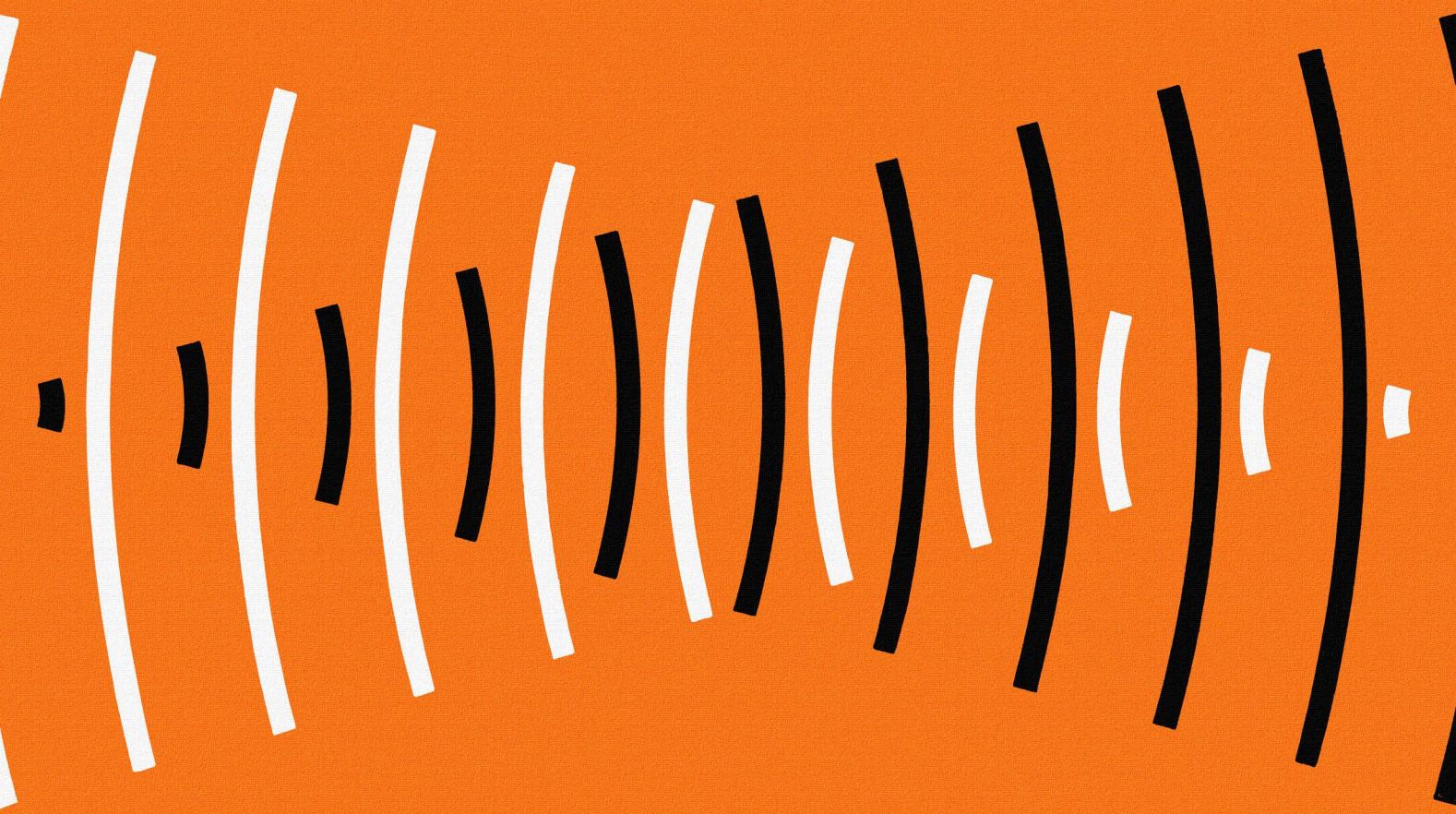


# HI-FI STEREO

## SERVICING GUIDE

by Robert G. Middleton



# **Hi-Fi Stereo Servicing Guide**

by

**Robert G. Middleton**



**HOWARD W. SAMS & CO., INC.**  
**THE BOBBS-MERRILL CO., INC.**  
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# Preface

Hi-fi stereo servicing is a major source of income for shops that are properly equipped and staffed. A reputation for reliability and competence in this field places the technician in a privileged position. Audio enthusiasts are well aware of the fact that true tonality is the chief vehicle of musical meaning; this segment of the home-entertainment market is much more demanding than the TV viewer or the casual radio listener. The dominant characteristics of a hi-fi stereo system are uniform frequency response over the entire audio-frequency range, a low percentage of harmonic and intermodulation distortion, and ample stereo separation. The acoustic properties of a hi-fi installation are also of basic concern. In turn, a working knowledge of hi-fi circuitry and troubleshooting procedures must be augmented by full understanding of audio test and measurement procedures, and a practical grounding in the principles of acoustics.

Hi-fi technicians often tend to specialize, because the field is so extensive. One who aspires to general practice in the hi-fi stereo field must acquire competence in troubleshooting of fm and a-m tuners, multiplex adapters, amplifiers, tape recorders, and record players, and must know the fundamentals of speaker repair. He must be fully familiar with the operation of audio oscillators, signal generators, sweep generators, stereo-multiplex generators, oscilloscopes, harmonic-distortion meters, and intermodulation analyzers. It is also very helpful to understand the fundamentals of square-wave generator application. The keystone of a professional reputation is the ability to provide critical customers with quantitative evaluation of frequency response, percentage of distortion, and dB separation.

This book covers all components of the hi-fi stereo system except record players and tape recorders, which are the subject of companion volumes in this series. The topics treated in this text include the troubleshooting of equipment (a-m tuners, fm tuners, stereo multiplex units, and audio amplifiers), installation of hi-fi speakers, system evaluation, and test-equipment performance verification. It is assumed that the reader is familiar with the basics of a-m and fm radios, and with the operation of basic electronic test equipment such as the VOM, the VTVM, and the oscilloscope. The book is functionally divided according to sections in the hi-fi system, and is further divided according to symptoms. It has been organized to serve as a ready reference to a particular part of the system when a given trouble symptom occurs.

It is my firm belief that this hi-fi stereo servicing guide will be a valuable addition to the libraries of all audio technicians.

ROBERT G. MIDDLETON

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## Chapter 1

# A-M Tuner Troubles

Although no industry standards have been established, it is generally agreed that high-fidelity reproduction entails a frequency response that is flat within  $\pm 1$  dB from 20 Hz to 20 kHz, approximately, and with a harmonic-distortion figure of less than 1 percent at maximum rated power output. We will find that harmonic-distortion (HD) and intermodulation-distortion (IM) values usually have the same order of magnitude. However, there are certain types of system defects that cause IM distortion to increase much more rapidly than harmonic distortion as the power output is increased. Various types of distortion and pertinent measurement techniques are explained subsequently. An a-m tuner is a refined version of a conventional a-m radio receiver, but without an audio amplifier and speaker. A hi-fi a-m tuner differs from an ordinary a-m tuner primarily in its greater bandwidth; a hi-fi unit also has certain minor refinements. In most cases, a hi-fi a-m tuner is provided as a section of a hi-fi am/fm receiver (see Fig. 1-1). Few hi-fi a-m tuners are individually merchandised because there is seldom more than one hi-fi a-m broadcast station in a given locality.

Most current-model hi-fi a-m tuners are solid state, although many tube-type tuners are still in use. Solid-state design is usually on a transistor-for-tube basis, except for systems employing IC packages. Printed-circuit construction is generally utilized. The superheterodyne configuration is universal, although design details vary considerably. A typical arrangement utilizes four transistors and a diode detector; two i-f stages are used, with no rf amplification (no preselection). In economy-type designs, we may find that a single transistor does double duty by operating as an oscillator and converter. Integrated circuitry is used in the i-f section by some manufacturers. De-

lux e a-m tuners are provided with at least one stage of preselection to improve selectivity and image rejection. Common trouble symptoms caused by defects in hi-fi a-m tuners are as follows:

1. Dead tuner.
2. Weak output.
3. Distorted output.
4. Incorrect dial indication.
5. Intermittent operation.
6. External interference.
7. Poor selectivity.
8. Drifting off-frequency.

### GENERAL DISCUSSION

A block diagram of a typical a-m tuner is shown in Fig. 1-2. This arrangement employs a preselector stage (rf amplifier), a converter, an i-f stage, and a diode detector. The tuning range is from 530 to 1620 kHz; no short-wave bands are provided. Note that a converter stage has a single transistor (or tube) that

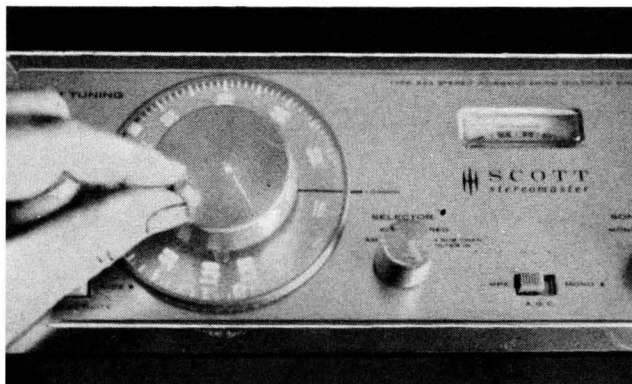


Fig. 1-1. Typical hi-fi am/fm tuner.

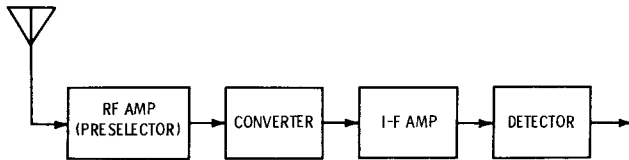


Fig. 1-2. Block diagram of a typical a-m tuner.

does double duty as an oscillator and mixer. On the other hand, some a-m tuners use two transistors in this section—one transistor operates as an oscillator, and the other operates as a mixer. An external antenna is utilized in the arrangement of Fig. 1-2. This circuit is resonated by means of a tuned loopstick; an external antenna is not required if the prevailing signal strength is comparatively high. The gain of this type of tuner is not as great as in designs that have two i-f stages and no preselector stage. On the other hand, a preselector stage minimizes the possibility of image interference.

Another arrangement for an a-m tuner is shown in Fig. 1-3. It includes two transistors, an integrated circuit, and a diode detector. No preselector is provided, and the converter transistor is driven directly by a loopstick antenna. Since ample system gain is available, no provision is made for connection of an external antenna. The second i-f stage employs an integrated circuit instead of a conventional transistor. An integrated circuit (IC) includes semicon-

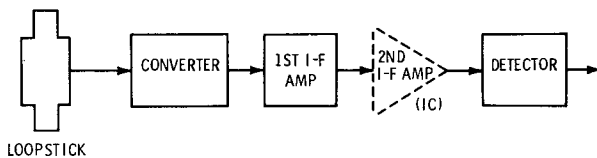


Fig. 1-3. A-m tuner with an IC i-f amplifier stage.

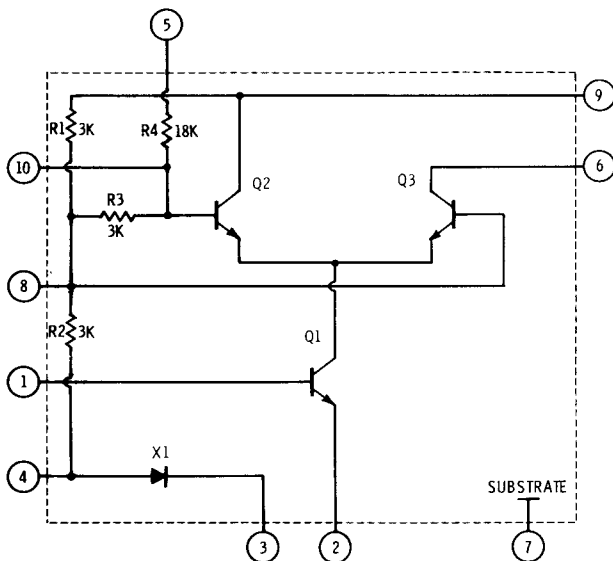


Fig. 1-4. Integrated amplifier package.

ductor devices and resistors, as exemplified in Fig. 1-4. An IC package may also include fixed capacitors; we will find that all i-f IC packages are connected externally to tuned coils or transformers. A diode detector in an a-m tuner often does double duty as a signal detector and agc rectifier; in de luxe designs, however, a separate agc rectifier diode is provided.

Fig. 1-5 shows a block diagram for an elaborate a-m tuner. This arrangement employs five transistors, a diode detector, and a diode agc rectifier. Reception is provided by a built-in loop, which may be supplemented by an external antenna, if desired. The incoming signal is stepped up by a preselector stage, followed by a mixer-oscillator section for frequency conversion. Note that the first i-f stage is agc-controlled; this transistor does double duty in that it provides i-f amplification and also operates as an amplifier for the agc current. Thus, amplified agc current is applied to the rf-amplifier transistor. The mixer transistor is biased by agc current that has been amplified through the rf-amplifier transistor.

