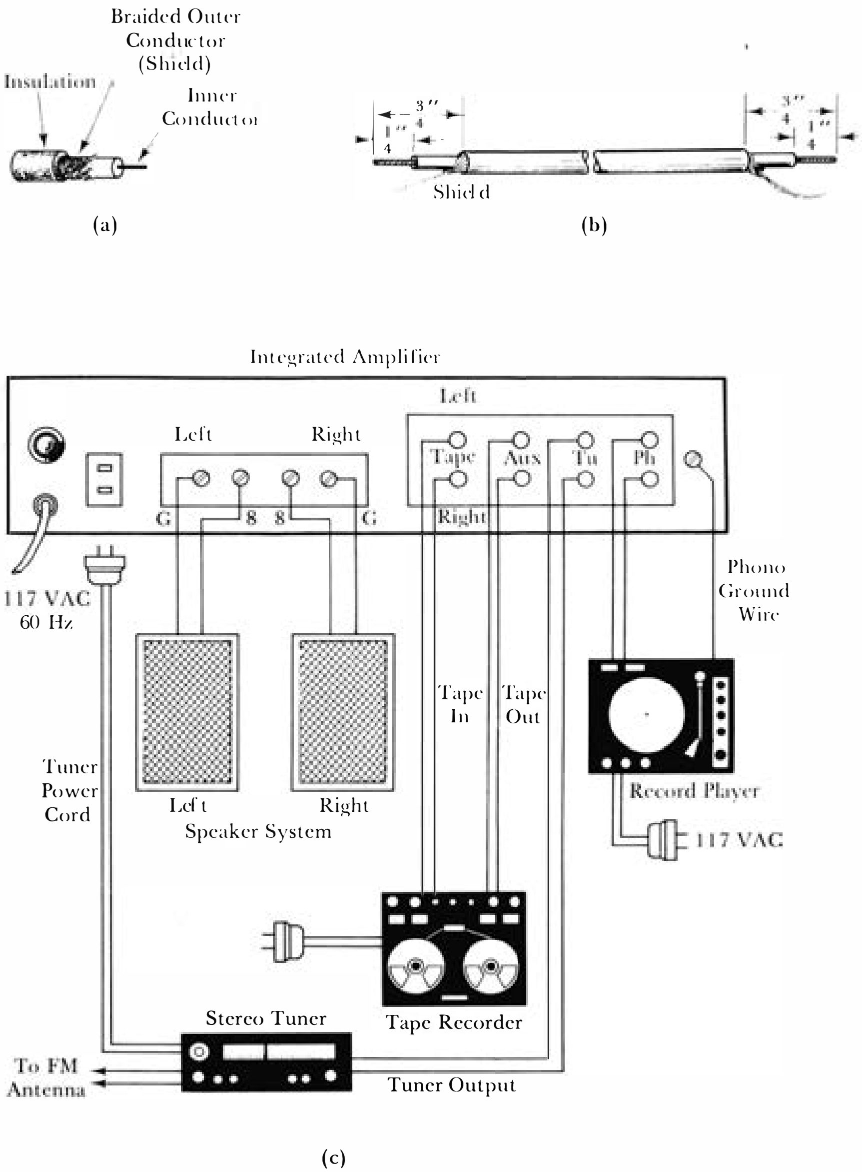


Figure 2-2. Basic cabling and wiring in a stereo system: (a) end of shielded cable with insulation stripped; (b) preparation of shielded cable for connection to terminals; (c) typical interconnections, all cables shielded except for wires to speakers (lamp cord is commonly used); a single copper phono ground wire is connected in addition to the chassis of the record player. (*Courtesy Kenwood*)

Figure 2-2. (cont.) (d) A high-quality integrated amplifier.

Particular care is necessary in the connection of braid of shielded cables to ground terminals. In other words, a stray "whisker" can easily escape notice and make a short circuit to an adjacent terminal. This fault makes the system inoperative, and there are a few situations in which short circuits can cause component damage. More workmanlike connections can be made if the ends of conductors are first soldered to spade lugs, such as pictured in Fig. 2-3(c). A small soldering gun such as that customarily used in electronic repair work is suitable.

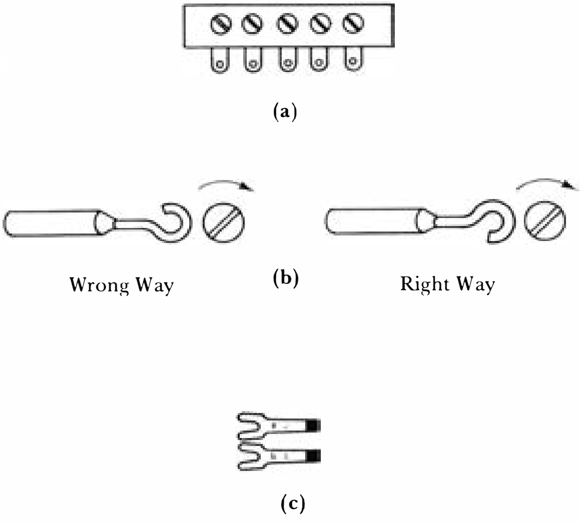
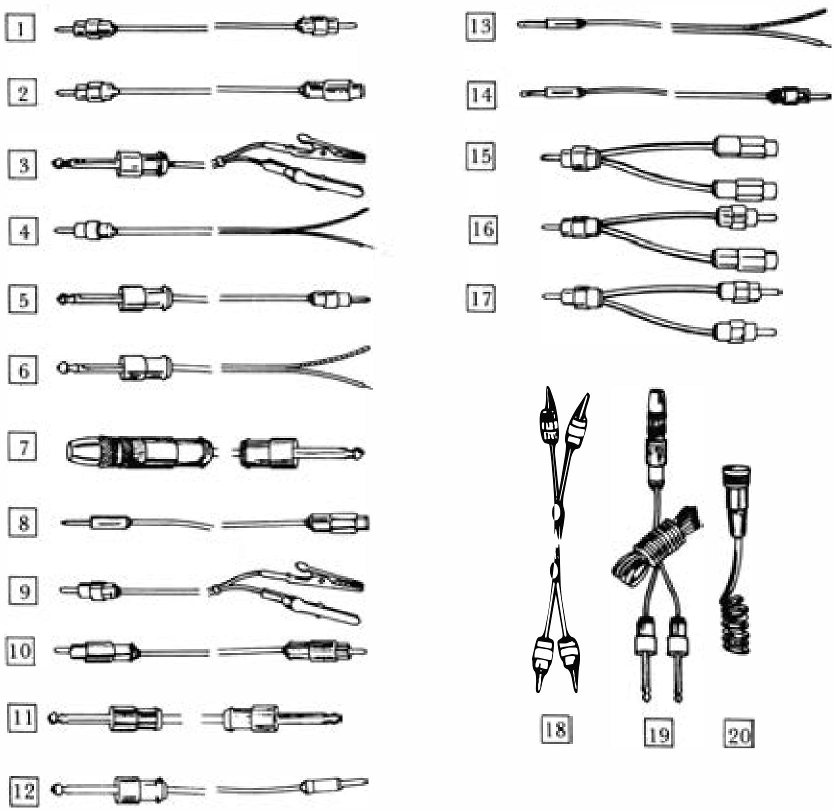


Figure 2-3. Terminal connections: (a) typical screw-terminal strip; (b) right and wrong ways of connecting a conductor to a screw terminal; (c) spade lugs may be soldered to the ends of conductors for connection to screw terminals.

Chart 2-1

AUDIO CONNECTORS



1. Phono pin plugs: 5 to 12 ft.
2. Phono pin/plug jack: 3 to 12 ft.
3. Phono plug/clips: 10 ft.
4. Phono pin plug: 3 to 15 ft.
5. Phono plug/pin plug: 6 to 10 ft.
6. Phono plug: 6 ft.
7. Phono plug/jack: 6 to 12 ft.
8. Mini phono plug/jack: 6 ft.
9. Mini phono plug/alligator clips: 10 ft.
10. Long handle pin plug: 15 ft.
11. Phono plugs: 6 to 10 ft.
12. Phono plug to mini plug: 3 to 6 ft.
13. Mini plug: 3 to 6 ft.
14. Mini plug to phono pin: 6 to 10 ft.
15. Adapter (pin plugs/jack): 4 ft.
16. Adapter (pin jacks/plug): 4 ft.
17. Adapter (pin plugs): 4 ft.
18. Pin plugs: 3 to 6 ft.
19. Phono plugs/3 con. jack: 4 ft.
20. Microphone connector: 6 ft.

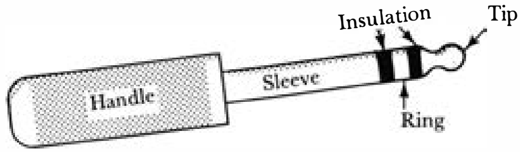
If a "package" component system is purchased, all necessary conductors will ordinarily be supplied. Cables, cords, and wires will have been cut to suitable lengths for the "average" installation, and appropriate plugs or connectors will have been soldered to the ends of the conductors to match the terminal facilities of the components (refer to Chart 2-1). The chief types of audio connectors are shown in the chart, and standard cable lengths are noted. Phono pin plugs and jacks are in very wide use. Phono plugs and miniature phono plugs will also be encountered. Alligator clips are ordinarily used only in experimental arrangements. Note that various adapters are available for mating various types of connectors. As an illustration, an adapter is utilized to mate a phono pin plug to a phono pin jack and to another phono pin plug. When unusually long "runs" are required, audio cables with suitable connectors can be connected in series. However, certain precautions must be observed, as explained under the circuit-impedance topic.

In addition to the basic types of audio connectors noted above, you may have occasion to use additional types of more specialized audio connectors, as follows:

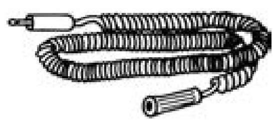
1. European-type Y adapter (five pins) for connecting DIN or Hirschman sockets to standard U.S. phono plugs and cables. A 6-ft length of cable is supplied.
2. European-type DIN patch cord; shielded cable with five-pin DIN male connectors on the ends of a 6-ft cable.
3. Phono plug to spade lugs.
4. Ninety-degree phono plug to phono plug.
5. Phone plug output to phono jack input.
6. Phone jack input to phono plug output.
7. Y adapter with three-conductor (stereo) phone jack to two phono plugs.
8. Y adapter with three-conductor (stereo) phone plug to two three-conductor phone jacks (see Fig. 2-4).
9. Y adapter with three-conductor (stereo) phone plug to two phono jacks.
10. Miniature phone plug to stripped leads.

(a)

Figure 2-4. Stereo earphones, with standard phone plug: (a) appearance (*courtesy Radio Shack, a Tandy Corporation Co.*).



(b)



(c)

Figure 2-4 (cont.) (b) Stereo phone plug terminology; (c) a 25-ft earphone extension cord with plug and jack terminations.

2-3 IMPEDANCE AND CURRENT-FLOW CONSIDERATIONS

As mentioned previously, it is sometimes desirable or necessary to use comparatively long cables or lines. In many situations, this is a matter of no technical concern. On the other hand, certain equipment characteristics may be encountered that limit the length of standard cable that can be used satisfactorily. As an illustration, some types of microphones will provide poor high-frequency response if they are operated with abnormally long shielded cables. Again, some types of phono cartridges are subject to the same limitation. These difficulties are the result of high internal-impedance characteristics of such microphones and cartridges. Next, consider the current demand of a high-power speaker system. The audio current that flows through a speaker system also flows through the loop resistance of the leads that connect the enclosure(s) to the power amplifier. Suppose that an 8-Ω speaker system is connected remotely to a power amplifier with a pair of leads that have a loop resistance of 8 Ω. In turn, half of the available audio power is dissipated (lost) in the resistance of the connecting leads.

To avoid difficulties in these categories, note first that microphones and phono cartridges with various impedance characteristics are available. Typical microphones are designed to work into impedances as low as 150 Ω and as high as 20,000 Ω. These are called *low-impedance* and *high-impedance microphones,* respectively. Consider a microphone of moderate impedance that works into ordinary microphone cable. As exemplified in Fig. 2-5, practically no high-frequency loss occurs in cable runs up to 20 ft. How-

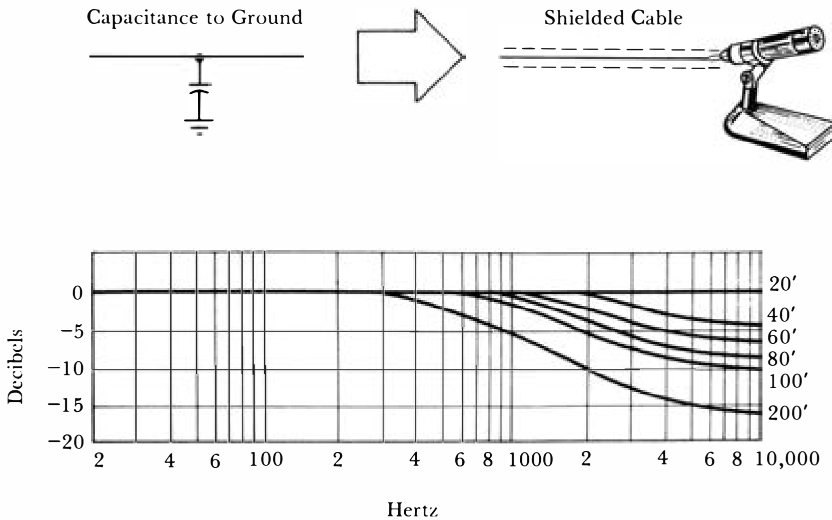


Figure 2-5. Frequency responses for a moderate-impedance microphone with various lengths of connecting cable.

ever, a 40-ft run incurs a loss of almost 5 dB at 10 kHz, compared with the output at low frequencies. This is approximately equivalent to a 50 percent loss at 10 kHz. If the same microphone is connected to a 200-ft cable, a loss of 5 dB occurs at 1 kHz, with a loss of more than 15 dB at 10 kHz. A 15-dB loss means that substantially less than 20 percent of the source voltage arrives at the output end of the cable. Therefore, the first guidepost to keep in mind is that low-impedance microphones should be used with unusually long cables. The same observation applies to phono cartridges.

When a transducer such as a microphone or a phono cartridge is supplied in a "package" audio system, the output level of the transducer will routinely meet the input requirement of the preamplifier. For example, a particular preamplifier may have an input connector for a low-impedance microphone, rated for a sensitivity of 350 microvolts (μV). This means that an input signal level of 350 μV will be required to drive the preamplifier to a normal output level (usually 1 V), and in turn to obtain normal sound output from the speaker(s) (see Fig. 2-6). Therefore, if another low-impedance transducer were utilized that has an output level of only 175 μV, the system would be "weak" and possibly unsatisfactory. On the other hand, if a low-impedance transducer with an output level of 700 μV were employed, abnormal sound output from the speaker(s) would occur unless the operator "backs off" on the volume-control setting. In most cases, resetting of the volume control is all that would be required. However, in the event that a transducer with a comparatively high output level overdrives the

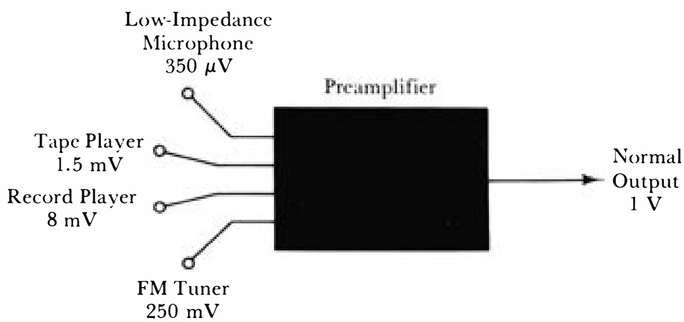


Figure 2-6. Typical rated input and output voltages for a preamplifier.

preamplifier, distortion will result. Therefore, the installer will occasionally need to check whether a particular transducer provides a reasonable "match" to the input requirements of the preamplifier to be utilized.

Consider next the practical aspects of long lines between speaker enclosures and power amplifiers. A speaker system is usually rated for 8 Ω of input impedance; some systems are rated at 16 Ω and some at 4 Ω. If a remote enclosure is connected to a power amplifier with a line that has appreciable resistance, it is evident that, from the viewpoint of power efficiency, a 16-Ω speaker system would be preferred to a 4-Ω system. In other words, power is equal to the product of voltage and current; a 16-Ω speaker draws only one-fourth as much current as a 4-Ω speaker at a given voltage level (see Fig. 2-7). Since the power loss in the connecting line increases as the square of the current flow, the advantage of a reduced cur-

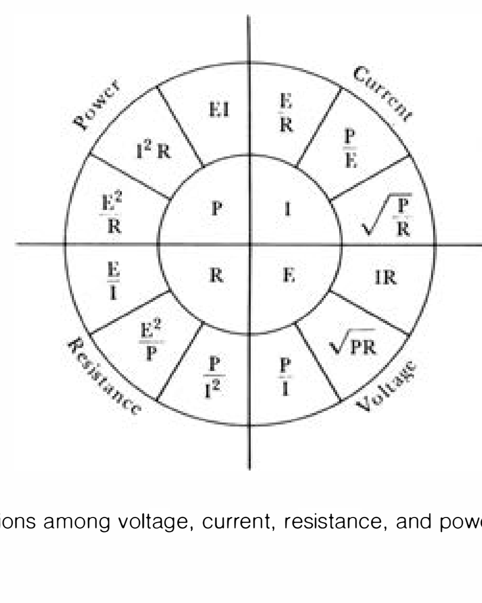


Figure 2-7. Relations among voltage, current, resistance, and power.

rent level is apparent. Efficiency can also be increased by utilizing conductors with ample cross section. As a general guideline, it may be observed that a line to a speaker at a distance of 50 ft will have a resistance of approximately 0.3 Ω if No. 14 wire is used, it will have a resistance of 0.65 Ω if No. 18 wire is chosen, and it will have a resistance of 2.6 Ω if No. 24 wire is utilized (see Fig. 2-8).

Whenever a run of more than a few feet must be installed between a power amplifier and a speaker system, avoid the use of "flat cable," which is sometimes used under carpets because of its very thin construction. The high resistance of this type of speaker cable will impose an excessive power loss on long runs.

In addition to installing speaker lines with suitably low resistance, it is also important to make low-resistance connections. Otherwise stated, con-

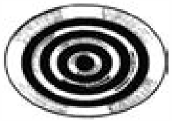
Figure 2-8. The loop resistance of a line is equal to the total resistance of the conductors: (a) representation of loop resistance; (b) loop resistance is measured with the ohmmeter section of a volt-ohm-milliammeter. (*Courtesy Triplett Electrical Instrument Co.*)

ductors must be clean and bright at connection points, and they must be firmly secured beneath terminal screws. Otherwise, the contact resistance represented by a poor connection could be many times the resistance of the line itself—contact resistance results in a power loss in the same manner that line resistance reduces efficiency. Contact resistance often develops sufficient heat that the metallic surfaces corrode rapidly, thereby aggravating the condition. Stranded conductors such as lamp cord are often used in speaker lines. In turn, it is essential to make certain that all the strands are included in a terminal connection. Note that it is usually of no advantage to install shielded cables in low-impedance high-level circuits such as speaker systems.

2-4 PRECAUTIONS IN CONVENTIONAL INSTALLATIONS

Conventional installations involve equipment that is not weatherproof. In turn, it is essential to avoid exposure of record players, tape decks, FM tuners, amplifiers, and speakers to rain or moisture. Not only is the equipment likely to become disfigured and damaged, but fire and shock hazards are also involved. As explained in greater detail in a subsequent chapter, specialized hi-fi speakers can be installed in patios or even in completely exposed locations. These speakers are designed to withstand almost any adverse weather conditions except flooding or complete immersion (see Fig. 2-9). Amplifier equipment, in particular, must be protected against excessive heat as well as excessive moisture. An integrated receiver, for example, should be placed on a suitable shelf or table, not in proximity to a

(a)



(b)

Figure 2-9. Specialized speaker arrangements: (a) folded-horn speaker with weatherproof construction for outdoor installation; (b) grille for ceiling or wall installation.

source of heat. It would be poor practice to place an integrated receiver next to a fireplace or radiator. In addition, it is advisable to allow at least 2 in of clearance above and behind the integrated receiver, to ensure adequate ventilation.

Nearly all power lines in the United States supply 60-Hz alternating current (ac). The principal exceptions are the use of direct current (dc) in isolated areas, or ac with a frequency other than 60 Hz is occasionally generated locally. Since standard hi-fi stereo equipment must be operated from a 117- to 120-V source of 60-Hz ac, the installer should make certain that this requirement is met. Most residences are wired with both 120- and 240-V circuits. However, the two types of outlets ordinarily have different mechanical construction, so there is little danger of accidentally connecting a hi-fi system into a 240-V circuit. If this error were made, catastrophic damage would possibly occur. Until the system is completely installed and all connections checked, it is good practice to leave the power cord(s) unplugged. This precaution ensures that all the components are "dead" and that damage to the system or shock hazard to the installer will not be incurred in case of various errors that may be made and subsequently corrected during the course on installation.

To avoid acoustic feedback and improper reproduction, record players should not be placed on top of speaker enclosures. A record player should be installed at a reasonable distance from a powerful speaker. Similarly, it is poor practice to place a tape deck on top of a speaker enclosure. Note that two basic types of phono cartridges are in general use; these are called the magnetic and the ceramic cartridge designs. Both types have a low-level output and in turn are subject to microphonic reaction in the presence of substantial mechanical vibration or even in a high-intensity sound field. A magnetic cartridge has an output of 5 or 10 millivolts (mV), whereas a typical ceramic cartridge has an output of 200 mV. Hence, magnetic cartridges are about 30 times lower in output level than are ceramic cartridges. For this reason, a magnetic cartridge tends to be more responsive to extraneous mechanical vibrations. If a record player employs a ceramic cartridge, for example, be sure that the cable from the player is plugged into the "ceramic" input of the preamplifier, not into the "magnetic" input. Similarly, it is essential to plug the cable from a magnetic-cartridge player into the "magnetic" input of the preamplifier. This type of connection error results in very poor reproduction, owing to the difference in output levels and frequency characteristics of these different kinds of cartridges (see Fig. 2-10).

Avoid plugging audio equipment into outlets that operate in overloaded branch circuits of the 120-V wiring system. Although audio equipment is designed to operate over a reasonable variation in line voltage, best reproduction is obtained when the supply voltage remains constant at the specified level. Fluctuation in line voltage can be avoided by utilizing a

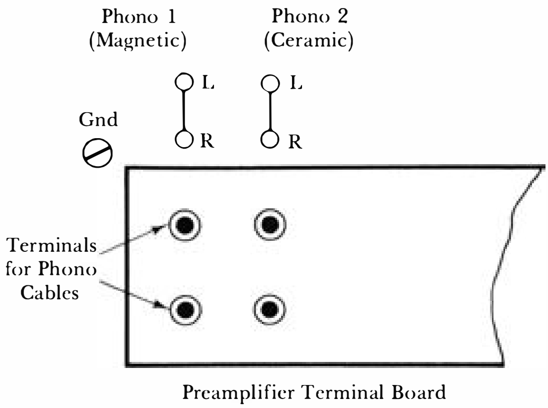


Figure 2-10. Typical terminal facilities for magnetic and for ceramic phono cartridge outputs.

separate branch circuit from the service panel that powers only the hi-fi system. Although a separate branch circuit can be added to the existing wiring arrangement by a do-it-yourselfer, it is generally desirable to enlist the services of a commercial electrician. Another approach to the problem of the overloaded branch circuit is to reserve one such circuit solely for powering the hi-fi system—although small additional loads such as floor or table lamps may be operated from the same branch circuit. Avoid the use of heavy loads, such as toasters or other appliances. Of course, if a branch circuit is very heavily loaded, its associated circuit breaker will trip, or the associated fuse will blow and the hi-fi system will suddenly "go dead."

In most cases, the installer will not know the locations of the outlets that are powered by a particular branch circuit from the service panel. However, it is a simple matter to make this determination. Plug a portable lamp into the outlet that has been chosen to power the hi-fi system. Then, turn off all the circuit breakers (or remove all the fuses) except the one that supplies power to the portable lamp. Next, check all the outlets in the wiring system with the portable lamp, to determine which outlets are "live" and which are "dead." All the "live" outlets are on the same branch circuit. Insofar as hi-fi system operation is concerned, the "live" outlets are declared to be "off limits" for powering household appliances that draw substantial amounts of current.

A record player should be placed on a level surface, so that the turntable does not slope in any direction. If the turntable is tilted, the tracking and antiskate adjustments of the record player will be upset, and poor groove tracking will result. In an aggravated situation, the tone arm will skip across the record grooves. It is undesirable to use an abnormally long cable between the record player and the preamplifier from the viewpoint of increased hum level, as well as capacitive loading of the cartridge. Even if a separate ground wire is run from the record player to the preamplifier chassis (see *Gnd* terminal in Fig. 2-10), an abnormally long shielded cable can develop noticeable hum in the sound output. This is a result of slightly different "ground" potentials at the record player and at the preamplifier.

Magnetic cartridges are susceptible to pickup of magnetic fields, and this factor can increase the hum level objectionably if the record player is mounted too close to the power supply in the power amplifier, for example. In other words, large power transformers often develop significant magnetic fields at a distance of several inches from the equipment cabinet. If a magnetic phono pickup is located within this area, it will have sufficient hum voltage induced into its inductors that 60-Hz hum becomes audible in the sound output. The same observation applies to tape decks; the playback heads will respond to significant magnetic fields in their vicinity. Note also that some tape decks will operate satisfactorily whether they are mounted horizontally or vertically; others operate properly only in a horizontal or in a vertical position, as specified in the instructions for the unit.

It was previously noted that the speakers or enclosures in a system should be connected so that they operate in phase. To check for proper phasing, first set all of the tweeter level controls (Fig. 2-11) to their normal

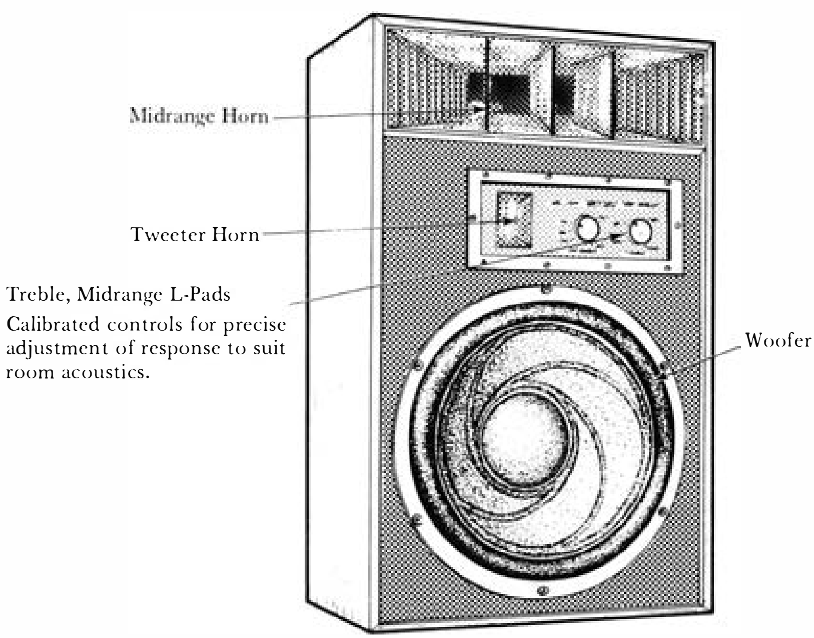


Figure 2-11. Example of treble and midrange L pads in a speaker enclosure.

operating positions. All bass and treble controls should be set to their "flat" positions. If a loudness-compensation switch is provided, it should be turned off at this time. Set the stereo function selector to a monaural position, and set the input selector switch to a phono position. If a balance control is provided, it is set to its normal operating position. Turn down the volume control. Then start the record player and touch up the balance control, if necessary, to obtain equal sound outputs from the L and R speakers. In turn, the L and R speakers will be radiating the same audio signal at the same level. To check the phasing, the two enclosures may be placed together (side by side). Make the following tests:

1. Listen to the sound output from the speaker enclosures and note the bass reproduction characteristics.
2. Disconnect one of the enclosures from the power amplifier and listen to the change in bass reproduction characteristics.
3. Reconnect the enclosure to the power amplifier, and again note the change in bass reproduction.
4. Reverse the wires to the terminals for the enclosure that was reconnected in the preceding step. The bass reproduction level will either increase or decrease.
5. Correct phasing corresponds to an increase in the bass reproduction level.

In addition, the phono cartridge should be checked for correct phasing. To do so, repeat the foregoing procedure with the function selector switch set to its stereo position. If a blend control is provided, it is set to its minimum blending position. If the stereo power amplifier has a stereo–mono switch, this switch should be set to its stereo position. In the event that the speakers appear to be incorrectly phased when the foregoing procedure is repeated, the difficulty will be found in a reversal of connections at the phono cartridge output terminals, or in a reversal of connections at the terminal board underneath the turntable.

If difficulty is encountered in distinguishing between the left- and right-channel audio cables, some installers prefer the following test. The cables are first connected at random and the record player is started. Either the Phono-1 or the Phono-2 setting of the function selector may be utilized. A *stereo* symphonic or orchestral recording should be employed. Adjust the volume control for a comfortable listening level, and listen for the apparent placement of the violins. If the sound of the violins seems to come from or near the left speaker (as viewed from the listening position), the audio cables have been connected properly. On the other hand, if the sound of the violins does not seem to come from the left speaker, the audio cables are improperly connected and should be reversed.

As noted previously, all the components in a system, with the exception of the speakers, require a source of ac power. Thus, a preamplifier, power amplifier, record player, tape recorder, and FM/AM tuner must each be plugged into an ac power outlet. As exemplified in Fig. 2-12, a preamplifier is usually provided with a sufficient number of ac outlets on its rear panel to serve the remaining components in the system. In turn, only the preamplifier itself need be plugged into a wall outlet. Four switched outlets and one unswitched outlet are denoted in the illustration. In other words, some of the outlets provided by a preamplifier may be controlled by its on–off switch, whereas others may not. It is desirable to plug the cords from the tuner, tape recorder, and power amplifier into the switched outlets of the preamplifier. The reason for this preference is that the power amplifier should necessarily be switched on when the preamplifier is turned on. Although the tuner and the tape recorder may be plugged into unswitched outlets, an FM listener often prefers for the tuner to turn on when the amplifiers are switched on. It is good practice to plug a record player into an unswitched outlet—this avoids the danger of developing "flat spots" on idler wheels.

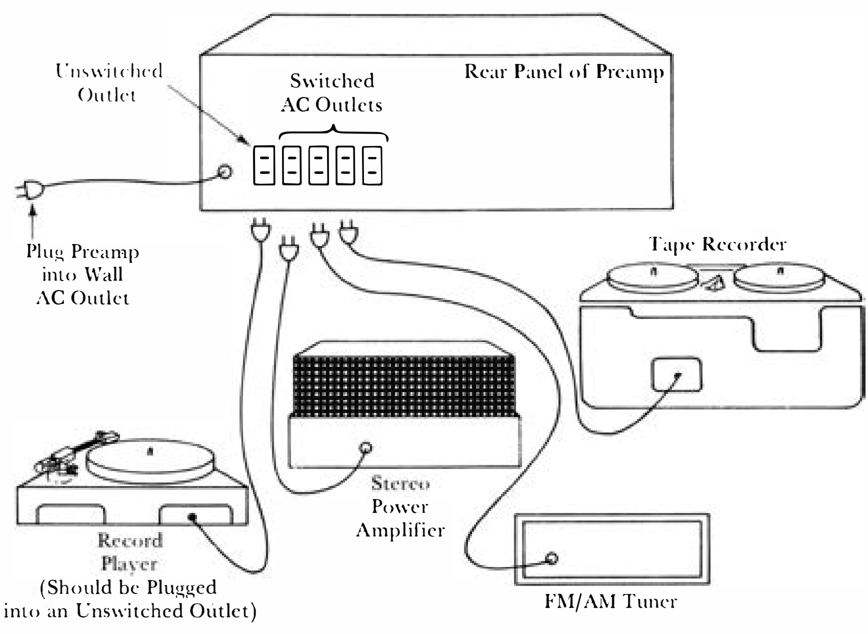


Figure 2-12. Both switched and unswitched outlets are usually provided.

2-5 INSTALLATION ADJUSTMENTS

In addition to tweeter level adjustments, installation personnel may be called upon to make record-player adjustments, such as stylus pressure, antiskate, tone-arm height, and stylus set-down adjustments. With reference to Fig. 2-13, the stylus pressure adjustment is made in the exemplified record player with the aid of a suitable stylus pressure gage. This type of gage indicates the stylus pressure in grams, and the pressure is adjusted to the value specified in the record-player manual. In this example, the stylus pressure knob (A) is turned clockwise to reduce the stylus pressure; it is turned counterclockwise to increase the stylus pressure. Note that when the stylus pressure is changed, the antiskate compensation knob must be adjusted correspondingly.

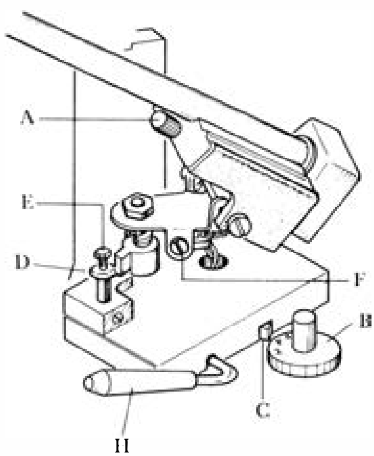


Figure 2-13. Typical changer mechanism adjustments. (*Courtesy Allied Electronics*)

This antiskate control (B) is set to the same value as the stylus pressure. In this example, the antiskate control is not calibrated for values less than 2 grams (g). If the stylus pressure adjustment should be set for less than 2 g, the antiskate control will then be set for 2 g. To adjust the antiskate knob (B), it is turned so that the desired number is lined up with the indicator point (C). Calibrations are provided for values of 2, 4, and 6 g. For an intermediate value, such as 5 g, the operator positions the knob so that a value of approximately 5 is lined up with indicator (C). To adjust the tone-arm height and to raise the arm, the plastic nut (D) is held securely with an end wrench and the screw (E) is turned counterclockwise as required. Conversely, to lower the arm, screw E is turned clockwise as required. The correct tone-arm height in this example provides a clearance of the arm support by ⅛ in during the change cycle. To adjust the stylus set-down position, the record-size selector knob is turned to its 12-in position. A single 12-in record is placed on the turntable, and the operator turns the set-down screw (F) to position the stylus in the record groove ⅛ in from the edge of the record.

2-6 INSTALLATION OF A CENTER-CHANNEL MIXER

As noted previously, when L and R speakers are installed a considerable distance from each other, an objectionable "hole-in-the-middle" effect may be experienced. In other words, the separation between the L and R channels becomes excessive for realistic stereo reproduction. An expedient that is utilized in this situation consists of a mono speaker installed midway between the L and R speakers. This mono speaker is energized by an L + R signal from the preamplifier. Since all preamplifiers do not provide an L + R output, the installer will occasionally need to provide an external mixer, as depicted in Fig. 2-14. This is a resistive pad that combines equal amounts of L and R signals, to form a center-channel mono signal. To provide an adequate level for the mono signal, the mixer is followed by a power amplifier, which in turn drives the mono speaker.

2-7 UTILIZATION OF AUXILIARY INPUT CONNECTORS

To increase the versatility of a stereo system, any chosen audio signal source may be fed into the "Aux In" connectors of the preamplifier. If the signal source ordinarily is connected to a low-fidelity speaker or speakers, connection of its audio output to the "Aux In" connector of a hi-fi preamplifier will usually improve the quality of the reproduced sound considerably. Among auxiliary sound sources are multiband radio tuners, TV receivers, sound-movie projectors, local telephone or intercommunication arrangements, electronic organs, and so on. Convenient connection can be made to an "Aux In" connector if the device has at least one audio-output jack. This jack will be typically marked "To Amplifier," "Line Output," "To Tape Recorder," or equivalent designation. In the event that the audio signal source does not provide an audio-output jack, a technically capable installer may add the necessary jack, and, if required, a switching circuit to turn the associated speaker(s) on or off as desired.

Some sources may be designed as ac/dc "transformerless" arrangements and have "hot chassis" that present a potential for electrical shock. A "hot chassis" is also a possible generator of hum voltage. If the installer is in

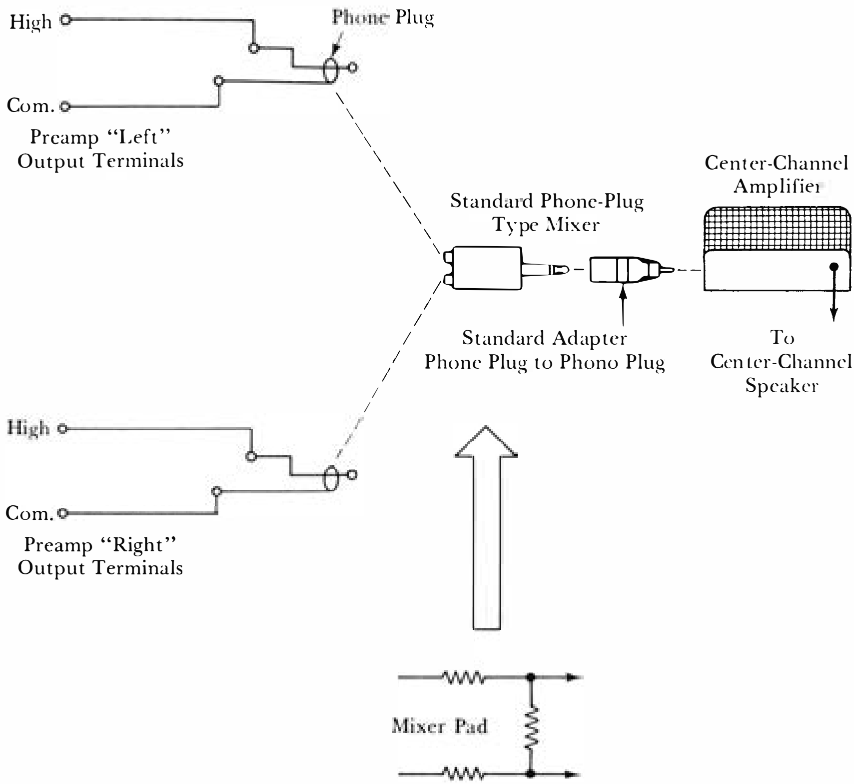


Figure 2-14. Mixer arrangement for a "hole-in-the-middle" speaker.

doubt concerning this point, the device should not be connected to an "Aux In" connector on the preamplifier unless a line-isolating transformer is first connected between the power outlet and the "hot-chassis" device. A line-isolation transformer effectively converts the device into a transformer power-supply arrangement, so that there is no danger of shock when the chassis or output lead of the device is touched. Then, proceed as follows:

1. Turn off the stereo system. If the device has mono audio output, connect its single output jack to the preamplifier's L *and* R "Aux In" jacks. It is convenient to employ a shielded Y-connector audio cable with an appropriate plug on each of its three branches.
2. If the device has stereo audio output, connect its Left, L, A, or 1 output jack to the preamplifier's "L Aux In" jack, and connect its Right, R, B, or 2 output jack to the preamplifier's "R Aux In" jack. It is convenient to utilize two shielded audio cables, with appropriate plug on their ends.

3. Connect the line-isolation transformer (or device) to a nearby electrical outlet (switched or unswitched, as desired). Route the power cord away from all the shielded audio cables.
4. Turn on the stereo system and the auxiliary device. Set the preamplifier's Function control to its "Aux" position, and adjust the preamp volume control to a suitable setting.
5. Turn the Function switch of the preamp to its "FM Stereo" position and then to its "Aux" position. Compare the relative sound-output levels of the two sources. It is desirable to have approximately equal sound levels without readjustment of the Volume control of the preamp. Therefore, if the auxiliary device has any controls that affect its volume as reproduced through the stereo system, adjust these controls to obtain the desired output level.

2-8 BASIC INSTALLATION PRECAUTIONS

When a component system is connected together, the power cord for the power amplifier is often plugged into a switched outlet on the rear of the preamplifier. This is a convenient arrangement, because the power amplifier is then turned on or off when the preamplifier's power switch is turned on or off. However, this mode of operation may not be feasible when a super-high-power amplifier is to be switched on and off. In other words, damage may occur to the ac switch in the preamplifier owing to the power amplifier's very high turn-on (in-rush) current demand. Although this very large in-rush current is drawn for only a very short time, its peak value is sufficiently great that small switches can be damaged. Therefore, a super-high-power amplifier should be plugged into a separate power outlet; in turn, its own power switch is used to turn the super-high-power amplifier on and off.

Power cords for hi-fi stereo systems are provided with three prongs. Under no circumstances should the third (center) prong on the ac line-cord plug be defeated, either by removing it or by utilizing a 3:2 contact-reducing adapter. This third prong provides a ground connection, which helps prevent pickup of RF interference and possible damage to amplifiers. This ground return is also legally required by Underwriters' Laboratories, a branch of the National Board of Fire Underwriters. Amplifiers are provided with line fuses. If a fuse blows, a replacement fuse will probably blow also. In other words, there is some fault that has imposed an excessive current demand. The installer should proceed to determine why a fuse has blown and to correct the fault in the system. Note that it is extremely poor practice to replace a blown fuse with a heavier fuse or to attempt to defeat the fuse in any way. Not only is the equipment very likely to be damaged, but a fire hazard is introduced.

Input or output connectors to a power amplifier should never be disconnected or connected while the ac power to the system is turned on. A system should be completely connected (hooked up), with all ac power turned off. Then, all connections should be carefully checked. Then the power switch(es) may be turned on. Beginners sometimes fall into the error of connecting the power amplifier to the speakers, connecting the input cables to the power amplifier, and then turning on the ac power to the power amplifier before the preamplifier is connected. The most hazardous practice, moreover, is to make a "hum test" or "thumb test" to determine whether the power amplifier is working. A beginner may make this damaging test by touching his thumb to the open end of an input cable while the power amplifier is turned on. Both amplifier failure and speaker damage can result from this malpractice. Some common trouble symptoms, with probable causes and possible remedies are listed in Chart 2-2.