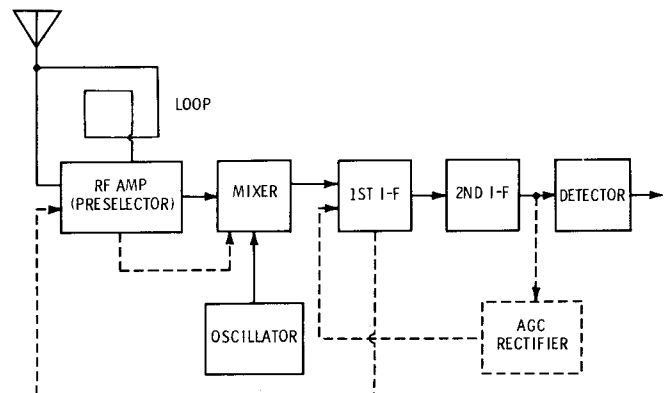


Fig. 1-5. Block diagram of an elaborate a-m tuner.

### Transistor Versus Tube-Type Tuners

Transistors provide several advantages in high-fidelity systems, among which we may note their small size, comparatively low power consumption, reliability, and absence of microphonics. Transistors are basically low-impedance devices and are relatively immune to pickup of stray fields from power supplies or other ac circuitry. Since a transistor has no filament or heater, it cannot develop internal hum. Fig. 1-6 shows a schematic for a transistor a-m tuner. It consists of an rf amplifier, local oscillator, mixer, two stages of i-f amplification, and a diode detector. Agc bias current is applied to the bases of the rf amplifier, mixer, and first i-f transistors. This agc bias current is introduced at the base of the first i-f transistor; in turn, stepped-up agc bias current is fed from the emitter of the first i-f transistor through a switching section to the base of the rf amplifier



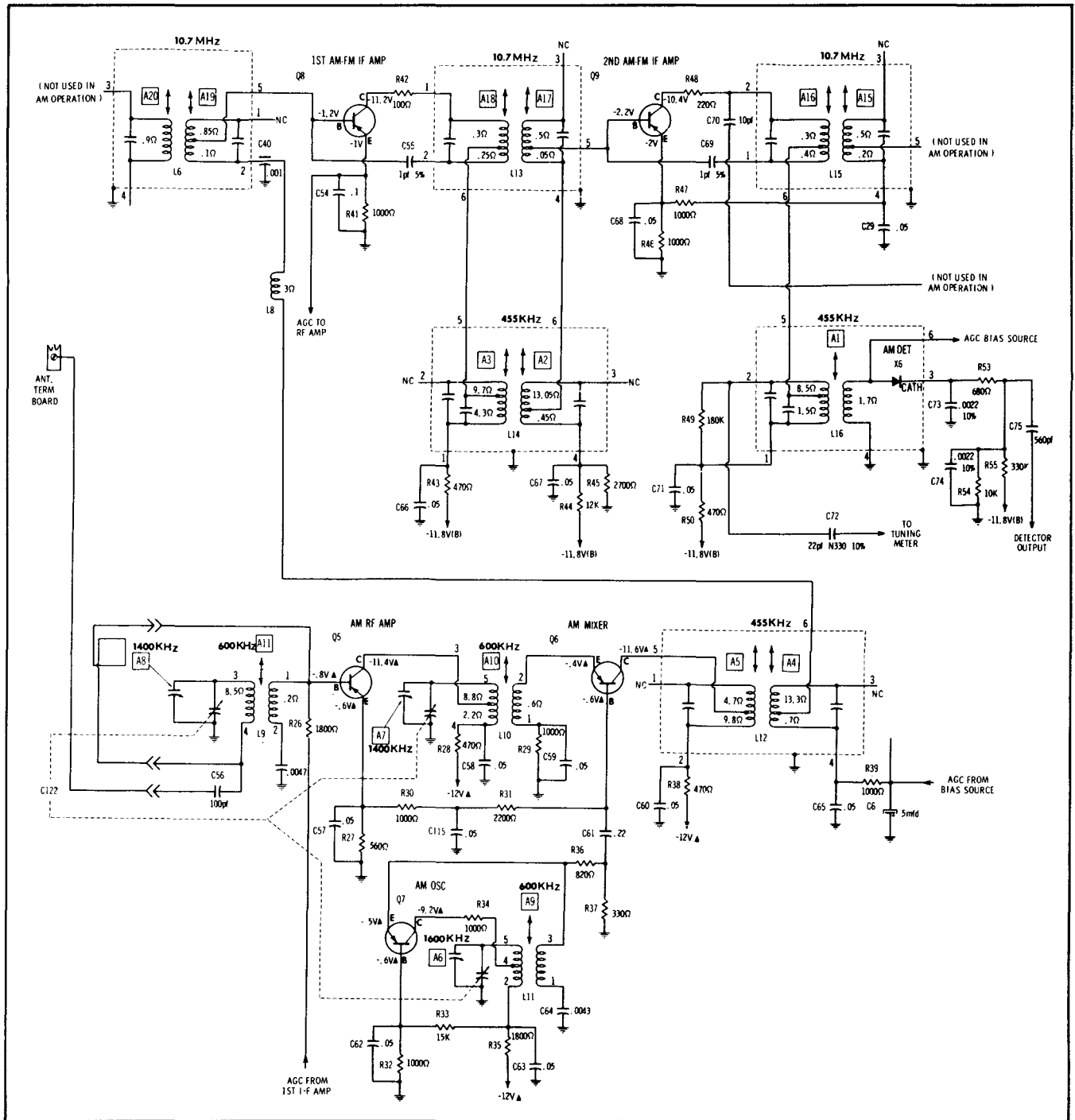


Fig. 1-6. Partial schematic of a transistor a-m tuner.

transistor. Stepped-up agc bias current is then applied from the emitter of the rf amplifier transistor to the base of the mixer transistor.

The second i-f transistor in Fig. 1-6 operates constantly at maximum gain. As is customary in combination receivers, a-m and fm transformers are operated in series (see L13/L14 and L15/L16). This series mode of operation is feasible because the in-

termediate frequencies are widely different in a-m and fm operation. Thus, the impedance of a 10.7-MHz winding is so small in 455-kHz operation that the winding "looks like" a short-circuit. The transistors operate in the common-emitter (CE) configuration, with the exception of the oscillator and mixer transistors, which operate in the common-base (CB) configuration. Note that so far as dc current amplifica-

# Hi-Fi Stereo Servicing Guide

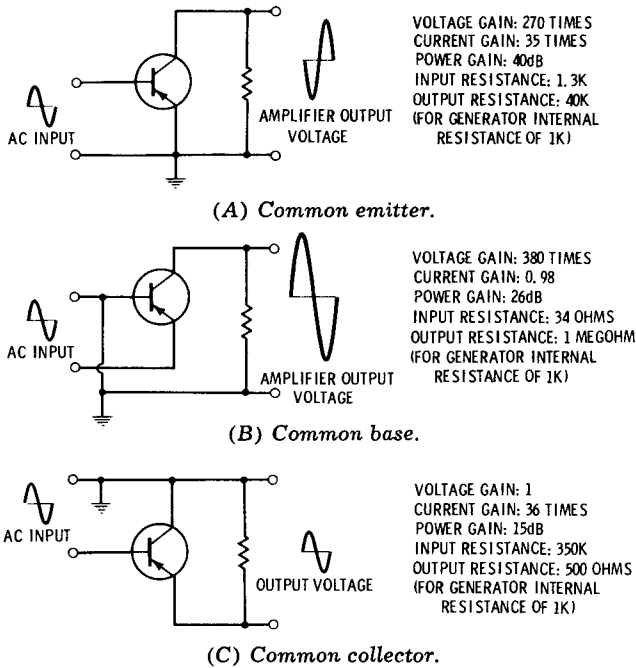


Fig. 1-7. Transistor circuit parameters.

tion in the agc system is concerned, the first i-f transistor and the rf-amplifier transistor operate in the common-collector (CC) configuration. Fig. 1-7 shows the basic configurations and their relative performance characteristics.

In addition to junction transistors and IC packages, we also find field effect transistors (FETs) in some a-m tuners, as exemplified in Fig. 1-8. An FET is similar to a vacuum tube in that it has an extremely high input impedance. As shown in Fig. 1-9, a field-effect transistor has three electrodes, called the *drain*, the *gate*, and the *source*; these are comparable respectively to the plate, grid, and cathode of a vacuum tube. An n-channel junction FET is exemplified in Fig. 1-9; a positive voltage is applied to the drain, and a negative control voltage is applied to the gate. When the gate voltage is zero, there is current from the source to the drain, as shown in Fig. 1-10. However, an increasing negative voltage on the gate reduces current because of the restricting action of the depletion layer. In normal operation, the gate is reverse-biased, so that the operation is

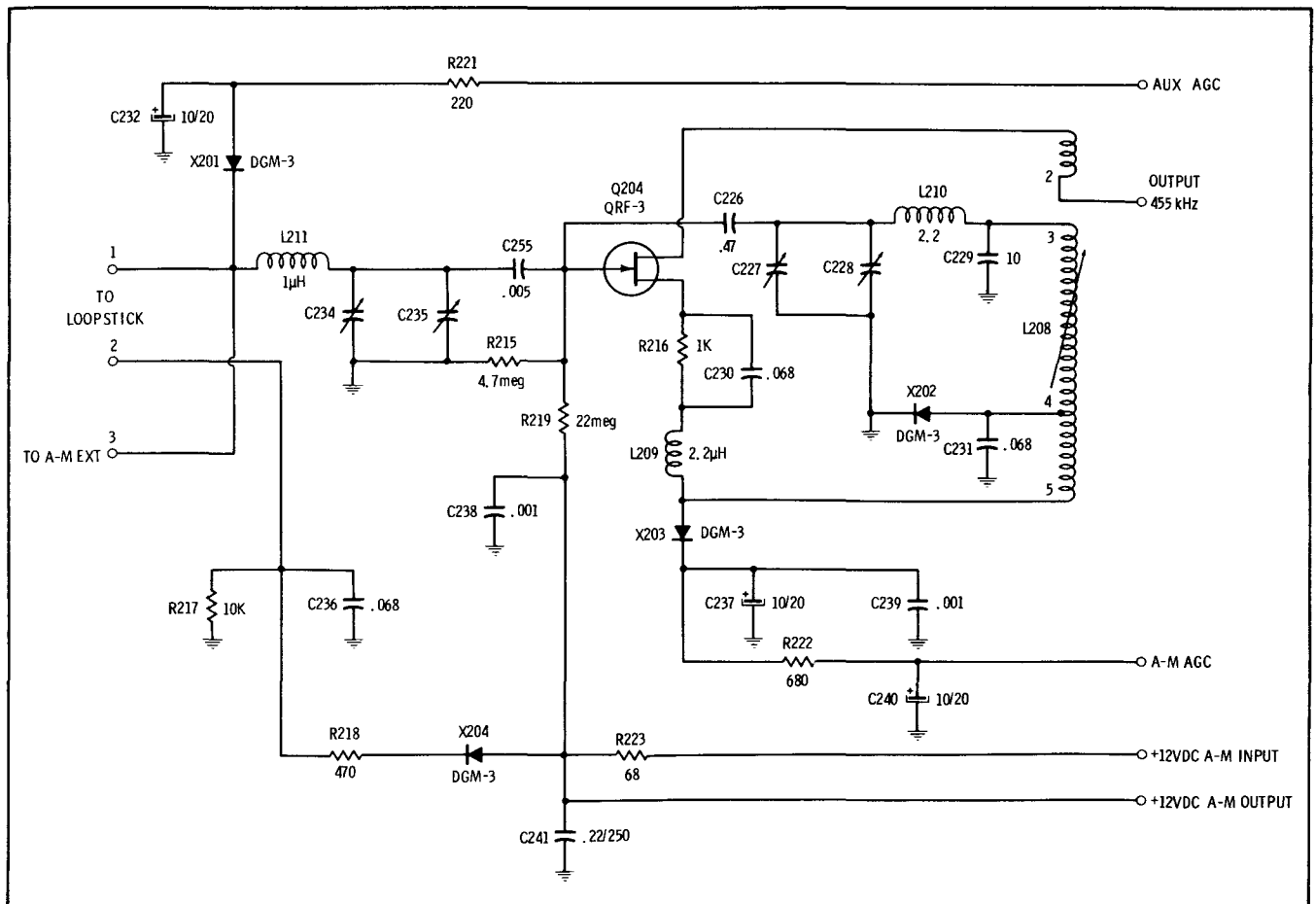


Fig. 1-8. Input section of an a-m tuner using an FET.

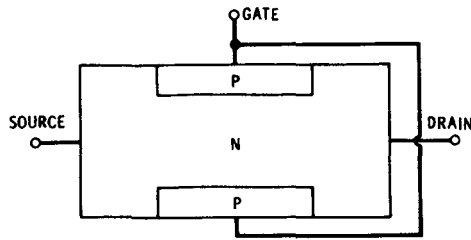
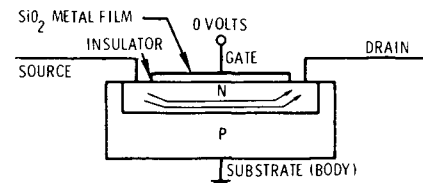


Fig. 1-9. N-channel junction FET.

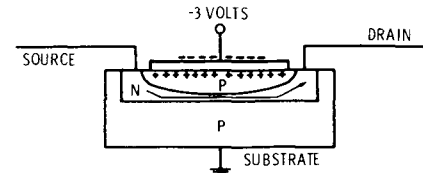
quite similar to that of a triode tube. A p-channel FET operates in the same manner, except that the bias voltages are reversed. This is the same distinction that is observed in pnp and npn junction transistors.

Another type of FET, called the MOSFET, is designed so that at zero gate voltage, drain-current conduction either will or will not occur. If a MOSFET permits current at zero gate voltage, it is called a *depletion* type; on the other hand, if current is stopped at zero gate voltage, it is called an *enhancement* type. Fig. 1-11 shows the cross-section of a depletion-mode MOSFET (n-channel version). Note that it has an additional connection called the substrate, which is usually connected to ground. "Channel" terminology derives from the fact that when the device is conducting, a channel of n-type material occurs between the n-type drain and the source. The basis of MOSFET operation is the capacitance formed by the metal gate, the silicon-dioxide insulator, and the silicon semiconductor.

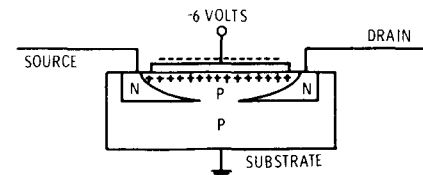
Since an ideal capacitor conducts no direct current, the gate electrode has zero leakage current whether



(A) Gate voltage 0 volts.



(B) Gate voltage -3 volts.



(C) Gate voltage -6 volts.

Fig. 1-11. N-channel depletion MOSFET.

a positive or a negative signal voltage is applied. On the other hand, the grid of a triode tube draws current when driven positive. When the gate voltage is zero, as shown in Fig. 1-11A, the existing n-channel conducts electrons from the source to the drain. If a negative gate voltage is applied, the MOS capacitor charges. This causes holes to be drawn toward electrons on the gate, thus forcing electrons out of the n-channel, as shown in Fig. 1-11B. Because the channel is now smaller, there is less current from source to drain. If sufficient negative voltage is applied to the gate, enough electrons are forced out of the channel to produce pinch-off, as shown in Fig. 1-11C. An npn junction is formed, but it does not conduct current because the pn junction is reverse-biased:

The enhancement type of MOSFET (Fig. 1-12) operates in an opposite manner. When the gate voltage is zero, there is no current-conducting channel. On the other hand, when the gate is positive, electrons are drawn toward the gate, which sets up an n-channel. There is then current from source to drain in accordance with the value of positive gate voltage that is applied. Since the input impedance to the gate is extremely high, static electricity from strong stray fields can build up enough voltage to damage an FET when it is not connected into a circuit. Therefore, the leads are usually short-circuited until the FET is connected into its circuit. Most hi-fi assemblies employ printed or plated circuits; a spectacle loupe such as is illustrated in Fig. 1-13 is helpful for inspecting connections, and also in localizing defects such as cracks in PC conductors.

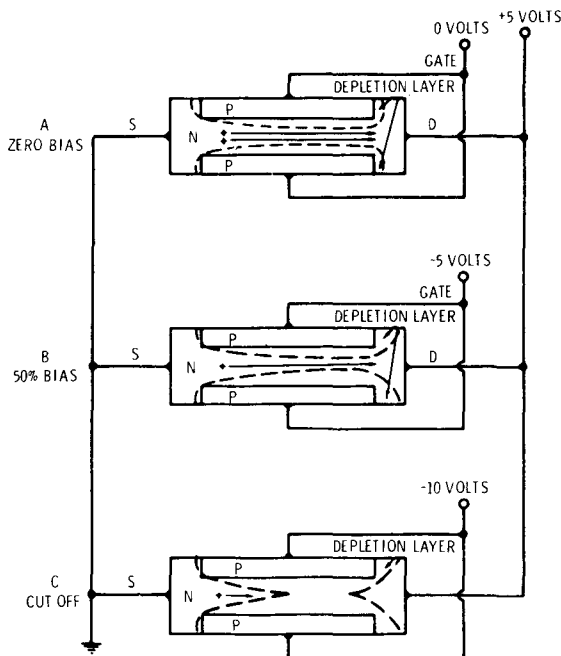
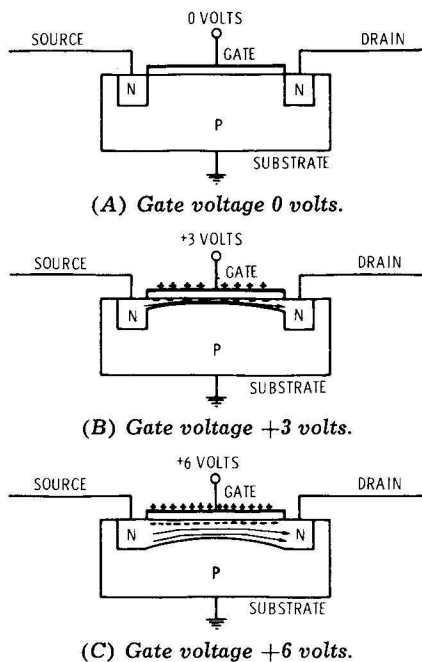
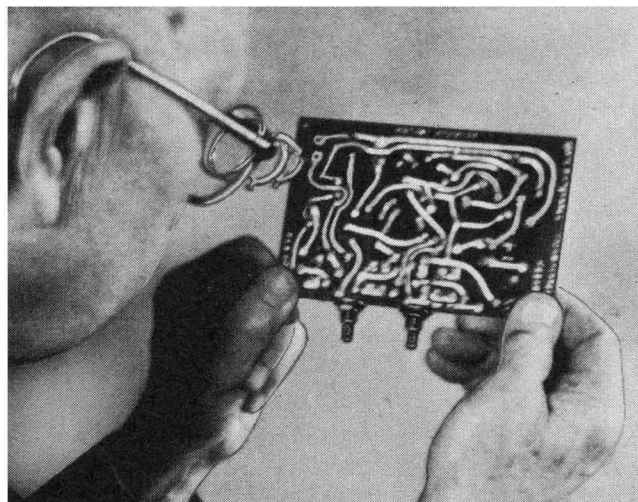


Fig. 1-10. Effect of bias on an FET.



**Fig. 1-12. N-channel enhancement MOSFET.**

Next, let us consider the tube-type a-m tuner configuration shown in Fig. 1-14. This arrangement employs a stage of preselection (rf amplifier), a converter, a stage of i-f amplification, and a diode detector. A bandwidth (bandpass) switch is provided, which operates in the rf-amplifier input and output circuits. This switch provides a choice of an audio-frequency bandwidth of either 7 kHz or 14 kHz. Although an af range of 7 kHz does not provide hi-fi response, narrow-band reception is desirable for tuning in weak or distant stations with a minimum of interference. On the other hand, an af bandwidth of 14 kHz provides the maximum available fidelity from hi-fi a-m broadcast stations. When wide-band



**Fig. 1-13. Spectacle loupe aids inspection of PC boards.**

reception is utilized on conventional a-m station signals, a 10 kHz beat note is developed; this beat note results from the fact that the stations have carrier allocations at 10-kHz intervals. Such 10-kHz beat notes are trapped by a high-Q filter (see T5-C7 in Fig. 1-14).

The rf bandpass in Fig. 1-14 is controlled by switching resistance into or out of the tuned circuits. Thus, a 27-ohm resistor, R204, can be switched into the rf grid-input circuit; also, an 18-ohm resistor, R210, can be effectively switched (unbypassed) in the rf plate-output circuit. Note that an af bandwidth of 7 kHz corresponds to an rf bandpass of 14 kHz, and an af bandwidth of 14 kHz corresponds to an rf bandpass of 28 kHz. In other words, the rf amplifier processes both upper and lower sidebands in an rf signal. We measure bandpass values between the 70.7 percent amplitude points on the frequency-response curve, as shown in Fig. 1-15. The way in which signal voltage and rf bandpass change with values of circuit resistance is seen in Fig. 1-16 for a basic example. A leaky capacitor can change the circuit resistance.

Technician apprentices often suppose that the selectivity of a tuner depends entirely upon its bandpass. However, selectivity also depends to an appreciable extent upon the steepness of the frequency-response curve. For example, if the bandpass is correct, but a tuner has a 50 percent response at the carrier frequency of an adjacent-channel station, it is obvious that objectionable interference will result. An ideal response curve would be rectangular (see Fig. 1-17). Although the ideal response cannot be obtained in practice, it can be approximated by suitable tuner design and correct alignment of the tuned circuits. The use of a double-tuned transformer with suitable coupling provides a double-humped response that approaches a flat-topped response, as seen in Fig. 1-18. The comparative steepness of response curves for a single tuned circuit and a tuned transformer (two identical, coupled, tuned circuits) is shown in Fig. 1-19. Troubles in tuned circuits are analyzed with the aid of a signal generator.

## Automatic Volume Control

Automatic volume or gain control (avc or agc) denotes an arrangement for maintaining the output from an a-m tuner at a comparatively constant level, in spite of variation in strength of an incoming signal. The basic agc circuit comprises a signal rectifier and low-pass filter; this signal rectifier is very often a diode that does double duty as a detector. An RC low-pass filter is always used because of its economy and because it serves the purpose as effectively as an

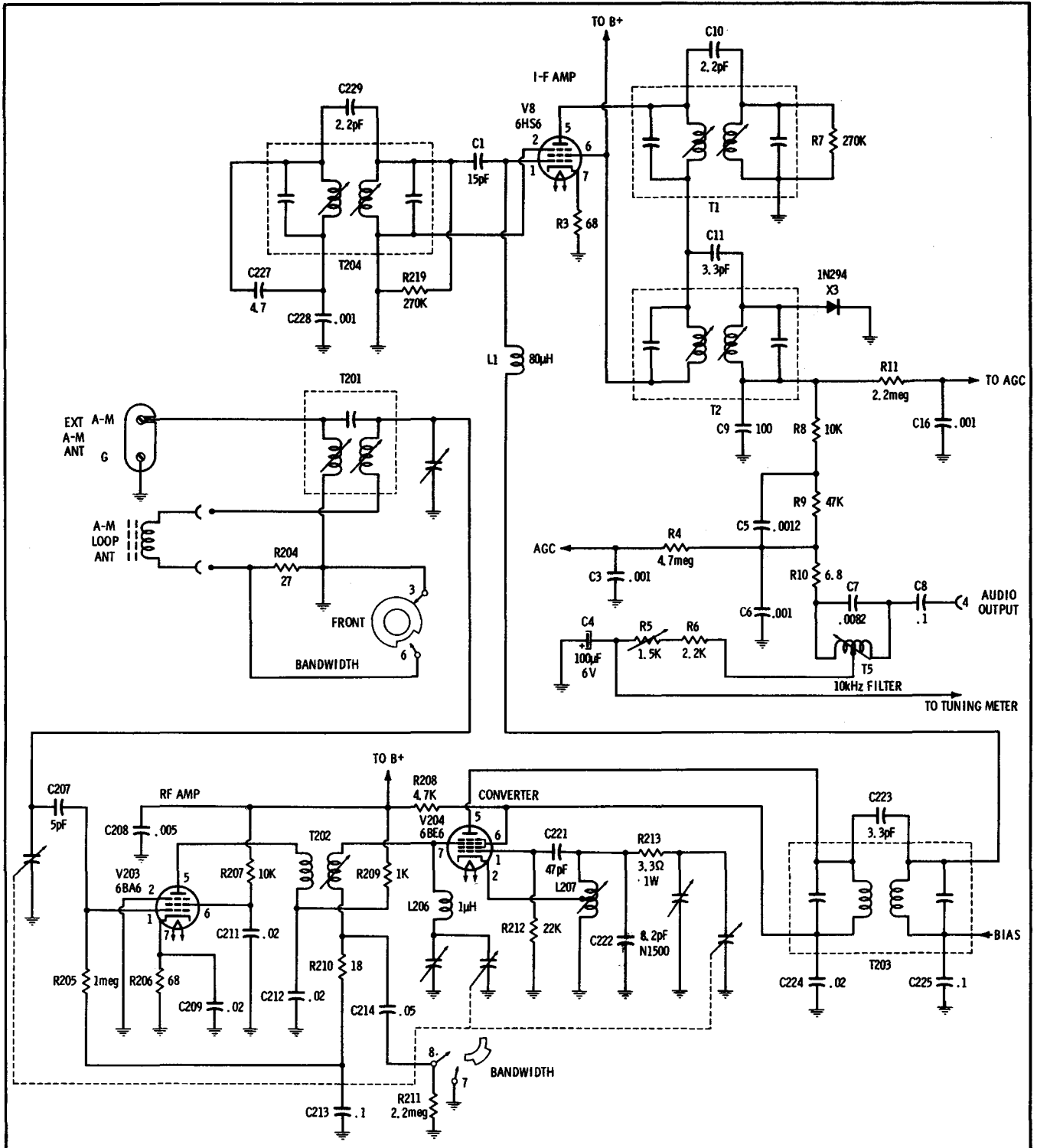


Fig. 1-14. Tube-type a-m tuner configuration.