Chart 2-2

BASIC INSTALLATION PROBLEMS

Symptom	*Probable Cause*	*Possible Remedy*
1. Unit will not turn on	a. Not plugged into ac outlet.	a. Connect ac line cord to outlet or try different outlet.
	b. Blown ac fuse.	b. Refer to General Maintenance section and replace fuse with proper value.
2. Unit turns on but no sound is heard.	a. Switches in wrong position on preamplifier.	a. Check positions.
	b. Cables not connected properly.	b. Check cable connections.
	c. Tape Monitor switch on preamplifier in Monitor position not allowing normal operation of phono, tuner, or aux.	c. Move switch to "Out" position.
3. One or both channels inoperative.	a. Bad cables.	a. Try other cables (or interchange).
4. Unit hums.	a. Unit too close to phono section of preamplifier.	a. Isolate the input function associated with the problem. Reorient the preamp in relation to the power amplifier.
	b. Lack of shielding between units.	b. Insert MU metal shield between units if reorientation is impossible.

Chart 2-2 (cont.)

Symptom	*Probable Cause*	*Possible Remedy*
	c. Phono cartridge.	c. Check cartridge ground connections, then move tone arm while operating to see if hum level varies. If so, reorient turntable.
5. RF interference. Radio: radio program heard. TV: rasping buzz.	a. Poor cable shielding.	a. Shorten cables or obtain cable with better shielding.
6. Meters do not move.	a. Inadequate signal level to deflect meters.	a. Select a more sensitive setting.

(Courtesy Scientific Audio Electronics, Inc.)

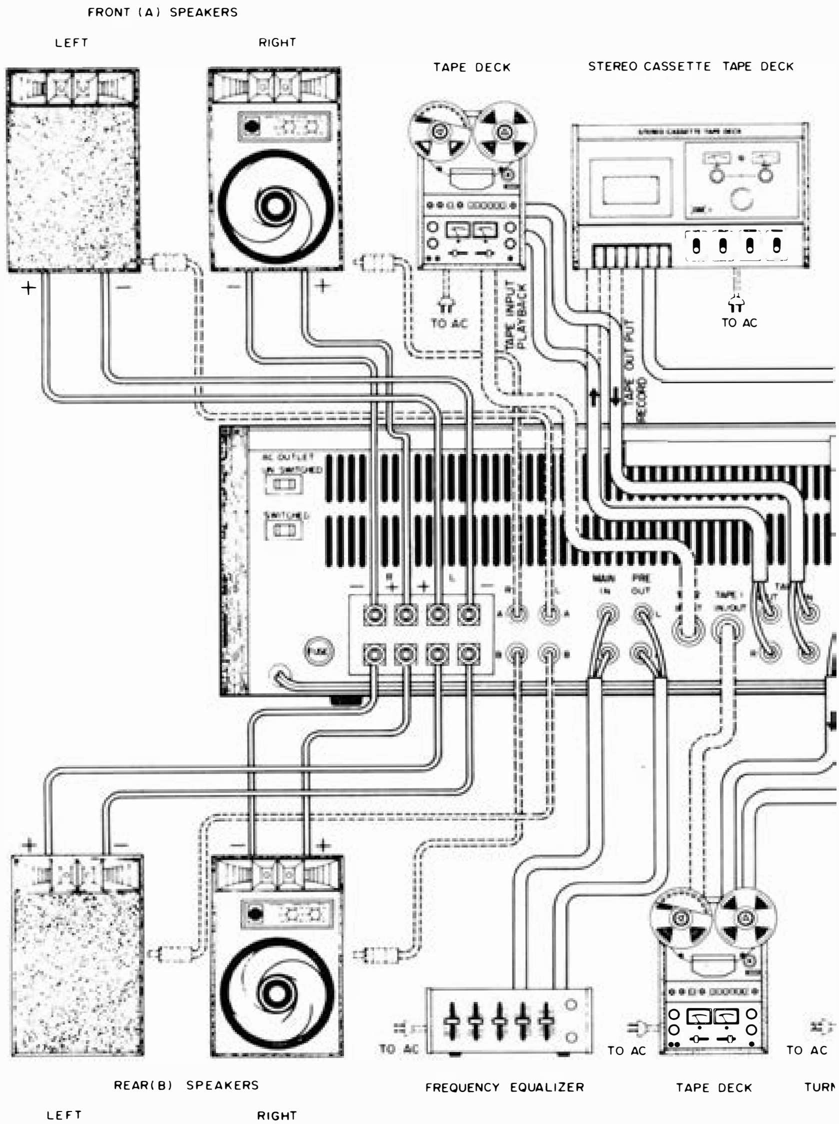
Chapter 3

Interconnections and Specifications for Commercial Audio Equipment

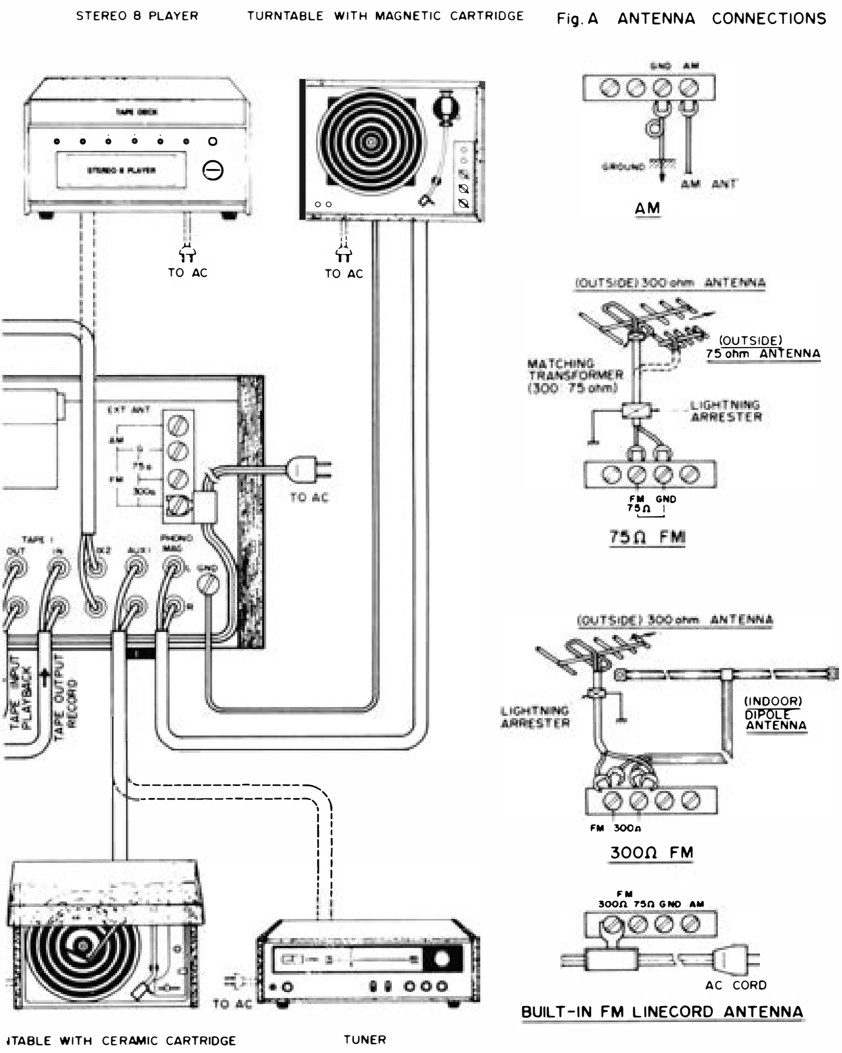
3-1 INTERCONNECTIONS AND PERFORMANCE SPECIFICATIONS

A sampling of representative high-fidelity equipment, with interconnections and performance specifications is presented in the following pages. (See Figs. 3-1 through 3-20.) These installation data are necessarily brief, and the reader who may acquire any of these units is advised to consult the detailed owner's instruction manual that is usually provided by the manufacturer. Detailed service data for many types of high-fidelity equipment are available from electronic manual publishers such as Howard W. Sams & Co., Inc. General checks and tests for audio components are explained in Chapter 7.

Figure 3-1. (a) Appearance of the Realistic STA-2000 AM/FM stereo receiver. (*Courtesy Radio Shack, a Tandy Corporation Co.*)

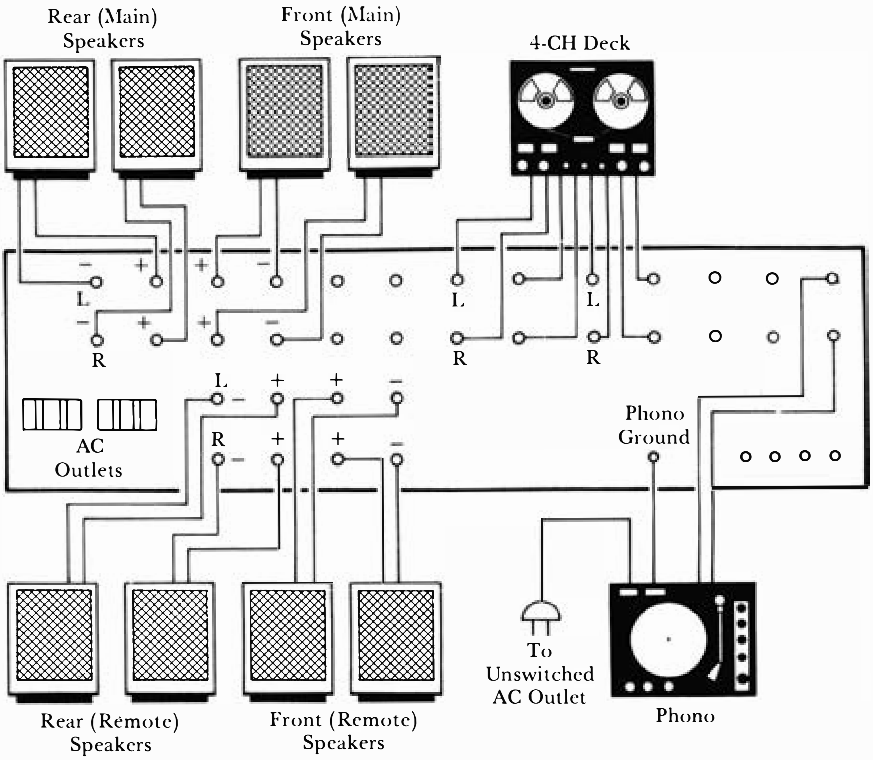


AMPLIFIER. Power Output at 8 Ohms: 75 W per channel, minimum rms 20–20,000 Hz, with no more than 0.25% total harmonic distortion. **Frequency Response:** 15–25,-000 Hz ±2 dB at 5 W. **Intermodulation Distortion:** 0.1% at 50 watts. **Signal-to-Noise Ratio:** 70 dB (phono); 75 dB (aux). **Phono Input Overload:** 200 mV. **FM TUNER.**



Sensitivity (IHF): 1.7 μV. **Capture Ratio:** 1.5 dB at 1 kHz. **Alternate Channel Rejection:** 1.7 dB. **Stereo Separation:** 48 dB at 1 kHz. **Total Harmonic Distortion:** 0.15% stereo; 0.15% mono. **Signal-to-Noise Ratio:** 70 dB. **AM TUNER. Sensitivity:** 200 μV for 20 dB S+N/N. **Image Rejection:** 60 dB. **Signal-to-Noise Ratio:** 45 dB. **POWER REQUIREMENT:** 120 V/ac, 60 Hz.

Figure 3-1. (cont.) (b) Interconnection diagram and specifications for the Realistic STA-2000 AM/FM stereo receiver.



TUNER SECTION: (FM)
IHF Sensitivity: 1.9 μV. *S/N Ratio*: 60 dB. *Selectivity*: 60 dB. *Capture Ratio*: 1.5 dB. *Harmonic Distortion*: (Mono): 0.8% or better; (Stereo): 1.0% or better. *Image Rejection*: 60 dB or better. *Stereo Separation*: 35 dB at 1 kHz.
TUNER SECTION: (AM)
Sensitivity: 250 μV/meter. *Image Rejection*: 50 dB.
AUDIO AMPLIFIER SECTION:
Continuous Power Output: (4-channels driven, 8 ohm loads, 20 Hz to 20 kHz): 25 watts per channel. *Rated Harmonic Distortion*: 1.0%. *Rated IM Distortion*: 1.0%. *Frequency Response*: 20 Hz to 20,000 Hz. *Signal-to-Noise Ratio*: (Phono) 60 dB; (Aux.): 70 dB.
GENERAL SPECIFICATIONS:
Power Requirements: 120 VAC, 60 Hz.

Figure 3-1. (cont.) (c) Interconnection diagram and specifications for the Realistic QTA-770 integrated receiver.

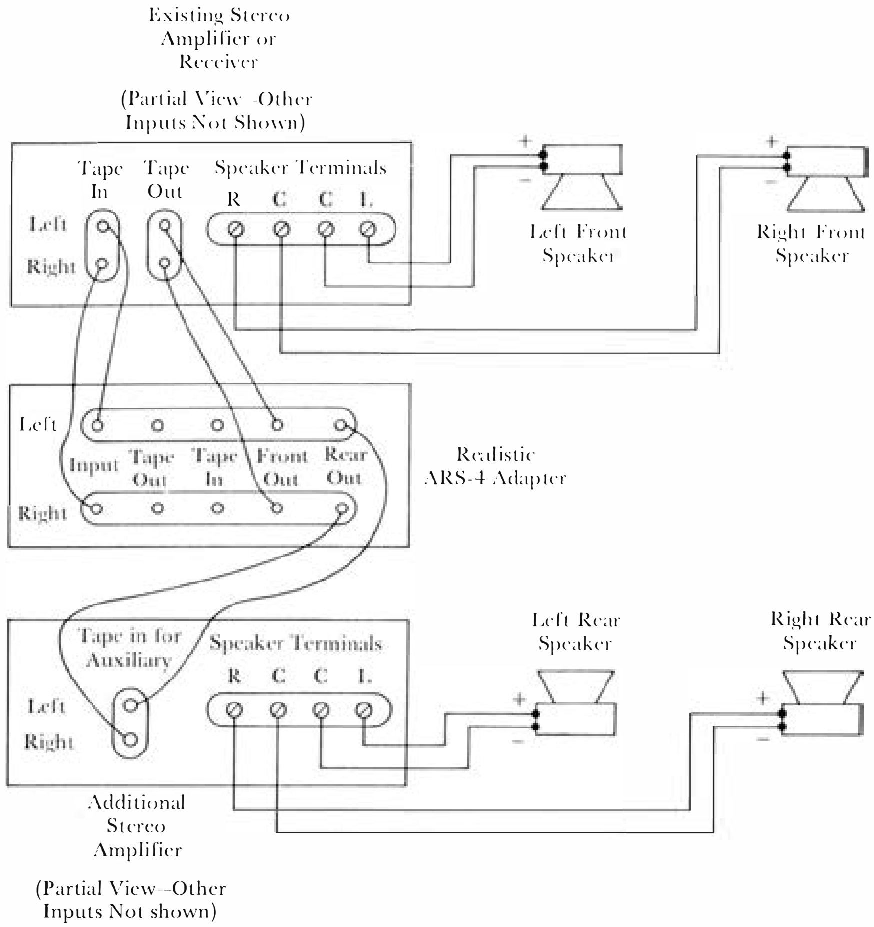


Figure 3-1. (cont.) (d) Interconnection diagram for the Realistic ARS-4 quadraphonic synthesizer.

Figure 3-2. (a) Appearance of the SAE Super High Power amplifier. (*Courtesy Scientific Audio Electronics, Inc.*)

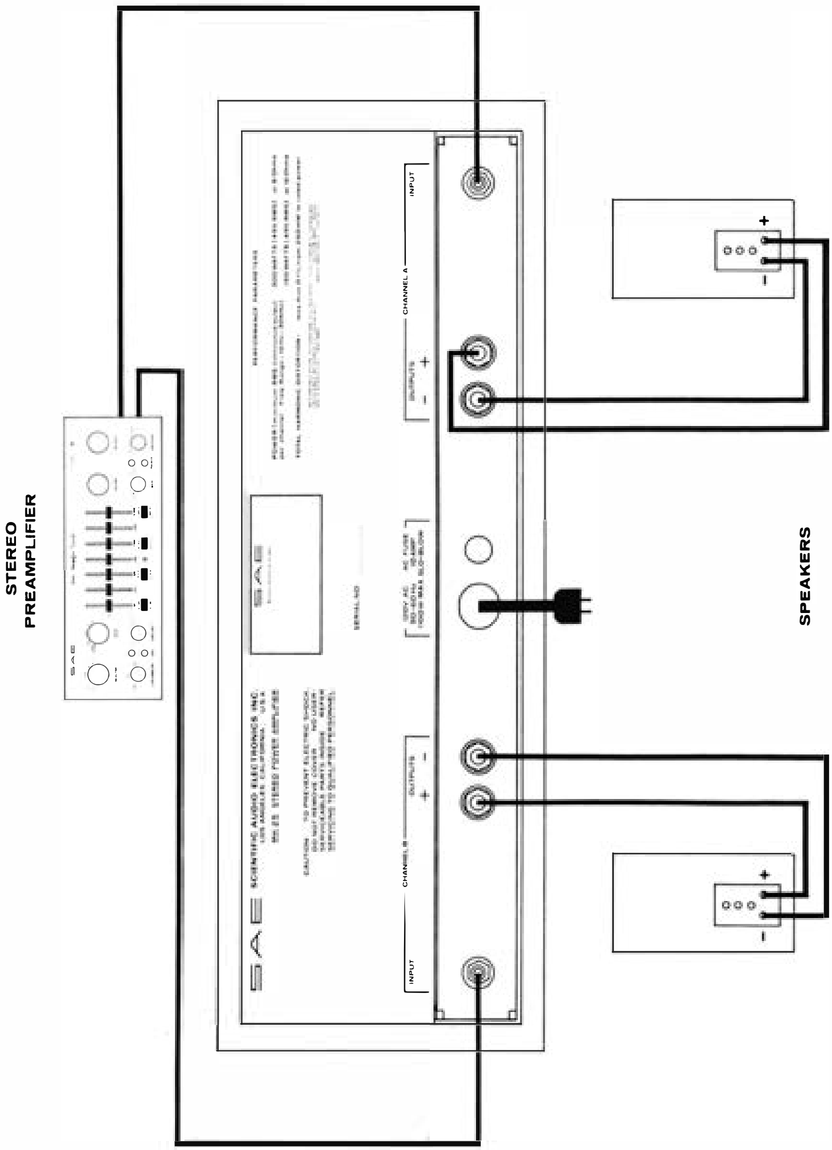


Figure 3-2. (cont.) (b) Interconnection diagram for the SAE Super High Power amplifier

RMS (min) continuous power output per channel 20 Hz to 20 kHz (both channels driven) at 8 Ω. 300 W at 0.05% total harmonic distortion
THD (total harmonic distortion) from 20 Hz to 20 kHz at 250 mW to rated power at 8 Ω. 0.05% max.
IM (intermodulation distortion) from 250 mW to rated power at 8 Ω with any two mixed frequencies between 10 Hz and 30 kHz at 4:1 voltage ratio. 0.05% max.
Frequency response at rated power. ±0.25 dB, 10 Hz to 30 kHz
Noise. Greater than 100 dB below rated power
Transient response of any square wave. 2.5 μs rise and fall time
Slew rate. 40 V/μs
Stability. Unconditionally stable with any type of load or no load including full-range electrostatic loudspeakers
Damping factor. 150 min (100 Hz)
Input sensitivity. 1.5 volts rms for rated output at 8 Ω
Input impedance. 50 kΩ
Semiconductor complement. 46 transistors, 49 diodes
Overload protection.
1. Low-impedance electronic-sensing circuit limits with output current below 2 Ω without limiting with 4 Ω or higher (or reactive loads).
2. Thermal sensing of inadequate ventilation.
3. Internal B+, B− supply fuses.

Loudspeaker protection. Relay circuit protects loudspeakers from low-frequency oscillations and plus or minus dc output. Five-second turn on/off delay eliminates on/off disturbances.
Power requirements. 110–125 V 50 Hz/60 Hz, 100 W at idling, to 1100 W at rated output.

Figure 3-2. (cont.) (c) Specifications for the SAE Super High Power amplifier.

No. 1 Make sure that local electric power is AC, and that the voltage matches the voltage specified on the rear of the receiver (or, in some cases, on a special tag on the power cord).

CAUTION: In any case, do not connect the power cord to an electrical outlet or turn on the unit yet.

No. 2 Place the set on any convenient shelf or table **away from all sources of heat.** Allow at *least* 2 inches clearance above and behind the set for ventilation.

No. 3 Place your two speakers 5-10 feet apart, *facing* your selected listening position, as close as possible to ear level. **Plug both speakers into the receiver *exactly* as shown. (Use two twin-conductor cables with the appropriate plugs at their ends.)**

No. 4 **Unwind the power cord *completely*, drape it *full length*, and plug it into an AC wall outlet.**

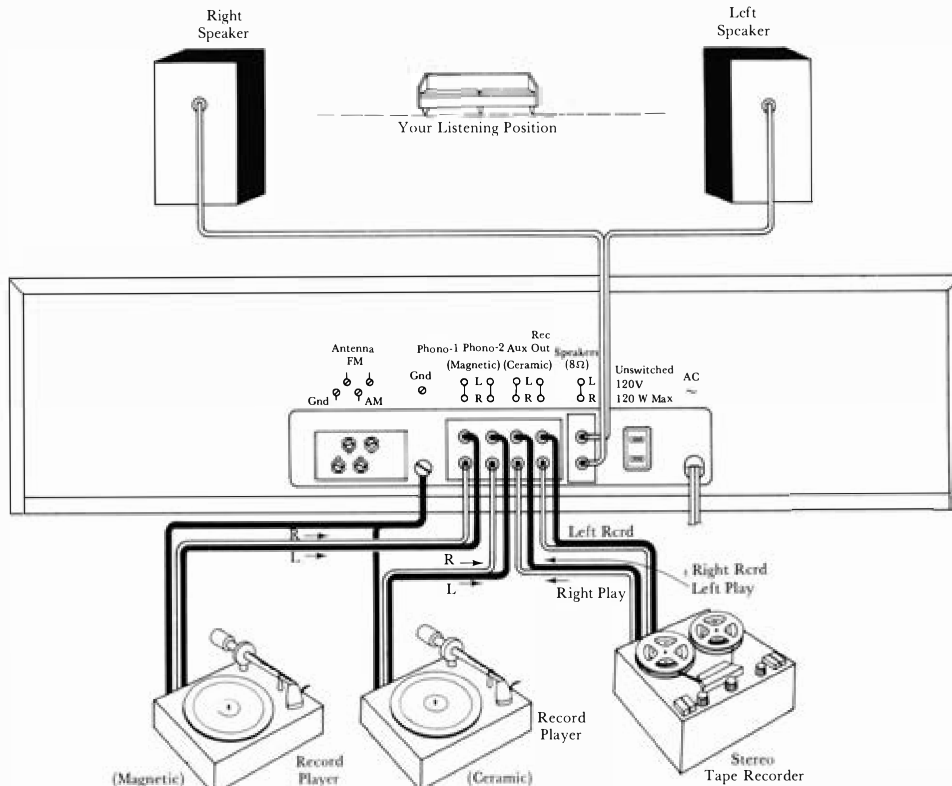


Figure 3-3. (a) Interconnection diagram for the Fisher MC-3010 integrated receiver.

FM TUNER SECTION

Usable sensitivity (IHF standard)	5 μV
Harmonic distortion (at 400 Hz, 100% modulation)	
Mono	1.0%
Stereo	1.5%
Signal-to-noise ratio (at 100% modulation and 1 mV input)	60 dB
Selectivity	36 dB
Image frequency rejection (at 88 MHz)	40 dB
IF frequency rejection (at 88 MHz)	70 dB
FM stereo separation (at 1 kHz)	30 dB
Capture ratio, IHF, at 1 mv	3 dB

AM TUNER SECTION

Usable sensitivity (IHF standard) S/N 20-dB loop antenna	250 μV/min
Selectivity (at 1 MHz and $\pm$10 kHz)	30 dB
Image frequency rejection (at 1 MHz)	50 dB
IF frequency rejection (at 1 MHz)	40 dB

AMPLIFIER SECTION

Minimum continuous rms sine-wave power per channel from 60 Hz to 15 kHz with less than 1% total harmonic distortion into an 8-Ω load.	5 watts
Sensitivities (for rated output power at 8 ohms, 1 kHz)/impedances	
Phono (magnetic)	2.5 mV/50 kΩ
Phono (ceramic)	300 mV/1M Ω
Auxiliary	200 mV/100 kΩ
Microphone	1.0 mV/600 Ω
Hum and noise (below rated output)	
Phono (magnetic)	62 dB
Phono (ceramic)	62 dB
Auxiliary	62 dB
Frequency response $\pm$2.0 dB	40–15,000 Hz
Bass control range (at 100 Hz)	$\pm$8 dB
Treble control range (at 10 kHz)	$\pm$8 dB
Loudness contour (at 40-dB volume attenuation fixed)	+8 dB at 100 Hz +5 dB at 10 kHz

TAPE SECTION

Tape speed	
Play and record	3.75 in/s
Fast forward	3.75 in/s $\times$ 2
Wow and flutter, maximum	0.2%
Signal-to-noise ratio (play and record)	40 dB
Channel separation	35 dB
Crosstalk	45 dB
Distortion	2.0%
Erasure ratio	55 dB
Recording sensitivity	
Microphone	1 mV
Auxiliary	200 mV
General	
Ac power	120 V
Frequency	50/60 Hz
Consumption	70 W

Figure 3-3. (cont.) (b) Specifications for the Fisher MC-3010 integrated receiver.

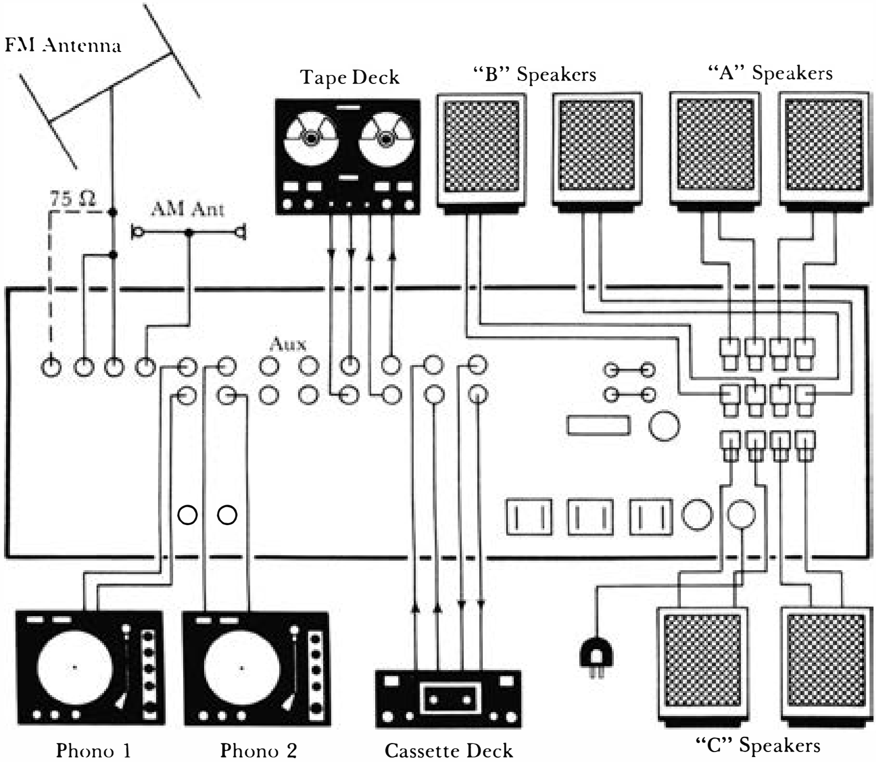
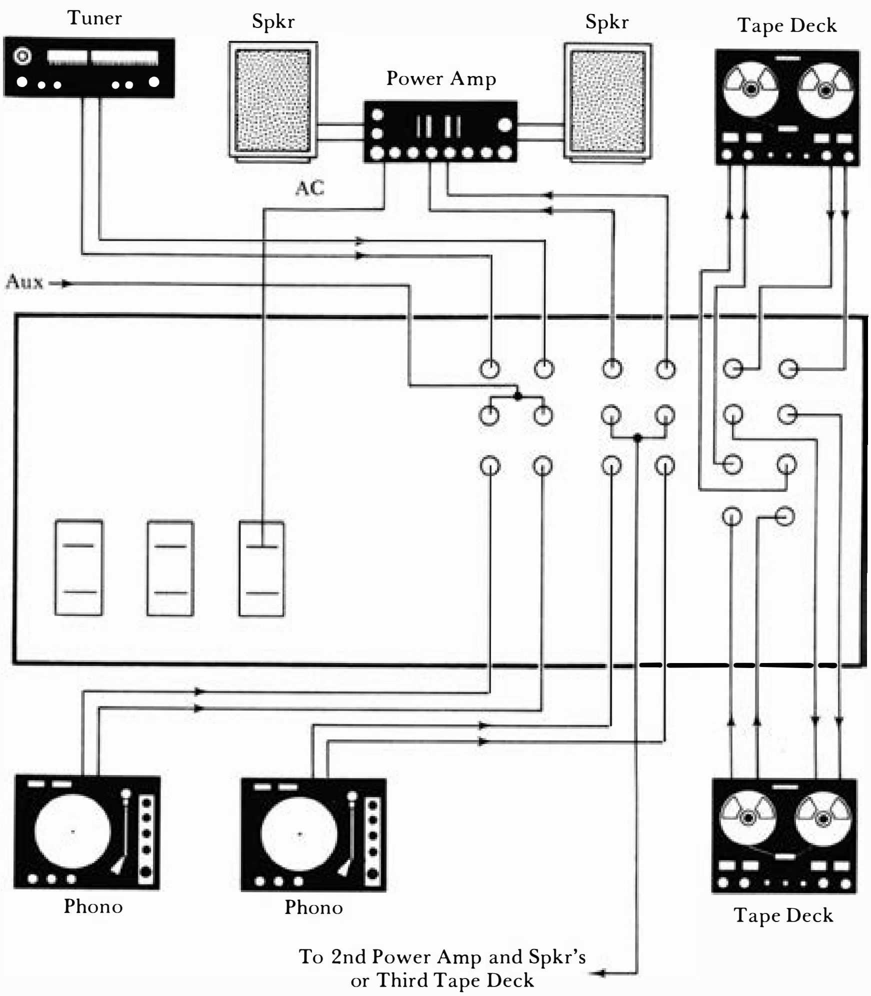
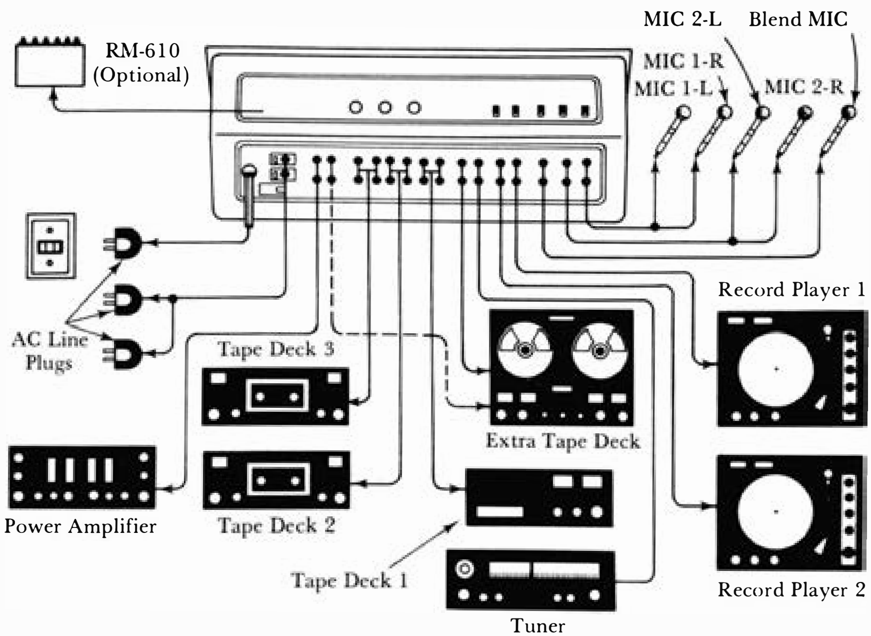


Figure 3-3. (cont.) (c) Interconnection diagram for the Fisher RS-1080 integrated receiver.



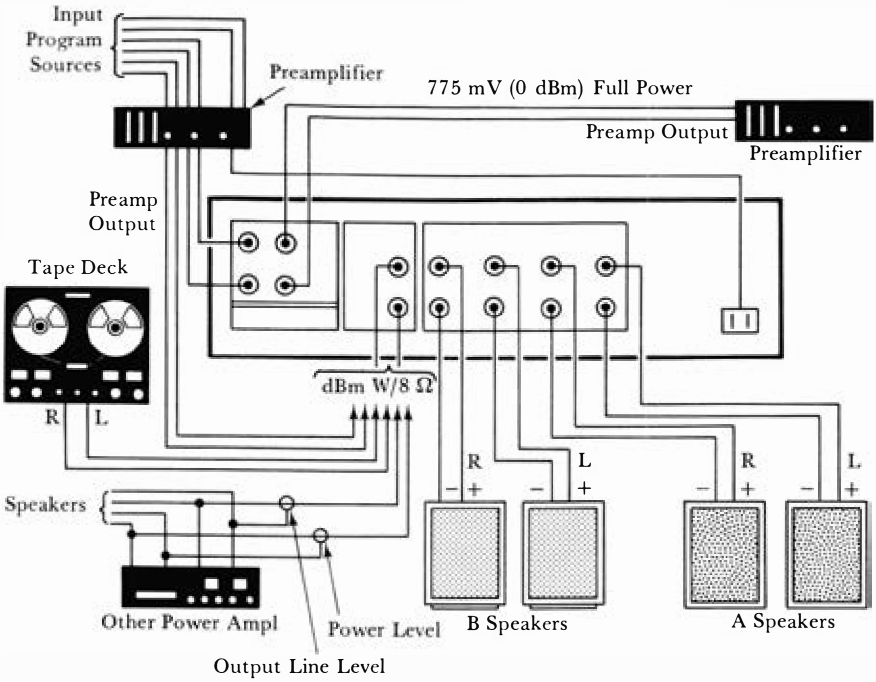
Rated Output: 2.5 volts. *Frequency Response*: 20 Hz to 20 kHz, ±0.25 dB. *Harmonic Distortion at Rated Output*: 0.005%. *IM Distortion at Rated Output*: 0.005%. *Phono Input Sensitivity*: 1.5 mV. *High-Level Input Sensitivity*: 100 mV. *Phono Overload*: 150 mV. *Maximum Output*: 7 volts into 100 K ohms; 3.5 volts into 600 ohms. *Filter Cut-Off Frequencies*: Low: 50 Hz at 12 dB per octave; High: 7.5 kHz at 12 dB per octave.

Figure 3-4. Interconnection diagram and specifications for the Epicure PR-4 amplifier.



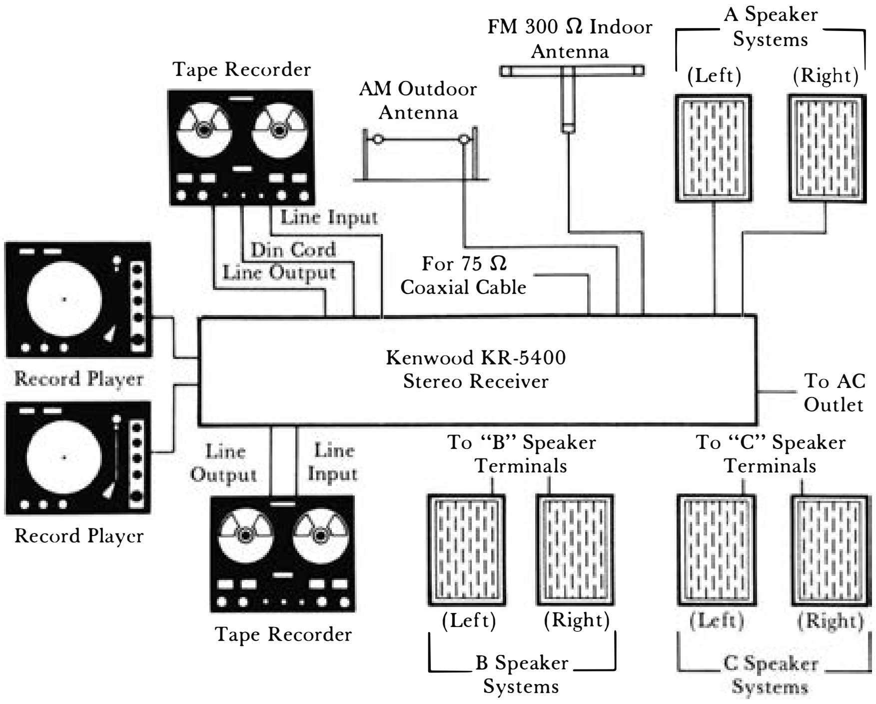
Frequency Response: Mike: 30 Hz–100 kHz, +0, −1.5 dB; Phono: 30 Hz–15 kHz, ±0.3 dB; High Level: 20 Hz–100 kHz, +0, −1.5 dB. *Input Sensitivity*: Mike: 0.2 mV; Phono: 1 mV; High Level: 75 mV. *Maximum Input Level*: Mike: 1 volt; Phono: 250 mV; High Level: 50 volts. *Signal-To-Noise Ratio*: (IHF A-weighting): Mike: Better than 53 dB, referenced to 0 dB; Phono: Better than 80 dB, referenced to 1 mV. *Distortion*: Mike: Less than 0.01% at all frequencies up to 10 kHz; Phono: less than 0.005% at all frequencies up to 10 kHz; High Level: Less than 0.005%.

Figure 3-5. Interconnection diagram and specifications for the Nakamichi 610 amplifier.



Rated Power Output: 100 watts per channel, 20 Hz to 20 kHz, with no more than 0.08% total harmonic distortion, 8 ohm loads (140 watts, 4 ohms). *IM Distortion*: 0.03%, at 50 watts, 4, 8 or 16 ohms. *Damping Factor*: 70 at 1 kHz. *Frequency Response*: (1 watt, normal setting): 10 Hz to 100 kHz, +0, −1 dB. *Input Sensitivity*: 0.75 V. *Input Impedance*: 25,000 ohms. *Signal-To-Noise*: (IHF "A" Weighting): 115 dB. *Meter Range*: −50 to +5 dB (0 DB = 100 W into 8 ohms, or 0 dBm). *Power Consumption*: 290 watts.

Figure 3-6. Interconnection diagram and specifications for the Yamaha B-2 power amplifier.



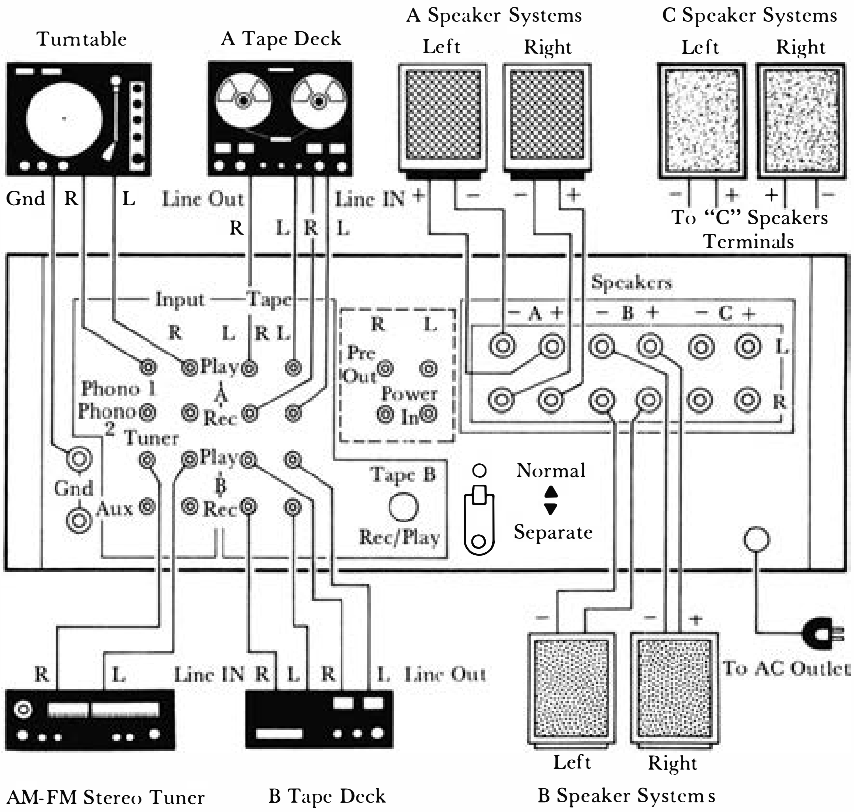
TUNER SECTION (FM)
IHF Sensitivity: 1.9 μV. *S/N Ratio*: 68 dB. *Quieting Slope*: 48 dB at 4 μV; 60 dB at 10 μV; 68 dB at 50 μV. *Selectivity*: 65 dB. *Capture Ratio*: 1.5 dB. *Image Rejection*: 70 dB. *IF Rejection*: 90 dB. *Spurious Rejection*: 90 dB. *Harmonic Distortion*: (mono): 0.3%; (Stereo): 0.5%. *Stereo FM Separation*: (1 kHz): 35 dB; (10 kHz); 27 dB.

TUNER SECTION (AM)
Sensitivity: (ext. antenna): 18 μV. *Selectivity*: 30 dB. *Image Rejection*: 60 dB. *IF Rejection*: 40 dB. *S/N Ratio*: 45 dB.

AUDIO AMPLIFIER SECTION
Continuous Power Output: (Both channels driven, 8 ohm loads, 20 Hz to 20 kHz): 35 watts-per-channel. *Rated Harmonic Distortion*: 0.5%. *Rated IM Distortion*: 0.5%. *Frequency Response*: 10 Hz to 40,000 Hz ± 1 dB. *Phono Equalization*: RIAA ± 1 dB. *Damping Factor*: 50 (at 8 ohms). *Input Sensitivity*: (Phono): 2.5 mV; (Aux and Tape): 150 mV. *Record Output Level*: 150 mV. *Hum and Noise*: (Phono 1 and 2): 70 dB ("A" Weighting); (High Level Inputs): 90 dB ("A" Weighting). *Tone Control Range*: (Bass): ± 10 dB @ 100 Hz; (*Treble*): ±10 dB ± 10 kHz. *Low Filter*: −5 dB @ 100 Hz. *High Filter*: −10 dB @ 10 kHz.

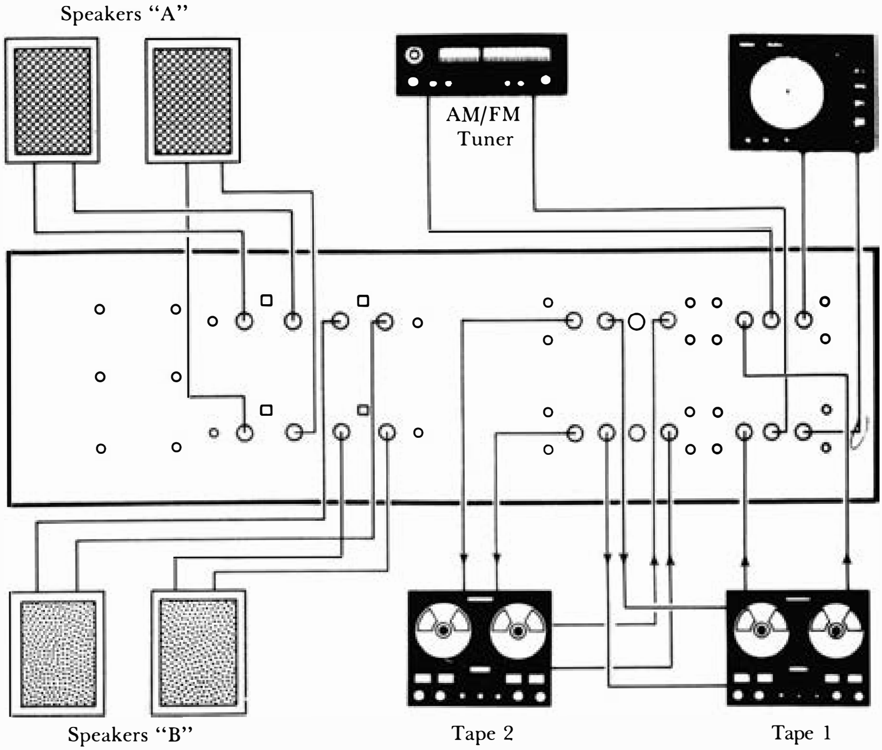
GENERAL SPECIFICATIONS
Power Requirements: 120 V, 50/60 Hz, 240 watts maximum consumption.

Figure 3-7. (a) Interconnection diagram and specifications for Kenwood KR-5400 integrated receiver.



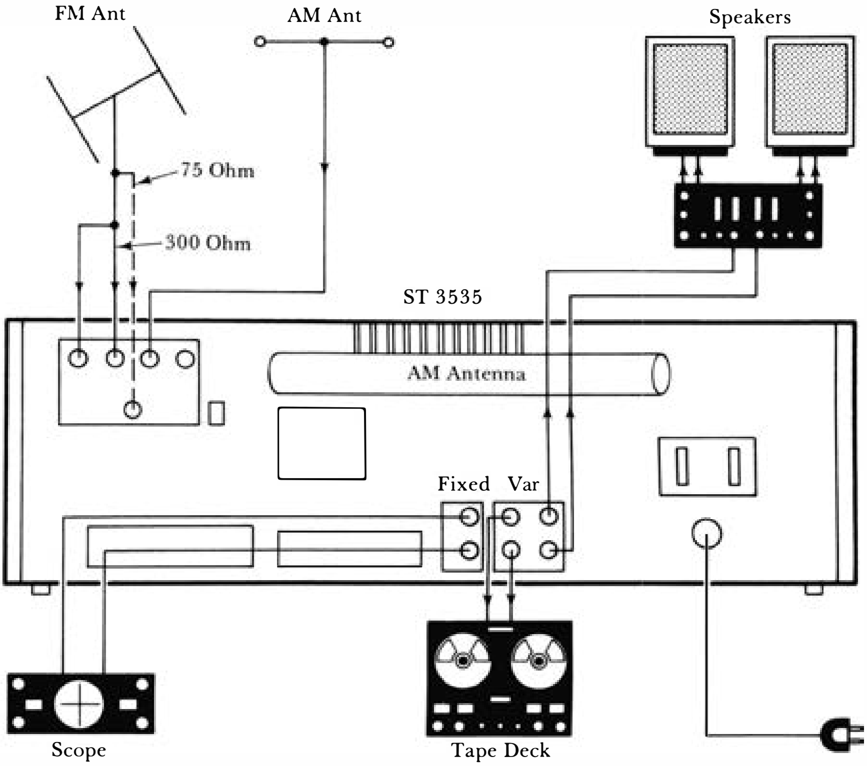
Power Output: 80 watts-per-channel, minimum continuous power, 8 ohm loads, from 20 Hz to 20 kHz. *Rated Harmonic Distortion*: 0.1% (0.04% at 1 watt). *IM Distortion*: 0.1% (0.04% at 1 watt). *Damping factor*: 50 at 8 ohms. *Residual Noise*, main amplifier: 100 dB. *Imput Sensitivity*: Phono 1 & 2: 2.5 mV; Tuner, Aux, Tape: 150 mV. *Signal-to-Noise Ratio (A-weighted)*: Phono: 72-dB below 5 mV; High Level: 90 dB *Phono Overload*: 260 mV. *Frequency Response*: Phono: RIAA ±0.3 dB; High Level: 20 Hz to 40 kHz, +0, −0.5 dB. *Bass Control Range*: ±7.5 dB at 100 Hz or 40Hz (depending upon turnover frequency selected); *Treble Control Range*: ±7.5 dB at 10 kHz or 20 kHz *Bass and Treble Turnover Frequencies*: 400 Hz, 150 Hz, 3 kHz and 6kHz. *Low Filter Cutoff*: −3dB at 40 Hz, 12 dB-per-octave. *High Filter Cutoff*:−3 dB at 8 kHz, 12 dB-per-octave. *Presence Control Range*: +6 dB at 800 or 3000 Hz. *Power Requirements*: 120 V, 60 Hz, 550 watts (maximum).

Figure 3-7. (cont.) (b) Interconnection diagram and specifications for the Kenwood KA-8300 integrated amplifier.



Power Output: 25 watts continuous per channel, 20 Hz to 20 kHz, 8-ohm loads. *Rated THD*: 0.25%. (*Power at "Clipping"*: 35 watts, 1 kHz, 8 ohms; 35 watts, 4 ohms; 20 watts, 16 ohms). *IM Distortion*: 0.1%. *Frequency Response*: Phono: RIAA ± 1.0 dB. High Level: 15 Hz to 45 kHz, ±0.5 dB. *Hum and Noise*: Phono: 72 dB below 10-mV reference; High Level: 89 dB below 0.5-volt input. *Phono Overload*: 100 mV at 1 kHz. *Input Sensivity*: Phono: 1.65 m V; High Level: 125 mV. *Tone Control Range*: ±10 dB at 50 Hz and 15 kHz. *Dimensions*: 13½ x 12 x 4¼ inches high. *Net Weight*: 15 lbs. *Power Consumption*: 20 watts at "no signal"; 240 watts maximum, 50 to 60 Hz, 120 or 240 volts.

Figure 3-8. Interconnection diagram and specifications for the Dynaco SCA-50 integrated amplifier

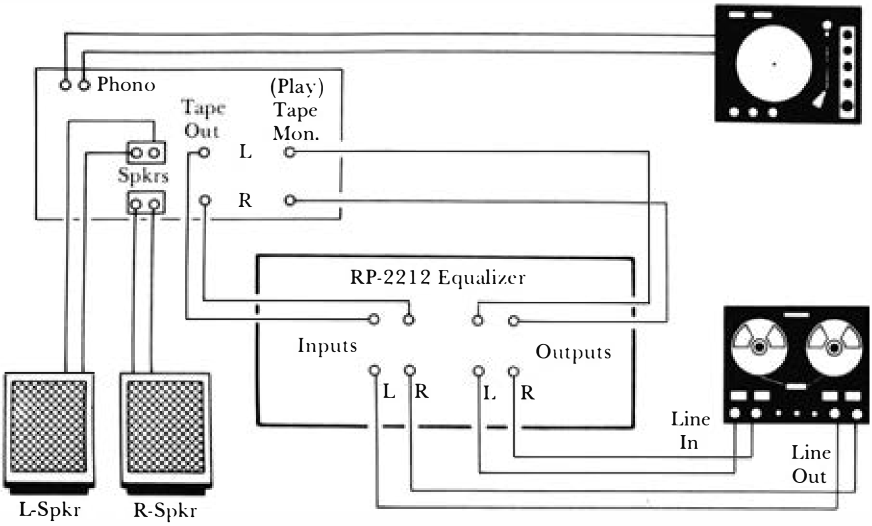


FM SECTION:
Usable Sensitivity: Mono: 1.8 μV (10.3 dBf). *S/N Ratio*: Mono: 70 dB. *Selectivity*: 75 dB. *Capture Ratio*: 1.0 dB. *Total Harmonic Distortion*: Mono: 0.2% at 1 kHz; Stereo: 0.4% at 1 kHz. *Image Rejection*: 90 dB. *IF Rejection*: 90 dB. *Spurious Response Rejection*: 90 dB. *AM Suppression*: 50 dB. *Frequency Response*: 40 Hz to 14 kHz, ±1.5 dB. *Stereo Separation*: 1 kHz: 38 dB at 1 kHz, 30 dB from 50 Hz to 10 kHz.

AM SECTION:
Sensitivity: 200 μV/M (Internal Antenna). *Selectivity*: 29 dB. *S/N Ratio*: 45 dB. *Image Rejection*: 60 dB. *IF Rejection*: 60 dB. *Total Harmonic Distortion*: 1.0%.

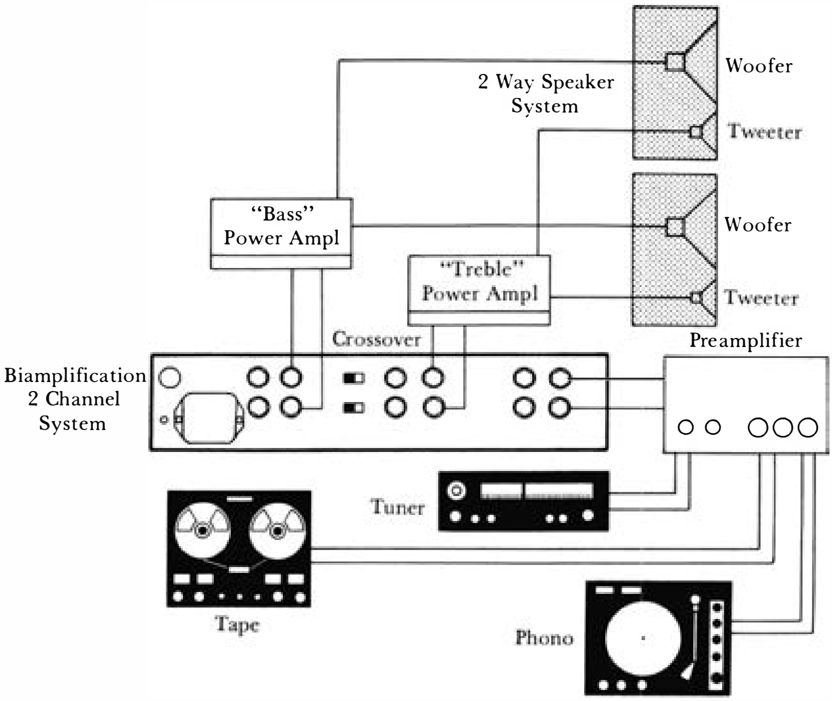
GENERAL SPECIFICATIONS:
Audio Output Level: Fixed, FM: 0.8 V; Variable FM: 0–1.0 V; Fixed, AM: 0.2 V for 30% modulation; Variable AM: 0–0.3 V for 30% modulation. *Power Consumption*: 24 watts.

Figure 3-9. Interconnection diagram and specifications for the Optonica ST-3535 tuner.



Frequency Response: 20-Hz to 20,000-Hz ±0.5-dB. *Harmonic Distortion*: Less than 0.1% at 2 volts (typically 0.05% at 1 volt in and out). *IM Distortion*: Less than 0.1% at 2 volts (typically 0.05% at 1 volt). *Signal-to-Noise Ratio*: Better than 90-dB referred to 2-volt input. *Input Impedance*: Operates from any signal source 100 K ohms or less. *Output impedance*: Operable into 3 K ohms or greater. *Control range*: ±12 dB, each octave.

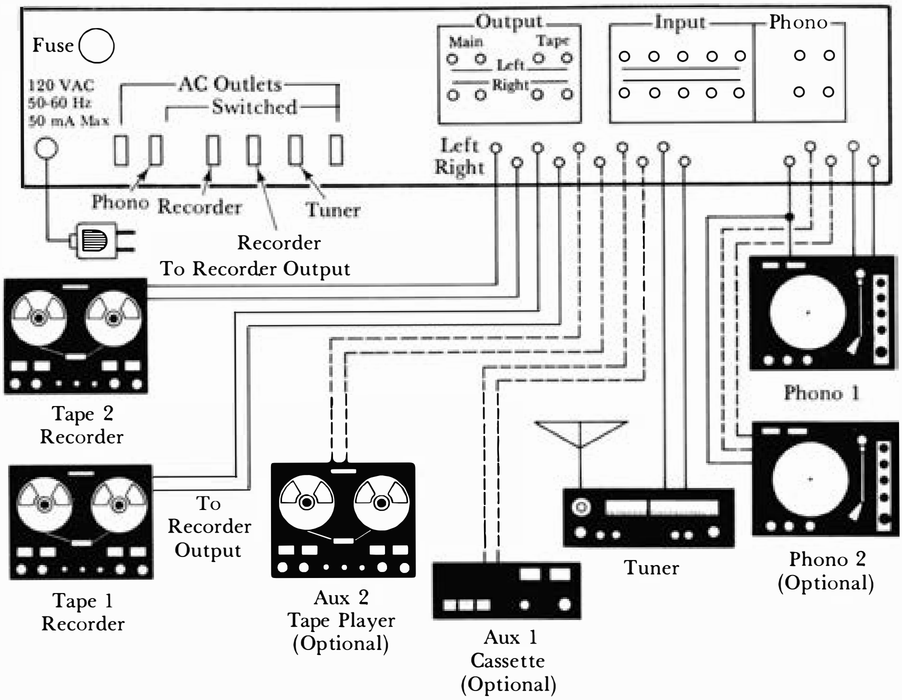
Figure 3-10. Interconnection diagram and specifications for the Soundcraftsmen RP2212 integrated amplifier.



Frequency Response: 18 Hz to 38 kHz ±0.5-dB into 600-ohm load.
Inputs: Bridging input, 20 K ohm balanced or 10 K ohm unbalanced, and 1 M ohm unbalanced; both using ¼ in. standard phone jack.
Output: 10 volts maximum before overload; 2.5 volts rated.
Gain: 0 to 15.5 dB from balanced/unbalanced input.
Hum and Noise: More than 100-dB below rated output, with 0-dB gain, over entire audio spectrum from 20 Hz to 20 kHz.
IM Distortion: Less than 0.01% at rated output.
Filter Characteristics: Separate 18-dB Butterworth highpass and lowpass, with adjustable corner frequencies. Can be internally cascaded to form band pass and band reject filters.
Controls: (front panel) Range and vernier controls for corner frequencies (high and low pass), power on/off switch. (rear panel): Screwdriver-adjustable input attenuators for each channel.

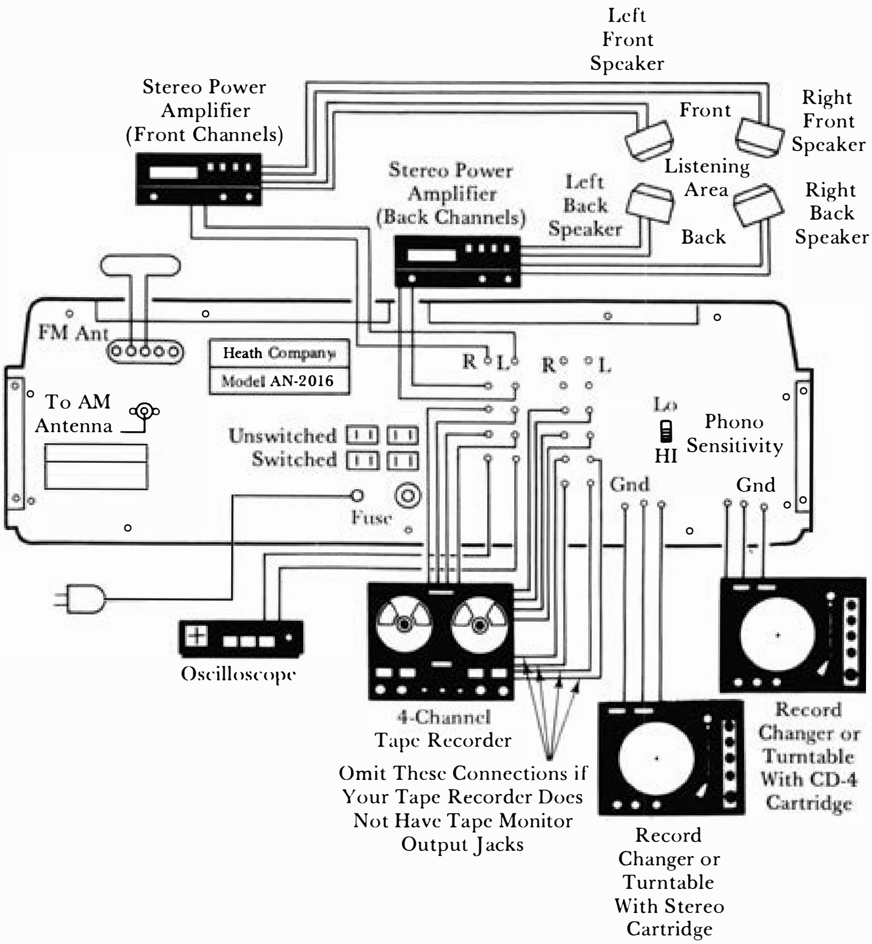
(a)

Figure 3-11. (a) Interconnection diagram and specifications for the Crown International VFX-2 power amplifier.



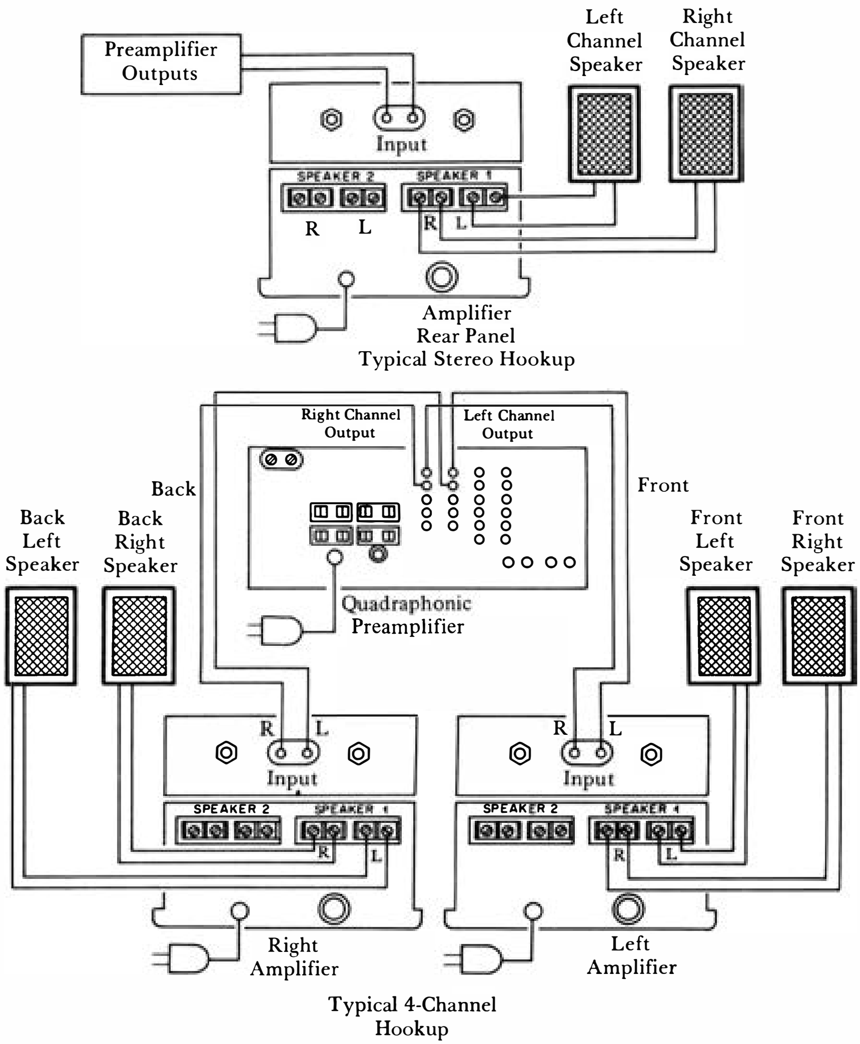
(b)

Figure 3-11. (cont.) (b) Interconnection diagram for the Crown International IC-150A preamplifier.



(a)

Figure 3-12. (a) Interconnection diagram for the Heath AN-2016 preamplifier.



(b)

Figure 3-12. (cont.) (b) Interconnection diagram for the Heath AN-2016 power amplifier.

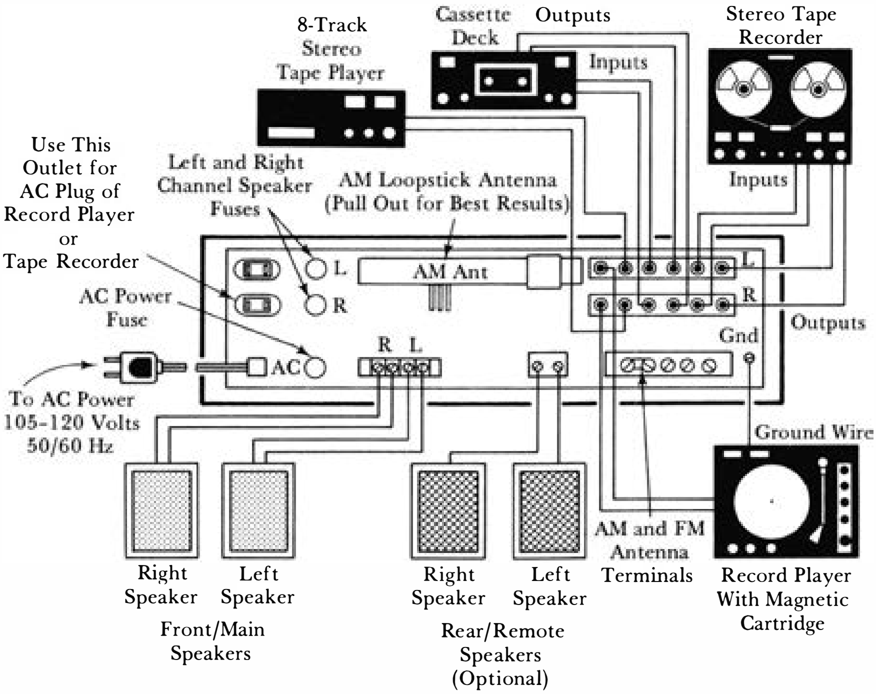


Figure 3-13. Interconnection diagram for the Lafayette LR-2200 integrated receiver.

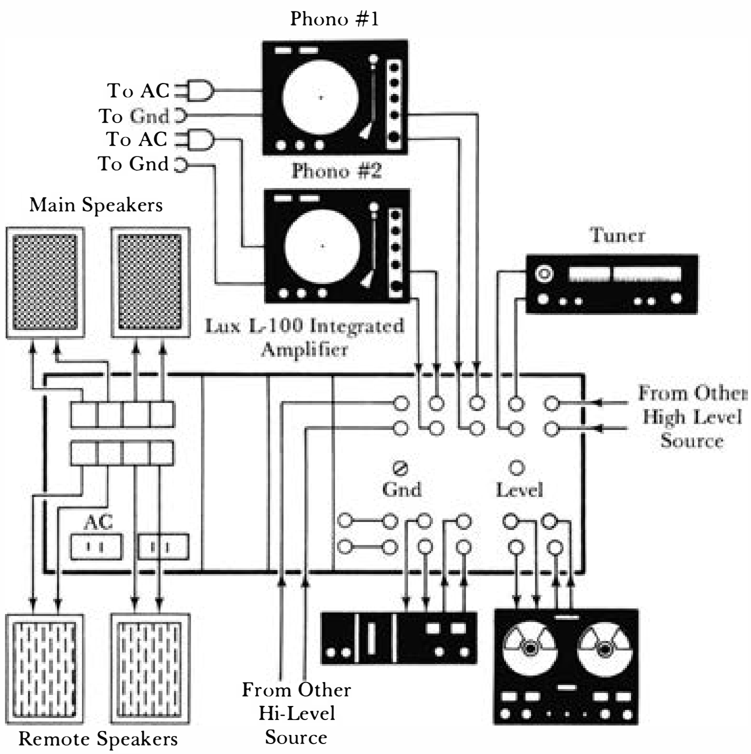


Figure 3-14. Interconnection diagram for the Lux L-100 integrated amplifier.

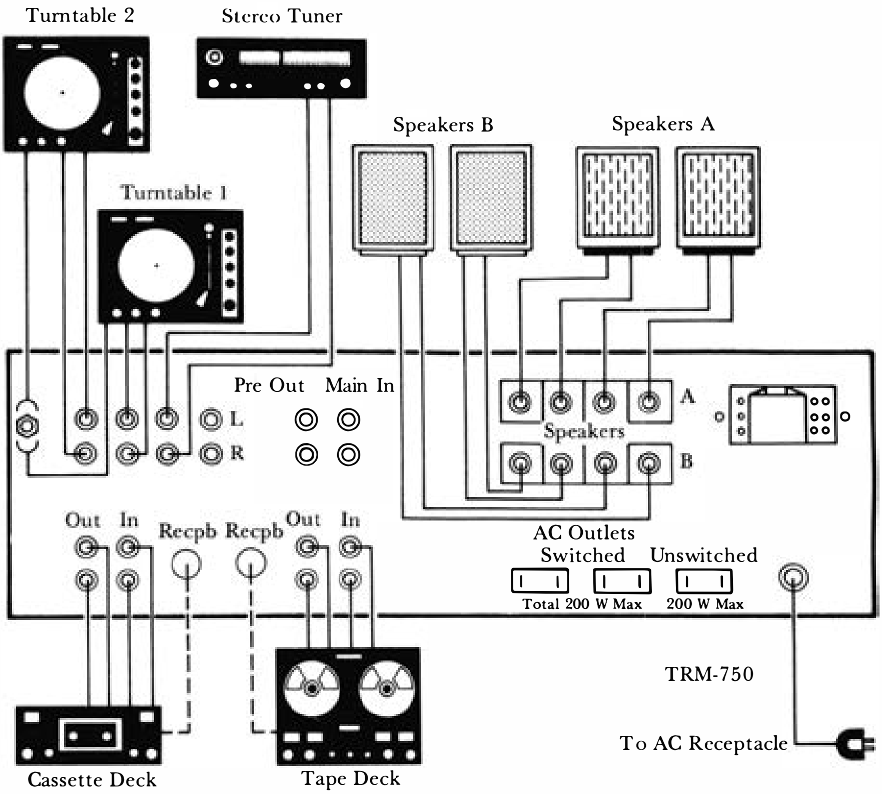


Figure 3-15. Interconnection diagram for the Nikko TRM-750 integrated amplifier.

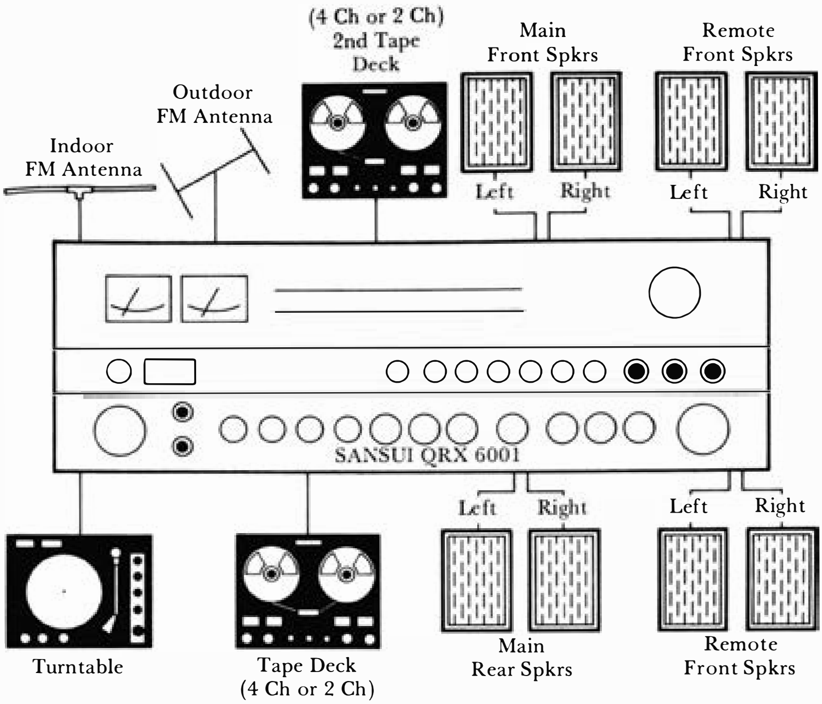


Figure 3-16. Interconnection diagram for the Sansui QR integrated receiver.

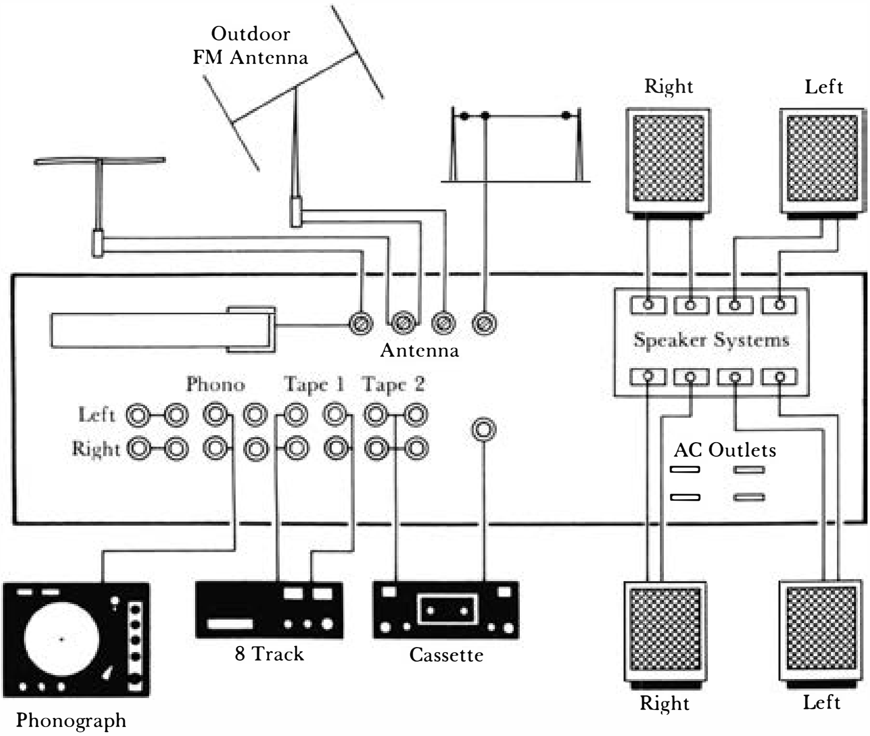


Figure 3-17. Interconnection diagram for the JVC S-300 integrated receiver.

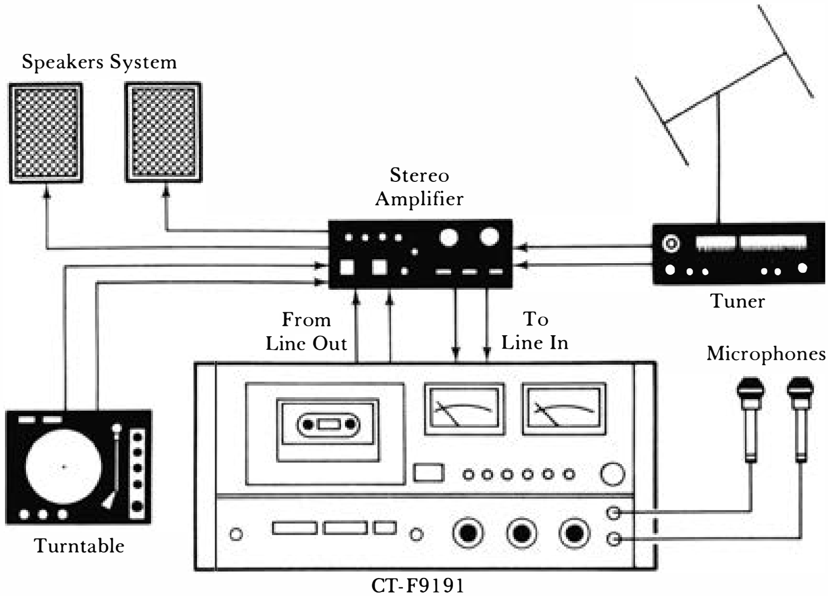


Figure 3-18. Interconnection diagram for the Pioneer CT-F9191 power amplifier.

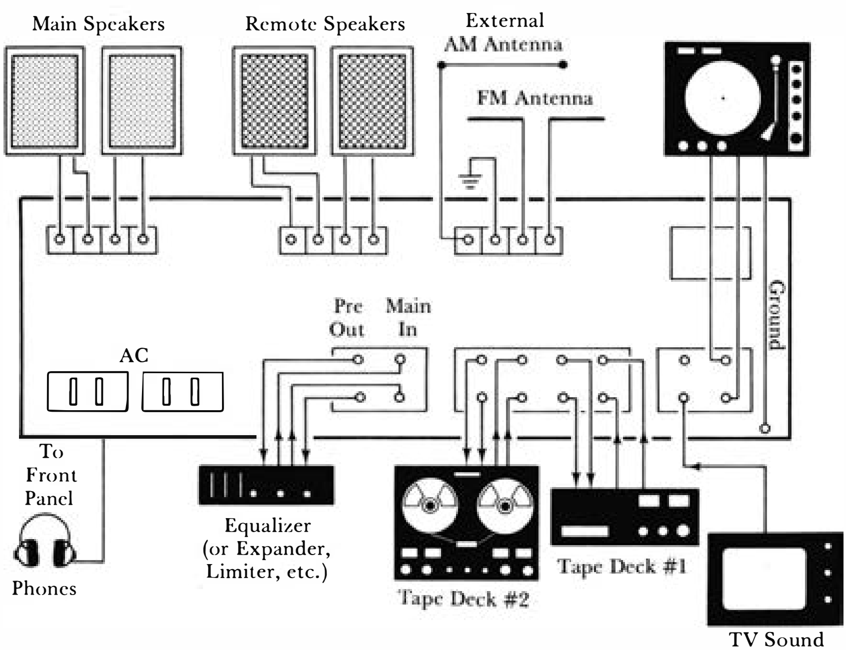


Figure 3-19. Interconnection diagram for the Marantz 2325 integrated receiver.

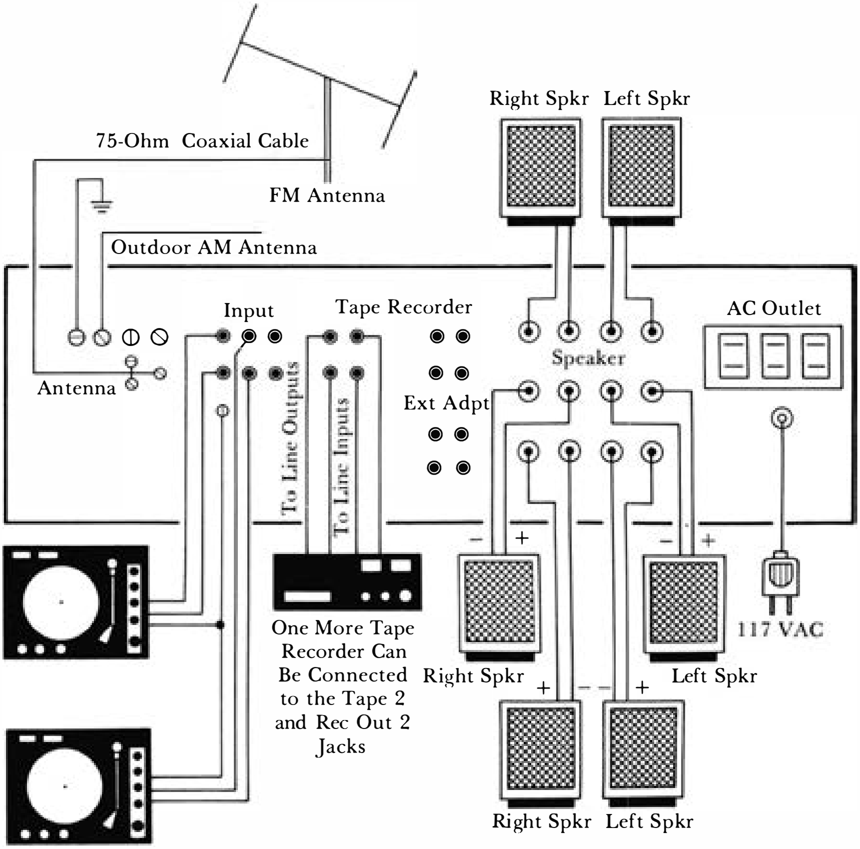


Figure 3-20. Interconnection diagram for the Sony STR-6800SD integrated receiver.

Chapter 4

Antenna Installation

4-1 GENERAL CONSIDERATIONS

Most FM/AM tuners contain a built-in ferrite-core antenna for AM reception, such as shown in Fig. 4-1. In most situations, a built-in antenna is satisfactory for standard broadcast reception. However, if AM reception is weak, noisy, or subject to interference in a particular location, the installer can supplement the built-in antenna for improved reception by means of an external antenna. Sometimes a short indoor external antenna will be found adequate. For example, a 10- to 20-ft length of insulated, flexible, single-conductor wire may be connected to the "Ext AM Ant" terminal on the back of the tuner. This antenna wire should be well separated from all speaker, audio, and power cables. Run the antenna wire in a straight line along a nonmetallic baseboard or under a rug. In some cases, improved reception may be obtained by draping the antenna wire out a window. In difficult situations, the installer may need to employ an elevated antenna wire. For example, a 50-ft length of antenna wire may be strung between insulators and supported by poles on the roof of the building. In turn, a lead-in wire is run from the antenna down the side of the building, and under a window sash (or equivalent) to the tuner. Installation details are covered later in this chapter.

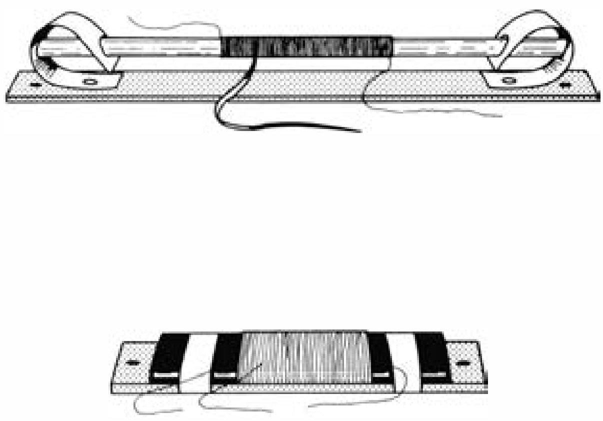


Figure 4-1. Two types of built-in ferrite-core antenna commonly provided in AM tuners.