LR or LCR filter. The output from this low-pass filter is a dc voltage (or dc current), which is applied to the grid of a controlled tube, or to the base of a transistor. Functionally, the low-pass filter permits the dc output to change level when the incom-

ing signal varies in strength, but prevents the dc output from changing as the modulation envelope of the signal rises and falls. That is, the time constant of the RC filter is chosen sufficiently long that the filter cannot respond to the short rise-and-fall

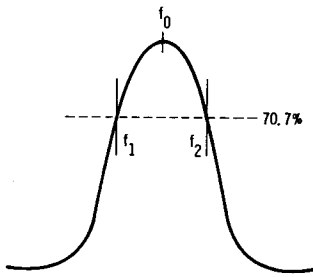


Fig. 1-15. Bandpass is defined as  $f_2 - f_1$  hertz.

intervals in the modulation envelope. Defects in agc filter components produce distinctive trouble symptoms.

An agc circuit normally supplies zero current to a controlled tube, since a grid is effectively an open circuit. Thus, the gain of the tube is controlled by the value of the agc voltage. Reverse agc is always utilized; in other words, the agc voltage becomes more negative as the signal strength increases. On

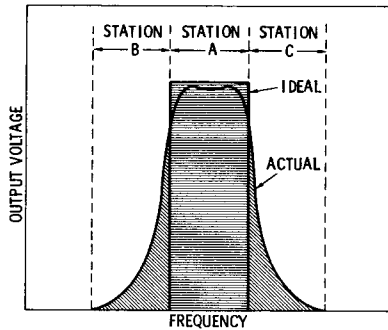


Fig. 1-17. Selectivity depends on the steepness of the frequency-response curve.

the other hand, the gain of a transistor is controlled primarily by variation of bias current; that is, the base of a transistor draws dc current from the agc line. Since the input resistance of the base is very low, a very small change in voltage produces a large change in current; this is one of the reasons that we characterize a transistor as a current-operated device. Also, any change in base current is multiplied by the beta value of the transistor in the collector circuit. This is another one of the reasons for regarding a transistor as a current-operated device.

Transistors differ from tubes in that they may be controlled either by reverse agc, or by forward agc. When reverse agc is used, an increase in signal strength results in reduction of the base current. However, when forward agc is used, an increase in signal strength results in an increase of base current. Some types of transistors are controlled to best advantage by reverse agc; others provide better control with forward agc. In either case, the designer employs the mode that gives the more gradual cutoff characteristic. A sharp cutoff characteristic is undesirable because it curves rapidly and may give rise to cross-modulation interference. Whenever forward

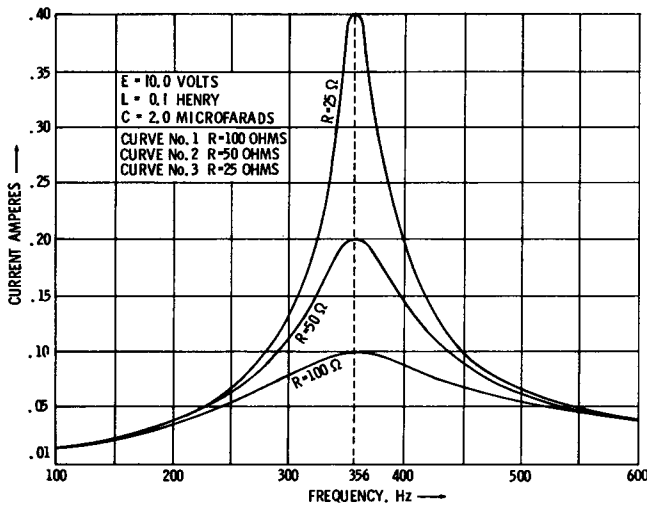
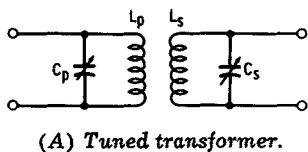
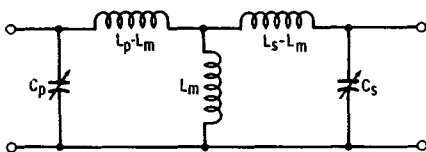


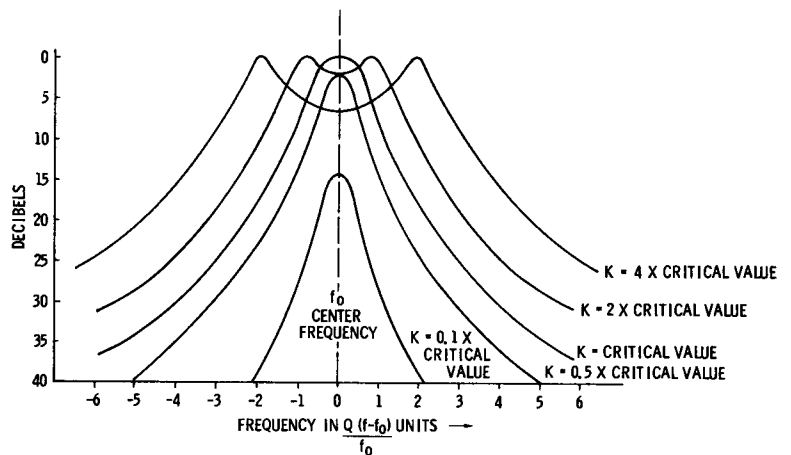
Fig. 1-16. Three frequency-response curves for an LCR series-resonant circuit.



(A) Tuned transformer.



(B) Equivalent circuit.



(C) Frequency-response curves for various coefficients of coupling ( $k$ ).

Fig. 1-18. Increased bandwidth is obtained by coupling two tuned circuits.

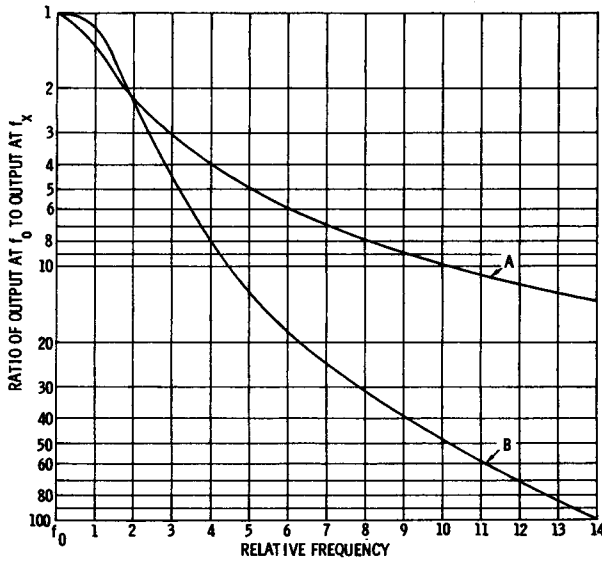
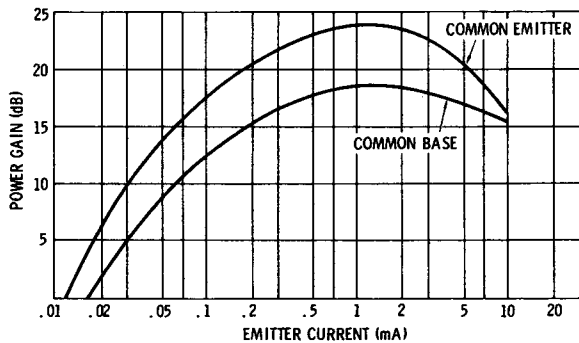
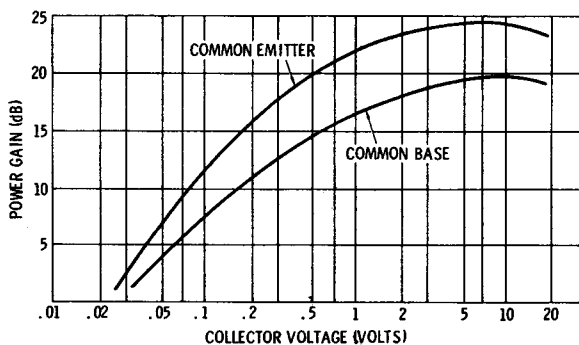


Fig. 1-19. Skirt of frequency-response curve.

agc is used, we will find a substantial amount of resistance present in the collector circuit. This series resistance is required because forward-agc action entails transistor saturation, and saturation takes place at very low collector voltage. In other words, when the base current is very high, the collector current is also very high. In turn, a large voltage drop is developed across the series resistance in the



(A) Reverse-agc curve.

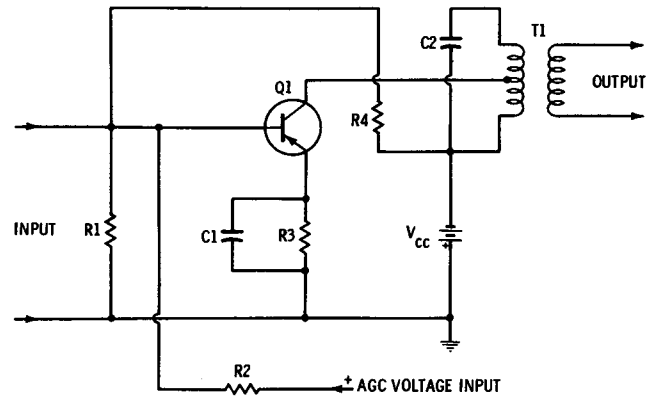


(B) Forward-agc curve.

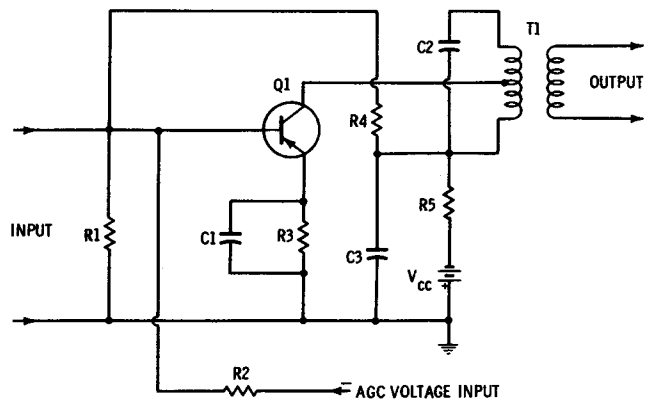
Fig. 1-20. Typical agc characteristic curves.

collector circuit, causing the collector-emitter voltage to fall to a very low value. Normal agc characteristics are shown in Fig. 1-20.

Basic examples of reverse-agc and forward-agc arrangements are shown in Fig. 1-21. Note that a series resistor, R5, is included in the collector circuit of the forward-agc arrangement. In Fig. 1-22, we see an i-f stage with a detector that does double duty as an agc rectifier. The agc current is filtered by R1



(A) Reverse agc.



(B) Forward agc.

Fig. 1-21. Schematics of agc circuits.

and  $C2$ ; as the signal strength increases, the base current decreases—that is, reverse agc is employed in this example. Conventional transistors are triode devices, and many of them require neutralization to avoid instability due to regeneration or oscillation. Thus, signal voltage is fed back in reverse phase to the base of  $Q1$  from  $T2$  via neutralizing capacitor  $C5$ . It is easy to confuse neutralizing trouble with agc trouble, because defects in a neutralizing circuit are associated with large changes in gain. To anticipate subsequent discussion, agc trouble causes little or no change in bandpass, whereas neutralizing trouble causes a large change in bandpass.

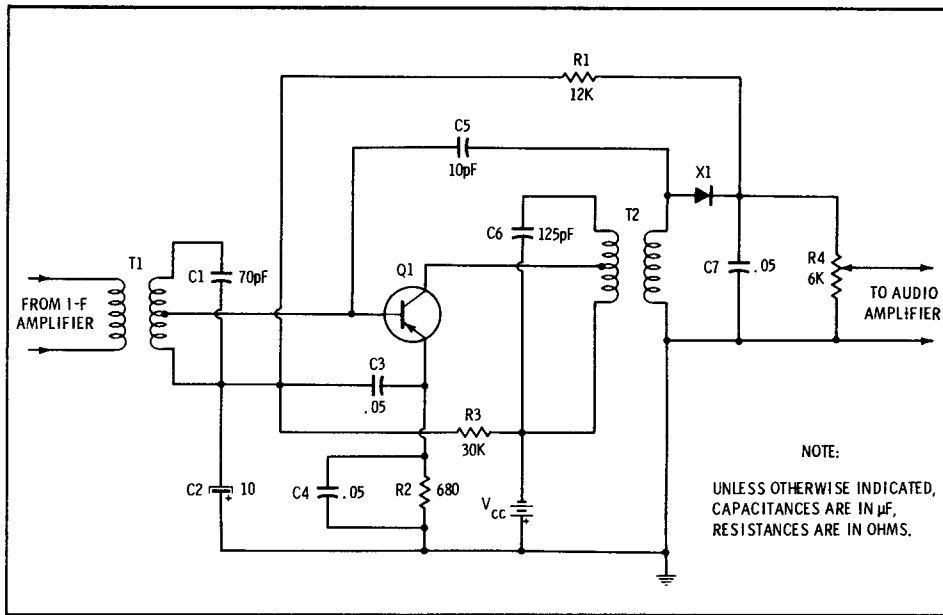


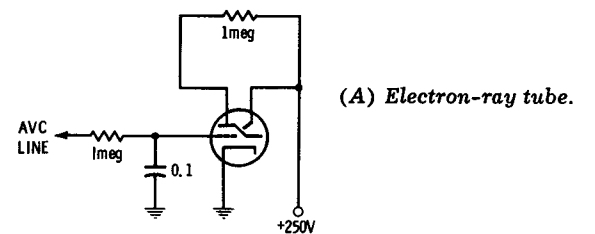
Fig. 1-22. I-f stage and detector, with agc and neutralizing circuits.