Most FM tuners do not have built-in antenna arrangements. Instead, it is general practice to connect an indoor dipole antenna to the "FM Ant" terminals on the back of the tuner (see Fig. 4-2). Note in passing that a dipole antenna (like a ferrite-core antenna) is directional to some extent. In turn, improved reception can sometimes be realized by orienting a dipole antenna in the optimum direction. An installer is occasionally confronted with the problem of *multipath reception.* In other words, in some strong-signal

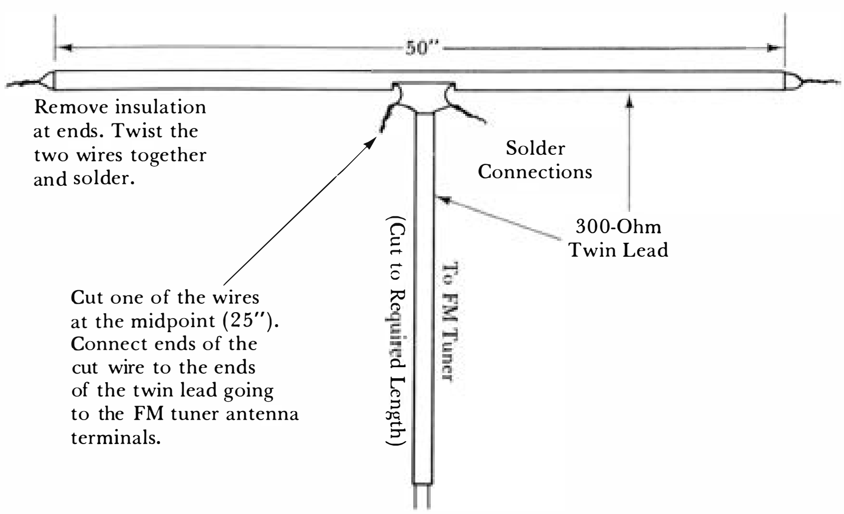
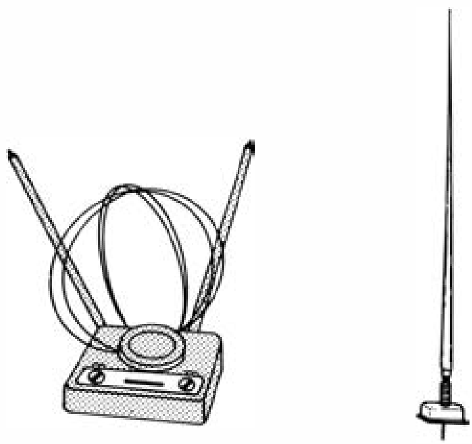


Figure 4-2. A folded dipole antenna constructed from a 300-Ω twin lead is generally used with an FM tuner.

locations, pronounced signal reflections from surrounding buildings, towers, or hills may cause severe multipath interference. This trouble is similar to "ghosts" in television picture reproduction; it causes distortion, "fuzziness," and reduced left-right separation in FM stereo reception. When this difficulty is encountered, it may be necessary to replace the indoor folded-dipole antenna with an indoor "rabbit-ear" antenna or with a telescoping dipole antenna that can be rotated for best reception of the desired signal and maximum rejection of the unwanted reflected signal(s). These specialized types of indoor antenna are illustrated in Fig. 4-3.



(a) (b)

Figure 4-3. Two commercial types of indoor antenna: (a) rabbit-ear antenna; (b) telescoping dipole antenna.

Sometimes an antenna appears to perform poorly because of a faulty connection, such as an accidental short circuit between the antenna-input terminals. A rabbit-ear antenna should also be oriented in various directions, to determine the direction of optimum reception. This orientation is not necessarily the same from one FM station to another. Reception in fringe areas may be unsatisfactory with any type of indoor antenna, owing to weak field strength and electrical interference. In such areas, an outdoor FM antenna must be employed. In some cases, an outdoor VHF television antenna will be available. If the FM broadcast signals arrive from the same general direction as the TV signals, the television antenna may serve satisfactorily for FM reception. Or, if the TV antenna is installed with a rotor, the direction of the arriving FM signals will be inconsequential. If a test shows that the TV antenna is adequate for FM reception, a two-set antenna coupler should then be installed so that both the TV receiver and the FM tuner can be operated simultaneously (see Fig. 4-4).

Figure 4-4. Two-set antenna coupler. (*Courtesy Winegard*)

4-2 ANTENNA INSTALLATION

A high-gain VHF/UHF/FM antenna has the construction exemplified in Fig. 4-5. This design is quite directional and usually requires a rotor. Directionality is a desirable feature in fringe-area installations, because interference and electrical noise are thereby reduced. Two-set couplers are usually installed at or near the receiver location; antenna rotor controls

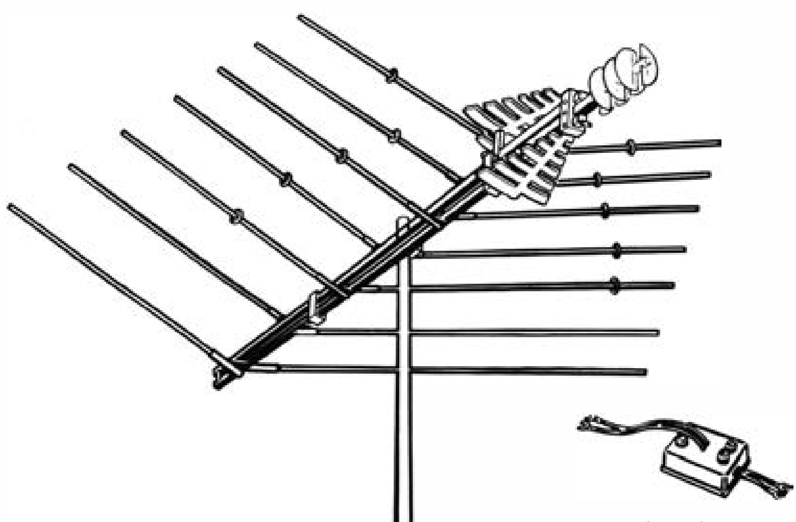


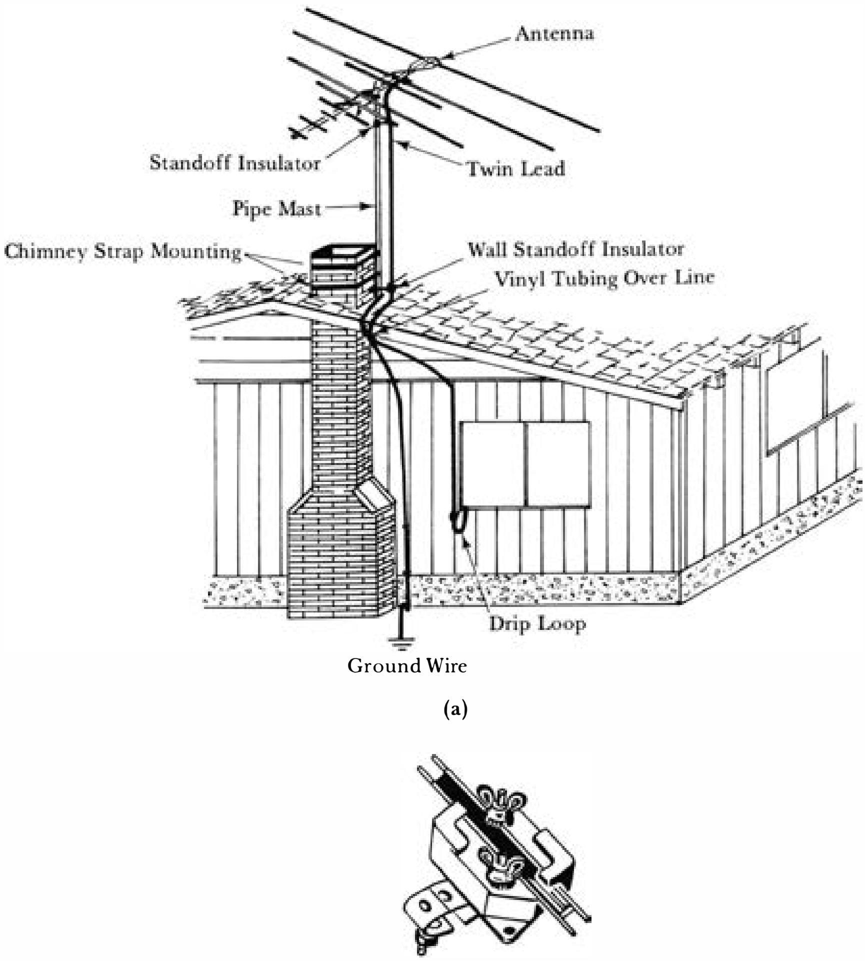
Figure 4-5. High-gain VHF/UHF/FM antenna. (*Courtesy JFD*)

(Fig. 4-6) are also installed near the receiver. Outdoor antennas are often installed without any lightning protection; however, this is very poor practice, and it is a violation of the National Electrical Code. Most outdoor an-

Figure 4-6. Antenna rotor control. (*Courtesy Alliance Corp.*)

tennas are mounted on metal tubing, with a metal boom that is secured to the tubing, or mast. In such a case, adequate lightning protection is obtained by grounding the mast as depicted in Fig. 4-7. The essential requirement is to connect the mast as directly as possible to a ground rod; this connection should be made with heavy copper wire. The ground rod ordinarily consists of a metal pipe driven several feet into the soil. If the ground wire can be run directly to a cold-water pipe, this type of ground may be used. Sometimes an outdoor antenna is not mounted on a metallic mast. In such a case, the National Electrical Code requires that a lightning arrester be connected in series with the lead-in near its point of entry into the building. A ground wire is then run as directly as possible to a ground rod or to a cold-water pipe.

When reception is marginal for one or more desired stations, three chief methods are available to increase the signal strength. If it is practical to increase the height of the antenna above ground, the signal strength can often be increased appreciably. It is also advantageous to install a special FM antenna of the Yagi type. If a Yagi antenna has eight or ten elements, its gain will be considerably greater than that of a conventional TV antenna. Its bandwidth is also considerably less; therefore such an antenna must be designed for reception in the 88- to 108-MHz range. An eight-element Yagi antenna is depicted in Fig. 4-8. It is highly directional and must



(b)

Figure 4-7. Outdoor antenna installation: (a) arrangement; (b) lightning arrester.

be installed with a rotor. Note that the strength of an antenna signal can be increased with the aid of a suitable amplifier, called a *booster,* as illustrated in Fig. 4-9. A booster should be installed high on the antenna mast, where the signal-to-noise ratio is best. In other words, a booster cannot improve the signal-to-noise ratio of its input signal; only an antenna with higher gain and sharper directivity can improve the signal-to-noise ratio. Since the lead-in tends to pick up appreciable noise voltage, a booster should be installed at the antenna end of the lead-in.

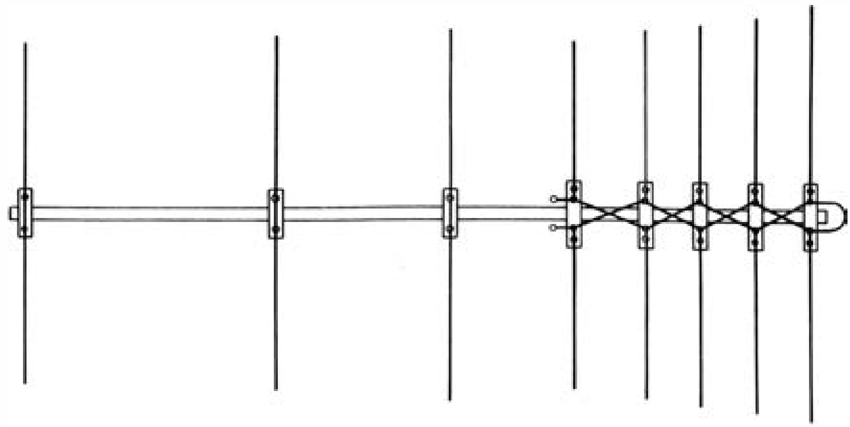


Figure 4-8. Example of a Yagi antenna with eight elements.

Figure 4-9. FM booster for fringe-area reception. (*Courtesy Winegard*)

If an FM antenna installation is made near a busy thoroughfare or industrial area and an outdoor antenna is connected to the tuner with conventional 300-Ω twin lead, interference from automotive ignition systems or from electrical machinery may radiate into the long lead-in and cause objectionable noise interference with FM reception. In such a case, the conventional twin lead should be replaced with *shielded* 300-Ω twin lead. This type of lead-in is available at many high-fidelity stores. Installation is made in the same manner as for conventional twin lead, except that the shield braid is connected to the "Gnd" terminal on the chassis near the "Phono In" connectors. Make certain that a stray strand from the shield braid does not make accidental contact with the antenna-input terminals.

4-3 ELECTRONIC FM ANTENNA

When reception is marginal with a conventional indoor folded-dipole antenna, it is often possible to obtain satisfactory reception with an indoor electronic FM antenna instead of erecting an outdoor antenna. An electronic FM antenna, exemplified in Fig. 4-10, contains a built-in preamplifier that increases the prevailing signal level. This type of antenna may be placed on top of an FM tuner or mounted on a wall with a bracket. It is advisable to move the electronic antenna about the listening room or area to determine the location that provides optimum reception. Note that this antenna design is omnidirectional and does not require orientation in the direction of the incoming signal(s). In the event that reception of all desired FM stations is not provided by an indoor installation, the electronic FM antenna may be installed outdoors on a mast. As noted previously, an antenna booster or preamplifier cannot improve the existing signal-to-noise ratio—it can only step up the amplitude of the signal and noise. Therefore, unless the antenna is installed in a location that provides a reasonable signal-to-noise ratio, a booster amplifier will serve no useful purpose.

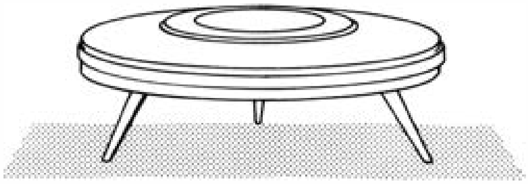


Figure 4-10. Appearance of an electronic FM antenna.

4-4 MASTER-ANTENNA INSTALLATION

Master-antenna installations, exemplified in Fig. 4-11, are used to energize several hi-fi units from a single antenna. To obtain ample signal strength at each FM/AM tuner, and to provide isolation between the receivers in the system, a broadband amplifier is utilized between the antenna and the individual receivers. Isolation is provided by means of multiple outputs from the amplifier; these output circuits have no common coupling, so the tuning of one receiver cannot affect reception of another receiver, whether it is tuned to the same or to a different FM channel. Only three hi-fi units are depicted in Fig. 4-11. However, an elaborate master-antenna installation can serve as many as 100 hi-fi units.

A community antenna system serves comparatively large numbers of subscribers at comparatively great distances from the antenna installation.

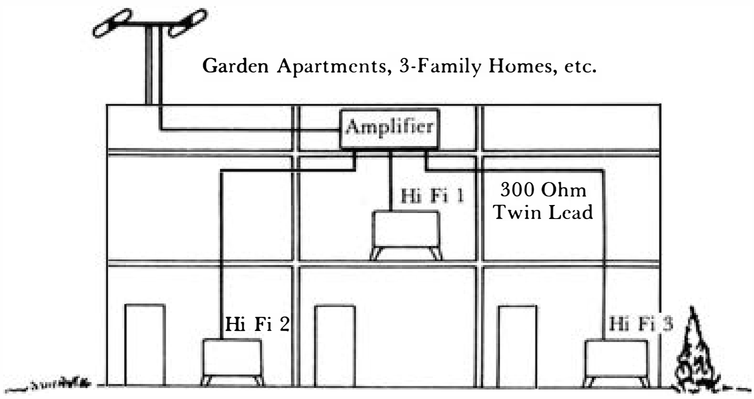


Figure 4-11. Master-antenna installation for a multiple dwelling.

As shown in Fig. 4-12, FM broadcast signals and television signals are processed. A head end in a community antenna system includes a VHF amplifier for each antenna and a signal combiner. A translator heterodynes a UHF signal into a VHF signal to minimize the cable loss on long runs. A trunk cable conducts the signal energy from its originating site, such as on top of a mountain, to the utilization site, such as a town. Coaxial cable imposes a signal loss of about 1 dB/100 ft. In turn, an amplification of approximately 50 dB/mile of cable is required. A bridger or bridging amplifier splits the signal into several portions for supplying feeder lines.

Feeder amplifiers are also called *line extenders.* Both the main trunk amplifiers and the feeder amplifiers provide a gain up to 25 dB over the frequency range 50 to 220 MHz (the FM broadcast band extends from 88 to 108 MHz). The impedance of a community antenna system is 75 Ω throughout. Amplifiers and signal splitters are usually provided with a tilt control. This is a filter of the bandpass type with an insertion loss that is a function of frequency. The high end of the band is usually attenuated to some extent at the end of a run, because coaxial cable has higher loss at higher frequencies. Distribution cables are tapped off from feeder cables. Tap points are called *subscriber taps.* A trunk line is continued through a bridging amplifier. On the other hand, the last amplifier on a trunk cable is called a *distribution amplifier.* Feeder lines can be run up to 1000 feet from a bridger before another amplifier (line extender) is required.

As indicated in Fig. 4-12, a *subscriber drop* is a cable that connects to a tap point along a feeder line. An isolating resistor may be used between the subscriber drop and the feeder line. A small capacitor may be used instead of a resistor; a capacitor is sometimes preferred because it introduces some tilt in the frequency response. The signal level provided by a community antenna system is 1500 μV. This is only a moderately strong signal level; however, if the system is operated at a high level, it is difficult to avoid objectionable interference (crosstalk) between channels. Community antenna systems are the practical solution to satisfactory FM reception in far-fringe areas. They are also the practical solution to satisfactory reception in some metropolitan areas that are plagued by intolerable interference from conventional indoor or rooftop antennas.

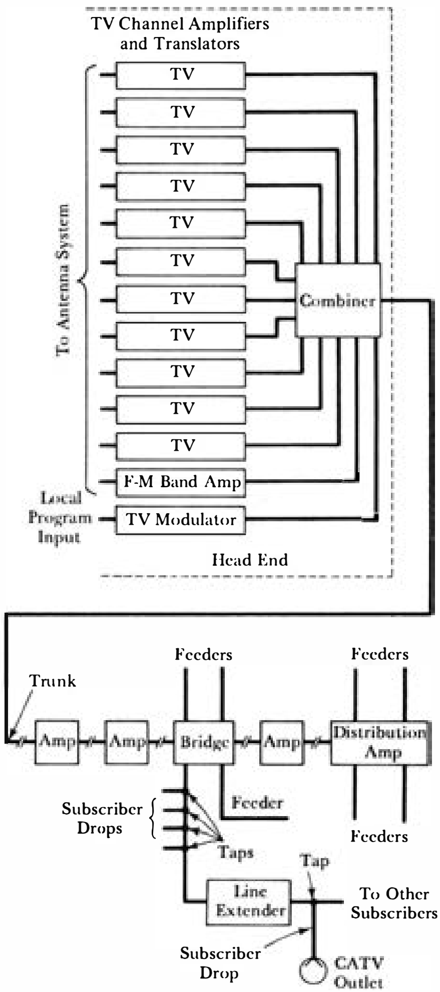


Figure 4-12. Block diagram for a community antenna system.

Chapter 5

Audio System Interference

5-1 GENERAL CONSIDERATIONS

Installation personnel are occasionally concerned with audio system interference, or radio-frequency interference (RFI). This type of interference should not be confused with noise or co-channel interference, which may occur during operation of the FM/AM tuner. As explained in Chapter 4, this difficulty stems from an inadequate antenna installation. RFI, on the other hand, results from pickup of radio-frequency energy by an audio amplifier or by a component connected to the amplifier. Although virtually any audio amplifier picks up more or less stray-field RFI, its level is ordinarily too low to cause system malfunction. However, in a high-level radio-frequency field, an audio system can process sufficient induced signal voltage that distorted sound output and audio interference becomes apparent. This undesired pickup can occur in a tape recorder or in any amplifier, such as in an electronic organ, public-address system, or even in a hearing aid.

Audio interference is the result of "audio rectification"; this term denotes unexpected demodulator (detector) action in a normally linear amplifier. Audio rectification is the result of overdrive (usually in the input stage); when overdrive occurs, a transistor or an integrated circuit can develop sufficient audio rectification that an objectionable amount of interference is passed through the remainder of the audio system to the speakers. Correction of RFI usually involves bypassing, shielding, filtering, and/or choking arrangements added into the amplifier circuitry, as exemplified in Fig. 5-1. An RF choke used for suppression of RFI may have a value from 5 to 7 microhenries (μH) for frequencies in the range from 30 to 110 MHz, or

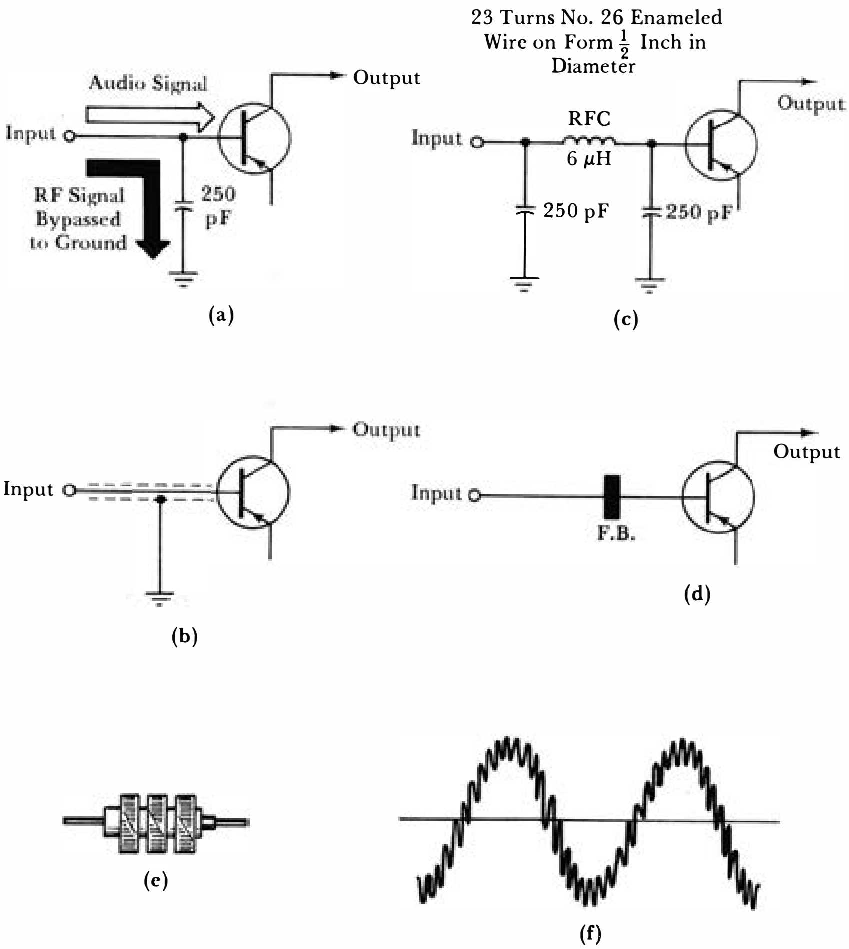
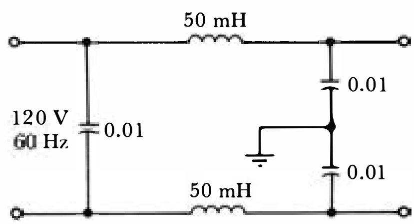


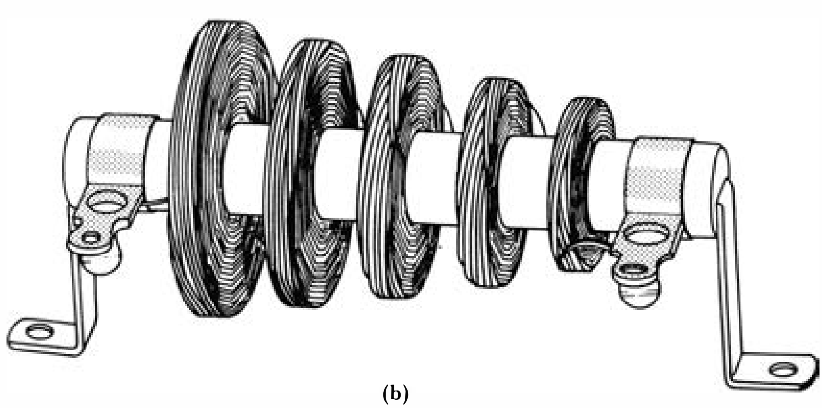
Figure 5-1. Skeleton amplifier circuits, with basic filter arrangements: (a) bypass capacitor added in base circuit; (b) shielded input lead utilized in base circuit; (c) pi-filter section inserted in base lead; (d) ferrite-bead choke added in base circuit; (e) a universal-wound RF choke coil; (f) audio waveform with RF interference.

1.5 μH in the range from 80 to 200 MHz. A 250-picofarad (pF) bypass capacitor is usually suitable for RFI suppression.

In most situations, audio rectification occurs in the first stage of a preamplifier, because the input stage operates at the lowest level and is most susceptible to overloading. The input stage is also followed by a high-gain amplifier system, so that any interference developed by the first stage is stepped up greatly in the following portion of the system. When RFI is encountered, it is often helpful to check its level as the setting of the volume control is varied. If the interference is minimized when the volume control is turned to its minimum position, the operator concludes that the point of entry is located ahead of the volume control. On the other hand, if the volume-control setting has no effect on the interference level, the operator concludes that the point of entry is in the circuitry following the volume control. In some situations, it will be found that part or all of the RFI voltage is entering via the power cord to the outlet. In this case, a pi-type power-line filter such as shown in Fig. 5-2 may be effective. Note that the



(a)



(b)

Figure 5-2. Pi-type line filter: (a) circuit; (b) typical heavy-current RF choke.

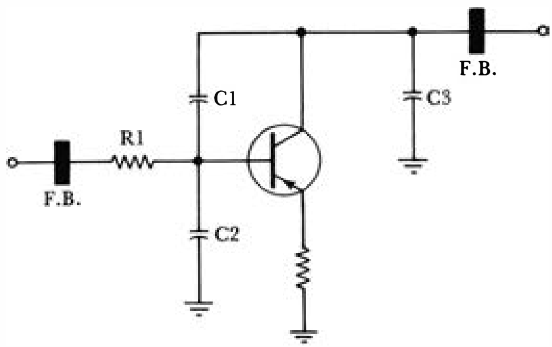


Figure 5-3. Ferrite beads function as RF chokes and supplement bypass-capacitor action for suppression of RFI.

50-mH choke coils must carry the total current demand of the power supplies in the audio system; therefore, they must be suitably rated for the application.

Consider next a situation in which a line filter reduces the audio interference, although its level remains objectionably high. Additional reduction can usually be obtained by means of RF filtering in the amplifier circuitry. A basic method is depicted in Fig. 5-3. Ferrite beads are inserted on the base and collector leads to function as high-frequency RF chokes. Bypass capacitors are provided from the base and collector terminals to ground. Values for C1, C2, and C3 must be determined experimentally for a particular amplifier. Also, the optimum value for R1 should be determined experimentally. These values represent a tradeoff between suppression of audio interference and loss in stage response. Excessive values of capacitance and/or resistance can also impair the high-frequency response of the stage.

Another possible route of entry (although less probable) for RFI is at the output end of an amplifier. As an illustration, speaker leads may happen to resonate at the interfering frequency and thereby develop much more interference potential than otherwise. Thus, an 8-ft length of speaker lead is resonant at approximately 30 MHz. In turn, it is occasionally helpful to bypass the audio output circuit to ground via small capacitors. Note that if an audio amplifier is properly stabilized, it will tolerate comparatively large values of capacitance across its speaker terminals without serious deterioration in performance. When a microphone or tape preamplifier at the input end of a system is the culprit, the first approach is to provide RF filtering for the input stage, as exemplified in Fig. 5-4. RFI is suppressed by a 1-kilohm (kΩ) resistor and a 200-pF capacitor in the case of a single transistor configuration. However, if a Darlington configuration is

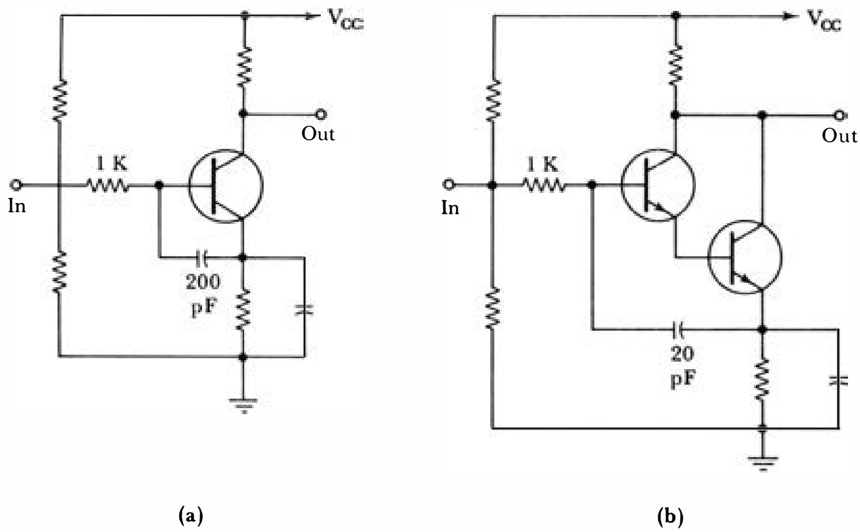


Figure 5-4. RFI filter configurations for a preamplifier stage: (a) single transistor circuit; (b) Darlington circuit.

utilized, a capacitance value of 20 pF is employed. These filter arrangements will attenuate interference over TV channels 4 through 6, and also through the 88- to 108-MHz FM broadcast band. If it is determined that RFI is being caused by AM broadcast fields, the 1-kΩ resistor is omitted and a 1-mH choke is used instead, with the capacitance value doubled. Or, in a case of UHF interference, a ferrite bead is utilized instead of a 1-kΩ resistor, and the capacitance value is halved.

5-2 METHODS OF SHIELDING AND FILTERING

On occasion, RFI becomes discernible only when the pickup arm or the turntable of a phono player is touched. In this situation, a ground wire should be run between the pickup arm and the preamp chassis ground. In case the pickup headshell is fabricated from Bakelite or plastic, a small metallic shield may be inserted between the cartridge and the pickup arm, as shown in Fig. 5-5. In turn, the shield plate should be grounded to the pickup arm. Sometimes, RFI becomes noticeable when a microphone is grasped. This difficulty can be corrected by connection of a ground wire between the microphone housing and the preamp chassis ground. Note also that "audio rectification" can result from cold-solder connections, poor ground connections, or poor solder bonds, as shown in Fig. 5-6. If a defec-

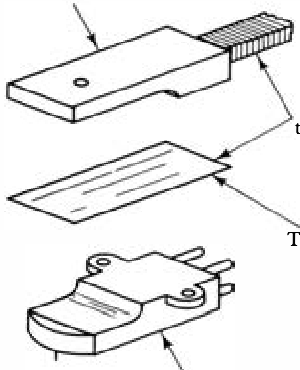


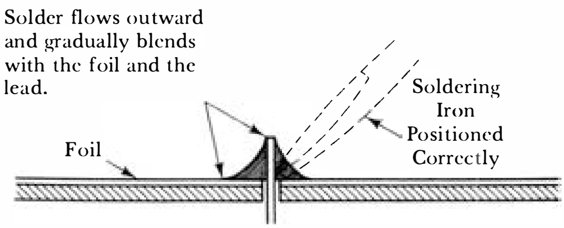
Figure 5-5. A grounded shield plate may be inserted between a cartridge and the pickup arm.

tive ground is checked with a low-ohms meter, it is often found that a different resistance reading is obtained when the meter leads are reversed. In other words, the poor connection develops a nonlinear resistance characteristic and partially rectifies the signal current.

In some situations, even if a ground connection is good and has a linear resistance characteristic, it may be helpful to increase the lead inductance and thereby provide some RF filter action to reduce RFI. As an illustration, suppose that the emitter of a preamp transistor is connected through a resistor to ground, as depicted in Fig. 5-3. To reduce the amplitude of the RFI voltage at the emitter terminal, it is helpful in some situations to connect an RF choke between the emitter terminal and the ground lead. This RF choke should have an inductance value in the range 1 to 10 mH; its optimum value may be determined experimentally. In effect, this RF choke impedes the flow of high-frequency current in the emitter lead, while permitting audio-frequency currents to pass. Thus, the emitter lead becomes substantially degenerative for RFI. In the event that the RFI is in the UHF range, a ferrite bead may prove more effective than a conventional RF choke.

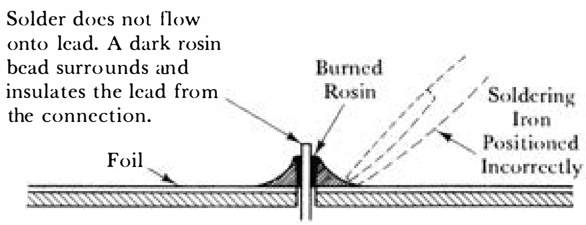
When a difficult RFI problem is encountered in a hi-fi system and cannot be satisfactorily controlled by the foregoing methods, it may become advisable to enclose some or all of the equipment in a screened structure. Thus, the hi-fi equipment may be installed in a decorative cabinet that has been lined with copper screen, for example. Radio-frequency fields cannot penetrate the screening. Note that it is necessary to ground the

A GOOD SOLDER CONNECTION

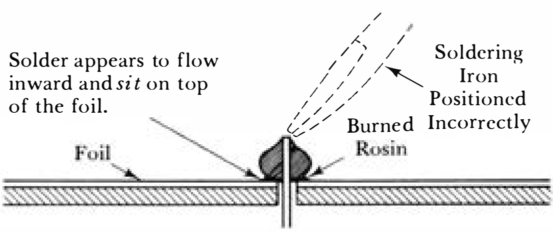


When both the lead *and* the circuit board foil are heated at the same time, the solder will flow onto the lead and the foil evenly. The solder will then make a good electrical connection between the lead and the foil.

BAD SOLDER CONNECTIONS



When the lead is *not* heated sufficiently, the solder will not flow onto the lead as shown above. Reheat the connection and, if necessary, apply a small amount of additional solder to obtain a connection as shown under "A Good Solder Connection."



When the foil is *not* heated sufficiently, the solder will blob on the circuit board as shown above. Reheat the connection and, if necessary, apply a small amount of additional solder to obtain a connection as shown under "A Good Solder Connection."

Figure 5-6. Poor solder connections can cause "audio rectification." (*Courtesy Heath Co.*)

screening to a good earth ground, such as a cold-water pipe. If "audio rectification" remains audible after the equipment has been shielded in the foregoing manner, it is indicated that the interfering voltage is being conducted into the equipment via the power cord or the speaker leads. In such a case, a power-line filter should be installed just inside the screening. Speaker leads can be bypassed to ground with small capacitors located on the inner wall of the screening at the point of exit for the speaker leads. Alternatively, shielded cables may be installed to the speakers, with the outer braid of the cable connected to the screening.

5-3 INTERFERENCE IN AUTOMOBILE INSTALLATIONS

Apart from "audio rectification," excessive interference and noise is likely to be encountered in automobile installations, unless the electrical system of the vehicle has been "quieted" and all poor connections in the grounding system have been corrected. Interfering noise sources include the ignition system, the alternator or dc generator, relay-type voltage regulators, electric motors, gas-gage circuits, loose metal parts that make intermittent ground connection, and various electrical and electromechanical accessories. A basic circuit arrangement for a standard ignition system is shown in Fig. 5-7(a); a basic solid-state ignition system is depicted in (b). When noise interference occurs, the installer must first localize its source. The radio receiver is tuned off-station (if the audio amplifier is driven by a tuner), and the noise output is then analyzed with the motor running. If it is observed that the noise level decreases when the headlights are switched on, the installer concludes that the voltage regulator is producing at least a part of the interference.

Next, it is helpful to remove the antenna lead from the radio; this test may result in reduction or elimination of the noise interference. In turn, the installer suspects that there may be an open shield braid on the coaxial antenna lead, or an ungrounded hood, an ungrounded fender, or other metal structure that is poorly grounded to the chassis of the vehicle. Another helpful test is made by turning the volume control to its minimum position. If the noise level remains unaffected, it is concluded that the noise voltages are probably gaining entry via the dc power lead. Remember that the case of the radio receiver must also be well grounded. Check for the possibility of a loose or a broken ground wire.

A suppressor resistor and a bypass capacitor for an ignition coil may be installed as exemplified in Fig. 5-8. In conventional ignition systems, carbon spark-plug wire may be utilized instead of a suppressor resistor. On the other hand, some solid-state ignition systems operate at extra-high volt-

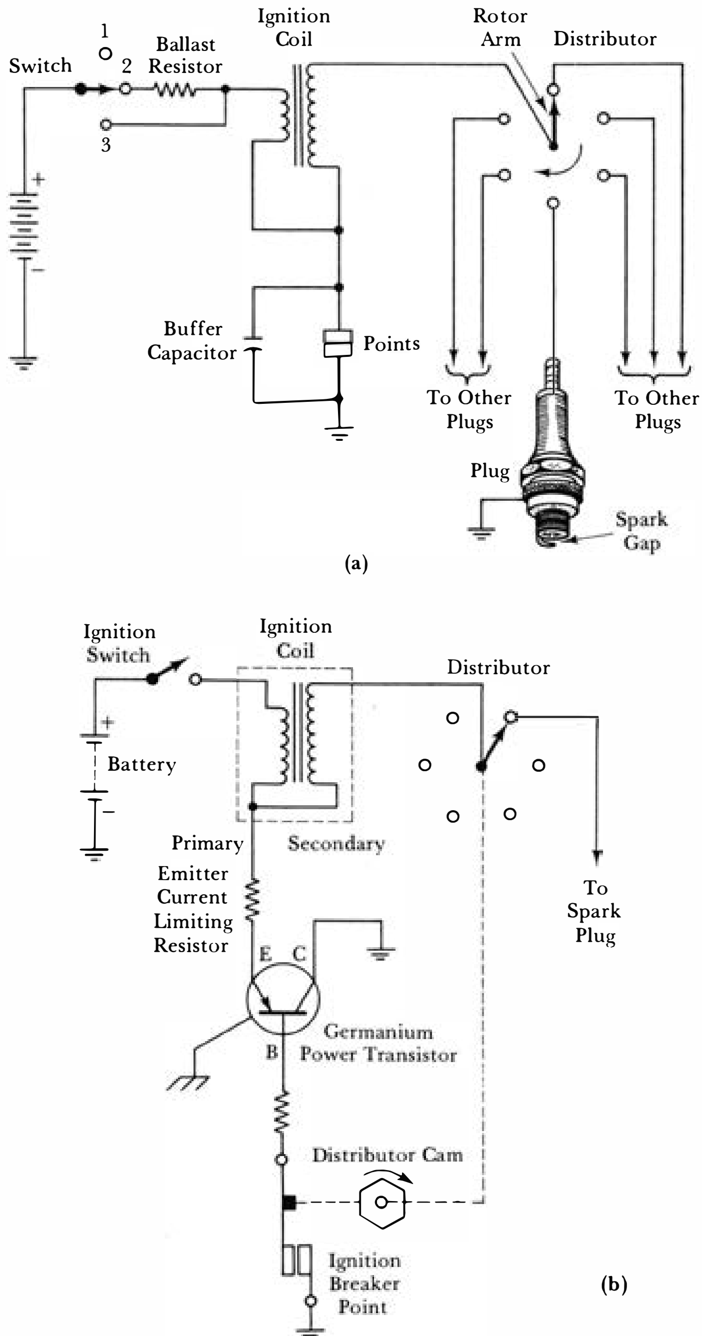


Figure 5-7. Basic ignition system configurations: (a) conventional arrangement; (b) basic solid-state arrangement.

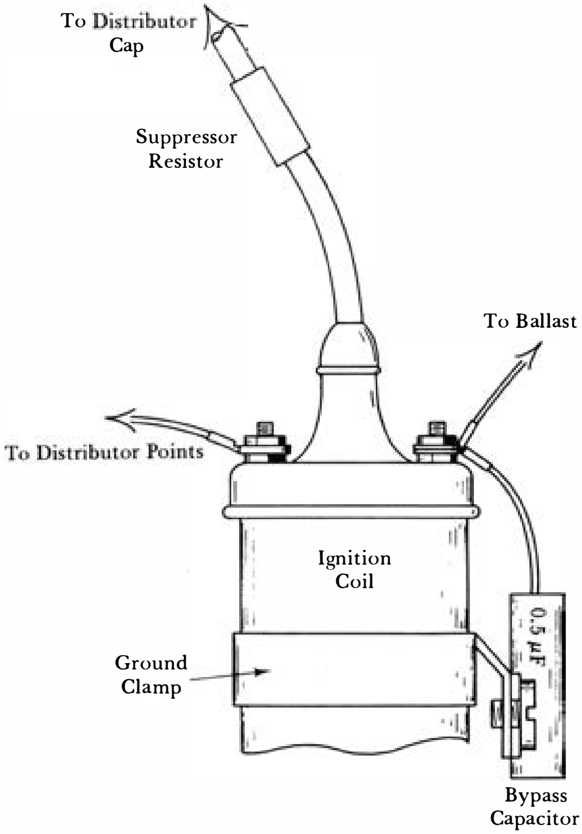


Figure 5-8. Example of suppressor resistor and bypass capacitor.

age, and carbon spark-plug wire will not withstand the necessary stress. In such a case, a suppressor resistor is mandatory. A typical length of carbon spark-plug wire has a resistance in the range 10 to 100 kΩ. Remember that an aged section of carbon spark-plug wire may develop an abnormally high resistance value and result in impaired firing; replace any suspicious sections. Bypass capacitors housed in metal cases are widely used for noise suppression and are usually satisfactory. In difficult situations, it may be found advantageous to install coaxial bypass capacitors. The appearance of a typical feed-through capacitor is seen in Fig. 5-9.

Emergency-brake cables and any power wires that pass through the firewall of the vehicle occasionally pick up noise voltages and reradiate interference to the radio receiver power leads. This difficulty can often be corrected by installing an L-section filter such as shown in Fig. 5-10. A filter

Figure 5-9. Appearance of a typical feed-through capacitor.

is particularly advantageous when the reradiating wire or cable cannot be grounded and cannot be bypassed. Because the winding in the filter inductor (Fig. 5-10) must carry appreciable current, the ferrite core should be wound with wire of the same diameter as that used for the power lead. This ferrite core may be improvised by unwinding the original wire from an AM ferrite-core antenna unit. It is good practice to install an interference filter on the chassis of the radio receiver so that a short connecting lead can be run to the load circuitry. As indicated by the dotted lines in Fig. 5-10, better noise suppression is sometimes obtained if the bypass capacitor is connected at the input end of the filter coil. If necessary, a bypass capacitor can be connected at each end of the filter coil to form a pi-filter configuration.

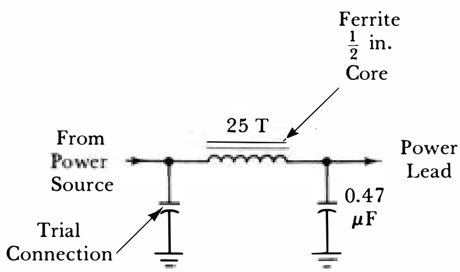


Figure 5-10. Typical L-section interference filter for a power lead.