**Tuning Indicators**

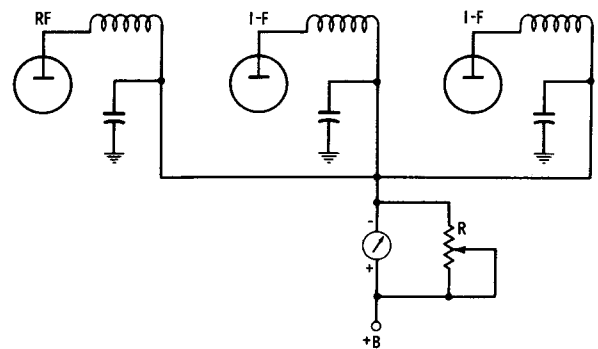
A tuning indicator is provided on nearly all hi-fi a-m tuners; it indicates when the unit is tuned exactly to a station carrier. Thus, a tuning indicator assists nontechnical persons in avoiding distortion due to sideband cutting. Technicians also find tuning indicators useful during alignment procedures. A tuning indicator may be some type of electron-ray tube, or a panel meter. We find electron-ray tubes mounted near the tuning scale, or alternatively, a miniature electron-ray tube may do double duty as a pointer on the tuning scale. An electron-ray (eye) tube is ordinarily energized from the agc line; occasionally, a stage of dc amplification may be provided for optimum sensitivity on weak signals. Tuning meters are also energized from the agc line, although the meter is occasionally connected into the plate or screen circuit of an agc-controlled tube (see Fig. 1-23). If a zero-center or zero-left reference is utilized, the meter is connected into a bridge circuit with a zero-adjustment potentiometer.

**Signal Sources for Troubleshooting**

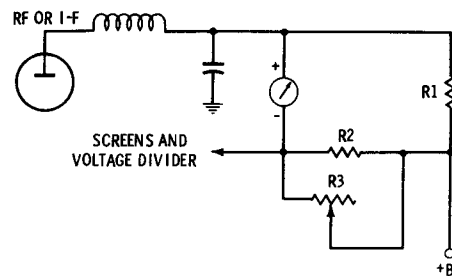
Preliminary troubleshooting of an a-m tuner often requires the localization of a dead stage. This can be done by means of either signal-tracing or signal-substitution techniques. In either method, it is desirable to have a controllable source of modulated rf signal. An economy-type a-m generator is suitable, inasmuch as we are not concerned with accurate frequency or signal-level measurements. Even a simple noise generator or harmonic oscillator will serve the purpose. If signal injection is utilized,



(A) Electron-ray tube.



(B) Meter in plate circuit.



(C) Meter in bridge configuration.

Fig. 1-23. Tuning-indicator arrangements.



the tuning meter will serve as an indicator; however, if the a-m tuner does not have a tuning meter, we can connect a dc voltmeter to the agc line. Signal injection is started at the detector input, and proceeds back step by step until the dead stage is located. Note that a dead local oscillator does not show up in a test made with a noise generator or harmonic oscillator. For this reason, tests with an a-m generator are sometimes more conclusive. In signal-tracing tests, we apply the signal at the antenna-input terminals, and check its progress through the a-m tuner step by step with a suitable indicator. An oscilloscope used with a low-capacitance probe is the most useful type of indicator.

Signal-tracing and signal-substitution tests will sometimes pinpoint a defective component—however, this type of test often serves only to localize the defective section. Dc voltage measurements are usually made to close in on the defective component. Resistance measurements and continuity checks are also helpful in various situations. After a defective component is replaced, and the hi-fi tuner is apparently operating normally, two basic proof-of-performance tests should then be made. First, the frequency response can be checked as indicated in Fig. 1-24. An accurate test requires a good-quality signal generator capable of hi-fi external modulation; a good-quality audio oscillator is also required that has good waveform and flat output over its entire range. If the a-m tuner has true hi-fi frequency response, the ac VTVM reading will not vary by more than 1 dB over the range from 20 Hz to 20 kHz. It is customary to use 30 percent modulation in this test.

Unsatisfactory frequency response can be caused by several types of defects. For example, one or more of the rf or i-f tuned circuits may be misaligned. In such a case, careful realignment in accordance with the tuner service data will correct the frequency response. However, capacitor defects can cause poor frequency response also, although the alignment is correct. For example, a defective neutralizing or bypass capacitor can seriously affect the frequency response. Again, if the detector load resistor increases in value, the high-frequency response will be impaired. In tube-type tuners, a small coupling capacitance may be provided between primary and second-

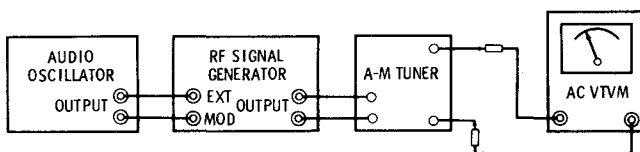


Fig. 1-24. Check of frequency response.

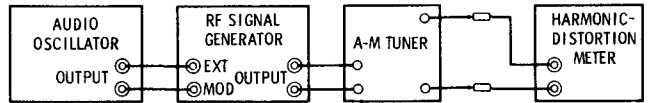


Fig. 1-25. Check of harmonic distortion.

ary of the i-f transformers (see C223 and C11 in Fig. 1-14). If one of these capacitors becomes defective, the bandwidth will be seriously impaired.

Next, we should note that a hi-fi a-m tuner requires a separate test for harmonic distortion. In other words, frequency response is not related to harmonic distortion. Harmonic distortion can be measured with the test setup shown in Fig. 1-25. The basic test is made with the audio oscillator set to a frequency of 1 kHz, and with an output level that modulates the signal generator 30 percent. A fairly high output is customarily used from the signal generator, such as 100,000  $\mu$ V. If the a-m tuner is operating properly, its harmonic distortion will be well below 1 percent. Note that if your rf signal generator does not have a meter to indicate modulation percentage, you can feed the generator output into a scope, and evaluate the screen pattern, as exemplified in Fig. 1-26. Since the ratio of B - A to B + A is equal to 0.65 in this example, the modulation percentage is 65 percent. A harmonic-distortion meter is illustrated in Fig. 1-29.

In case a harmonic-distortion test shows that objectionable distortion is present, we look for defects that can cause nonlinear operation. Incorrect bias voltages are suspected first, and are often caused by leaky capacitors. Defective transistors can also cause excessive harmonic distortion. In most situations, abnormal terminal voltages are produced by transistor defects. However, if the normal dc voltage values are not known, an in-circuit transistor tester will give a useful preliminary test (see Fig. 1-27). In many cases, a simple control-action check can be made with a voltmeter to determine if a transistor is in good operating condition. With reference to Fig. 1-28, a resistor is present in series with the collector circuit.

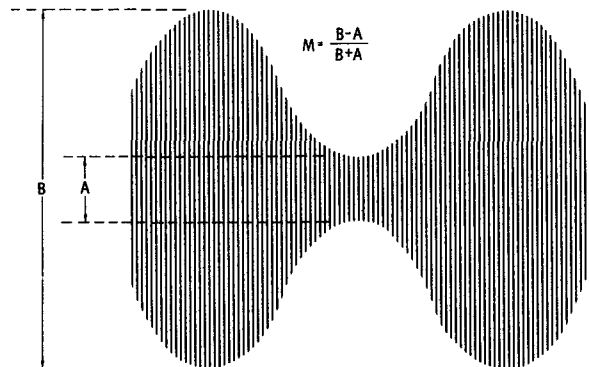


Fig. 1-26. Percent of modulation.

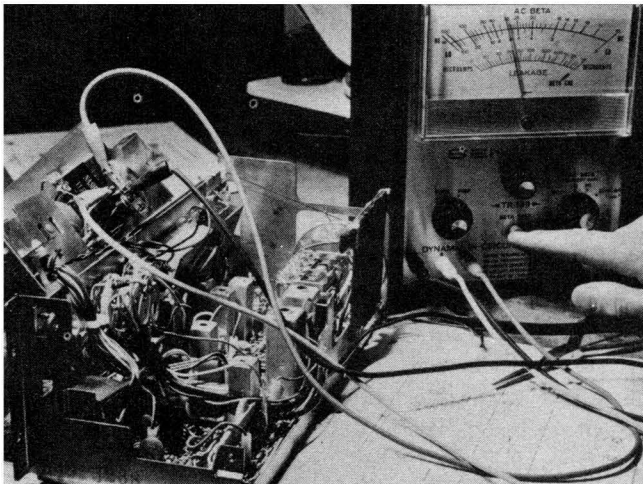
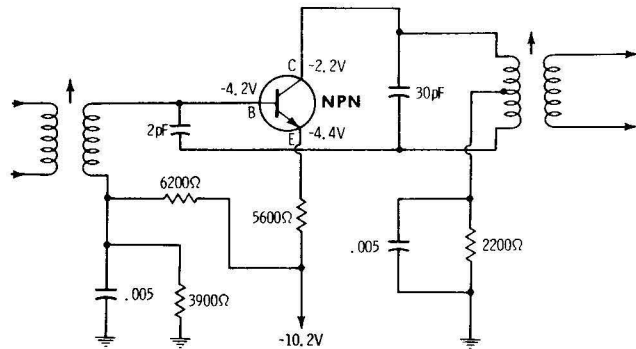


Fig. 1-27. Checking beta and leakage with an in-circuit transistor tester.

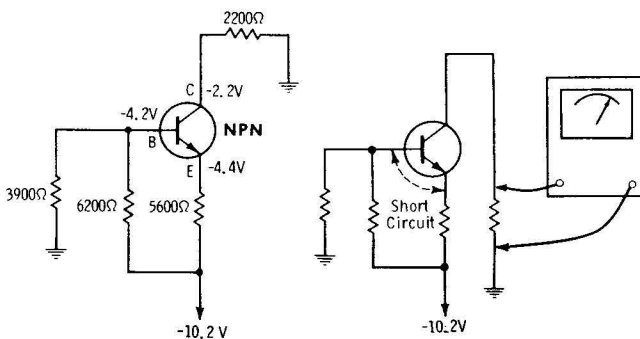
We connect a dc voltmeter across the collector resistor, and short-circuit the base and emitter terminals. If control action is normal, the voltmeter reading drops to zero.

**TROUBLESHOOTING PROCEDURE**

As noted previously, the technician is usually concerned with an am/fm stereo combination. Fig. 1-30 shows a block diagram of a typical chassis. If a-m



(A) With collector dropping resistor.



(B) Equivalent dc circuit.

(C) Control-action test.

Fig. 1-28. Transistor quick-check with voltmeter.

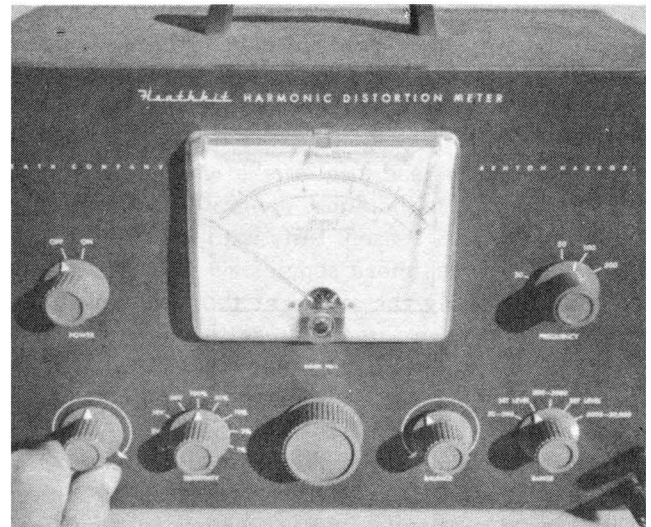


Fig. 1-29. Harmonic-distortion meter.

reproduction is defective, but fm reproduction is normal, we know that the trouble will be found in the a-m converter or detector section. On the other hand, if both a-m and fm reception are defective, the trouble will be found in the i-f or audio sections. In case the phono operation is also defective, we know that the trouble is located in the audio section. However, if phono operation is normal, with both a-m and fm operation faulty, it is evident that the trouble is located in the first or second i-f section. A dead stage is easily localized by signal-injection tests at the bases of the transistors with a noise generator, harmonic oscillator, or modulated-rf generator. Apprentice technicians should be reminded to make certain that a series capacitor is included in the "hot" lead from a signal source, to prevent dc drain-off from transistors under test.