As noted, the voltage regulator may also generate noise voltages (refer to Fig. 5-11). This filtering arrangement is very effective; in many situations, installation of C1 alone will suffice. However, if residual noise interference persists, include C2, and if necessary, C3 and R; resistor R has a typical value of 4 ohms. Sometimes the gasoline-gage mechanism becomes noisy. This source of noise interference becomes apparent when the automobile is being driven over a rough road. The usual method of coping with this problem is to bypass one or both terminals of the gas-gage meter to ground with a 0.5-μF capacitor. If it is discovered that the horn circuit produces a popping type of interference when traveling over rough roads, the horn connecting lead should be bypassed to ground with a 0.5-μF capaci-

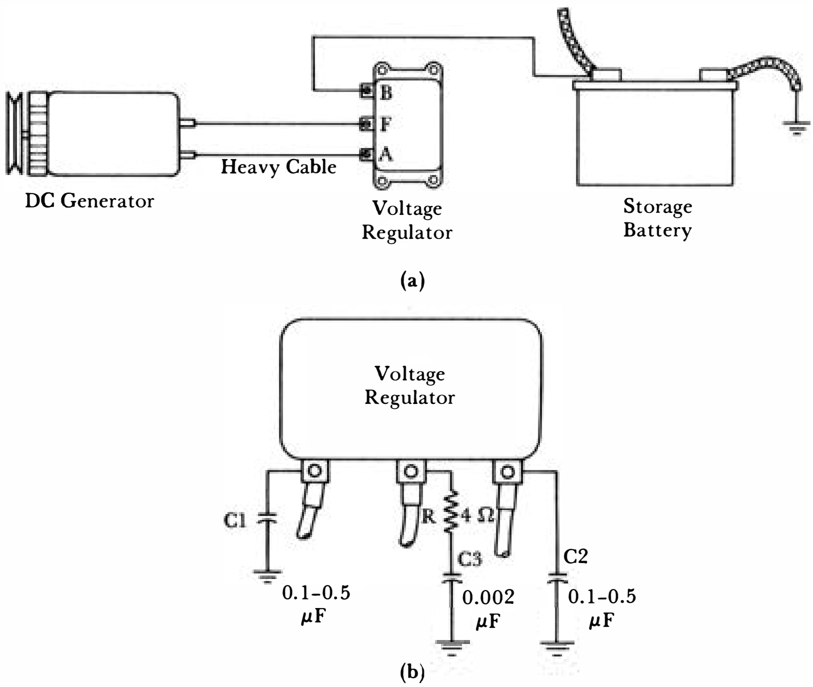


Figure 5-11. Generator type of battery-charging arrangement: (a) regulator circuitry; (b) connection of bypass capacitors.

tor. A noise problem is sometimes tracked down to a structural part of a vehicle that is improperly grounded. Flexible copper braid is generally used to bond such structural parts to the grounded frame.

5-4 TRAP ACTION OF SHIELDED CABLE

In the vicinity of an FM broadcasting station, RFI is likely to be encountered with conventional phono pickups. In other words, the phono input cable can pick up sufficient radio-frequency energy that "audio rectification" becomes noticeable. In this situation, the RF filter network installed in the preamp input circuit should be supplemented by a shielded input cable that is cut to a length such that it provides trap action with respect to the frequency of the interfering signal (refer to Fig. 5-12). Cut the shielded phono cable to an odd multiple or to an odd submultiple of a quarter-wavelength corresponding to the frequency of the interference. For example, one quarter-wavelength at 100 MHz is equal to a length of 0.75

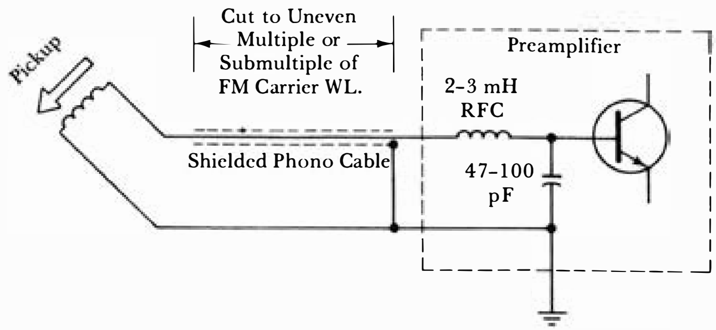


Figure 5-12. RFI trap and filter arrangement for a phono installation.

meter (m), or 29.5 in. In trap application, the braid of the shielded cable is grounded only at the amplifier end. The pickup end of the shielded cable is left ungrounded so that the RF energy will "see" a very high impedance at the input end of the cable. Ground connection to the braid of the shielded cable should be made both to the chassis of the preamp and to a good earth ground such as a cold-water pipe. Characteristics of tuned stubs are summarized in Fig. 5-13.

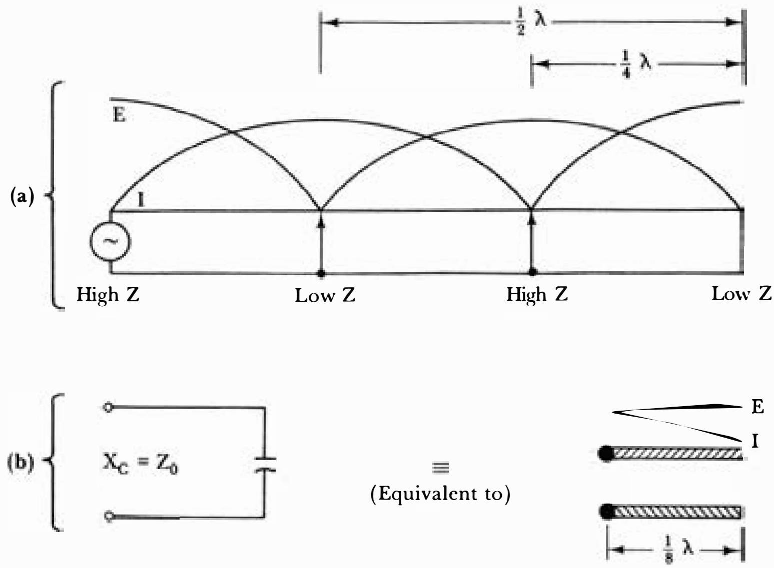


Figure 5-13. Short-circuited and open-circuited line sections (stubs) are equivalent to resonant circuits.

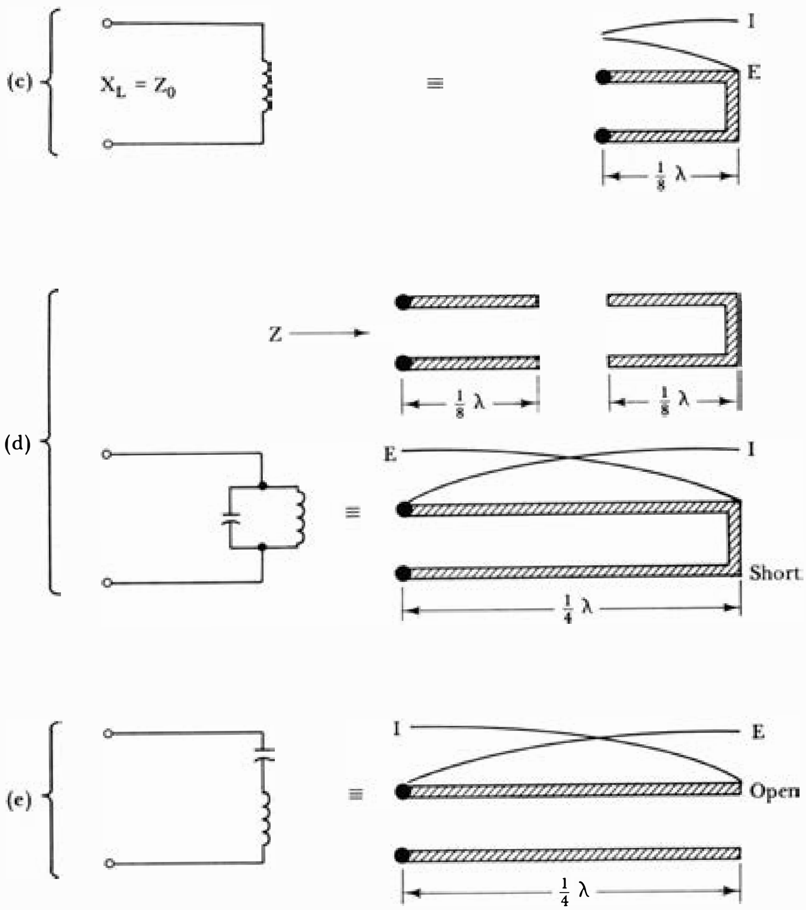


Figure 5-13. (cont.)

Chapter 6

Residential High-Fidelity Wiring Systems

6-1 GENERAL CONSIDERATIONS

Residential high-fidelity wiring systems are installed principally for the operation of speaker enclosures in various rooms. Sometimes intercommunication facilities are included. Two chief types of wiring are utilized, termed *concealed wiring* and *exposed wiring*. Concealed wiring is installed in the same general manner as for electric lighting, although a different type of plate is utilized on the outlet boxes. Concealed wiring is run behind walls, under floors, and over ceilings. On the other hand, exposed wiring is enclosed in metal surface raceways that are secured to the surfaces of walls or ceilings. Wiring installations are classified as new work or old work. New work consists of wiring in buildings that are under construction. Conversely, old work involves alterations or additions to wiring systems in completed buildings. Concealed wiring is ordinarily installed in new work, whereas exposed wiring is often employed in old work. Although the installation procedure is comparatively difficult, concealed wiring may be utilized in old work, if desired.

6-2 FEATURES OF EXPOSED WIRING

In most circumstances, the installer will be concerned with old work, and surface metal raceways, as exemplified in Fig. 6-1, may be used for purposes of economy and convenience. Raceways are available in various

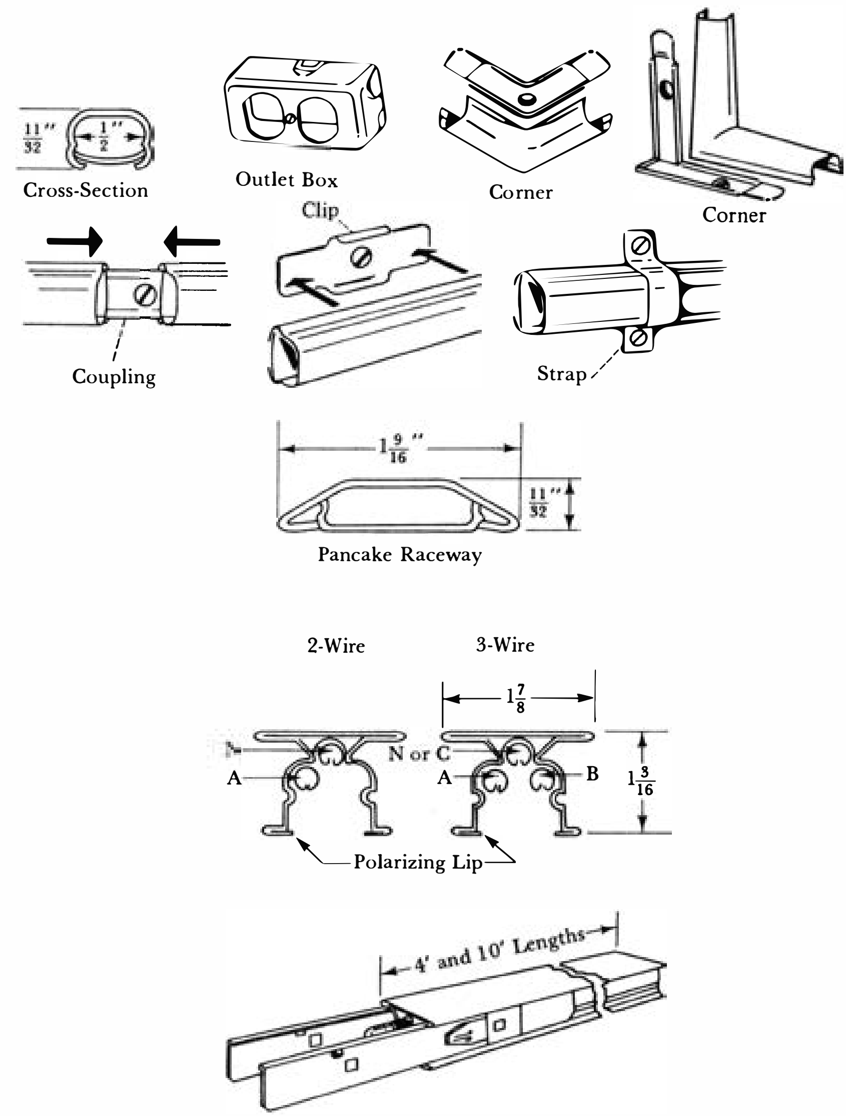


Figure 6-1. Typical surface metal raceways and fittings.

colors, and it is ordinarily desirable to match the color of the wall, baseboard, or ceiling across which the raceways are installed. Inasmuch as a raceway is constructed of metal and is customarily grounded to a cold-water pipe, it provides considerable shielding action for the enclosed conductors. In turn, standard nonmetallic electric cable is suitable; it has a very low resistance and will operate a high-powered speaker efficiently at a considerable distance. Note that standard nonmetallic cable has a third conductor; this is a grounding wire that is connected to the outlet box at the time of installation. A raceway is never bent; suitable fittings are used wherever the direction of run changes. Metal straps are installed to secure a raceway to the surface of a wall or ceiling. Wires are "fished" through the raceway after it is installed.

It is impossible to install concealed wiring in brick walls, for example. Hence, surface metal raceways are the logical choice. In general, it is preferable to install raceways in corners between the molding and wall, and between a door jamb and wall, where it is least noticeable. Raceway stock is available in 4- and 10-ft lengths. It is cut to length with a hacksaw and terminated with appropriate fittings or outlet boxes. When a surface metal raceway must be installed across a floor, the pancake type is utilized, as depicted in Fig. 6-1. It has a low contour, so it imposes minimum obstruction to persons walking over the raceway. Methods of "fishing" cable through raceways are detailed subsequently.

6-3 WIRING INSIDE WALLS AND CEILINGS

Audio wiring inside walls, as exemplified in Fig. 6-2, is classified as concealed wiring. Studs are bored to pass the cable (ordinarily nonmetallic sheathed cable). Alternatively, the studs may be notched so that the cable is installed next to the wallboard. When cable is run up or down a stud or joist, the cable should be strapped every 4½ ft. It should also be strapped within 12 in of every outlet. However, in the case of old work, where the cable must be fished behind the walls, strapping of cables is impractical. After a cable is installed, it is cut with at least 8 in of conductor extending out of the box, and the protruding portion of the cable is stripped for making connections.

A square receptacle box with a side-mounting bracket is pictured in Fig. 6-3(a). This type of bracket has inverted (protruding) points that help to secure the box in position on the stud. In addition, nails are driven into the stud through the holes in the bracket. In some cases, the installer may use wood screws instead of nails. In Fig. 6-3(b), placement of a standard outlet box with a side-mounting bracket against a stud is depicted. As seen in (c), a steel box support is used for mounting a receptacle box between

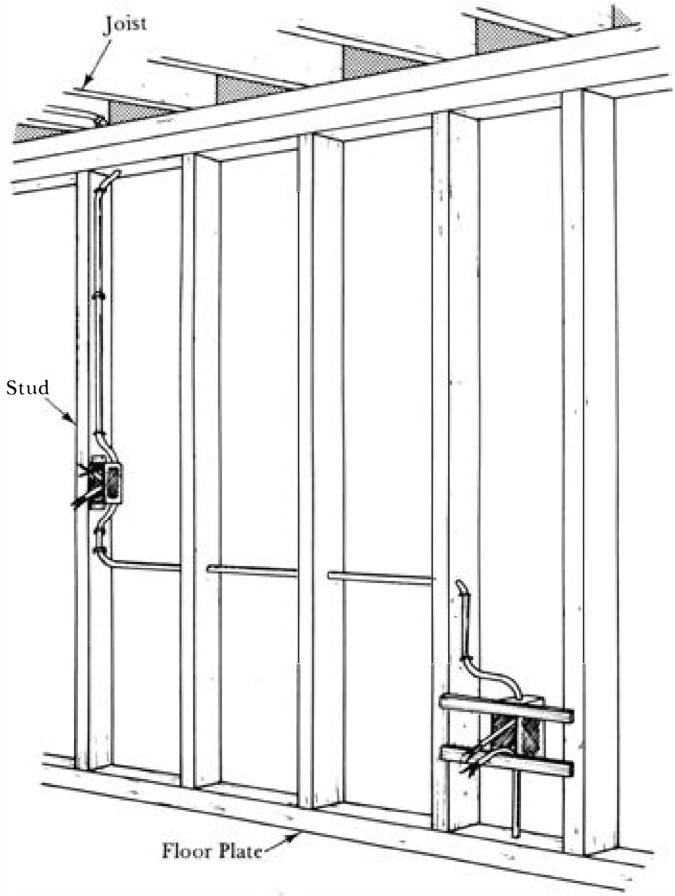


Figure 6-2. Example of concealed wiring

studs. It is also used to mount one or more boxes in any position. A square box serves as a junction for wire splices, if provided with a plain cover. In old-work procedures, a box is installed as shown in Fig. 6-4. Drill four ½-in holes, as shown, to permit entry of a hacksaw blade. Mount the blade so that the saw cut is made by drawing the blade outward. Either wallboard or plaster is cut in the same manner. In the case of a plaster wall, the installer should hold his hand against the plaster, to minimize the possibility of cracking. If plaster is accidentally chipped, the damage can be repaired by spackling.

When a box is installed in building board, a box with expansion anchors is utilized, as shown in Fig. 6-5. After the cable has been secured in

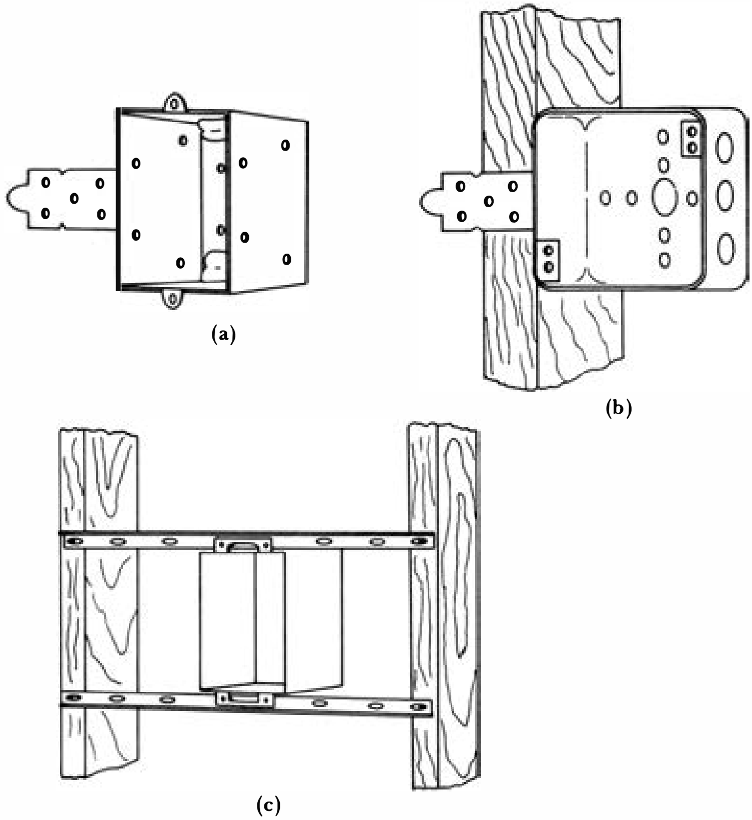
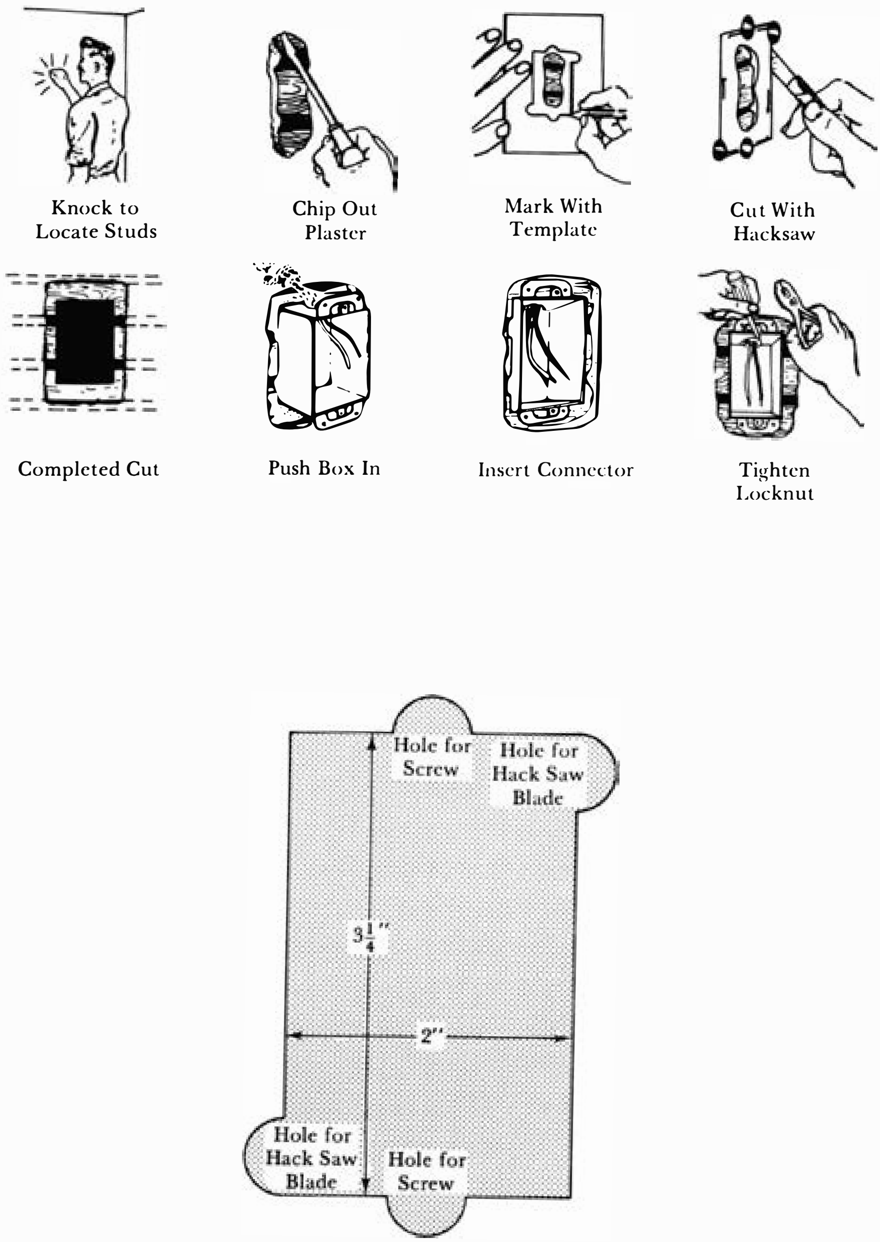


Figure 6-3. Outlet boxes: (a) square receptacle box with side-mounting bracket positioned on stud; (b) placement of a standard outlet box with side-mounting bracket; (c) steel box support for mounting a receptacle box.

the box, as pictured in Fig. 6-6, push the box into the opening so that its front brackets are against the wall. Then tighten the side screws on the box until the side brackets are pressing snugly against the inner surface of the wall. Audio wiring is usually installed in boxes 2½ in deep. However, if there happens to be less than 2½ in of clearance, a shallow box may be used. Any box less than 1½ in deep is called a shallow box. Wiring procedures are facilitated, however, by utilizing large boxes. Note that two single boxes can be combined or ganged, if desired, as depicted in Fig. 6-7, to double the available space.

Old houses sometimes have plastered walls that are papered. In such a case, the paper is cut away with the plaster and lath when the opening is sawed out. If difficulty is encountered in fishing the cable, one or more ad-



Dimensions for Template

Figure 6-4. Installing a box in a wall.

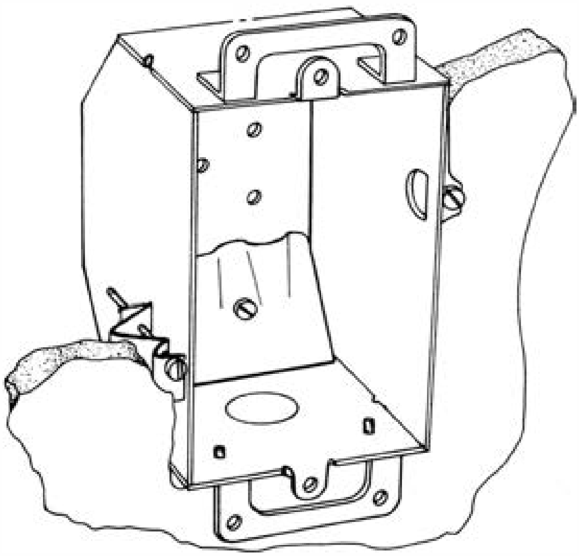


Figure 6-5. Old-work box with expansion anchors.

ditional openings may need to be cut through the wall in order to pull or push the cable into place. In this process, the wallpaper must be properly loosened, cut, and finally replaced. Wallpaper paste can be softened in the work area by applying a damp cloth against the paper. Then a three-sided flap can be cut in the paper with a razor blade. This permits the flap to be folded up, so that a hole can be cut through the plaster and lath. Finally, after the cable has been installed, the hole must be repaired and plastered and the flap pasted back in place.

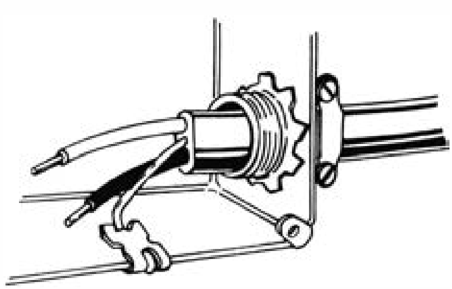


Figure 6-6. Cable installed in an outlet box.

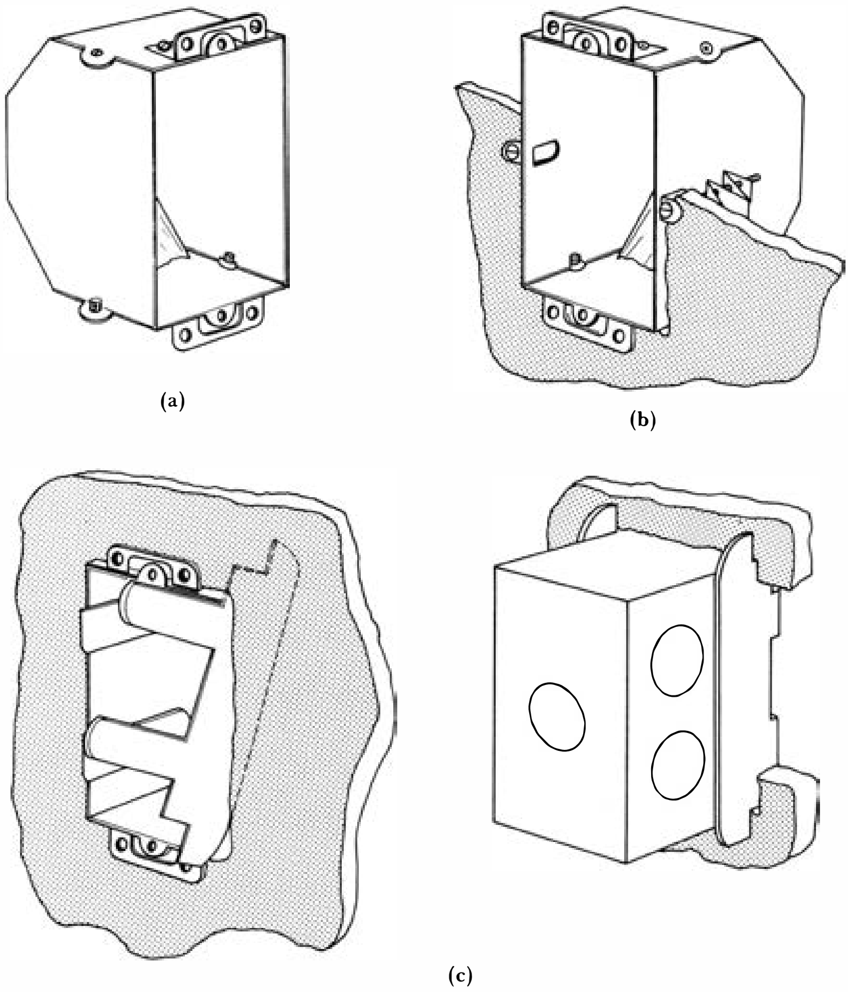
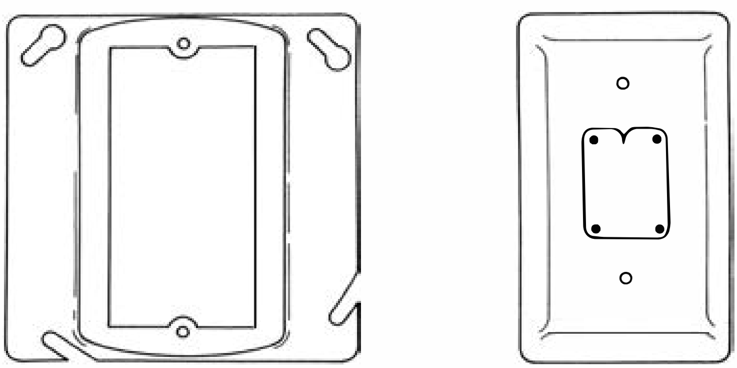


Figure 6-7. Special old-work boxes for sheetrock or plaster walls: (a) beveled corner box; (b) Grip-Tite® box; (c) installation of metal box supports.

Various special types of boxes are available for old work. Previous mention has been made of the expansion-type box. Beveled-corner boxes, depicted in Fig. 6-7(a), are easier to install in old work and do not utilize connectors. Clamps are provided for nonmetallic sheathed cable. This type of box is also available with knockouts to accommodate connectors (exemplified in Fig. 6-6). In Fig. 6-7(b), note the Grip-Tite® clamps at the sides of the box. After the box is seated in place, the side screws are tightened, which expands the clamps and brings them up snugly against the wall behind the box. This type of box is also very useful for installation in sheetrock walls. Conventional boxes can be secured in place to good advantage by inserting additional metal box supports on each side, as depicted in Fig. 6-7(c). This box support is worked in beside the box so that its ends rest against the wall behind the box. Then the support is drawn forward snugly and the two tabs are bent down over the edge and inside the back of the box.

Large boxes 2½ in deep are preferred, because they are easier to work with. However, small and/or shallow boxes must sometimes be installed. Note that a 4-in-square box can be fitted with a suitable cover, as shown in Fig. 6-8, and used as a smaller outlet box. Wall plates with suitable receptacles are finally attached to the boxes, as exemplified in the illustration. A cleat-type box that is somewhat similar to a Grip-Tite® box is shown in Fig. 6-9. A cleat-type box has a pair of folding side cleats that are unfolded and drawn up against the back of the wall by tightening a pair of screws. Boxes as shallow as ½ in are permitted by the National Electrical Code for installation in difficult situations. If a shallow box cannot be installed in a wall, surface wiring must be utilized, as explained previously.



(a) (b)

Figure 6-8. A cover may be used with a 4-in square box to serve as a smaller outlet box: (a) appearance of cover; (b) a wall plate is attached to an outlet box.

6-4 INSTALLATION OF OUTLET CABLES

Installation of outlet cables in old work involves various procedures, depending upon circumstances. As shown in Fig. 6-10, cable can be fished through walls, utilizing a hole bored through from the cellar. Fish tapes, or

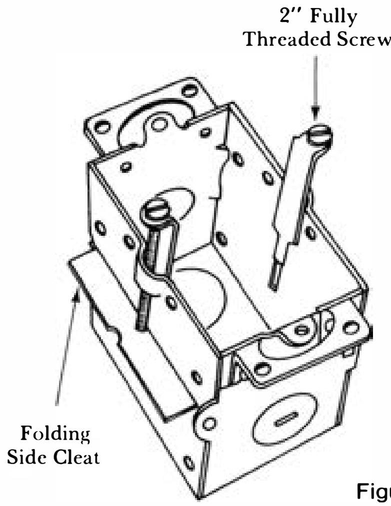


Figure 6-9. Cleat type of old-work box.

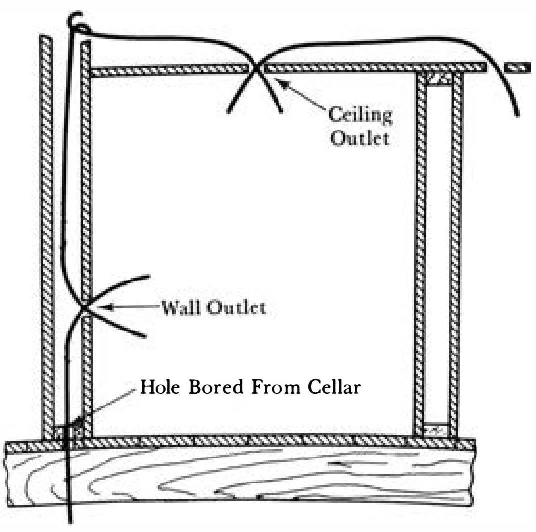


Figure 6-10. Audio cable can be pushed through a hole bored from the cellar and fished through the walls.

ordinary iron wire with a hook bent on the end, may be used to draw the cables through the walls. A mouse, or small plumb bob, is very useful to check for cross studs, as depicted in Fig. 6-11. To bypass a stud, the installer bores two holes through the wall, as shown in Fig. 6-12. These holes are drilled at 45° angles to the cross studs so that the cable can be passed

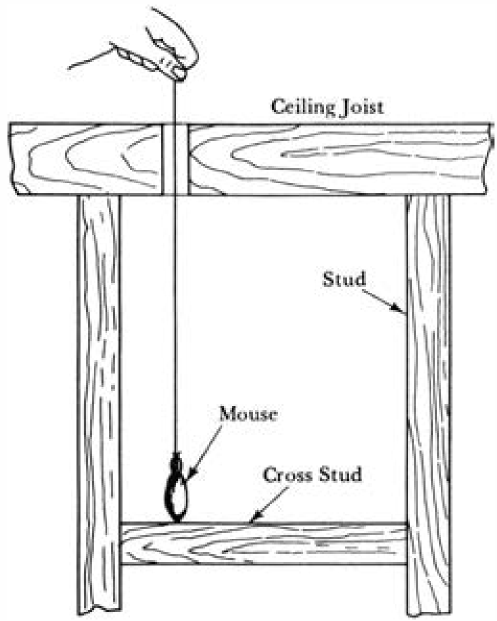


Figure 6-11. Locating cross studs with a mouse.

around the stud. If plaster finish is present, it must sometimes be spackled to repair minor chipping after the cable has been fished through. If the spackling is too light, it may be tinted as required to match the existing plaster color. When a cable or wires must be fished for a considerable distance, the conductors must be firmly secured to the fish tape, as depicted in Fig. 6-13. Otherwise, the hook is likely to be pulled open and the job must then be started all over again.

Details of fishing a wire are pictured in Fig. 6-14. In this example, two diagonal holes are drilled. In (a), a hole is cut in the wall, and another hole is cut in the ceiling. A horizontal hole is drilled to join the two diagonal holes (b). Then the installer pushes a 12-ft fish wire, with hooks on each end, through the hole on the second floor. One end of the wire is then pulled out from behind the wall and through the opening for the outlet on the first floor (c). Next, a 25-ft fish wire, with hooks on each end, is pushed through the ceiling outlet, as shown by the arrows in (d). This wire is worked back and forth until it hooks the first wire. Either one of the fish wires may then be withdrawn to slide the hooks together (e). Finally, the shorter wire is pulled through the opening for the outlet. After the shorter fish wire is removed, the cable is attached to the long fish wire for drawing through the wall and ceiling spaces (f).

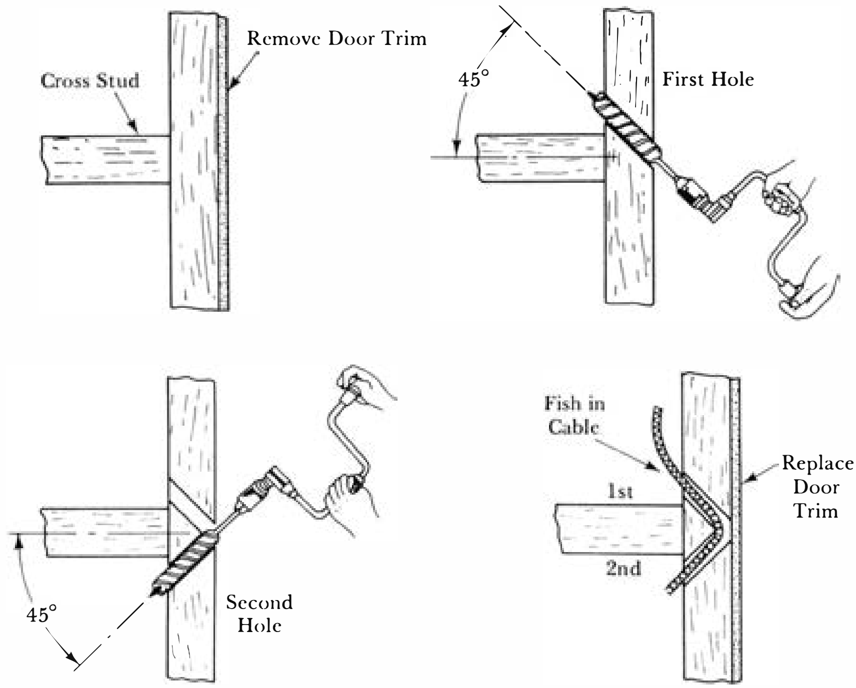


Figure 6-12. Two holes are bored in the wall to bypass a cross stud.

When audio cable is installed across joists or beams, attic floor boards must be lifted, as illustrated in Fig. 6-15, so that the joists can be notched and holes bored through obstructions. Boards can be lifted without undue difficulty if they do not have tongues and grooves. Small holes are drilled at the ends of a section to be lifted, and the board is cut with a thin keyhole saw. In turn, the end of the section can be pried up and lifted out. After the cable is installed, the board is replaced, with cleats tapped in as shown in

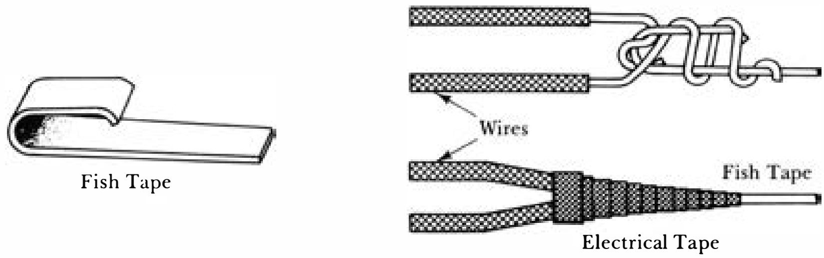


Figure 6-13. How to secure wires firmly into a fish-tape hook.

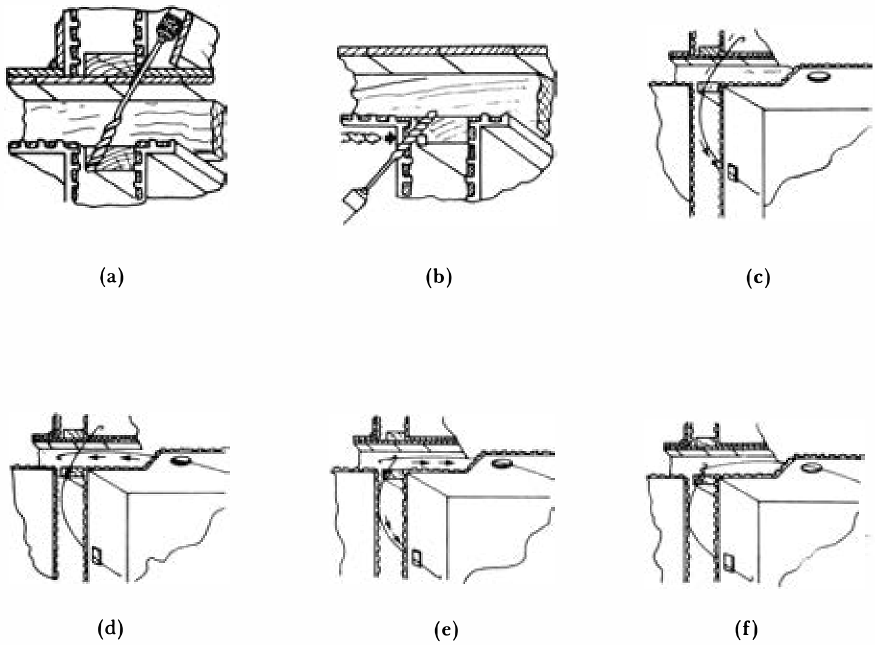


Figure 6-14. Typical cable-fishing procedure: (a) boring the first diagonal hole; (b) boring the second diagonal hole and the horizontal hole; (c) insertion of the shorter fish wire; (d) insertion of the longer fish wire; (e) bringing the hooks together; (f) shorter wire is pulled to bring long wire through outlet opening.