Weak stages are localized to best advantage by means of stage-gain measurements. A dc voltmeter is connected at the detector output, and a generator signal is injected at the bases of successive transistors. The agc line must be clamped to obtain accurate output-voltage indications. In the case of i-f stages, a comparative check is sufficient; that is, inject the i-f signal at the detector input and note the voltmeter reading—then inject the i-f signal at the base of the second i-f stage and observe the increase in voltmeter reading. Finally, inject the i-f signal at the base of the first i-f stage and again observe the increase in voltmeter reading. The ratios of the successive readings should be in the same order of magnitude. A significantly low ratio points to a weak stage. The gain of an rf stage is not quite as great as that of an i-f stage in normal operation, and the gain of a converter stage is about half as much.

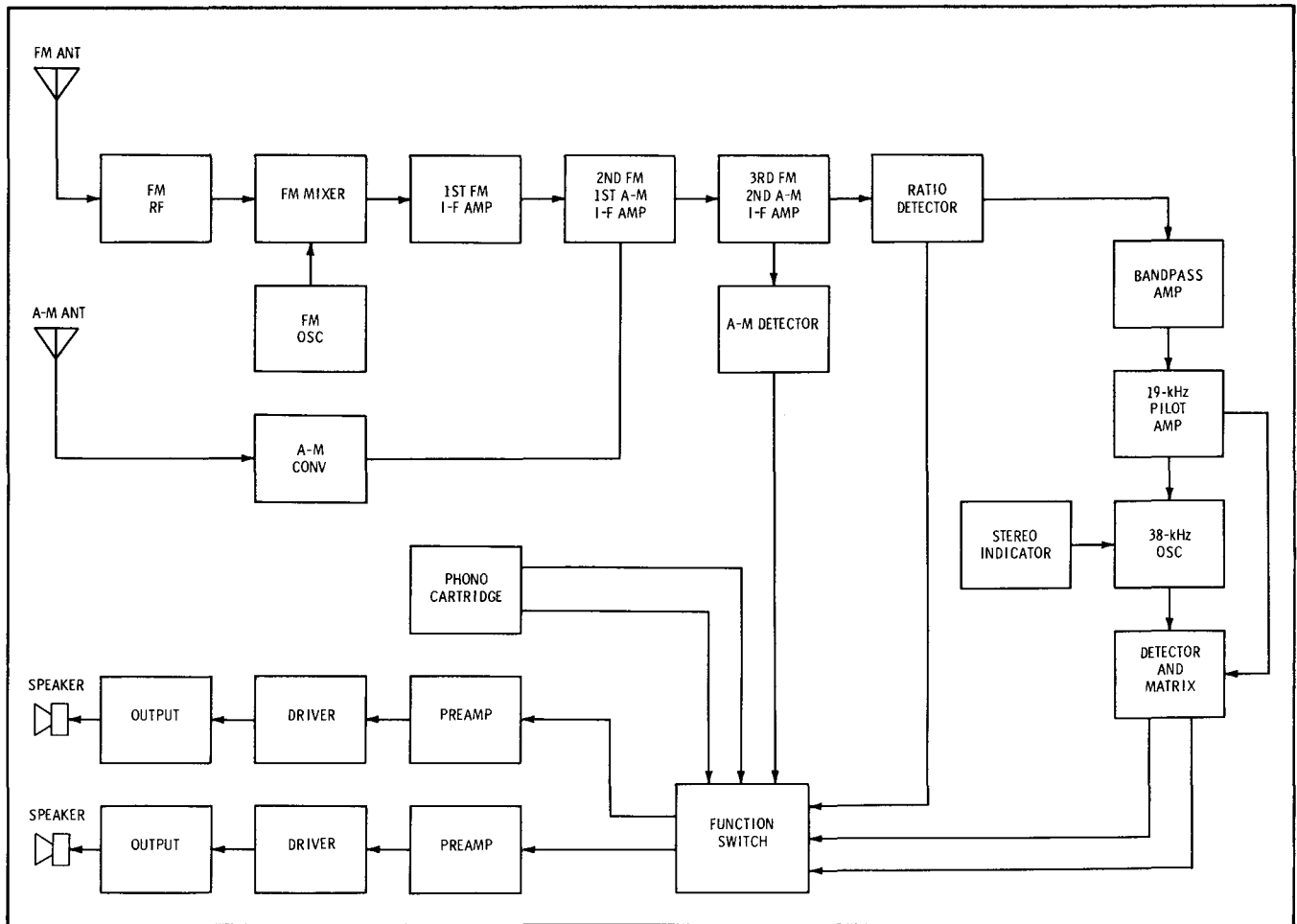


Fig. 1-30. Block diagram of a typical am/fm stereo chassis.

Distortion is most likely to occur in a high-level stage, such as the last i-f amplifier. However, regeneration can cause distortion in low-level stages. In some cases, distortion is the result of high-level ripple on the B+ line. The detector stage occasionally produces distortion if a load resistor increases greatly in value. If regeneration is present, the distortion will increase at low values of agc override voltage. The reason for this response is that the beta of a transistor or the mu of a tube increases when the agc voltage is reduced. In turn, regeneration and distortion are increased. To localize a regenerative stage, it is advantageous to check the bandwidth step-by-step with a signal generator; the tuning meter may be used as an indicator, as noted previously. We start by injecting the signal at the base of the last i-f transistor, and noting the 70.7 percent frequencies. Then we proceed to the next-to-the-last stage, and so on. A regenerative stage shows up as subnormal bandwidth between the 70.7 percent frequencies.

Intermittent troubleshooting is often the most

time-consuming and baffling type of job that must be confronted by the technician. An intermittent is a trouble condition that is not continuously present, but switches in and out or varies in degree from time to time. Thermal intermittents are actuated by heat; for example, a thermally intermittent resistor may change value suddenly when its temperature passes a certain critical level. Another thermally intermittent resistor may change value smoothly but rapidly as its temperature increases. Defective components in this category can be localized to best advantage by varying their temperature with a heat lamp followed by a spray coolant.

In a puzzling troubleshooting situation, the value of a thorough visual inspection cannot be overemphasized; there is always a tendency for the highly skilled technician to overlook the obvious possibilities, such as a broken conductor inside an insulated lead, or frayed insulation on interconnecting leads. Cracked PC conductors and cold-soldered connections (Fig. 1-13) are common causes of intermittent operation. Line cords should be checked, particu-

## Hi-Fi Stereo Servicing Guide

larly in installations that employ long extension cords. Since the voltage drop produced by an extension cord aggravates any line-voltage fluctuation that occurs, monitoring should be done at the a-m tuner—not at the power outlet. If the supply voltage fluctuates sufficiently, the local oscillator can “drop out” mysteriously when a refrigerator starts up, or when a disposal unit is turned on.

Other common troublemakers are worn switches, erratic controls, incompletely inserted plugs, defective capacitors, overheated power transformers, poorly grounded coil shields, loosely mounted PC boards, and failing rectifiers. Experienced technicians know that it is easy to overlook the soldering of a connection after a component has been replaced and its connecting wires inserted in eyelets or hooked around lugs. Mechanical intermittents of this general type can often be localized by “tapping” procedures. In most cases, the intermittent condition becomes increasingly responsive when the trouble area is approached. Case histories show that an a-m transformer can develop leakage or a short-circuit between windings, or a winding can become intermittent. Try pushing on the i-f transformer terminals, and press against the shield can while listening to the signal output. In most cases, a defective transformer tunes broadly, and several turns of the core will be required to change the signal level noticeably.

As a general rule, transistors are very long-lived, and tend to fail catastrophically. However, there are also the exceptions that prove the rule. For example, all experienced technicians can recall an occasional encounter with an intermittent transistor. The intermittent often develops as an “on-or-off” switching action. Again, an intermittent transistor can fluctuate between normal operation and weak operation. A defective transistor in the i-f section can produce regeneration and can masquerade as a defective neutralizing capacitor. If stations can be tuned in only at the high or at the low end of the band, the oscillator transistor is a ready suspect. Nearly all transistor defects are associated with changed values of terminal voltage. Technician apprentices may discover that a transistor trouble symptom can be “cured” by operating the circuit with more or less overvoltage. However, it is very poor practice to “troubleshoot” a circuit by providing excessive operating voltage; an imminent catastrophic failure can be anticipated under these conditions.

Troubleshooting an interference symptom requires an accurate a-m generator and starts with a check of the overall frequency-response curve for the tuner. Sometimes you will find that interference is the result of “cockpit trouble” in which a semitech-

nical customer has attempted to align the tuner by ear. In such case, it is only necessary to follow the alignment procedure specified in the receiver service data. However, the majority of interference symptoms will be tracked down to defective components such as leaky or open capacitors, shorted coil turns or layers, a damaged built-in antenna, and so on.

### ANALYSIS OF COMMON SYMPTOMS

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

#### 1. Dead Tuner

Any defect that completely stops the signal will result in a “dead tuner” symptom. It is sometimes helpful to evaluate the noise level of the system with the volume control turned to maximum. For example, the noise level will be higher if the rf stage is dead, than if the last i-f stage is dead. In the case of a tube-type tuner, the tubes should be checked first.

Possible causes of a dead a-m tuner are as follows:

- Defective or erratic function switch (see Fig. 1-30).
- Oscillator drop-out. Check by substituting a generator signal, as shown in Fig. 1-31.
- Shorted capacitor, such as C73 in Fig. 1-6.
- Open capacitor, such as C75 in Fig. 1-6.
- Leaky capacitor, such as C69 in Fig. 1-6.
- Defective transistor; make control-action test, measure dc terminal voltages, or use in-circuit transistor checker.
- Break in PC wiring; make voltage measurements and continuity checks of suspected conductors.

As noted previously, preliminary localization of a dead stage is made on the basis of signal-tracing or signal-substitution tests. If the entire tuner is dead, the trouble is most likely to be found in the detector

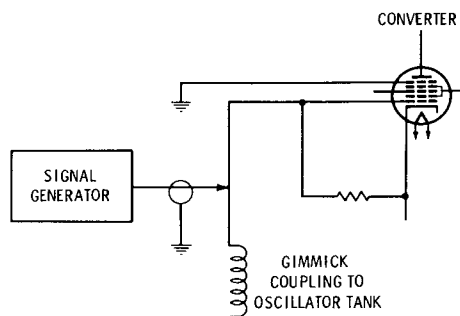


Fig. 1-31. Substituting signal generator output for the local oscillator signal.

stage—for example, X6 in Fig. 1-6 might be open or shorted. This type of trouble will show up in signal-injection tests, and is easily confirmed by signal-tracing tests.

## 2. Weak Output

Weak output in a-m reception, with normal output in fm reception, points to specific areas of circuitry in a tuner. For example, if an a-m transformer winding is defective or misaligned, fm reception remains unaffected. Similarly, a defect in the a-m detector section will affect a-m reception only. On the other hand, collector leakage in an i-f transistor will cause weak output on both a-m and fm functions.

Possible causes of weak output are:

- a. Defective capacitor, such as C59 in Fig. 1-6.
- b. Misaligned rf, i-f, or oscillator tuned circuits.
- c. Diode with poor front-to-back ratio (X6 in Fig. 1-6).
- d. Leaky capacitor; e.g., C73 in Fig. 1-6.
- e. Defective i-f transformer (tuned circuits cannot be aligned properly).
- f. Transistor with collector leakage; make control-action test, measure dc terminal voltages, or use in-circuit transistor checker.
- g. Incorrect agc bias (tuner operates properly with over-ride bias); check for defective capacitors in agc section.

## 3. Distorted Output

It was pointed out previously that distortion can be caused by poor frequency response, by nonlinear operation, or by both defects. Incorrect frequency response is the most likely source of distortion in an a-m tuner. This may be due to a component defect or to misalignment. Nonlinear operation is usually associated with other defects, such as weak or noisy output.

Possible causes of distorted output are:

- a. Defective neutralizing capacitor (see C55 and C59, Fig. 1-6).
- b. Resistor greatly increased in value, such as R55 in Fig. 1-6.
- c. Deteriorated detector diode (X6 in Fig. 1-6).
- d. Failing transistor; measure terminal voltages.
- e. Incorrect replacement transistor (see Table 1-1 for examples of standard rf, i-f, and audio types).
- f. Fault in agc system, such as open C6 in Fig. 1-6.

## 4. Incorrect Dial Indication

In all superheterodyne receivers, the dial indication is determined by the local-oscillator frequency. The reason for this is that most of the selectivity is provided by the i-f amplifier. Since the incoming signal frequency is unchanged by the receiver tuning,

Table 1-1. General Replacement Transistors

Stage and Type	RCA	GE	SYL	MOT	INR	SEM	WTV
A-m converter or rf, (pnp germanium)	SK3005 SK3008	GE-1 GE-9	ECG100 ECG126	HEP1 HEP2 HEP3	TR-11 TR-12	HF12H,M HF20H,M	AA1
(nnp, germanium)	SK3011	GE-5 GE-6	ECG101	HEP641	TR-08	NR10	AA2
Fm-rf amp (pnp germanium)	SK3006	GE-9	ECG106	HEP636 HEP637	TR-12 TR-17	HF12H,M HF20H,M	AA3
Fm i-f amp or converter (nnp, silicon)	SK3018	GE-11	ECG107 ECG108	HEP718 HEP719 HEP720	TR-21 TR-22		AF4
Af output or driver (pnp, germanium)	SK3003 SK3004	GE-2	ECG102 ECG123	HEP629 HEP634	TR-14	AT10H,M,N AT20H,M,N	
(nnp, germanium)	SK3010 SK3020	GE-8	ECG103	HEP641	TR-09		
Audio output (pnp, germanium)	SK3009 SK3012 SK3014	GE-3 GE-4 GE-16	ECG104 ECG105 ECG121	HEP234 HEP623	TR-01 TR-03	PT25 PT50	AA4
Audio output, Hi-fidelity amp (matched pair)	SK3013 SK3015	GE-13MP					

RCA Radio Corporation of America  
 GE General Electric Company  
 SYL Sylvania  
 MOT Motorola  
 INR International Rectifier  
 SEM Semitronics Corp.  
 WTV Workman Electronics Products, Inc.

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and the i-f amplifier responds to a fixed frequency, the signal can be passed only when the local oscillator is tuned to a frequency that produces the correct beat frequency. In turn, if the dial indication is incorrect, either the local oscillator is at fault, or the dial assembly is defective in some manner that causes incorrect pointer location.

Possible causes of incorrect dial indication are as follows:

- a. Dial-cord stringing incorrect, slack, or loose. See receiver service data.
- b. Dial plate incorrectly mounted or shifting in position.
- c. Local oscillator incorrectly aligned. Check the tracking per receiver service data.
- d. Defective capacitor in oscillator section.
- e. Shorted turns or other defect in oscillator transformer.
- f. Incorrect type of replacement transistor in oscillator circuit.
- g. Incorrect type of replacement tuning capacitor.

Tracking is a technical term that denotes the ability of a receiver to maintain correct resonant frequencies in the rf, oscillator, and mixer circuits, at any point on the tuning dial. Dial-tracking refers to the ability of the system to indicate correct frequency values on the tuning dial when the receiver is in correct alignment. Therefore, we start by checking the alignment of the tuned circuits, as specified in the receiver service data. After the trimmers and slugs have been properly adjusted, we then turn to dial-tracking considerations, and any mechanical faults that might be involved.

### 5. Intermittent Operation

Whenever an intermittent “ties up” an a-m tuner on the bench, the most useful approach is to monitor the operation of the various circuit sections for a suitable length of time. Specialized monitoring instruments can be employed, or conventional meters and scopes will serve the purpose. It is advisable to energize the tuner with a steady signal from an a-m generator and to monitor the outputs at the rf amplifier, mixer, i-f amplifier, and detector with VTVMs or scopes. In turn, when the intermittent occurs, informative data is provided without disturbing the circuits—next, the monitoring instruments can be connected at various points in the defective section, to close in on the defective component.

Possible causes of intermittent operation in an a-m tuner are:

- a. Local oscillator “drop-out” due to low supply voltage or a defective transistor.
- b. Mechanical intermittent such as a poor connection; can often be localized by tapping.
- c. Microscopic break in PC wiring; shows up clearly when the PC board is flexed slightly up and down.
- d. Thermally intermittent resistor or transistor; localize by using a heat lamp, followed by a spray coolant.
- e. Intermittent capacitor; monitoring procedures will pinpoint.
- f. Marginal defect in tuned coil or transformer; tap with a tuning rod or pencil; press assembly gently from side to side; press terminals while observing output from the tuner.
- g. Broken conductor inside an insulated lead; flex lead while monitoring.
- h. Frayed insulation, permitting intermittent short circuit.
- i. Plug not fully inserted into receptacle.

### 6. External Interference

External interference is sometimes due to “cockpit trouble”; that is, if the customer connects the a-m tuner to an excessively high and long external antenna, it may be impossible to prevent powerful local stations “feeding through” when the receiver is tuned to a weak station. There is particular need for restricting the antenna signal level when no pre-selection is provided in an a-m tuner. In any case, the experienced technician will determine the prevailing conditions in the given locality, and whether all the receivers in the area might be affected by external interference. Antenna trap circuits may need to be provided. When an interference complaint is legitimate, the trouble will usually be a result of poor alignment, or a component failure that disturbs alignment.

Possible causes of external interference in a-m tuner operation are:

- a. Previous sloppy alignment job; check alignment adjustments carefully per receiver service data.
- b. Defective capacitor, such as C58 in Fig. 1-6.
- c. Incorrect replacement loop antenna.
- d. Defective rf or i-f transformer; shows up as inability to tune, or as an abnormally broad peak.
- e. Incorrect replacement type of rf transformer.
- f. Defect in agc section that causes tuner to operate at maximum gain regardless of the incoming signal level.

**7. Poor Selectivity**

Poor selectivity is generally due to the same defects or misadjustments that were noted under Topic 6. In addition, poor selectivity sometimes is the result of a defective IC in systems such as shown in Fig. 1-3. When a defect develops in an IC, we can usually confirm the suspicion by means of dc voltage measurements at the package terminals. This procedure is helpful in distinguishing between a defective i-f transformer and a defective IC package.

**8. Drifting Off Frequency**

Off-frequency drift can occur in any section of an a-m tuner; however, it is most common in the local-oscillator section. The frequency of oscillation is affected by the condition of components other than

coils and capacitors; for example, a deteriorating transistor can cause frequency drift. A leaky capacitor may cause frequency drift as the leakage resistance fluctuates. Bias variation on a transistor due to capacitor leakage or resistor instability often causes frequency drift. This is the result of the extremely high value of Q in an oscillatory feedback circuit.

Possible causes of off-frequency drift in an a-m tuner are:

- a. Leaky capacitor, such as C62 in Fig. 1-6.
- b. Deteriorating oscillator transistor.
- c. Unstable resistor—e.g., R32 in Fig. 1-6.
- d. Marginal defect in oscillator coil.
- e. Mechanical defect in tuning assembly.
- f. Operation of tuner in poor location, such as a patio exposed to fog or mist.

## Chapter 2

# FM Tuner Troubles

Practically all current-model fm tuners are transistorized, although many tube-type tuners are still in use. Many fm tuners are combined with a-m tuners in combinations; quite a few fm tuners are merchandised as separate units for hi-fi buffs. The superheterodyne configuration is universal; however considerable variation of detail is employed. For example, lower-priced fm tuners use a single transistor as a combined oscillator and mixer (converter), whereas de-luxe type tuners provide separate oscillator and mixer transistors. Again, economy-type designs generally utilize ganged tuning capacitors (Fig. 2-1), while the more elaborate designs provide permeability-tuned arrangements (Fig. 2-2). Solid-state designs are based on the transistor-for-tube principle, except for systems employing IC packages.

Fig. 2-3 shows a block diagram for a typical fm tuner. This arrangement includes six transistors, two ratio-detector diodes, an afc diode, and an agc diode. Only the rf stage is agc-controlled. The afc circuit minimizes the effect of any frequency-drift tendency in the oscillator section. No limiter is provided, because a ratio detector has self-limiting action. Nearly all fm tuners have a stage of rf amplification (preselection). This provides improved input selectivity, with the result that the possibilities of cross modulation and image interference are minimized. To ensure that cross modulation will not occur in the rf stage when high-level adjacent-channel signals are present, agc control is provided.

Common trouble symptoms caused by defects in fm tuners are as follows:

1. No fm reception.
2. Weak and/or noisy output.
3. Poor sound quality.

4. Afc failure.
5. Co-channel and adjacent-channel interference.
6. Incorrect dial calibration.
7. Intermittent operation.

### GENERAL DISCUSSION

An fm broadcast signal is basically a high-fidelity transmission, whether the mono or the stereo mode is used. Note that some fm broadcast stations also transmit a multiplexed subsidiary carrier-assignment (SCA) signal; this is not a high-fidelity transmission. An SCA signal cannot be reproduced by a conventional fm receiver; in fact, traps are provided to eliminate any possibility of SCA interference. An fm signal not only provides better sound quality than an a-m signal, but it also provides quieter reception in normal operation. That is, natural and manmade noise is chiefly an a-m disturbance, to which fm receivers are largely unresponsive. The high-fidelity capability of an fm system is due to its comparatively wide channels (see Fig. 2-4). Each fm broadcast channel has a bandwidth of 150 kHz, or  $\pm 75$  kHz. Although a high-fidelity fm signal can be transmitted in a narrower channel, wide channels provide a better signal-to-noise ratio.

It follows from Fig. 2-4 that an fm receiver should have a bandwidth of 150 kHz. The fm broadcast band extends from 88 to 108 MHz, encompassing a total range of 20 MHz; an fm station occupies (including its guard band) a range of 200 kHz. Thus, a maximum of 100 fm broadcast stations can be accommodated in a given service area. Co-channel interference from distant areas is much less troublesome in the case of fm, than for a-m. This comparative immunity of an fm receiver to co-channel interference



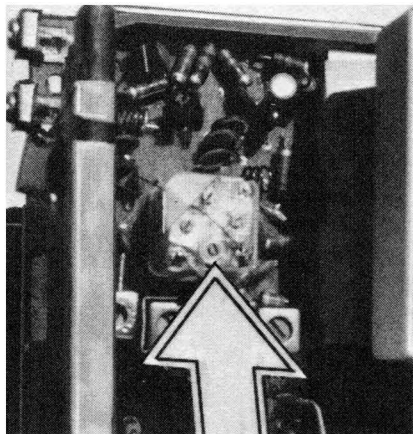


Fig. 2-1. Arrow points to gang tuning capacitor.

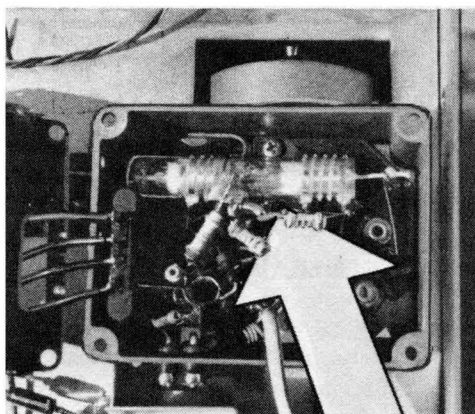


Fig. 2-2. Arrow points to permeability-tuned rf and oscillator coils.

is due to the "capture effect," which means that the stronger signal tends to suppress the weaker signal. If one signal is twice as strong as the other signal, the suppression is practically complete. Even if one signal is only 50 percent stronger than the other, the suppression is very effective, provided that the receiver is in good adjustment.

All fm receivers use an intermediate frequency of 10.7 MHz. At the 50 percent of maximum response points (6 dB down) on the i-f response curve, the

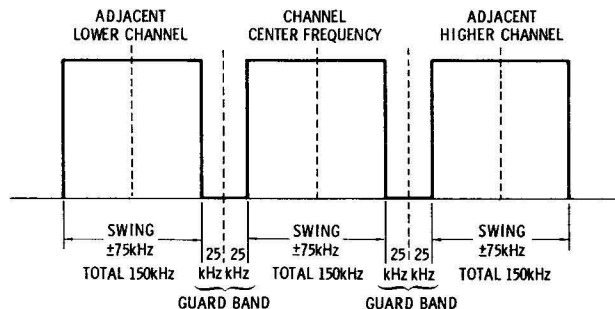


Fig. 2-4. Bandwidth specifications for commercial fm.

bandwidth is typically 200 kHz, as shown in Fig. 2-5. The comparatively high value of intermediate frequency reduces the circuit complexity that is required. For example, if a lower intermediate frequency were used, such as 455 kHz, elaborate circuitry would be required to obtain a flat-topped response over a bandwidth of 150 kHz. When the bandwidth of an amplifier is increased, its gain is decreased. Therefore, the i-f stage gain is considerably less in an fm receiver than in an a-m receiver. Therefore, an fm receiver employs more i-f stages. At least one more i-f stage is provided in most tuners. When two more fm i-f stages are included, the last stage operates as a limiter; since a limiter is essentially an overdriven amplifier, its gain is considerably less than that of the other i-f stages.

Some fm tuners employ integrated circuitry, as exemplified in Fig. 2-6. As noted previously, a typical IC package has 10 terminals, with normal operating voltages specified. We often find groups of two terminals, or groups of three terminals tied together in an i-f configuration. When an IC section develops a defect, the terminal voltages become abnormal in

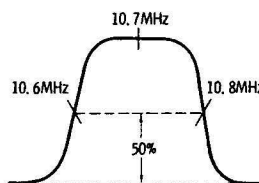


Fig. 2-5. Frequency response of an i-f amplifier in a typical fm tuner.