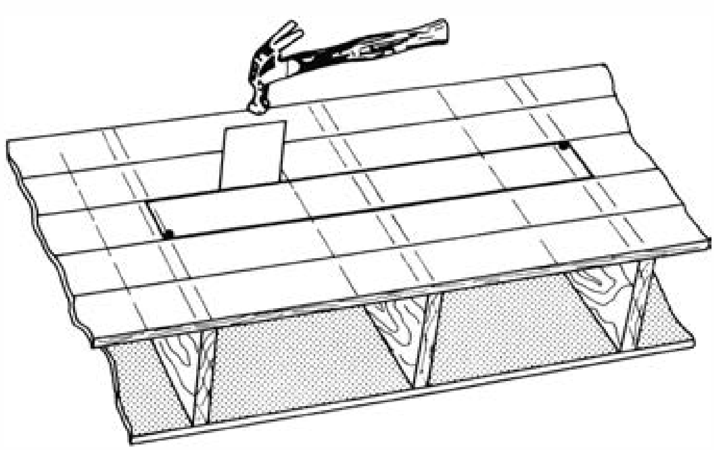


Figure 6-15. Tongues are cut from floor boards with a painter's scraper and hammer.

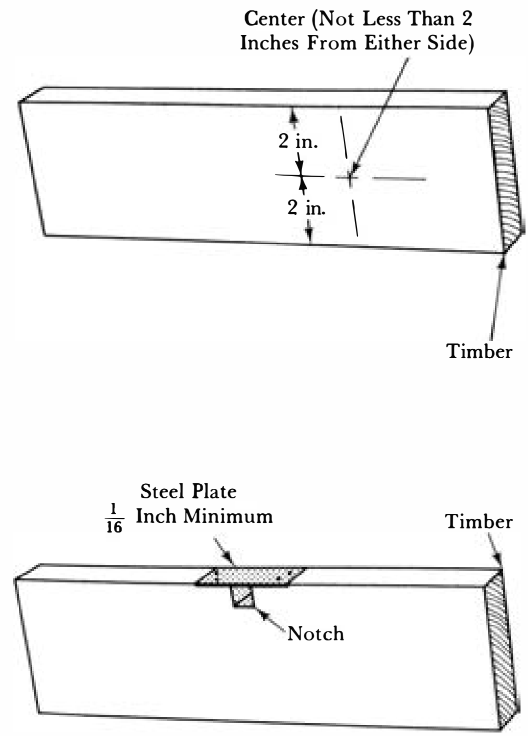


Figure 6-16. Approved method of drilling and notching timbers.

the diagram, so that the board has a firm footing. Finally, the drill holes are filled with wooden plugs. When tongue-and-groove flooring is to be lifted, it is necessary first to cut the tongues free. This is done with a painter's scraper and a hammer. Some installers prefer to start the job with the scraper and then to do most of the cutting with an extra-thin hacksaw blade. This method has the advantage of cutting through any stray nails without difficulty. It is good practice to lift boards over at least three joints, so that the replaced board has a firmer footing. All building codes must be observed when passing cables through studs, joists, or rafters. In addition, the National Electrical Code requires that the holes must be drilled as near the center of the timber as possible, and that the holes must be at least 2 in from an edge. This requirement is depicted in Fig. 6-16. Notches must be covered by steel plates not less than 1/16 in thick, to protect the cable from nails. It is general practice to make conductor connections with solderless connectors and solderless lugs, as shown in Fig. 6-17.

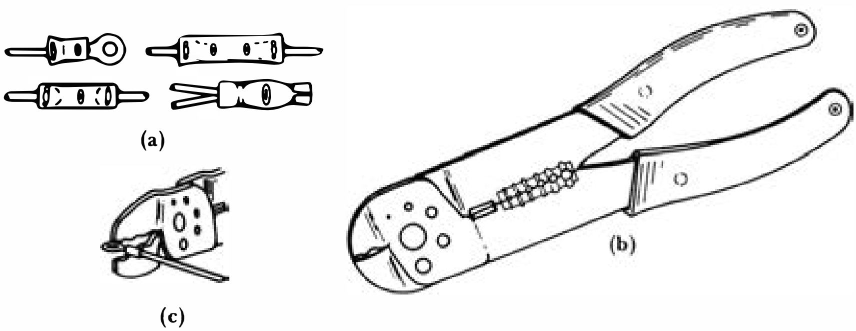


Figure 6-17. Solderless connectors: (a) examples of solderless connectors and a solderless lug; (b) crimping tool; (c) use of crimping tool.

6-5 INTERCOMMUNICATION AND MUSIC UNITS

Intercommunication units installed in residences are often combined with FM/AM radios that may be supplemented by tape decks and/or high-fidelity record players. Installation instructions and color-coded wire are generally supplied by the manufacturer of the intercommunication units. The National Electrical Code stipulates that the connecting wires or cables must not be run closer than 2 inches to any ac line. This applies to ac lines enclosed by conduit, in armored cable, or in nonmetallic sheathed cable, and also to low-voltage ac lines. This requirement not only ensures safety in the event of electrical malfunctions, but also minimizes hum pickup by intercom wires. Similarly, intercom wires must not be run close to and parallel with telephone utility wires, to minimize crosstalk between the systems.

The National Electrical Code also requires that a lightning arrestor be installed on each intercom line whenever there is, owing to a support or insulation failure, any possibility of contact with a light or power line. Note that intercom units are usually of the in-wall type. Other designs are designated as on-wall types. Again, the master unit (with the FM/AM radio) may be built into the wall, whereas the substations are on-wall types. Many designs are supported by wallboard or plaster-lath surfaces, although some designs utilize metal boxes. Elaborate master units are comparatively heavy and must be secured to studs or to appropriate wood framing. In addition to substations in various rooms, a front-door substation is sometimes installed. This permits the resident to talk to a caller from any interior substation or from the master intercom unit. An intercom and music system is energized from a 120-V line. Therefore, the installer may need to provide a power outlet where the amplifier is located (usually at the master station). This is typically a permanent behind-the-wall outlet. However, on-wall designs may be powered from a convenience outlet, particularly in old-work installations. Intercom cable runs between master and substation units typically employ three conductors. One of these is a common line that is energized with a second line for outgoing audio signals, and also is energized with a third line for incoming audio signals. In new work, junction boxes are generally installed in the same manner as for light or power outlets.

Typical intercom and audio cables are labeled type A, B, C, or D. Type A has three vinyl-insulated conductors. One conductor has a tinned copper shield. The other two conductors are unshielded and the outer insulation is a chrome vinyl jacket. Shielding of the common conductor is occasionally helpful in reducing the noise level without resorting to conduit or to metal raceways. Type B audio cable has three polyethylene insulated conductors with aluminum-Mylar shielding over two of the conductors. The third conductor is unshielded. A No. 20 stranded ground wire is provided and the outer insulation is a chrome vinyl jacket. Type C audio cable consists of four polyethylene insulated conductors and is used in the more elaborate intercom/audio installations. One pair of the conductors is tinned-copper-shielded. The other pair is unshielded. The outer insulation of this cable is a chrome vinyl jacket.

Type D audio cable is another design of three-conductor cable in which each conductor is aluminum-Mylar shielded. A ground wire is also provided for each conductor, consisting of a tinned cadmium-bronze ribbon. This is a comparatively costly type of cable. Type E is another design of four-conductor audio cable, employing polypropylene insulation with Beldfoil shielding over one pair of conductors. This cable is also provided with a ground wire. Elaborate intercom/audio installations utilize multi-conductor cables, with 12 conductors cabled in three quads (groups of four). Beldfoil shielding is provided over two conductors of each quad, with a clear Mylar shield over the entire quad. The shields are color-coded and are provided with a ground wire. The outer insulation consists of a chrome vinyl jacket.

Chapter 7

Audio Component Checks and Tests

7-1 GENERAL CONSIDERATIONS

When an additional component is installed in an existing audio system, it is sometimes found that the new component does not perform as expected, owing to a fault or a malfunction in the system. In turn, a technically competent installer may proceed to make various checks and tests in order to localize the fault. One basic technique consists of unit interchange, as shown in Fig. 7-1. For example, suppose that the L channel of a stereo system is "dead" and that the R channel operates normally. Initially, it is unknown whether the fault is in the signal source, in the amplifiers, or in the speaker. Since the trouble symptom indicates that the fault is somewhere in the L channel and that the R channel is functioning normally, an informative checkout can be made by means of unit interchange tests. This procedure is practical, because the same kinds of units are utilized in both the L and R channels.

With reference to Fig. 7-1, a signal-source fault could involve one-half of a tape head or one-half of a stereo decoder. In the case of a phono cartridge, it is often a simple procedure to reverse the L and R lead connections on the cartridge terminals (see Fig. 7-2). Then, if the L channel starts

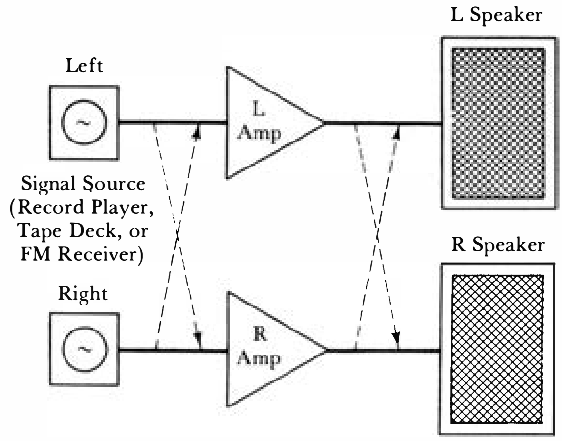


Figure 7-1. Component checkout by interchange tests.

working and the R channel is "dead," it is concluded that one-half of the cartridge is defective. On the other hand, if the trouble symptom remains unchanged, it is evident that the trouble will be found in the L amplifier section or in the L speaker. Note that if the phono cartridge is a plug-in type, its L and R leads may not be easily accessible in the pickup arm. In such a case, the L and R leads can be reversed without undue difficulty at the input terminals of the L and R preamplifiers. It is advisable to turn the system off while leads are being connected or disconnected.

Next, the possibility of a defective speaker may be checked. It is a simple procedure to reverse the L and R speaker leads to the L and R

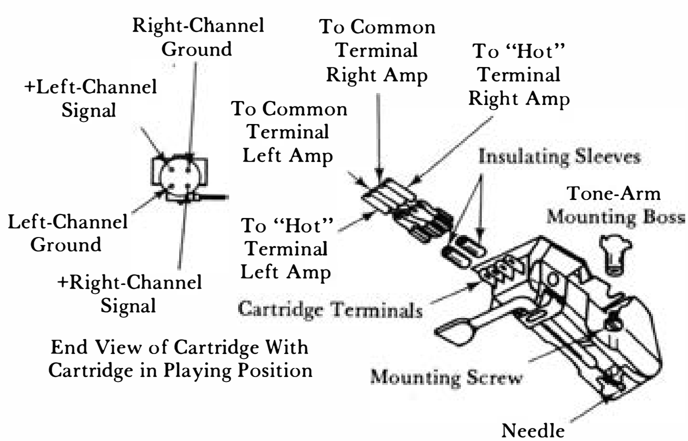


Figure 7-2. Stereo cartridge with removable stylus.

power amplifiers. Then, if the R channel becomes "dead" and the L channel operates normally, it is recognized that the fault is in the L speaker. Conversely, if the trouble symptom remains unchanged, it is indicated that the fault is in the amplifier system. In this example, it is known that there is a fault in the L channel, and the checkout procedure serves to localize the defective section. Coupling between various points in the L and R channels is provided by a 0.5-μF capacitor connected to a pair of test leads. Procedural steps are as follows.

First, an input signal is applied to the L channel only (Fig. 7-3). For example, the R lead may be disconnected from the input terminal of the R amplifier. Then, the signal from the collector of the L first preamp transistor is coupled into the base of the R second preamp transistor. Then, if there is no sound output from the R speaker, it is concluded that the fault will be found in the L first preamp. On the other hand, if sound output is obtained, it is indicated that the L first preamp is operating, and we proceed to step two to determine whether the fault is in the L second preamp. This is a signal-tracing procedure that is effective, and it can be a rapid procedure if the proper test points are readily identifiable. In most cases, test-point identification is made to best advantage by consulting the service literature for the unit(s). It is good practice to turn off the audio system before a test lead is connected or disconnected; this precaution avoids the possibility of damage resulting from accidental short circuits between terminals or to current surges in operating semiconductor devices. It is also advisable to discharge the test capacitor each time that it is disconnected from a circuit.

7-2 RESISTANCE CHECKS

After an amplifier malfunction has been localized to a particular section, resistance checks may be made to assist in pinpointing the defective component or device. Sometimes a defective connection will be found responsible for the difficulty. Resistance measurements are made with an ohmmeter; an ohmmeter is usually one of the functions provided by a volt-ohm-milliammeter (VOM). A scale plate for a VOM is exemplified in Fig. 7-4. Resistance values are indicated on the top scale. An ohmmeter is customarily provided with several ranges, and the resistance scale is direct-reading on the first range (R×1). Observe in Fig. 7-5 that the exemplified VOM has both a function switch and a range switch. To measure resistance values, the function switch is set for dc operation, and the range switch is set to a suitable resistance range, such as R×1. Connection resistance is measured to best advantage on the first range of an ohmmeter.

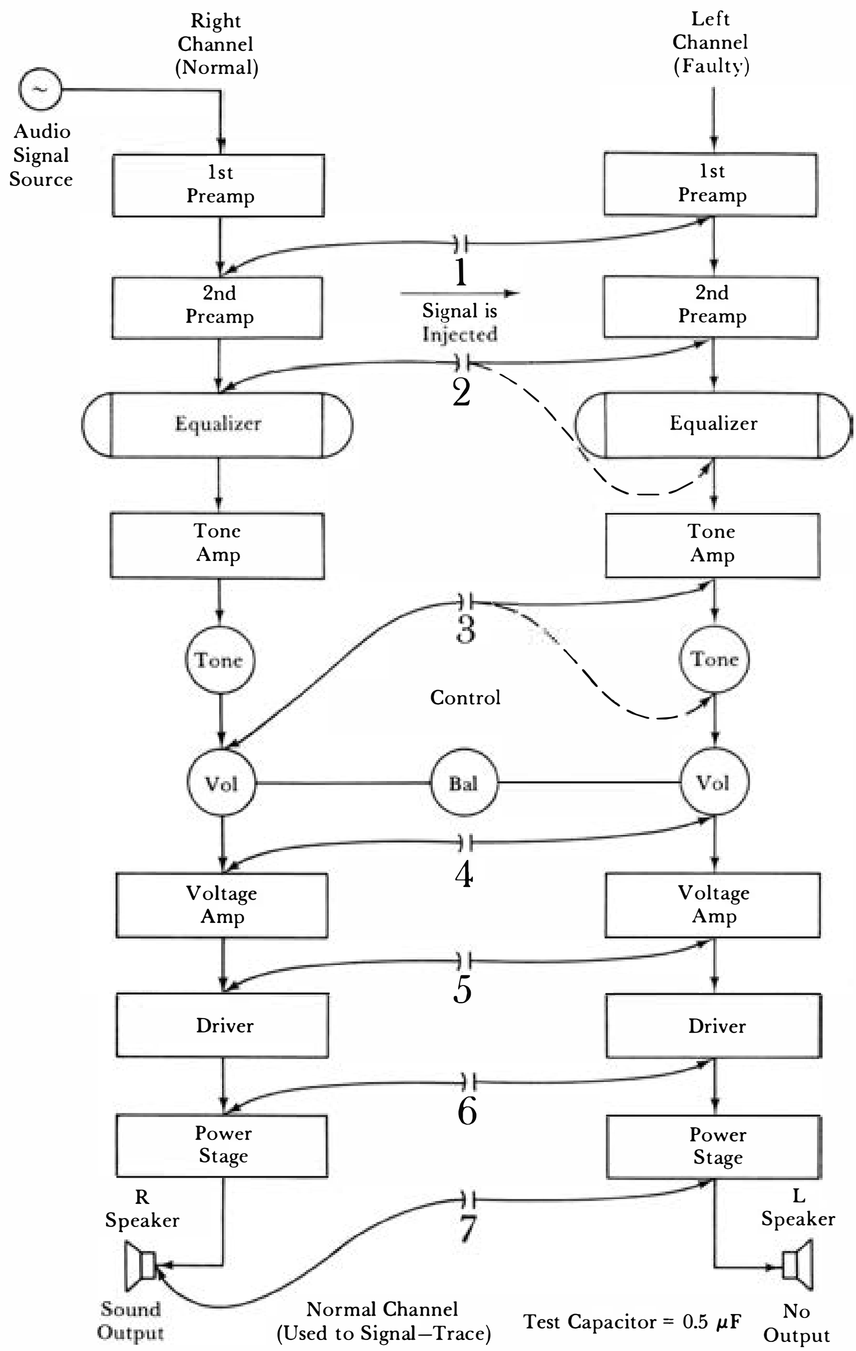


Figure 7-3. Quick checks are made by using one stereo channel to cross-check the other channel.

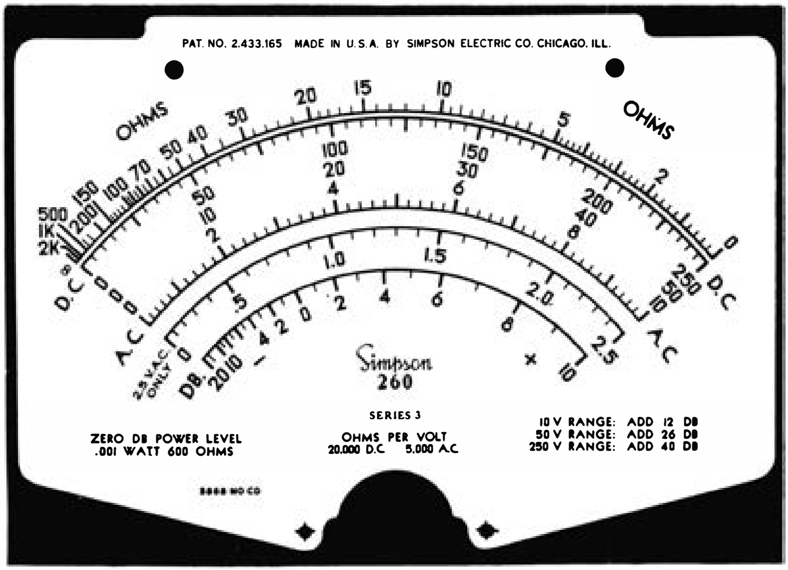
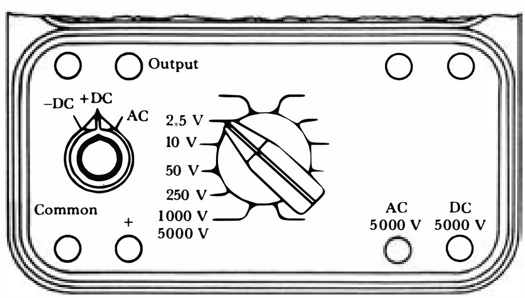
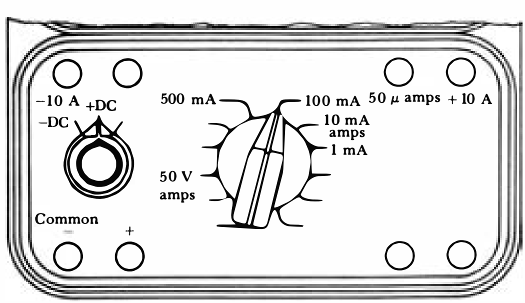


Figure 7-4. Scale plate for a volt-ohm-milliammeter. (*Courtesy Simpson Electric Co.*)

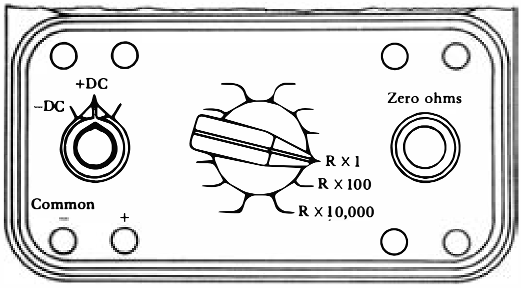


(a)



(b)

Figure 7-5. Dc voltage, current, and resistance ranges associated with the scale plate shown in Fig. 7-4.



(c)

Figure 7-5. (cont.).

A good connection reads zero ohms on a conventional ohmmeter when the instrument's test leads are applied at the input and output ends of the connection. If circuit boards in an amplifier have eyelet construction (Fig. 7-6), check the resistance of the throughboard connections—eyelets sometimes become loose and form poor contacts that measure a substantial value of resistance. If a defective eyelet is found, it is advisable to add a short stub of copper wire through the eyelet, and to add solder to the connection on the top and the bottom of the circuit board, as depicted in Fig. 7-6. An ohmmeter check should then indicate a value of zero ohms for the

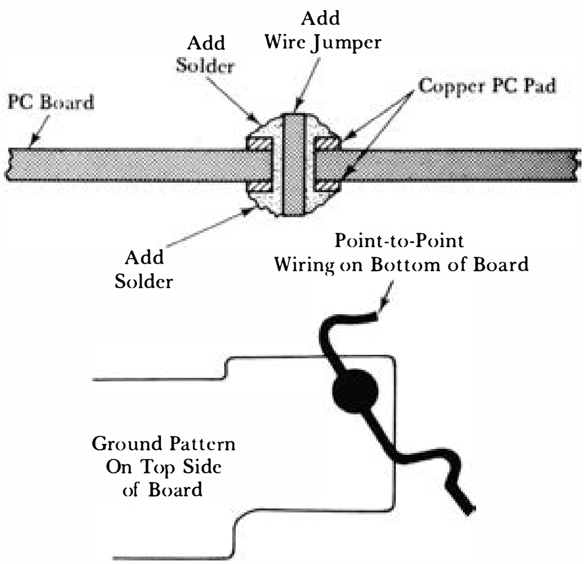


Figure 7-6. Correction of a defective eyelet connection on a circuit board.

throughboard connection. Cold-soldered joints are deceptive, because they appear to be good connections, whereas the solder was not heated sufficiently to form a good bond between the metal surfaces.

Another fault in this category consists of a broken foil conductor on a circuit board. Sometimes a break is very small and can be verified visually only under a magnifying glass. Such cracked printed-circuit (PC) conductors are often intermittent and cause the amplifier to stop operation and then to resume operation at arbitrary intervals. Sometimes an intermittent connection of this type can be localized by gently flexing the circuit board with the eraser end of a pencil. In any event, whenever a conductor is open-circuited, an ohmmeter will indicate an infinite resistance across the break. Note, however, that a misleading indication could be obtained in the event that the ends of the broken conductor are connected to circuitry that provides a resistive return path. This point can be established by inspection of the circuit diagram for the amplifier. A broken PC conductor is repaired with a jumper wire, as pictured in Fig. 7-7.

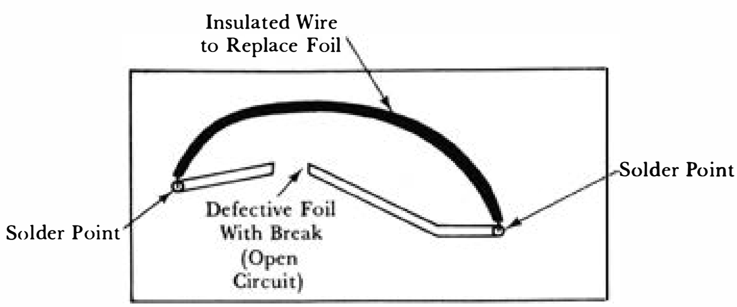


Figure 7-7. Repair of a broken foil conductor on a circuit board.

7-3 NOISE QUICK-CHECKS

Noise can originate anywhere in an audio system. However, the most disturbing noise sources usually occur in the preamplifier section. The input stage, in particular, falls under preliminary suspicion, because any noise generated in this stage is amplified by all the following stages. Noise may be minimized in an input stage by selecting a low-noise transistor. Although collector leakage (defective collector junction) is often responsible for noise generation, other possibilities are resistors that have deteriorated, leaky capacitors, and defective diodes. Coupling transformers occasionally become noisy. An imperfect connection can become a noise source; it can often be pinpointed by tapping various components and devices or by gently flexing a circuit board.

To close in on any type of noise source, the quick-check depicted in Fig. 7-8 is very helpful. A 0.1-μF capacitor is shunted step by step across each component and device in the preamplifier circuit. If a resistor is noisy, for example, the noise output from the speaker will stop, or will be reduced to a low level, when the test capacitor is shunted across the faulty resistor. Note that the test capacitor should be connected or disconnected only with the power to the audio system turned off. Also, the test capacitor should be discharged before it is reconnected into the circuit. This precaution will avoid possible surge damage to semiconductor devices. If an amplifier has modular construction, preliminary localization of a noise source can be made by unplugging a suspected module and replacing it with a corresponding module from the other channel of the audio system.

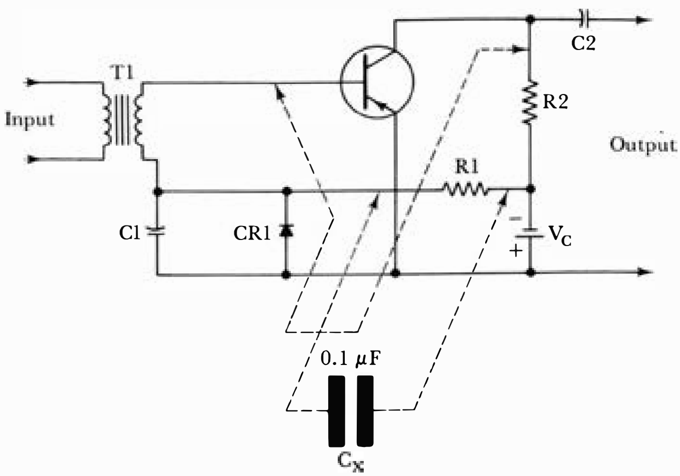


Figure 7-8. Noise quick-checks in an audio amplifier.

7-4 CLICK TESTS

When the signal is being stopped by a fault somewhere in an amplifier channel, successive stages can be quick-checked by means of "click" tests, as exemplified in Fig. 7-9. A jumper lead is applied between the base and emitter terminals of each transistor in turn. For example, it might be observed that when a click test is made at Q3, a sound output is obtained from the speaker (or from headphones). Then, when a click test is made at Q2, it might be observed that no sound output is obtained. This result indicates either that Q2 is defective or that if Q2 is normal, some circuit defect has occurred that removes the forward bias from Q2. In other words, a click test is based upon bringing a normal bias voltage temporarily to zero. This voltage change generates a pulse that produces a clicking sound from the speaker.

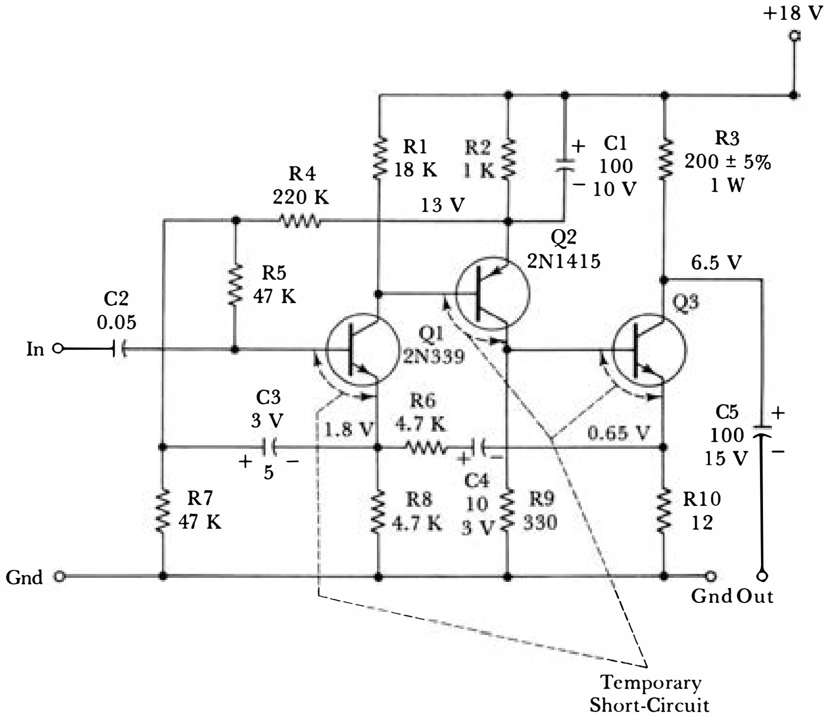


Figure 7-9. Click tests in an audio amplifier network.

7-5 CIRCUIT TESTS WITH A VOM

Dc voltage and resistance measurements are basic in audio servicing procedures. Current measurements are also made on occasion, although it is ordinarily easier to measure voltage or resistance values. As exemplified in Fig. 7-9, normal operating voltage and resistance values are specified in audio service data. In addition, the service data may provide resistance charts; these charts specify normal resistance values from each semiconductor terminal to chassis ground (or to the common power-supply bus). Such charts are particularly helpful in preliminary analysis of system malfunctions. Circuit malfunctions are often accompanied by changes in dc voltage and/or resistance values. As an illustration, defective transistors can often be localized by means of in-circuit dc-voltage measurements. This approach applies to transistors in high-frequency circuits, as in FM tuners, as well as to transistors in audio-frequency circuits.

Typical NPN and PNP bipolar transistor circuits, with specified operating voltages, are exemplified in Fig. 7-10. Germanium transistors are usually operated with a base-emitter bias of 0.2 V. On the other hand, silicon transistors are usually operated with a base-emitter bias of 0.6 V. Bias polarity is such that class A operation is provided; in other words, transistors are usually forward-biased. If a transistor develops collector-junction leakage, excessive collector-emitter current flows. The stage gain decreases, and the transistor may generate excessive noise. With reference to Fig. 7-10, excessive collector-emitter current flow produces an abnormal voltage drop across the emitter resistor R1. In turn, the emitter voltage becomes higher than normal. An abnormal voltage drop also occurs across the collector series resistor R2, with the result that the collector-emitter voltage of the transistor becomes lower than normal. In an extreme condition, with the collector junction short-circuited, practically all of the supply voltage drops across R1 and R2, and the collector-emitter voltage of the transistor is reduced to virtually zero.

Current values can be measured indirectly by measuring the voltage drop across a series resistor and applying Ohm's law to calculate the current that is flowing through the resistor. If it is desired to measure a current value directly, the method shown in Fig. 7-11 may be utilized. A razor slit is made through a printed-circuit conductor to temporarily open the circuit through which the current is flowing. In turn, the test leads from a VOM operated on its milliampere function are applied at the ends of the opened

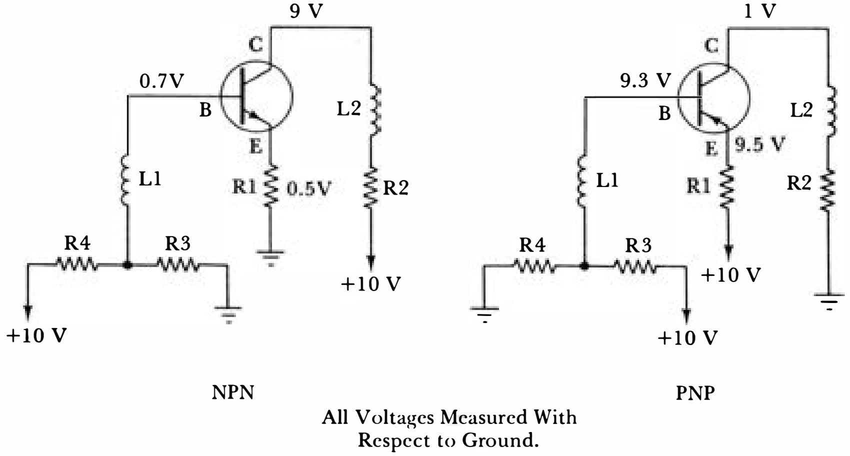


Figure 7-10. Typical operating voltages in germanium bipolar-transistor circuitry.

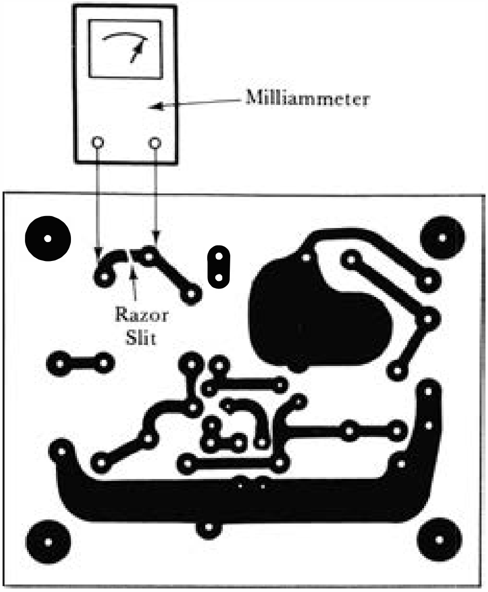


Figure 7-11. Printed-circuit conductor may be cut to make an in-circuit current measurement.

conductor. After the current measurement has been made, the PC conductor is repaired by closing the slit with a small drop of solder. Note that this method can also be made to measure resistance values in-circuit, when one resistor (or device) is shunted by another. There are situations in which current measurements are "forbidden." For example, in Fig. 7-12 the microammeter has an internal resistance of 5000 Ω. If this instrument resistance is inserted in series with the base circuit of the transistor, the normal current flow is substantially reduced and an erroneous measurement results.

Sometimes the collector junction of a transistor burns out and becomes open-circuited. In such a case, the collector-emitter voltage of the transistor becomes abnormally high. As a practical note, off-value transistor terminal voltages do not necessarily prove that the transistor is defective. That is, a change in transistor terminal voltages can result from defects in associated circuit components. Therefore, it is often necessary to analyze and to interpret the dc voltage distribution that is found in a malfunctioning circuit (refer to Fig. 7-13). There is more than one possible cause for loss of base-emitter voltage (base and emitter terminals are found to be at zero volts with respect to ground). Inasmuch as the base-bias voltage is applied via R4, it is evident that a short circuit in C3 is a probable cause of bias-voltage loss. On the other hand, in the event that C3 checks

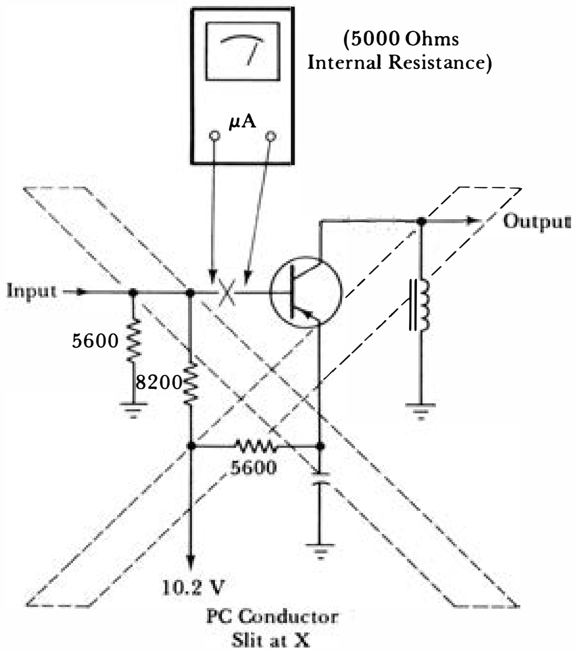


Figure 7-12. "Forbidden" current measurement.

out normally, attention would be turned to R4 and L1. In other words, an open-circuit condition in either R4 or L1 will result in bias-voltage loss.

From the viewpoint of probability, capacitor defects are more likely to occur than are resistor or inductor defects. Circuit analysis will indicate in this situation that if the transistor had a base-to-emitter short circuit, the base-emitter voltage would be zero but the emitter voltage would measure above ground potential. That is, with a base-emitter short circuit, there will still be current flow through R1 in Fig. 7-13, and the resulting voltage drop across R1 will raise the emitter potential above ground. Observe that since the emitter is connected to ground through R1, when the base voltage is brought to zero for any reason, the transistor will normally be cut off and in turn no collector current will flow. Accordingly, there will be zero current flow through R1 and the emitter potential will rise to practically the value of the supply voltage.

The most probable transistor defect is leakage from collector to base through the collector junction. Collector-junction leakage changes the dc voltage distribution in typical PNP and NPN bipolar transistor circuitry, as exemplified in Fig. 7-14. Observe that collector leakage has the effect of increasing the leakage current through the transistor from collector to

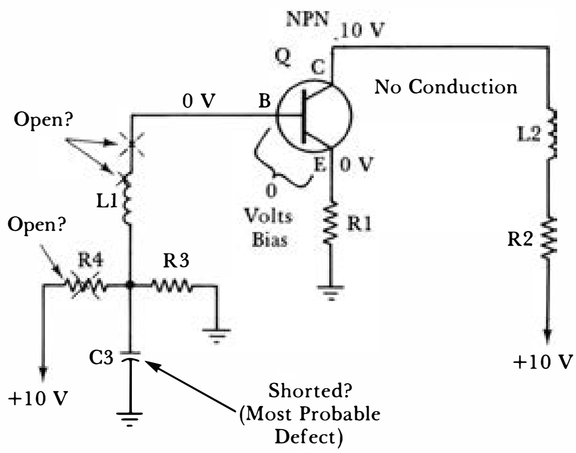
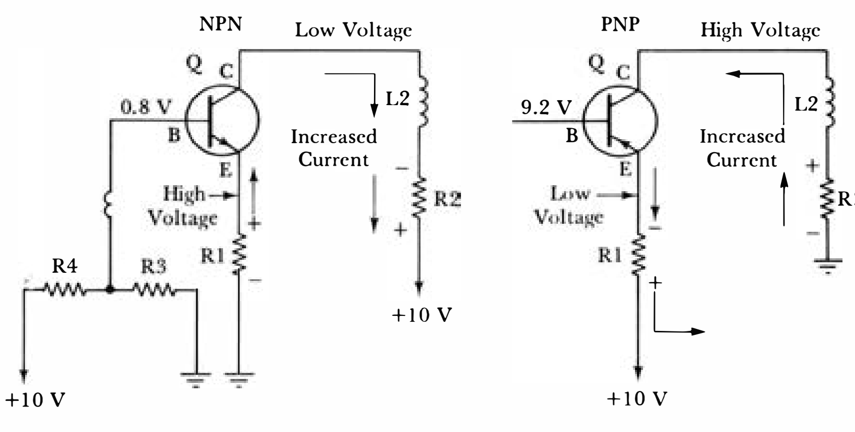


Figure 7-13. Component failure can cause loss of bias voltage.

emitter. In turn, collector leakage causes an increase in the voltage drops across R1 and R2. This increased voltage drop across R1 causes the emitter voltage to become abnormal in the configuration of Fig. 7-14(a). In addition, the increased voltage drop across R2 causes the collector voltage to become subnormal. On the other hand, when the configuration depicted in Fig. 7-14(b) is being checked, observe that the increased voltage drop across R1 causes the emitter voltage to become subnormal; the increased voltage



(a) Grounded Emitter. (b) Grounded Collector.

Figure 7-14. Collector leakage changes the dc voltage distribution.

drop across R2 causes the collector voltage to become subnormal.

In the foregoing configurations, if collector leakage is quite serious, the transistor can become zero-biased or reverse-biased, although the collector current is abnormal. Note that some of the collector leakage current is diverted into the base circuit. This diversion current has a tendency to produce a reversed base-emitter bias voltage. Consequently, whenever it is observed that the bias on a normally class A transistor is reversed, collector leakage should be suspected at the outset. In the event that the base-emitter junction is short-circuited, it is apparent that the same dc voltage value will be measured at the base and collector terminals of the transistor. Or, if the base-collector junction becomes open-circuited, the collector voltage will either measure zero, or it will measure practically the same value as the supply voltage, depending upon the circuit arrangement.

7-6 TRANSISTOR QUICK-CHECKS

Two useful quick-checks can often be made of bipolar transistor control action, as exemplified in Fig. 7-15. A *turn-off test* is made by bringing the transistor's base-emitter bias voltage to zero, while the collector-emitter voltage is observed with a dc voltmeter. In circuits of the class shown in Fig. 7-15, a turn-off test is made by temporarily short-circuiting the base and emitter terminals together, after a dc voltmeter has been connected between the collector and ground terminals of the transistor. When the base and emitter terminals are connected together in the configuration of Fig. 7-15(a), the collector voltage jumps up to the supply-voltage value if the transistor has normal control action. Note, however, that in the configuration of Fig. 7-15(b), the collector voltage will drop to zero in a turn-off test if the transistor has normal control action.

Another type of control-action test for a bipolar transistor is called a *turn-on test.* Typical turn-on tests are exemplified in Fig. 7-16. This test is made by increasing the forward bias voltage on a transistor while the collector or emitter voltage is monitored with a dc voltmeter. Whether the collector voltage or the emitter voltage is employed as an indicator depends upon the circuit arrangement. A 50-kΩ bleeder resistor is temporarily applied from the base to a supply-voltage terminal in order to increase the forward bias potential. If the transistor has normal control action, the collector current will increase when the forward bias voltage is increased. This increased current flow results in a decrease of collector voltage in the configuration of Fig. 7-16(a). However, owing to the difference in the circuit arrangement in the example of Fig. 7-16(b), increased current flow results in an increase in the collector voltage value if the transistor has normal control action. In-circuit quick-checks of transistor control action are very useful, because transistor leads are ordinarily soldered into circuit boards.

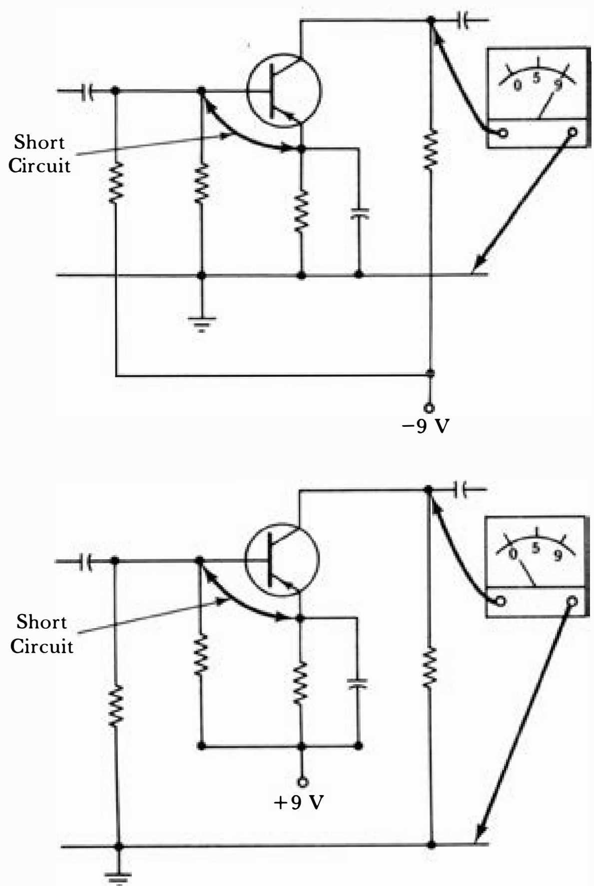
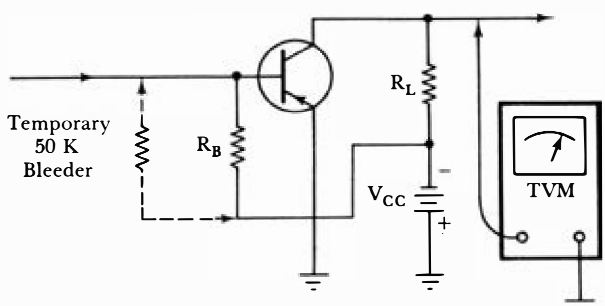
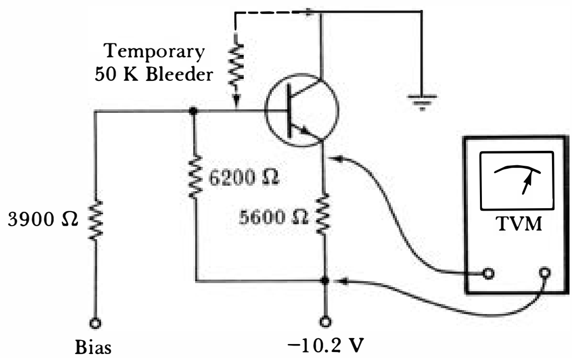


Figure 7-15. Basic transistor "turn-off" test procedure: (a) grounded emitter; (b) grounded collector.



(a) Grounded Emitter.

Figure 7-16. Examples of "turn-on" test procedure.



(b) Grounded Collector.

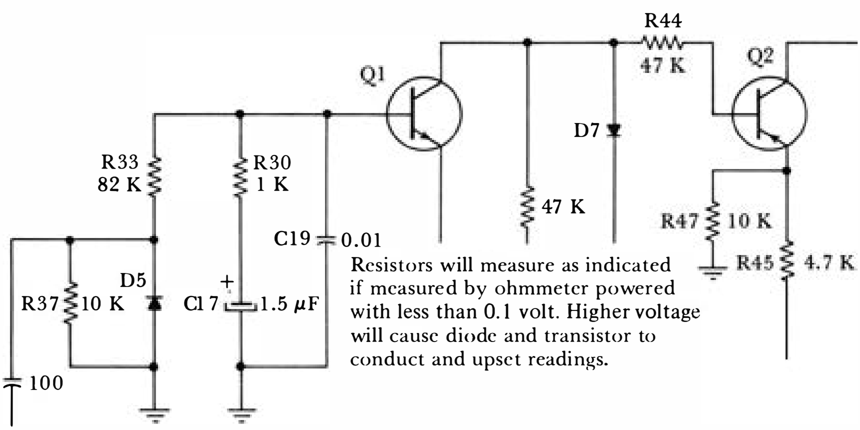
Figure 7-16. (cont.).

7-7 LOW-POWER OHMMETER FUNCTION

Various solid-state VOMs such as illustrated in Fig. 7-17 are provided with a low-power ohmmeter function in addition to a conventional ohmmeter function. A low-power resistance-measuring function has an advantage over a conventional ohmmeter in that it applies a test potential of less than 0.1 V between the points under test in a circuit. Accordingly, this test voltage is less than the threshold value that is required to turn on a normal transistor or diode junction. Therefore, many resistors can be checked for correct value in solid-state circuitry with disconnection. With reference to Fig. 7-18, a low-power ohmmeter will accurately check the resistance values of R37, R33, R30, and R44 in-circuit. Of course, if a capacitor, diode, or transistor happened to be defective, an inaccurate resistance measurement will result. Therefore, it becomes necessary upon occasion to analyze and to interpret a pattern of incorrect resistance values that have been measured in-circuit.

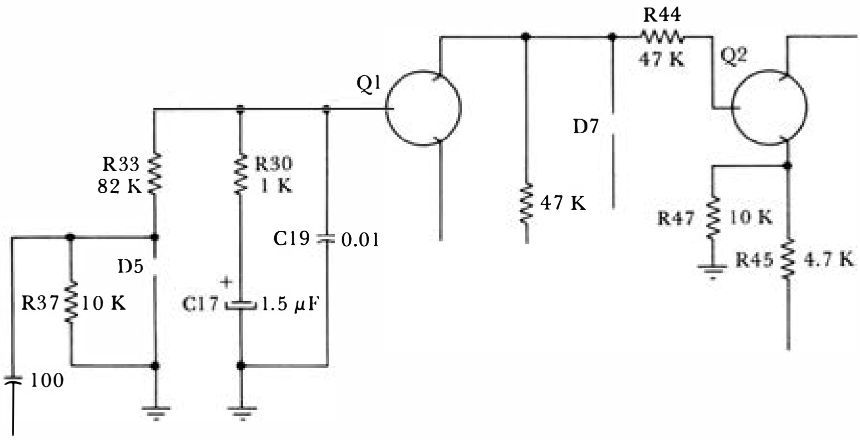
A conventional or high-power resistance-measuring function is also required in an ohmmeter, in order to check the front-to-back ratios of semiconductor device junctions. Note that the resistance values that are measured in a front-to-back ratio test of a device junction can vary considerably, depending upon the particular ohmmeter that is utilized, and upon the resistance range to which the instrument has been set. That is, the junction is characterized by a nonlinear resistance relation, and the resistance value that is indicated by an ohmmeter will depend upon the value of test voltage that is applied to the junction. Nevertheless, a simple front-to-back ratio test with an ohmmeter is adequate to "weed out" junc-

Figure 7-17. Solid-state VOM with high- and low-power ohmmeter functions. (*Courtesy Triplett Electrical Instrument Co.*)



(a)

Figure 7-18. Resistance measurement in solid-state circuitry: (a) device junctions will conduct when forward-biased by test voltage from conventional ohmmeter.



(b)

Figure 7-18. (cont.) (b) how the circuit "looks" to a low-power ohmmeter.

tion devices that are definitely defective. Forward-resistance readings are normally quite low, and reverse-resistance readings are normally quite high for semiconductor junctions.

Transistor Terminal Identification

Sometimes the terminal identification on a bipolar transistor has been obliterated. In this event, the ohmmeter checks shown in Fig. 7-19 will enable you to identify the terminals of the device. The first step is to make resistance measurements between each pair of the transistor leads in both the forward-current and reverse-current (forward-resistance and back-resistance) directions. A resistance reading below 500 Ω indicates that the ohmmeter test voltage is forward-biasing a junction. Note that the highest forward-resistance reading is obtained when the ohmmeter is applied between the emitter and collector leads of the transistor. In turn, the third lead, which is not connected to the ohmmeter test leads, is identified as the base lead of the transistor. At this point, it is not known whether the transistor is a PNP or an NPN type, and it is not known which of the two leads is the collector and which the emitter.

In the second step, a resistance measurement is made between the base lead and one of the other leads of the transistor. If a forward-resistance value is indicated when the negative lead of the ohmmeter is connected to the base lead, it is indicated that the transistor is a PNP type. On the other hand, if a forward-resistance value is indicated when the positive lead of the ohmmeter is connected to the base lead, it is indicated that the transis-

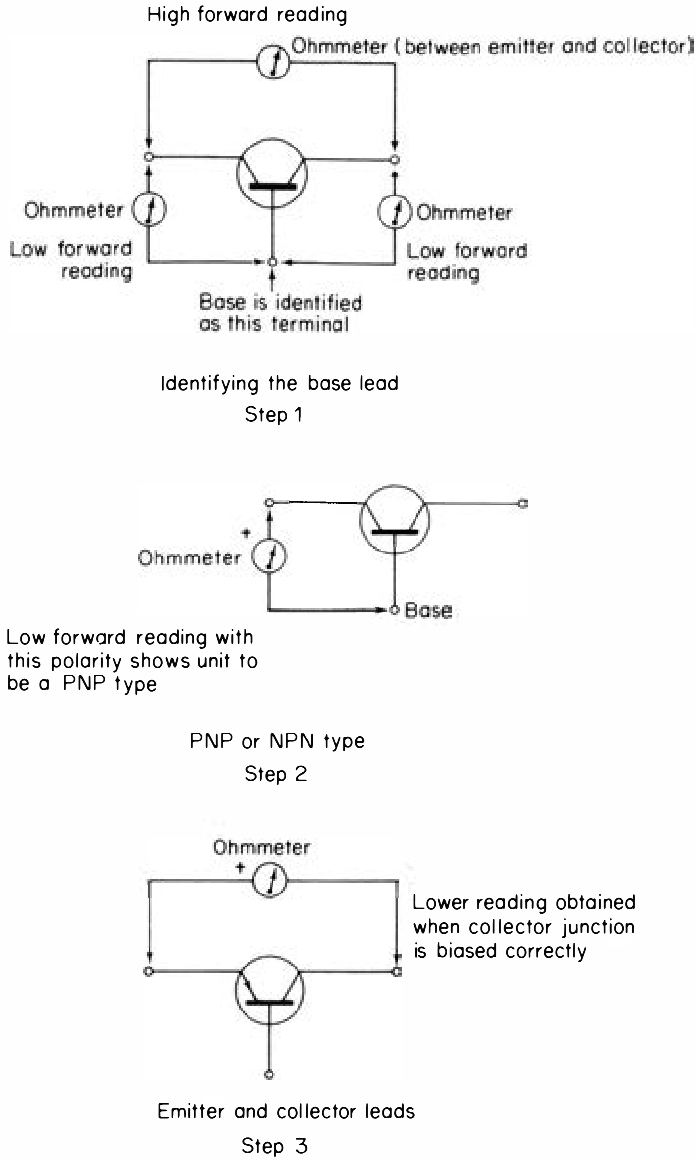


Figure 7-19. Identification of bipolar-transistor type and terminals.

tor is an NPN type. Next, in the third step, we determine which of the unknown leads of the transistor is the collector and which the emitter. Two resistance measurements are made between these leads, the ohmmeter polarity being reversed for the second measurement. Note the connections that are utilized in the lower resistance indication. In turn, for a PNP type of transistor, the negative lead of the ohmmeter will be connected to the collector lead. Conversely, for an NPN transistor, the positive lead of the ohmmeter will be connected to the collector lead.

To summarize the reasons for the foregoing test result, observe that more current flows and a lower resistance reading results when the ohmmeter applies test voltage in normal polarity for the device (as in normal operation)—in other words, when the test voltage is applied to the emitter and collector terminals in the same polarity as for normal operation. The reasons for this response are:

1. The emitter doping is almost always heavier than the collector doping. In turn, there is a slightly higher alpha value (current amplification) for the transistor when the emitter junction is forward-biased.
2. The collector junction has a larger area than the emitter junction. In turn, the collector junction has more leakage current than the emitter junction.
3. The total leakage current in the test situation is equal to the collector-junction leakage multiplied by the beta value (current amplification) of the transistor. This product has a greater value when the ohmmeter applies test voltage in the polarity that is employed in normal operation.

7-8 SIGNAL TRACING WITH THE OSCILLOSCOPE

An oscilloscope such as pictured in Fig. 7-20 is the most useful instrument to trace an audio signal step by step through a hi-fi system. It permits the operator to see the waveform of the signal at the point under test; the vertical-input channel of the oscilloscope can be calibrated to measure the voltage of the audio waveform. In addition, the displayed pattern shows whether the waveform is distorted; if distortion has occurred, the pattern also depicts the type of distortion that is present. In turn, useful clues can be obtained concerning malfunctioning circuits. These clues point to faulty circuit branches in most cases; in turn, defective components or devices are usually pinpointed by dc voltage and/or resistance measurements. Sometimes an oscilloscope display will indicate a defective component, such as an open coupling capacitor or an open bypass capacitor.

7-9 DECIBEL MEASUREMENTS

Decibel measurements are made with a volt-ohm-milliammeter; either the direct-reading decibel scale can be used, or ac-voltage ratios can be related to decibel values, as tabulated in Chart 7-1. When the decibel scale of a

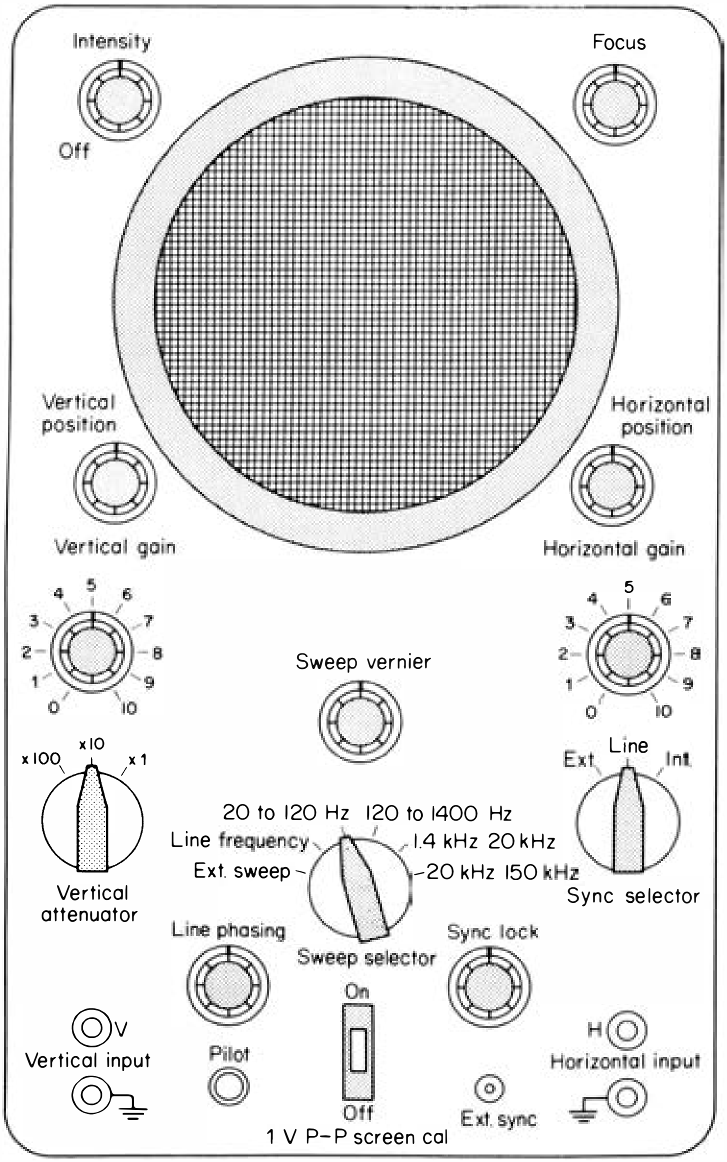


Figure 7-20. Front-panel arrangement of an audio-frequency oscilloscope.

VOM is utilized, it is important to note that the scale is direct-reading only on the first ac-voltage range of the instrument. As seen in Fig. 7-4, a suitable scale factor must be added to the scale indication when the instrument is operated on other than its first ac-voltage range. Required scale factors are printed on the scale plate of the VOM. The gain of an audio amplifier, for example, may be expressed either as a voltage ratio, or as a dB value. Thus, if an amplifier has an input–output voltage ratio of 150, its dB gain value is 43.5 dB.

Chart 7-1

DECIBEL, VOLTAGE, CURRENT, AND POWER RELATIONS

dB	Current or Voltage Ratio		Power Ratio	
	Gain	Loss	Gain	Loss
0	1.000	1.0000	1.000	1.0000
.1	1.012	.9886	1.023	.9772
.2	1.023	.9772	1.047	.9550
.3	1.035	.9661	1.072	.9333
.4	1.047	.9550	1.096	.9120
.5	1.059	.9441	1.122	.8913
.6	1.072	.9333	1.148	.8710
.7	1.084	.9226	1.175	.8511
.8	1.096	.9120	1.202	.8318
.9	1.109	.9016	1.230	.8128
1.0	1.122	.8913	1.259	.7943
1.1	1.135	.8810	1.288	.7762
1.2	1.148	.8710	1.318	.7586
1.3	1.161	.8610	1.349	.7413
1.4	1.175	.8511	1.380	.7244
1.5	1.189	.8414	1.413	.7079
1.6	1.202	.8318	1.445	.6918
1.7	1.216	.8222	1.479	.6761
1.8	1.230	.8128	1.514	.6607
1.9	1.245	.8035	1.549	.6457
2.0	1.259	.7943	1.585	.6310
2.1	1.274	.7852	1.622	.6166
2.2	1.288	.7762	1.660	.6026
2.3	1.303	.7674	1.698	.5888
2.4	1.318	.7586	1.738	.5754
2.5	1.334	.7499	1.778	.5623
2.6	1.349	.7413	1.820	.5495
2.7	1.365	.7328	1.862	.5370
2.8	1.380	.7244	1.905	.5248
2.9	1.396	.7161	1.950	.5129
3.0	1.413	.7079	1.995	.5012
3.1	1.429	.6998	2.042	.4898
3.2	1.445	.6918	2.089	.4786
3.3	1.462	.6839	2.138	.4677
3.4	1.479	.6761	2.188	.4571
3.5	1.496	.6683	2.239	.4467
3.6	1.514	.6607	2.291	.4365
3.7	1.531	.6531	2.344	.4266
3.8	1.549	.6457	2.399	.4169
3.9	1.567	.6383	2.455	.4074
4.0	1.585	.6310	2.512	.3981
4.1	1.603	.6237	2.570	.3890
4.2	1.622	.6166	2.630	.3802
4.3	1.641	.6095	2.692	.3715
4.4	1.660	.6026	2.754	.3631
4.5	1.679	.5957	2.818	.3548
4.6	1.698	.5888	2.884	.3467
4.7	1.718	.5821	2.951	.3388
4.8	1.738	.5754	3.020	.3311
4.9	1.758	.5689	3.090	.3236
5.0	1.778	.5623	3.162	.3162
5.1	1.799	.5559	3.236	.3090
5.2	1.820	.5495	3.311	.3020
5.3	1.841	.5433	3.388	.2951
5.4	1.862	.5370	3.467	.2884
5.5	1.884	.5309	3.548	.2818
5.6	1.905	.5248	3.631	.2754
5.7	1.928	.5188	3.715	.2692
5.8	1.950	.5129	3.802	.2630
5.9	1.972	.5070	3.890	.2570
6.0	1.995	.5012	3.981	.2512
6.1	2.018	.4955	4.074	.2455
6.2	2.042	.4898	4.169	.2399
6.3	2.065	.4842	4.266	.2344
6.4	2.089	.4786	4.365	.2291
6.5	2.113	.4732	4.467	.2239
6.6	2.138	.4677	4.571	.2188
6.7	2.163	.4624	4.677	.2138
6.8	2.188	.4571	4.786	.2089
6.9	2.213	.4519	4.898	.2042
7.0	2.239	.4467	5.012	.1995
7.1	2.265	.4416	5.129	.1950
7.2	2.291	.4365	5.248	.1905
7.3	2.317	.4315	5.370	.1862
7.4	2.344	.4266	5.495	.1820
7.5	2.371	.4217	5.623	.1778
7.6	2.399	.4169	5.754	.1738
7.7	2.427	.4121	5.888	.1698
7.8	2.455	.4074	6.026	.1660
7.9	2.483	.4027	6.166	.1622
8.0	2.512	.3981	6.310	.1585
8.1	2.541	.3936	6.457	.1549
8.2	2.570	.3890	6.607	.1514
8.3	2.600	.3846	6.761	.1479
8.4	2.630	.3802	6.918	.1445
8.5	2.661	.3758	7.079	.1413
8.6	2.692	.3715	7.244	.1380
8.7	2.723	.3673	7.413	.1349

Chart 7-1 (cont.)

dB	Current or Voltage Ratio		Power Ratio		dB	Current or Voltage Ratio		Power Ratio	
	Gain	Loss	Gain	Loss		Gain	Loss	Gain	Loss
8.8	2.754	.3631	7.586	.1318	14.4	5.248	.1905	27.54	.03631
8.9	2.786	.3589	7.762	.1288	14.5	5.309	.1884	28.18	.03548
9.0	2.818	.3548	7.943	.1259	14.6	5.370	.1862	28.84	.03467
9.1	2.851	.3508	8.128	.1230	14.7	5.433	.1841	29.51	.03388
9.2	2.884	.3467	8.318	.1202	14.8	5.495	.1820	30.20	.03311
9.3	2.917	.3428	8.511	.1175	14.9	5.559	.1799	30.90	.03236
9.4	2.951	.3388	8.710	.1148	15.0	5.623	.1778	31.62	.03162
9.5	2.985	.3350	8.913	.1122	15.1	5.689	.1758	32.36	.03090
9.6	3.020	.3311	9.120	.1096	15.2	5.754	.1738	33.11	.03020
9.7	3.055	.3273	9.333	.1072	15.3	5.821	.1718	33.88	.02951
9.8	3.090	.3236	9.550	.1047	15.4	5.888	.1698	34.67	.02884
9.9	3.126	.3199	9.772	.1023	15.5	5.957	.1679	35.48	.02818
10.0	3.162	.3162	10.000	.1000	15.6	6.026	.1660	36.31	.02754
10.1	3.199	.3126	10.23	.09772	15.7	6.095	.1641	37.15	.02692
10.2	3.236	.3090	10.47	.09550	15.8	6.166	.1622	38.02	.02630
10.3	3.273	.3055	10.72	.09333	15.9	6.237	.1603	38.90	.02570
10.4	3.311	.3020	10.96	.09120	16.0	6.310	.1585	39.81	.02512
10.5	3.350	.2985	11.22	.08913	16.1	6.383	.1567	40.74	.02455
10.6	3.388	.2951	11.48	.08710	16.2	6.457	.1549	41.69	.02399
10.7	3.428	.2917	11.75	.08511	16.3	6.531	.1531	42.66	.02344
10.8	3.467	.2884	12.02	.08318	16.4	6.607	.1514	43.65	.02291
10.9	3.508	.2851	12.30	.08128	16.5	6.683	.1496	44.67	.02239
11.0	3.548	.2818	12.59	.07943	16.6	6.761	.1479	45.71	.02188
11.1	3.589	.2786	12.88	.07762	16.7	6.839	.1462	46.77	.02138
11.2	3.631	.2754	13.18	.07586	16.8	6.918	.1445	47.86	.02089
11.3	3.673	.2723	13.49	.07413	16.9	6.998	.1429	48.98	.02042
11.4	3.715	.2692	13.80	.07244	17.0	7.079	.1413	50.12	.01995
11.5	3.758	.2661	14.13	.07079	17.1	7.161	.1396	51.29	.01950
11.6	3.802	.2630	14.45	.06918	17.2	7.244	.1380	52.48	.01905
11.7	3.846	.2600	14.79	.06761	17.3	7.328	.1365	53.70	.01862
11.8	3.890	.2570	15.14	.06607	17.4	7.413	.1349	54.95	.01820
11.9	3.936	.2541	15.49	.06457	17.5	7.499	.1334	56.23	.01778
12.0	3.981	.2512	15.85	.06310	17.6	7.586	.1318	57.54	.01738
12.1	4.027	.2483	16.22	.06166	17.7	7.674	.1303	58.88	.01698
12.2	4.074	.2455	16.60	.06026	17.8	7.762	.1288	60.26	.01660
12.3	4.121	.2427	16.98	.05888	17.9	7.852	.1274	61.66	.01622
12.4	4.169	.2399	17.38	.05754	18.0	7.943	.1259	63.10	.01585
12.5	4.217	.2371	17.78	.05623	18.1	8.035	.1245	64.57	.01549
12.6	4.266	.2344	18.20	.05495	18.2	8.128	.1230	66.07	.01514
12.7	4.315	.2317	18.62	.05370	18.3	8.222	.1216	67.61	.01479
12.8	4.365	.2291	19.05	.05248	18.4	8.318	.1202	69.18	.01445
12.9	4.416	.2265	19.50	.05129	18.5	8.414	.1189	70.79	.01413
13.0	4.467	.2239	19.95	.05012	18.6	8.511	.1175	72.44	.01380
13.1	4.519	.2213	20.42	.04898	18.7	8.610	.1161	74.13	.01349
13.2	4.571	.2188	20.89	.04786	18.8	8.710	.1148	75.86	.01318
13.3	4.624	.2163	21.38	.04677	18.9	8.811	.1135	77.62	.01288
13.4	4.677	.2138	21.88	.04571	19.0	8.913	.1122	79.43	.01259
13.5	4.732	.2113	22.39	.04467	19.1	9.016	.1109	81.28	.01230
13.6	4.786	.2089	22.91	.04365	19.2	9.120	.1096	83.18	.01202
13.7	4.842	.2065	23.44	.04266	19.3	9.226	.1084	85.11	.01175
13.8	4.898	.2042	23.99	.04169	19.4	9.333	.1072	87.10	.01148
13.9	4.955	.2018	24.55	.04074	19.5	9.441	.1059	89.13	.01122
14.0	5.012	.1995	25.12	.03981	19.6	9.550	.1047	91.20	.01096
14.1	5.070	.1972	25.70	.03890	19.7	9.661	.1035	93.33	.01072
14.2	5.129	.1950	26.30	.03802	19.8	9.772	.1023	95.50	.01047
14.3	5.188	.1928	26.92	.03715	19.9	9.886	.1012	97.72	.01023

Chart 7-1 (cont.)

dB	Current or Voltage Ratio		Power Ratio	
	Gain	Loss	Gain	Loss
20.0	10.00	0.1000	100.00	0.01000
25.0	17.78	0.0562	3.162×10^{2}	3.162×10^{-3}
30.0	31.62	0.0316	10^{3}	10^{-3}
35.0	56.23	0.0178	3.162×10^{3}	3.162×10^{-4}
40.0	100.00	0.0100	10^{4}	10^{-4}
45.0	177.8	0.0056	3.162×10^{4}	3.162×10^{-5}
50.0	316.2	0.0032	10^{5}	10^{-5}
55.0	562.3	0.0018	3.162×10^{5}	3.162×10^{-6}
60.0	10^{3}	10^{-3}	10^{6}	10^{-6}
65.0	1.778×10^{3}	5.623×10^{-4}	3.162×10^{6}	3.162×10^{-7}
70.0	3.162×10^{3}	3.162×10^{-4}	10^{7}	10^{-7}
75.0	5.623×10^{3}	1.78×10^{-4}	3.162×10^{7}	3.162×10^{-8}
80.0	10^{4}	10^{-4}	10^{8}	10^{-8}
85.0	1.778×10^{4}	5.623×10^{-5}	3.162×10^{8}	3.162×10^{-9}
90.0	3.162×10^{4}	3.162×10^{-5}	10^{9}	10^{-9}
95.0	5.632×10^{4}	1.78×10^{-5}	3.162×10^{9}	3.162×10^{-10}
100.0	10^{5}	10^{-5}	10^{10}	10^{-10}
110.0	3.162×10^{5}	3.162×10^{-6}	10^{11}	10^{-11}
120.0	10^{6}	10^{-6}	10^{12}	10^{-12}
130.0	3.162×10^{6}	3.162×10^{-7}	10^{13}	10^{-13}
140.0	10^{7}	10^{-7}	10^{14}	10^{-14}
150.0	3.162×10^{7}	3.162×10^{-8}	10^{15}	10^{-15}
160.0	10^{8}	10^{-8}	10^{16}	10^{-16}
170.0	3.162×10^{8}	3.162×10^{-9}	10^{17}	10^{-17}
180.0	10^{9}	10^{-9}	10^{18}	10^{-18}

(Courtesy Radio Shack, a Tandy Corporation Company)

***Example 1*—Find the dB equivalent of a power ratio of .631.**

***Answer*—2-dB loss.**

***Example 2*—Find the current ratio corresponding to a gain of 43 dB.**

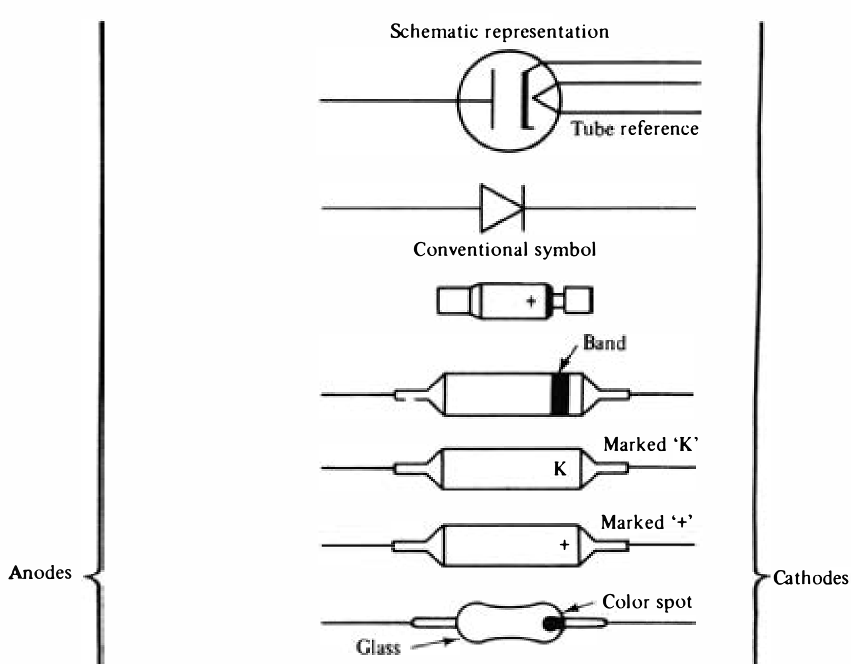
***Answer*—141. [First find the current ratio for 40 dB (100); then find the current ratio for 3 dB (1.41). Multiplying, $100 \times 1.41 =$ 141.]**

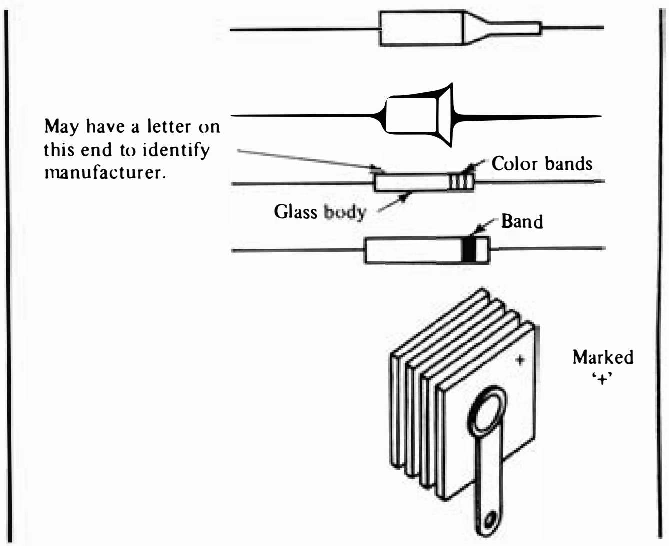
***Example 3*—Find the dB value corresponding to a voltage ratio of 150.**

***Answer*—43.5. [First factor 150 into 1.5×100. The dB value for a voltage ratio of 100 is 40; the dB value for a voltage ratio of 1.5 is 3.5 (approximately). Therefore, the dB value for a voltage ratio is $40 + 3.5$ or 43.5 dB.]**

Appendix III

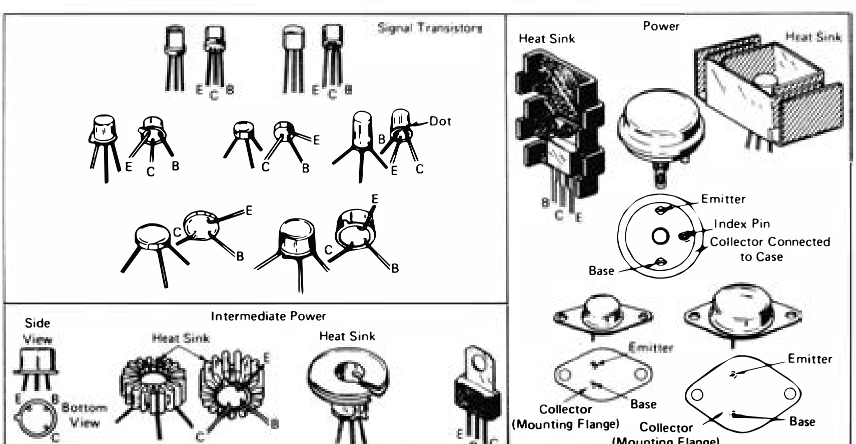
Diode Polarity Identification




May have a letter on this end to identify manufacturer.
Color bands
Glass body
Band
Marked '+'
+

Appendix IV

Transistor Identification

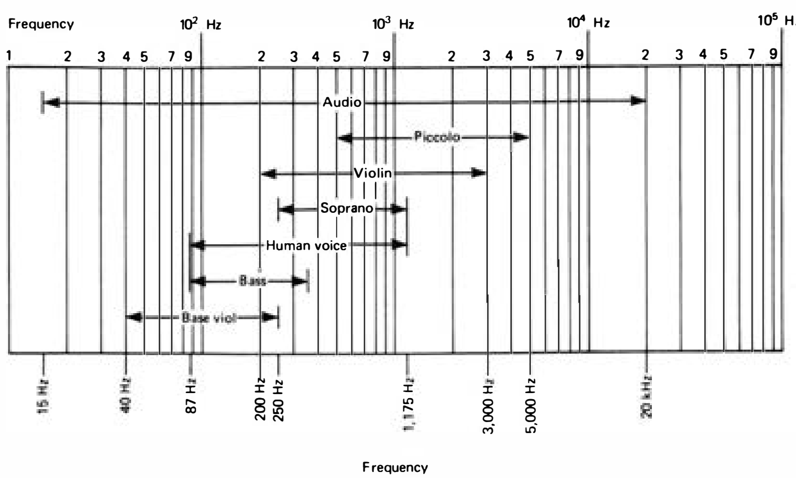


To Test a Transistor You Need to Know Three Things

1. The Basing Configuration (E, B, C, or S, G, D). The Diagram Above Shows Some of the More Common Configurations. If the Transistor Type Number Is Available, the Basing Configuration Can Be Found in the Manufacturer's Handbook. Also, a Schematic May Provide This Information.
2. The Type (NPN or PNP). This Information Can Come From the Circuit, Schematic, or Manufacturer's Handbook.
3. The Power Class. See Diagram Above. (Signal, Intermediate Power, or Power.)

Appendix V

Audio-Frequency Spectrum



Glossary

Acoustic, Audio-Frequency, and Sound Terms

A-B Test. Comparison of sound from two sources, such as comparing original program to tape as it is being recorded by switching rapidly back and forth between them.

Accompaniment. Also called *lower* or *great*. The lower manual of an organ, which provides the musical harmony to the solo or melody.

Acetate Backing. A standard plastic base for magnetic recording tape.

Acoustic. Pertaining to sound or to the science of sound.

Acoustic Absorption Loss. The energy lost by conversion into heat or other forms when sound passes through or is reflected by a medium.

Acoustic Absorptivity. The ratio of sound energy absorbed by a surface to the sound energy arriving at the surface. Equal to 1 minus the reflectivity of the surface.

Acoustic Attenuation Constant. The real part of the acoustic propagation constant; neper per section, or unit distance.

Acoustic Capacitance. In a sound medium, a measure of volume displacement per dyne per square centimeter. The unit is the centimeter to the fifth power per dyne.

Acoustic Clarifier. A system of cones loosely attached to the baffle of a speaker and designed to vibrate and absorb energy during sudden loud sounds, thereby suppressing them.

Acoustic Compliance. The measure of volume displacement of a sound medium when subjected to sound waves. Also, that type of acoustic reactance which corresponds to capacitive reactance in an electrical circuit.

Acoustic Delay Line. A device that retards one or more signal vibrations by causing them to pass through a solid (or liquid).

Acoustic Dispersion. The change in speed of sound with frequency.

Acoustic Elasticity. The compressibility of the air in a speaker enclosure as the cone moves backward. Also, the compressibility of any material through which sound is passed.

Acoustic–Electric Transducer. A device designed to transform sound energy into electrical energy, and vice versa.

Acoustic Feedback. Also called *acoustic regeneration.* The mechanical coupling of a portion of the sound waves from the output of an audio-amplifying system to a preceding part or input circuit (such as a microphone) in the system. When excessive, acoustic feedback produces a howling sound in the speaker.

Acoustic Filter. A sound-absorbing device that selectively suppresses certain audio frequencies while allowing others to pass.

Acoustic Frequency Response. The voltage-attenuation frequency measured into a resistive load, producing a bandwidth approaching sufficiently close to the maximum.

Acoustic Generator. A transducer, such as a speaker, which converts electrical or other forms of energy into sound.

Acoustic Horn. Also called a *horn.* A tube of varying cross section having different terminal areas which change the acoustic impedance to control the directivity of the sound pattern.

Acoustic Impedance. Total opposition of a medium to sound waves. Equal to the force per unit area on the surface of the medium, divided by the flux, (volume velocity or linear velocity multiplied by area) through that surface. Expressed in ohms and equal to the mechanical impedance divided by the square of the surface area. One unit of acoustic impedance is equal to a volume velocity of 1 cubic centimeter per second produced by a pressure of 1 microbar. Acoustic impedance contains both acoustic resistance and acoustic reactance.

Acoustic Inertance. A type of acoustic reactance that corresponds to inductive reactance in an electrical circuit. (The resistance to movement or reactance offered by the sound medium because of the inertia of the effective mass of the medium.) Measured in acoustic ohms.

Acoustic Intensity. The limit approached by the quotient of acoustic power being transmitted at a given time through a given area divided by the area as the area approaches zero.

Acoustic Labyrinth. A special speaker enclosure having partitions and passages to prevent cavity resonance and to reinforce bass response.

Acoustic Lens. An array of obstacles that refract sound waves in the same way that an optical lens refracts light waves. The dimensions of these obstacles are small compared to the wavelengths of the sounds being focused. Also, a device that produces convergence or divergence of moving sound waves. When used with a speaker enclosure, an acoustic lens widens the beam of the higher-frequency sound waves.

Acoustic Line. Mechanical equivalent of an electrical transmission line. Baffles, labyrinths, or resonators are placed at the rear of a speaker enclosure to assist in reproduction of very low audio frequencies.

Acoustic Mode. A mode of crystal-lattice vibration that does not produce an oscillating dipole.

Acoustic Ohm. The unit of acoustic resistance, reactance, or impedance. One acoustic ohm is present when a sound pressure of 1 dyne per square centimeter produces a volume velocity of 1 cubic centimeter per second.

Acoustic Phase Constant. The imaginary part of the acoustic propagation constant. The commonly used unit is the radian per section or unit distance.

Acoustic Pickup. In nonelectrical phonographs, the method of reproducing a recording by linking the needle directly to a flexible diaphragm.

Acoustic Radiator. In an electroacoustic transducer, the part that initiates the radiation of sound vibration. A speaker cone or an earphone diaphragm are examples.

Acoustic Reactance. That part of the acoustic impedance due to the effective mass of the medium—that is, to the inertia and elasticity of the medium through which the sound travels. The imaginary component of acoustic impedance, expressed in acoustic ohms.

Acoustic Reflectivity. The ratio of the rate of flow of sound energy reflected from the surface on the side of incidence to the incident rate of flow.

Acoustic Refraction. A bending of sound waves when passing obliquely from one medium to another in which the velocity of sound is different.

Acoustic Resistance. That component of the acoustic impedance which is responsible for the dissipation of energy due to friction between molecules of the air or other medium through which sound travels. Measured in acoustic ohms, analogous to electrical resistance.

Acoustic Resonance. An increase in sound intensity as reflected waves and direct waves combine in phase. May also be due to the natural vibration of air columns or solid bodies at a particular audio frequency.

Acoustic Resonator. An enclosure that intensifies those audio frequencies at which the enclosed air is set into natural vibration.

Acoustic Scattering. The irregular reflection, refraction, or diffraction of a sound wave in many directions.

Acoustic System. Arrangement of components in devices designed to reproduce audio frequencies in a specified manner.

Acoustic Transmission System. An assembly of elements adapted to the transmission of sound.

Acoustic Treatment. Use of certain sound-absorbing materials to control the amount of reverberation in a room, hall, or other enclosed space.

Acoustic Wave. A traveling vibration by which sound is transmitted in air or other medium. The characteristics of these waves may be described in terms of change of pressure, or particle displacement, or of density.

Acoustic Wave Filter. A device designed to separate sound waves of different frequencies. (Through electroacoustic transducers, such a filter may be associated with electric circuits.)

Acoustics. Science of production, transmission, reception, and effects of sound. Also, in a room or other locations, those characteristics that control reflections of sound waves, and thus the sound reception in the room.

Acoustoelectric Effect. Generation of an electric current in a crystal lattice by a longitudinal sound wave.

Action. An organ action that denotes the assembly of key contacts and couplers.

Aeolian. A very soft organ stop of mild string quality.

AES. Abbreviation for Audio Engineering Society.

AF. Abbreviation for audio frequency, a range that extends from 20 Hz to 20 kHz.

AFC. Abbreviation for automatic frequency control, a circuit commonly used in FM receivers to compensate for frequency drift, to keep the tuner "locked" to a selected station.

Air Column. The air space within a horn or an acoustic chamber.

AM. Amplitude modulation; a method of superimposing intelligence on an RF carrier by amplitude variation of the carrier.

Ambient Noise. Acoustic noise in a room or other location. Usually measured with a sound-level meter.

Amplification. Magnification or enlargement.

Amplifier. An electronic device that magnifies or enlarges audio voltage or power signals.

Amplitude. Also called *peak value;* the maximum value of a waveform (with respect to one polarity).

Anechoic Enclosure. A low-reflection audio-frequency enclosure.

Anechoic Room. A room in which reflected sound energy is negligible, used for measurement of speaker and microphone characteristics.

Anode. The electrode at which electrons leave a device to enter the external circuit.

Arpeggio. Technique of playing the notes of a chord in rapid sequence instead of simultaneously; sometimes accomplished by automatic circuit action.

Articulation. The percentage of speech units understood by a listener.

Attack (related to *rise time*). The period of time during which a tone increases to full amplitude after a musical instrument starts to emit a tone.

Attenuation. Opposite of amplification; reduction of audio voltage or power.

Audio. Pertaining to frequencies corresponding to a normally audible sound wave. These frequencies range approximately from 15 Hz to 20 kHz.

Audio Level Meter. An instrument that measures audio-frequency power with reference to a predetermined level. Usually calibrated in dB.

Audio Rectification. Overdrive of an input stage in an audio amplifier by stray radio-frequency fields, resulting in production of interference in the output from the amplifier.

Audiophile. One who enjoys experimenting with high-fidelity equipment and who is likely to seek the best possible reproduction.

Autotransformer. A transformer designed with a single, tapped winding that serves as both primary and secondary.

Background Noise. Noise inherent in any electronic system.

Back Loading. A form of horn loading particularly applicable to low-frequency speakers; the rear radiating surface of the speaker feeds the horn and the front part of the speaker is directly exposed to the room.

Backloaded Horn. A speaker enclosure arrangement in which the sound from the front of the cone feeds directly into the room, while the sound from the rear feeds into the room via a folded horn.

Baffle. A partition or enclosure in a speaker cabinet that increases the length of the air path from the front to the rear radiating surfaces of the speaker.

Baroque. Baroque music is a basic form of composition characterized by ornamentation and powerful climaxes.

Bass. The lower or pedal tones provided by an organ.

Bass-Reflex Enclosure. A speaker cabinet enclosure in which a portion of the radiation from the rear of the cone is channeled to reinforce the bass tones.

Bass Response. The extent to which a speaker or audio amplifier processes low audio frequencies.

Bassy. A term applied to sound reproduction in which the low-frequency tones are overemphasized.

Beat. A successive rising and falling of a wave envelope caused by alternating reinforcements and cancellations of two or more component frequencies.

Bias. A high-frequency signal applied to a tape-recording head simultaneously with the audio signal to offset the effect of hysteresis in the core of the head. Its adjustment is critical—too much bias results in loss of the high audio frequencies, too little in increased distortion. The exact amount of bias depends on the tape formulation, and ideally the adjustment should be made for each type of tape used.

Binaural. A type of sound recording and reproduction. Two microphones, each representing one ear and spaced about 6 inches apart, are used to pick up the sound energy to be recorded on separate tape channels. Playback is accomplished through separate amplifiers (or a two-channel amplifier) or special headphones wired for binaural listening.

Blocked Impedance. The input impedance of a transducer when its output is connected to a load of infinite impedance.

Blocked Resistance. Resistance of an audio-frequency transducer when its moving elements are restrained so that they cannot move.

Boffle. A Hartley speaker enclosure that contains a group of stretched, resilient, sound-absorbing screens.

Bourdon. A low-pitched wood-flute organ pipe.

Brass. A generalized term that denotes tones resembling those from brass instruments such as the tuba, trumpet, or cornet.

Bridge. A precision electrical instrument for the measurement of resistance, capacitance, and inductance values.

Buffer. A device, such as an electron tube or transistor, employed between an ac source and its load, principally for the purpose of isolation.

Bus Bar. A bare electrical conductor that connects to various tone sources or that distributes voltages to various points in an organ system.

Cabinet, Tone. A speaker enclosure designed for operation with an electronic organ.

Capacitor (obsolete: *condenser*). Any device designed for storage of electrostatic field energy.

Capstan. The spindle or shaft of a tape transport mechanism that pulls the tape past the heads.

Capture Ratio. An FM tuner's ability to reject unwanted co-channel signals. If an undesired signal is more than 2.2 dB lower than a desired signal, the undesired signal will be completely rejected.

Cardioid Pattern. A heart-shaped directional pickup pattern for a microphone that assists in reducing background noise.

Carillon. A bell-tower voice actuated from an organ keyboard; the bell tones are electronically generated.

Cartridge. A transducer device used with a turntable to convert a mechanical channel in a disc into electrical impulses.

Cassette. A small plastic housing which encloses a length of tape and two reels (supply and take-up). It incorporates openings for the heads and the tape-drive mechanism. The standard tape speed for cassettes is 1⅞ inches per second (ips).

CD-4. A phonograph record that can store four channels of discrete sound.

Celeste. An organ stop characterized by a slow beat of 3 or 4 Hz; it is used in the upper register, usually in the diapason family.

Cent. An interval between two tones, with a value of approximately 1/100 semitone.

Ceramic. A piezoelectric element that is used as the basis of some phonograph pickups; it generates a potential difference when stressed or strained.

Changer. A record-playing device that automatically accepts and plays up to 10 or 12 discs sequentially.

Channel. A complete sound path. A monophonic system has one channel; a stereophonic system has two channels; a quadraphonic system has four channels. Monophonic material may be played through a stereophonic system, and quadraphonic material may be played through a stereophonic system. An amplifier may have several inputs, such as microphone(s).

Channel Balance. Equal response from left and right channels of a stereo amplifier. A balance control in a stereo amplifier permits adjustment for uniform sound volume from both speakers or a hi-fi system.

Chassis. Metal frame, or box, that houses the circuitry of an electronic unit or system.

Chimes. A bell-like tone produced by striking metal tubes or rods with a hammer, or by an equivalent electronic synthesis.

Choir. An organ voice produced by blending several tones (of the same family) that have practically the same pitch but differing phases. Sometimes a choir effect is simulated by blending several tones with a phase difference produced by frequency-modulating one or more of the tones.

Chord. A combination of harmonious tones that are sounded simultaneously.

Chord Coupling. An organ coupling mode wherein all tones for a specific chord can be played by depressing a single button or key.

Chord Organ. An organ arranged for playing a variety of chords in harmony with solo tones. Each chord is played by depressing a single button or key.

Chorus Effect. Same as *choir.*

Chromatic Keyboard. A keyboard with the black notes placed at the same height as the white notes, and with the same widths, to facilitate playing of chromatic scales.

Chromatic Percussion. Percussive effects that are applied to notes of an organ, to simulate struck strings, plucked strings, marimba, or xylophone voices.

Chromatic Scale. A scale composed entirely of half-steps.

Cipher. A tone that sounds when no key is depressed, owing to malfunction.

Clarinet. An organ stop for a voice that simulates clarinet tones.

Clavier. Any keyboard or pedal board operated with either the hands or feet. A hand-operated clavier is more often termed a *manual.*

Compact. A record player and amplifier, mounted on a common base; also called a *modular system.*

Compensator. A fixed or variable circuit built into a preamplifier that compensates for bass and treble alterations made during the recording process.

Complex Tone. An audio waveform composed of a fundamental frequency and a number of integrally related harmonic frequencies (a pitch and a number of related overtones).

Compliance. Physical freedom from rigidity that permits a stylus to track a record groove precisely, or of a speaker to respond to an audio signal precisely.

Concordant. A series of musically meaningful tones.

Cone. The diaphragm that sets the air in motion to generate a sound wave in a direct-radiator speaker; usually conical in shape.

Conical Horn. A horn the cross section of which increases as the square of its axial length.

Console. A cabinet that houses an electronic organ; also, a high-fidelity system completely enclosed in a cabinet with two speakers.

Contra. When prefixed to the name of a musical instrument, this term signifies that the tones have been lowered one octave.

Cornopean. An organ voice with a rich and horn-like tone color.

Counterbass. Also termed *contrabass;* this term denotes a second bass note that will harmonize with a particular chord.

Coupler. A stop or tab that permits the tones on one manual of an organ to be played with the tones of another manual, or that permits the sounding of octavely related tones on the same manual.

CPS. Abbreviation for cycles per second; *see* Hertz, Cycle, and Cycles per Second.

Crescendo. A pedal or equivalent control for an electronic organ that rapidly brings all stops into play; an increase in voice output to maximum power capability.

Crossover Distortion. Distortion that occurs in a push-pull amplifier at the points of operation where the signals cross over the zero axis.

Crossover Frequency. In reference to electrical dividing networks, the audio frequency at which equal power is delivered to each of the channels or speakers.

Crossover Network. Filtering circuit that selects and passes certain ranges of audio frequencies to the speakers that are designed for the particular ranges.

Crosstalk. In stereo high-fidelity equipment, crosstalk signifies the amount of left-channel signal that leaks into the right channel, and vice versa.

Crystal. A natural piezoelectric element that is used in some phono pickup cartridges and microphones.

Crystal Loudspeaker. A speaker in which piezoelectric action is used to produce mechanical displacement. Also termed a *piezoelectric loudspeaker.*

Cycle. One complete reversal of an alternating current, including a rise to maximum in one direction, a return to zero, a rise to maximum in the other direction, and another return to zero. The number of cycles occurring in 1 second is defined as the frequency of an alternating current. The word *cycle* is commonly interpreted to mean cycles per second, in which case it is a measure of frequency. The preferred term is hertz.

Cycles per Second. An absolute unit for measuring the frequency or "pitch" of a sound, various forms of electromagnetic radiation, and alternating electric current. *See* Hertz.

Cymbal. A high-pitched metallic organ stop that simulates the metallic clashing sound of orchestra cymbals.

Damping. Controlling of vibrations, response, or resonances which if unchecked would cause distortion.

Dead End. The end of a sound studio with the greater sound-absorption characteristic.

Dead Room. A room for testing the acoustic efficiency or range of electroacoustic devices such as speakers and microphones. The room is designed with an absolute minimum of sound reflection, and no two dimensions of the room are the same. The walls, floor, and ceiling are lined with sound-absorbent material.

Decay. A period of time over which a tone decreases from peak volume to inaudibility. It is characterized as an exponential function that defines the natural law of decay (and growth).

Decibel (dB). A unit for measuring relative power levels. One dB is equal to one-tenth of a bel and is about the smallest change that can be detected by a critical listener.

Decoder. Circuitry for recovering information that has been encoded. Present fm stereo receivers use decoder circuits to obtain the second channel. Various decoding systems are in use for obtaining four-channel information from records.

Deemphasis. An attenuation of certain frequencies; in playback equalization, deemphasis offsets the preemphasis given to the higher frequencies during the recording process.

Delay line. An electromechanical transmission line (or equivalent) for delaying a signal or impulse in passage between the input and output terminals; often terminated in comparatively high or low impedances, to obtain energy reflections (reverberation).

Diapason. The basic tone color of traditional organ voices, as produced by open or stopped pipes.

Diaphragm. Thin, flexible sheet that vibrates when struck by sound waves, as in a microphone, or which produces sound waves when moved back and forth at an audio-frequency rate, as in a headphone or a speaker.

Diffracted Wave. A sound wave that has struck an object and has been bent or deflected, other than by reflection or refraction.

Diffraction. The bending of sound waves as they pass through an object or barrier, thereby producing a diffracted wave. Also, the phenomenon whereby waves traveling in straight paths bend around an obstacle.

Diode. A unilateral electronic device that is used in rectification, waveshaping, switching, and other circuit applications.

Direct Radiator Speaker. A speaker in which the radiating element acts directly on the air instead of relying on any other element, such as a horn.

Directivity Factor. Of a transducer used for sound emission, the ratio of the intensity of the radiated sound at a remote point in a free field on the principal axis to the average intensity of the sound transmitted through a sphere passing through the remote point and concentric with the transducer.

Directivity Index. Also termed *directional gain.* A measure of the directional properties of a transducer. It is the ratio, in dB, of the average intensity over the whole sphere surrounding the projector to the intensity on the acoustic axis.

Discordant. Tones that are unrelated by established principles of harmony.

Discrete. Four-channel sound handled as such without conversion to two-channel sound; four independent sound sources on tape or disc played back through two stereo amplifiers into four speakers.

Distortion. Deviations from an original sound that occur in the reproduction process. Harmonic distortion disturbs the original relationship between a tone and other tones naturally related to it. Intermodulation distortion introduces new tones that result from the beating of two or more original tones.

Divider. A circuit, device, or arrangement that reduces a signal voltage to a certain fraction of its input value or that generates a subharmonic of an input signal frequency.

Dividing Network. Same as *crossover network.*

"Dolbyized" Tape. A prerecorded tape that was made using the Dolby noise-reduction process. Such tapes should be played back only on machines equipped with Dolby playback circuitry if accurate frequency response is desired. If Dolbyized tapes are played back on ordinary tape machines, they will tend to sound shrill and exhibit an overabundance of high frequencies. The Dolby noise-reduction system is also applicable to FM broadcasting.

Doppler Tone Cabinet. A tone-cabinet design in which one or more speakers are rotated or in which a baffle is rotated to produce a mechanical vibrato/tremolo effect.

Double Touch. A key-contact design for an electronic organ that provides actuation of an additional circuit when somewhat more than normal finger pressure is applied.

Doubling. The generation of a large amount of second-harmonic distortion as the result of a nonlinear motion of a speaker cone.

Drone Cone. An undriven speaker cone mounted in a bass-reflex enclosure.

Ducted Port. A form of bass-reflex speaker enclosure in which a tube is mounted behind the reflex port.

Dulciana. A flute voice with a small and slightly stringy tone.

Dynamic Cartridge (electrodynamic). A magnetic phono pickup in which a moving coil in a magnetic field generates voltages to form an audio signal.

Dynamic Microphone. A microphone that operates on the same basic principle as a dynamic cartridge.

Dynamic Speaker. Also termed a *moving-coil speaker.* The moving diaphragm is attached to a coil, which is conductively connected to the source of electric energy and placed in a constant magnetic field. The current through the coil interacts with the magnetic field, causing the coil and diaphragm to move back and forth in accordance with the current variations through the coil.

Dyne per Square Centimeter. The unit of sound pressure. Originally called a bar, but now termed by the full expression.

Eccles–Jordan Oscillator. Also termed a *flip-flop* or *bistable multivibrator;* used for frequency division in electronic organ networks.

Echo. A delayed repetition (sometimes several rapid repetitions) of the original sound.

Effective Current. The value of alternating or varying current that will produce the same amount of heat as the same value of direct current. Also called *rms current.*

Effective Sound Pressure. The root mean square of the instantaneous sound pressure at one point over a complete cycle. The unit is the dyne per square centimeter.

Efficiency. In a speaker, the ratio of power applied to the input terminals, expressed as a percentage.

Electroacoustic. Pertaining to a device, such as a speaker, which involves both electric current and sound-frequency pressures.

Electroacoustic Transducer. A device that receives excitation from an electric system and delivers its output to an acoustic system, or vice versa.

Electrodynamic Speaker. A speaker consisting of an electromagnet termed the field coil, through which a direct current flows.

Electromagnetic. Pertaining to a phenomenon that involves the interaction of electric and magnetic field energy.

Electrostatic Speaker. A type of speaker in which sound is produced by charged plates that are caused to move while one is changed from positive to negative polarity, resulting in forces of attraction or repulsion.

Electrostatic Tweeter. A speaker with a movable flat metal diaphragm and a nonmovable metal electrode capable of reproducing high audio frequencies. The diaphragm is driven by the varying high voltage that is applied to the plates.

Enclosure. A housing that is acoustically designed for a speaker or speakers. Also called a *tone cabinet* in electronic organ technology.

Encoding. A process for conveying additional information without disturbing the original format. Encoded four-channel records can be played on standard two-channel equipment.

Equal-Loudness Contours. *See* Fletcher–Munson Curves.

Erase Head. The leadoff head in a tape recorder that erases previous recordings from the passing tape by generating a strong and random magnetic field.

Excess Sound Pressure. The total instantaneous pressure at a point in a medium containing sound waves, minus the static pressure when no sound waves are present. The unit is the dyne per square centimeter.

Expression Control. An organ volume control, usually operated with the right foot.

Extended Octave. A tone above or below a note on a standard keyboard that sounds when a specific coupler is actuated.

Fast Decay. A rapid attenuation of a tone after its keyswitch has been released.

Feed Reel. The reel in a tape recorder that supplies the tape.

FET (field-effect transistor). A transistor of the voltage-operated device classification, instead of the current-operated type as a bipolar transistor.

Fidelity. The faithfulness of sound reproduction.

Filter Network. A reactive network that is designed to provide specified attenuation to signals within certain frequency limits; basic filters are termed low-pass, high-pass, bandpass, and band-reject designs.

Flare Factor. A number that expresses the degree of outward curvature of a speaker horn.

Flat. A note that is a half-step or semitone lower than its related natural pitch.

Flat Response. A characteristic of an audio system whereby any tone is reproduced without deviation in intensity for any part of the frequency range that it covers.

Fletcher–Munson Curves. Also called *equal-loudness contours.* A group of sensitivity curves showing the characteristics of the human ear for different intensity levels between the threshold of hearing and the threshold of feeling. The reference frequency is 1 kHz.

Flute. A basic electronic organ tone color that simulates the orchestral flute.

Flutter. A form of distortion caused when a tape transport or a turntable is subject to rapid speed variation.

FM. Frequency modulation.

FM Stereo. Broadcasting over FM frequencies of two sound signals within a single channel. A *multiplexing* technique is utilized.

Folded Horn. A type of speaker enclosure that employs a horn-shaped passageway that improves bass response.

Force Factor (of an electroacoustic transducer). The complex quotient of the force required to block the mechanical or acoustic system, divided by the corresponding current in the electrical system. The complex quotient of the resultant open-circuit voltage in the electric system, divided by the velocity in the mechanical or acoustic system.

Force-Summing Device. In a transducer, the element directly displaced by the applied stimulus.

Formant Filter. A waveshaping network or device that changes the waveform of a tone-generator signal into a desired musical tone waveform.

Forte. A forte tab (solo tab) increases the volume of other tabs that are depressed at the time; a forte tab has no voice of its own.

Foundation Voice. A definitive organ voice, such as the diapason and dulciana voices.

Free Impedance. Also called *normal impedance.* The input impedance of a transducer when the load impedance is zero.

Free Motional Impedance. The complex remainder after the blocked impedance of a transducer has been subtracted from the free impedance.

Free-Running Oscillator. An oscillator that generates an output in the absence of a synchronizing signal or a trigger signal.

Free Sound Field. A field in a medium free of discontinuities or boundaries. In practice, it is a field in which the boundaries cause negligible effects over the region of interest.

Frequency. The number of complete vibrations or cycles completed in 1 second by a waveform, and measured in Hertz.

Frequency Modulation. A method of broadcasting that varies the frequency of the carrier instead of its amplitude. FM is the selected high-fidelity medium for broadcasting high-quality program material.

Frequency Range. The limiting values of a frequency spectrum, such as 20 Hz to 20 kHz.

Frequency Response. The frequency range over which an audio device or system will produce or reproduce a signal within a certain tolerance, such as ± 1 dB.

Fundamental. The normal pitch of a musical tone; usually, the lowest frequency component of a tonal waveform.

Gain. The value of amplification that a signal obtains in passage through an amplifying stage or system.

Gate Circuit. A circuit that operates as a selective switch and permits conduction over a specified interval.

Gemshorn. A flute organ voice with a bright tone color.

Generator. A tone or signal source, such as an oscillator, frequency divider, or magnetic tone wheel.

Glide. Also termed *glissando.* A rapid series of tones, produced by a slight shift in pitch of successive tones.

Glockenspiel. Also called *orchestra bells.* An electromechanical arrangement that simulates the bells used in orchestras.

Great Manual. Also called *accompaniment manual* or *lower manual.* A keyboard used for playing the accompaniment to a melody.

Grille. A decorative and protective sound-transparent structure and/or mesh that forms the front surface of a speaker enclosure.

Half-Tone. Also called *semitone.* The relation between adjacent pitches on the tempered scale.

Harmonic. A frequency component of a complex waveform that bears an integral relation to the fundamental frequency. Also called *overtone.*

Harmonic Distortion. *See* Distortion.

Harmony. Musical support for a melody, consisting of two or more notes played simultaneously.

Head. Electromagnetic device used in magnetic tape recording to convert an audio signal to a magnetic pattern, and vice versa.

Headphones. Small sound reproducers resembling miniature speakers used either singly or in pairs, usually attached to a headband to hold the phones snugly against the ears. Available in monophonic or stereophonic design.

Helmholtz Resonator. An acoustic enclosure with a small opening that causes the enclosure to resonate. The frequency of resonance is a function of the resonator geometry.

Hertz. A unit of frequency equal to 1 cycle per second.

High Fidelity. The characteristic that enables an audio system to reproduce sound as nearly like the original as possible.

Hole-in-the-Middle Effect. The lower volume or absence of sound between the left and right speakers of a stereo system.

Horn. Also called an *acoustic horn.* A tubular or rectangular enclosure for radiation of acoustic waves.

Horn Cutoff Frequency. A frequency below which an exponential horn will not function correctly because it fails to provide for proper expansion of the sound waves.

Horn Loading. A method of coupling a speaker diaphragm to the listening space by an expanding air column that has a small throat and a large mouth.

Horn Mouth. The wide end of a horn.

Horn Speaker. A speaker in which a horn couples the radiating element to the medium.

Horn Throat. The narrow end of a horn.

Hum. Noise generated in an audio or other electronic device by a source or sources of electrical disturbance.

IC. Abbreviation for *integrated circuit.* Integral solid-state units that include transistors, resistors, semiconductor diodes, and often capacitors, all of which are formed simultaneously during fabrication.

Ideal Transducer. Theoretically, any linear passive transducer which, if it dissipated no energy and, when connected to a source and load, and presented its combined impedance to each would transfer maximum power from source to load.

IHFM (IHF). Refers to the Institute of High Fidelity Manufacturers, now called the Institute of High Fidelity, Inc. This group devises and publishes standards and ratings for high-fidelity equipment.

Image Rejection. The ability of a receiver to reject interference that is produced by an undesired input frequency which beats with the local-oscillator frequency, to produce an abnormal IF frequency.

Impedance. An electrical unit, expressed in ohms, that denotes the amount of opposition to alternating-current flow by a device or a circuit.

Infinite Baffle. A speaker mounting arrangement in which the front and back waves from a cone are totally isolated from each other.

Input. Connection through which an electric current is fed into a device, circuit, or system.

Integrated Amplifier. An audio preamplifier and power amplifier housed in a single cabinet.

Integrated Receiver. An integrated receiver contains an FM/AM tuner, preamplifier, and power amplifier in the same cabinet.

Intermodulation Distortion (IM). Two distinct and separate test frequencies are mixed in an amplifier, and their difference-frequency output is measured in IM percentage. *See* Distortion.

Interval. The difference in pitch between two musical tones.

Jack. A female receptacle for a plug-type connector.

Keybed. A shelf or horizontal surface on which a keyboard is mounted.

Keyboard. A bank of keys, comprising black-and-white sets, arranged in ascending tones.

Keynote. The tonic, or first note of a particular scale.

Keyswitch. A switch that closes when a key is depressed, thereby actuating a tone generator.

Kinura. A reed stop that has dominant harmonics and a subordinate fundamental.

Labyrinth. A speaker enclosure with absorbing air chambers at the rear to eliminate acoustic standing waves.

Lateral System. A system of disc recording in which a stylus moves from side to side (laterally).

Leslie Speaker. A generic term, originally a trade name, denoting a tone cabinet with a mechanical tremolo-vibrato assembly.

Level Indicator. A neon bulb, meter, or "eye" tube, used to indicate recording levels.

Lissajous Figures. An *XY* plot of voltage or current phase relations, usually produced automatically on the screen of a cathode-ray tube.

Load. A device, circuit, or system that absorbs or converts power from an electrical source, as a speaker converts power from an amplifier.

Loudness Control. An audio-frequency filtering arrangement that boosts the treble and particularly the bass tones in an amplifier as the volume level is reduced; it compensates for the listener's reduced sensitivity to tones at the extreme ends of the audio range at low volume levels.

Loudspeaker. Equivalent term for speaker.

Loudspeaker Dividing Network. Equivalent term for crossover network.

Loudspeaker Impedance. Equivalent term for speaker impedance.

Loudspeaker System. Equivalent term for speaker system.

Louver. The grille of a speaker.

Magnetic Armature Speaker. A speaker comprising a ferromagnetic armature actuated by magnetic attraction.

Magnetic Speaker. A speaker in which acoustic waves are produced by mechanical forces resulting from magnetic reaction.

Magnetic Tape. Plastic tape with an iron-oxide coating for magnetic recording.

Manual. Also termed a *clavier.*

Manual Player. Manual record-playing device used with a changer-type machine.

Master Oscillator. A source of a tone signal; it may be utilized directly, or it may be processed through one or more frequency dividers; these are also oscillators but are of the driven type.

Mean Free Path. The average distance that sound waves travel between successive reflections in an enclosure.

Mechanical Tone Generator. A mechanical unit for generation of electrical impulses that are subsequently converted into audible tones.

Mechanicals. Organ effects that are not voices in the strict sense of the term; thus, forte (solo), percussion effects, and couplers.

Megohm. A multiple unit that denotes 1 million ohms.

Mel. A unit of pitch; a simple 1-kHz tone, 40 dB above a listener's threshold, produces a pitch of 1000 mels. The pitch of any sound that is judged by the listener to be n times that of a 1-mel tone is denoted as n mels.

Melodia. An organ solo stop of the flute family.

Melody. Also called a *tune;* usually played sequentially note by note on the swell or solo manual.

Micro. A prefix that denotes one-millionth.

Microbar. A unit of pressure commonly used in acoustics. One microbar is equal to 1 dyne per square centimeter.

Milli. A prefix that denotes one-thousandth.

Mixing. A blend of two or more electrical signals or acoustic waves.

Modulation. A process wherein low-frequency information is encoded into a higher-frequency carrier or subcarrier; subdivisions include amplitude, frequency, and phase modulation, with various combinations and derivatives thereof.

Monophonic. A recording and reproduction system in which all program material is processed in one channel.

Monorange Speaker. A speaker that provides the full spectrum of audio frequencies.

Moving-coil Speaker. Also termed a *dynamic speaker.* A speaker in which the moving diaphragm is attached to a coil, which is driven by audio-frequency currents. These currents interact with a fixed magnetic field and cause the diaphragm to vibrate in unison.

Multiplexing. A system of broadcasting in which two or more separate channels are transmitted on one FM carrier, as in stereophonic broadcasting.

Multivibrator. A relaxation oscillator, usually developing a semisquare waveform. Subclassifications include the astable, monostable, and bistable types.

Muting. A silencing process or action.

NAB Curve. Tape-recording equalization curve established by the National Association of Broadcasters.

Nazard. An organ voice that simulates a piccolo-type pipe-organ voice.

Near Field. The acoustic radiation field close to the speaker or some other acoustic source.

Neon Lamp. A gas diode that emits an orange glow, and operates as an indicator, protective switch, regulator, relaxation oscillator, or divider.

Network. A comparatively elaborate electrical or electronic circuit arrangement.

Nonchromatic Percussion. A percussion effect that has no dominant pitch, such as wood-block, drum, castanet, or cymbal effects.

Note. A single musical tone, identified by the letters A through G, plus sharp or flat superscripts.

Octave. A pair of tones are separated by an octave if one has twice the frequency of the other.

Octave Coupling. An organ coupling arrangement wherein the depression of a key causes another note, an octave higher or lower in pitch, to sound simultaneously.

Ohm. The unit of electrical resistance, defined as a unitary voltage/current ratio.

Oscillator. An electronic, electrical, or mechanical generator of an electrical signal.

Outphasing. An organ voicing method wherein specified harmonics or subharmonics are added to or subtracted from a tone signal prior to its application to a formant filter. In a *chiff* outphasing circuit, certain harmonics are added to the tone signal during its attack period.

Output. A connection or conductor through which an electrical signal emerges from an electrical or electronic device, circuit, or system.

Overall Loudness Level. A measure of the response of human hearing to the strength of a sound. It is scaled in phons and is an overall single evaluation calculated for the levels of sound pressure of several individual bands.

Overtone. Same as *harmonic.*

Partial. Any one of the various frequencies contained in a complex waveform that corresponds to a musical tone.

Patch Cord. A shielded cable utilized to connect one audio device to another.

Peak Sound Pressure. The maximum absolute value of instantaneous sound pressure for any specified time interval. The most common unit is the microbar.

Pedal. The pedal keyboard of an electronic organ; also termed *clavier.*

Pedal Clavier. A pedal keyboard.

Pedal Divider. A frequency-divider section associated with the tone generators actuated by the foot pedals.

Pedal Generator. A tone generator utilized to produce the bass notes of an organ.

Pedal Keyboard. Same as *pedal clavier.*

Percussion. Characteristic tones, as produced by plucking or striking strings.

Permanent-Magnet Speaker. A moving-conductor speaker in which the steady magnetic field is produced by a permanent magnet.

pF. Abbreviation for *picofarad.*

Phase. Position occupied at any instant in its cycle by a periodic wave; a part of a sound wave or signal with respect to its passage in time. One signal is said to be in phase, or to lead, or to lag, another reference signal.

Phase Inverter. An amplifier that provides an output that is 180 degrees out of phase with its input, or an amplifier that provides a pair of output voltages that are 180 degrees out of phase with each other.

Phon. The unit for measurement of the apparent loudness level of a sound. Numerically equal to the sound-pressure level, in decibels relative to 0.0002 microbar, of a 1-kHz tone that is considered by listeners to be equivalent in loudness to the sound under consideration.

Pickup Cartridge. A device used with a turntable to convert mechanical variations into electrical impulses.

Picofarad. A unit equal to 1 micromicrofarad.

Piezoelectric Speaker. A speaker that employs a piezoelectric substance as a driver or motor.

Piston. An organ stop that is operated by pulling or pushing a knob. A piston generally operates groups of conventional stops.

Piston Action. The movement of a speaker cone or diaphragm when driven at the bass audio frequencies.

Pitch. That characteristic of a sound which places it on a musical scale.

Pizzicato. An organ sound effect that simulates the rapid plucking of strings.

Playback Head. The last head of a tape recorder, or the only head on a tape player, which converts the magnetic pattern impressed on a passing tape into an audio signal.

Plug-Type Connector. A mating connector for a jack.

PM. Permanent magnet.

Polyester Backing. A plastic material used as a base for magnetic recording tape.

Port. An opening in the baffle of a bass-reflex speaker enclosure for selective radiation of sound waves.

Power. A unit of the rate at which work is done, or energy is consumed, or energy is generated; electrical power is measured basically in rms watts.

Power Amplifier. An amplifier that drives a speaker in an audio system.

Power Output. The signal power delivered by an audio amplifier, measured in watt units.

Power Supply. A source of electrical energy; usually an arrangement that converts alternating current into virtually pure direct current.

Preamplifier. Amplifying arrangement that steps up a very weak input signal to a suitable level for driving an intermediate amplifier or a power amplifier.

Preemphasis. A deliberate exaggeration of the high-frequency components in an audio signal.

Presence. The quality of naturalness in sound reproduction. When the presence of a system is good, the illusion is that the sounds are being produced intimately at the speaker.

Preset. A control that turns on a group of voices or that turns them off without actuating any tabs.

Print-Through. Magnetization of a layer of tape by an adjacent layer.

Pulse. An electrical transient or a series of repetitive surges.

Quadraphonic. A system whereby sound that is picked up by four separate microphones is recorded on separate channels and played back through separate channels that drive individual speakers.

Quality. Relates to the harmonic content of a complex tonal waveform; also termed *timbre.*

Quarter-Track Recorder. A tape recorder that utilizes one-quarter the width of the tape for each recording; in stereo operation, two of the four tracks are used simultaneously.

Quieting. Standard of separation between background noise and the program material from a tuner.

Record Head. The second head of a tape recorder; used to convert an audio signal to a magnetic pattern on the passing tape.

Record-Playback Head. A head on a tape recorder that performs both recording and playback functions.

Recording Amplifier. An amplifying section in a tape recorder that prepares an audio signal for application to the record head and bias current to the erase head.

Reed. One of the basic tone-color groups of organ voices that simulates orchestral reeds.

Reference Acoustic Pressure. That magnitude of a complex sound that produces a sound-level meter reading equal to the reading that results from a sound pressure of 0.0002 dyne per square centimeter at 1 kHz. Also called *reference sound level.*

Register. A range of notes included by a clavier or manual; range of notes employed in playing a particular musical composition.

Relay. An electromagnetically operated switching device.

Reproducer. A device used to translate electrical signals into sound waves.

Resultant. Denotes a tone that is produced when two notes a fifth apart and an octave higher than the desired note are sounded to produce the desired pitch; a mode of generating *synthetic bass.*

Reverberation. The persistence of sound due to the repeated reflections from walls, ceiling, floor, furniture, and occupants in a room.

Reverberation Period. The time required for the sound in an enclosure to decay to one-millionth (60 dB) of its original intensity.

Reverberation Strength. The difference between the level of a plane wave that produces in a nondirectional transducer a response equal to that produced by the reverberation corresponding to a 1-yard range from the effective center of the transducer.

Reverberation Time. For a given frequency, the time required for the average sound-energy density, originally in a steady state, to decay to one-millionth (60 dB) of its initial value after the source is stopped.

Rhythm Section. An organ section that generates nonchromatic percussion effects in a periodic manner, either automatically or manually.

RIAA Curve. Standard disc-recording curve specified by the Record Industry Association of America.

Ribbon Tweeter. A high-frequency speaker, usually horn-loaded, in which a stretched, straight flat ribbon is used instead of a conventional voice coil.

Rolloff. The rate at which a frequency-response curve decreases in amplitude; it is usually stated in dB per octave, or dB per decade.

Rumble. A low-frequency vibration originating from a vibrating electric motor in a turntable.

Rumble Filter. A low-frequency filter circuit designed to minimize or to eliminate rumble interference.

Sabin (square-foot unit of absorption). A measure of the sound absorption of a surface. It is equivalent to 1 square foot of a perfectly absorptive surface.

Scale. A series of eight consecutive whole notes.

Scratch Filter. A high-frequency filter circuit that minimizes scratchy sounds in playback of deteriorated discs.

Sectoral Horn. A horn with two parallel and two diverging sides.

Selectivity. A measure of the ability of an electronic device to select a desired signal and to reject adjacent interfering signals; also termed *bandwidth.*

Semitone. The relation between adjacent pitches on the tempered scale.

Sensitivity. The minimum value of input signal that is required by an electronic unit, such as a tuner, to deliver a specified output signal level.

Separation. The degree to which one channel's information is excluded from another channel; customarily expressed in dB units.

Sforzando. A form of *crescendo,* but also employing discordant tones.

Sharp. Removed by a semitone from a reference pitch.

Signal-to-Noise Ratio. The extent to which program material exceeds the background noise level; customarily expressed in dB units.

Sine Wave. Graphical representation of simple harmonic motion.

Soft-suspension Speaker. A speaker design without inherent springiness; it utilizes the reaction of a trapped backwave for restorative force.

Solo Manual. The upper manual of a two-manual organ; also called a *swell manual.*

Sone. A unit of loudness; a simple 1-kHz tone, 40 dB above a listener's threshold, produces a loudness of 1 sone. The loudness of any sound that is judged by the listener to be *n* times that of the 1-sone tone is *n* sones.

Sound. Also called a sound wave. An alteration in pressure, stress, particle displacement, or velocity, propagated in an elastic medium. Also called a *sound sensation.* The auditory sensation evoked by a sound wave.

Sound Absorption. The conversion of sound energy into some other form (usually heat) in passing through a medium or on striking a surface.

Sound Absorption Coefficient. The incident sound energy absorbed by a surface or a medium, expressed as a fraction.

Sound Pressure Level. In dB, 20 times the logarithm of the ratio of the pressure of a sound to the reference pressure, which must be explicitly stated. (Usually, either 2×10^{-4} or 1 dyne per square centimeter.) Also, the pressure of an acoustic wave stated in terms of dynes/square centimeter, or microbars.

Sound Reflection Coefficient. Also called *acoustical reflectivity.* Ratio at which the sound energy reflected from a surface flows on the side of incidence, to the incident rate of flow.

Sound-Reproducing System. A combination of transducers and associated equipment for reproducing prerecorded sound.

Sound Spectrum. The frequency components included within the range of audible sound.

Speaker. An electroacoustic transducer that radiates acoustic power into the air.

Speaker Efficiency: Ratio of the total useful sound radiated from a speaker at any frequency to the electrical power applied to the voice coil.

Speaker Impedance. The rated impedance of the voice coil in a speaker.

Speaker System. A combination of one or more speakers and all associated baffles, horns, and dividing networks used to couple the driving electric circuit and the acoustic medium together.

Speaker Voice Coil. In a moving-coil speaker, the component that is moved back and forth in response to the applied audio voltage.

Specific Acoustic Impedance. Also called *unit-area acoustic impedance.* The complex ratio of sound pressure to particle velocity at a point in a medium.

Specific Acoustic Reactance. The imaginary component of the specific acoustic impedance.

Specific Acoustic Resistance. The real component of the specific acoustic impedance.

Squawker. A midrange speaker.

Standing Waves. Reflected waves that alternately cancel and reinforce at various distances.

Stereophonic Sound. A system wherein sound energy that is picked up by two separated microphones is recorded on separate channels and is then played back through separate channels that drive individual speakers.

Stop. A tab or other switch form that selects and/or mixes various voices and footages in an electronic organ system.

Strength of a Simple Sound Source. The RMS magnitude of the total air flow at the surface of a simple source in cubic meters per second, where a simple source is taken to be a spherical source, the radius of which is small compared with one-sixth wavelength.

Strength of a Sound Source. The maximum instantaneous rate of volume displacement produced by the source when emitting a sinusoidal wave.

String. One of the four basic tone-color groups that simulates orchestral strings.

Stroboscopic Disc. A cardboard or plastic disc with a specialized printed design suitable for checking turntable speed.

Stylus. Same as phonograph needle.

Subharmonic. An integral submultiple of the fundamental frequency in a tonal waveform.

Supertweeter. A speaker designed to reproduce the highest frequencies in the audio range.

Sustain. An effect produced by a note that diminishes in intensity gradually after the key has been released.

Swell Manual. The upper manual of an organ; also termed the *solo manual.*

Synthetic Bass. A method of bass-tone generation that depends on the nonlinear response of the ear; pertinent harmonics are intensified, and although the fundamental is not present, the listener obtains the impression that the tone is complete.

T Pad. A three-element fixed attenuator.

Tablet (Tab). A rocker-type switch control that selects an organ voice or footage.

Take-up Reel. A reel on a tape recorder that winds the tape after it passes the heads.

Tape Deck. A tape unit without a power supply or speaker.

Temperament. A mode of tuning an instrument scale so that successive tones correspond to specified intervals.

Tempered Scale. An arrangement of musical pitches such that successive notes have equal frequency ratios.

Terminal. Electrical connection point.

Tibia. An organ voice that simulates flute tones.

Timbre. Also termed *tone color.* The distinguishing quality of a sound that depends primarily upon harmonic content, and secondarily upon volume.

Tone. The fundamental frequency or pitch of a musical note.

Tone Arm. A pivoted arm on a turntable that houses the pickup cartridge.

Tone Burst. A test signal comprising short sequences of sine-wave energy.

Tone Color. Also termed *timbre.* Classified as *diapason, flute, string,* or *reed.*

Tone Control. A control that provides variation of an amplifier's frequency responses.

Tone Generator. An organ section that generates the basic voice waveforms.

Tracking. The path of a phono stylus within the grooves of a disc.

Transducer. A device that converts one form of energy into another form.

Transient. An electrical surge.

Transient Response. The ability of a speaker to follow sudden changes in signal level.

Tremolo. An amplitude modulation of a tone at a rate of approximately 7 Hz.

Triaxial Speaker. A dynamic speaker unit consisting of three independently driven units combined into a single speaker.

Tuning Fork. A precision source of pitch, usually designed as a U structure, and supported at its nodal point.

Turntable. Same as *record player.*

Tweeter. A speaker designed to reproduce the higher audio frequencies, usually those above 3000 Hz.

Varistor. A voltage-dependent resistor.

Vibrato. A frequency modulation of a tone at a rate of approximately 7 Hz.

Voice. An organ tone of specified timbre.

Volume. Same as *expression.* A relative sound level.

Watt. A power unit, equal to the product of 1 volt and 1 ampere.

Woofer. A speaker designed to reproduce bass tones.

Wow. A form of distortion that occurs when a magnetic tape varies back and forth in speed or a turntable varies similarly in rpm.

Wow-wow. A very slow vibrato effect.

Index

The often-neglected topic of antenna installation is covered in some detail, and the reader is guided in coping with reception problems in fringe areas. Management of audio system interference is also explained, with attention to details of shielding and filtering. Approved techniques for audio wiring systems are detailed and illustrated.

For benefit of the more technically minded reader, the author has included a chapter on audio component checks and tests. Both quick tests and basic instrument applications are described.

HI-FI STEREO INSTALLATION SIMPLIFIED includes a comprehensive glossary and several appendices. It has been cross-indexed for easy reference by readers.

Audio technicians and managers in hi-fi shops will find a large volume of useful reference data in this book. It will be especially appreciated by sales personnel because they will find understandable information which will make them more knowledgeable and more effective in customer relations. Installation personnel will find practical information on "problem" situations, as well as in routine installations. In addition, this handbook is suitable for any technical school that offers audio courses, and, of course, it is invaluable for audiophiles, from beginners to experts.

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ANOTHER BOOK OF INTEREST FROM RESTON . . .

HOW TO BUILD YOUR OWN STEREO SPEAKERS
Construction, Applications, Circuits and Characteristics

CHRISTOPHER ROBIN

This unusually comprehensive "how to" book explains how you can plan, construct, install, adjust, and test a modern stereo speaker system. It discusses the essentials of room acoustics, and how to control acoustical characteristics, so that the speaker system you construct will be correct for its location.

The book is designed to take the audio enthusiast in an orderly and concise manner through each step in the process of building modern speaker enclosures. Dimensioned drawings are provided throughout to eliminate guesswork. Detailed procedural instructions are included as well as hints on how to avoid common errors.

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