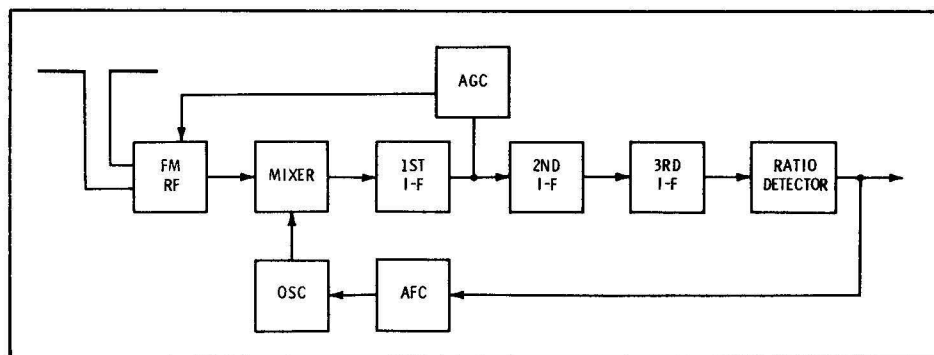
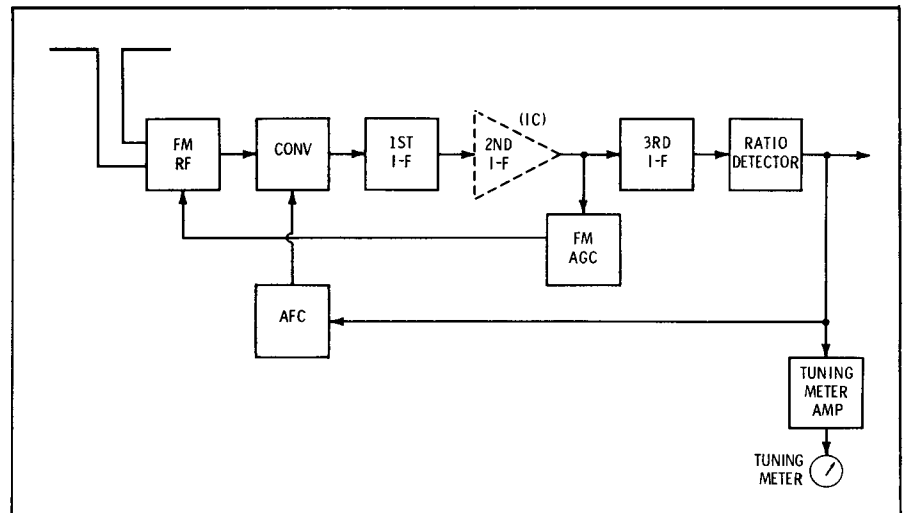


Fig. 2-3. Block diagram for a typical fm tuner.

Fig. 2-6. Block diagram of an fm tuner using an integrated circuit.



most cases. This situation is analogous to abnormal terminal voltages for a defective transistor. The receiver service data will usually specify the normal dc current demand of an IC package, such as 6 mA. Therefore, a current check can be made in addition to voltage checks. Technician apprentices must be on guard to avoid confusion between IC defects and leaky capacitors. That is, a leaky capacitor connected to an IC package terminal can make the IC "look bad." Therefore, we check capacitors first.

Many hi-fi fm tuners include a tuning meter or a tuning indicator such as some form of electron-ray tube. In the example of Fig. 2-6, a zero-center meter is used as a tuning indicator. The meter is energized from the output of the ratio detector, and high sensitivity is provided by a transistor that operates as a tuning-meter amplifier. As the converter stage is tuned through the center frequency of the incoming fm signal, the output from the ratio detector swings from positive to negative. In turn, the pointer on the tuning-meter scale deflects from above center to below center. The output from the ratio detector also energizes an afc diode (reverse-biased diode) to minimize converter frequency drift. A reverse-biased junction diode operates as a small capacitor, due to its junction capacitance. When the value of the reverse-bias voltage changes, the junction capacitance also changes. To avoid overload of the rf stage on strong incoming signals, an fm agc circuit is provided; this minimizes the possibility of cross-modulation interference.

Fig. 2-7 shows the configuration for a typical transistor fm tuner. In this example, the fm agc circuit is supplemented by a local-distant switch in the base-input circuit of the rf amplifier. This is a desirable feature, because the base circuit is untuned, and cross-modulation can become a problem when strong

incoming signals are present. In the "Local" position of the switch, substantial signal attenuation is provided by a resistive pad. Automatic frequency control is provided by diode X3, which is shunted across the oscillator tank. An afc defeat switch is included to facilitate tuning-in of weak signals; that is, if afc is used while attempting to tune-in a weak signal, the oscillator will often sweep through the weak carrier too fast to lock in. As in nearly all fm receivers, a ratio detector is used because of its self-limiting action; some pre-limiting is also provided by R35 in the collector circuit of Q5.

### Alignment Principles

Ratio detectors do not respond to amplitude modulation unless the modulation percentage is quite high, or unless the detector is defective or misaligned. The following general discussion will serve as a guide; however, the service data for a particular receiver should be read carefully before proceeding with frequency checks or alignment adjustments. An accurate signal generator is basic; its output may be applied to the grid of the last i-f stage. With reference to Fig. 2-8, an unmodulated rf signal is applied at A-B, and a VOM or VTVM is connected across R1. Using a 10.7-MHz signal, the primary of transformer T is adjusted for maximum meter indication. Next, to check the secondary alignment, it is necessary to divide the load circuit into two symmetrical parts. For example, we can connect two 100-k resistors (R2 and R3) across R1. A voltmeter is then connected from the junction of R2 and R3 to the audio output point (from Y to X1). Then, the secondary of the transformer is tuned for zero indication on the voltmeter.

To check the i-f alignment, connect the voltmeter across the load resistor of the ratio detector, and in-

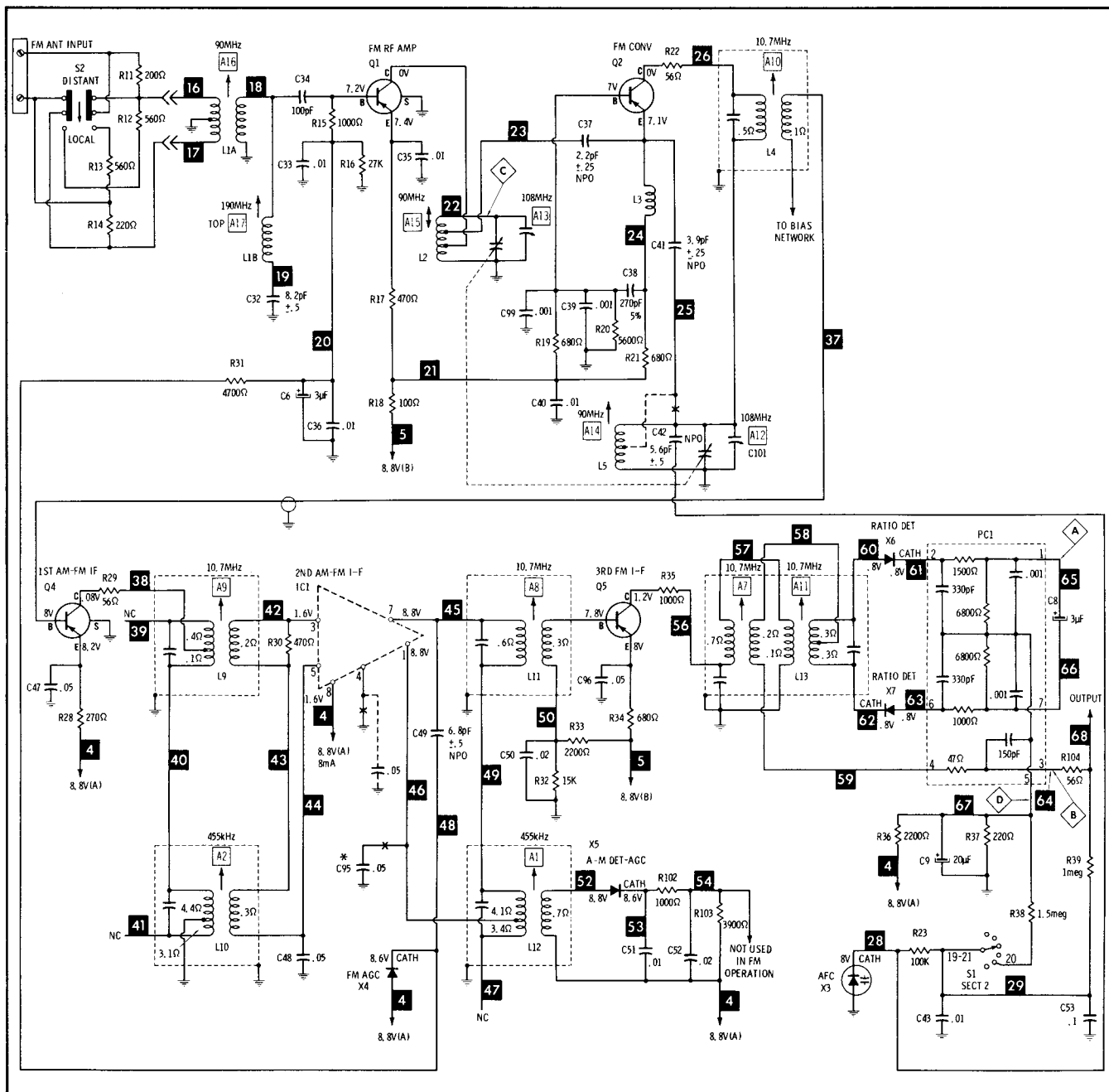


Fig. 2-7. Configuration for a transistor fm tuner.

ject an unmodulated 10.7-MHz signal at the input of the mixer stage. Then, peak each i-f stage for maximum indication on the meter. Finally, alignment of the rf, mixer, and oscillator (or converter) stages is similar to the procedure previously described for a-m receivers; the only difference is in the higher frequencies that are used in aligning the fm stages. You will find that an oscilloscope and sweep generator are more practical for checking frequency response, because the bandwidth of the circuits is easily determined. To anticipate subsequent discussion, various

trouble symptoms are caused by either subnormal or excessive bandwidth, although the center frequency of the stage is normal. Receiver service data will specify the connections to be used in sweep alignment. The basic considerations are as follows:

1. To check the center frequency and the bandwidth of the response curve, accurate marker signals must be used. A marker signal produces a pip (birdie) on the curve at the frequency to which the marker generator is set.

Fig. 2-8. Ratio-detector circuit, showing modification for alignment.

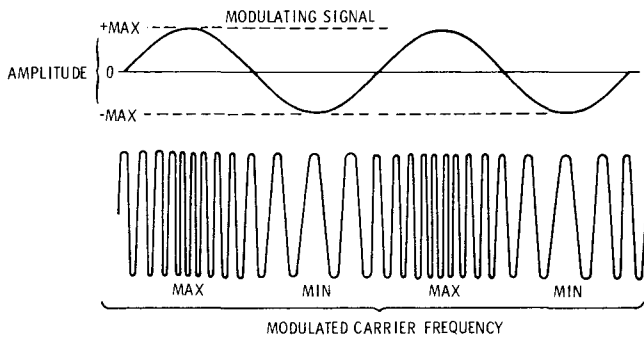
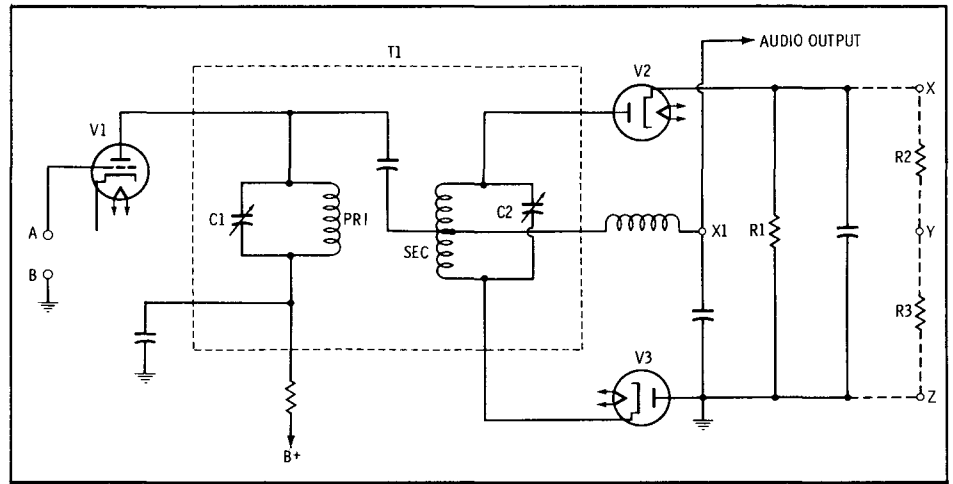


Fig. 2-9. The fm sweep signal.

2. Marker generators are crystal-calibrated to ensure maximum accuracy. The more elaborate types of generators will provide two or more markers simultaneously.
3. A conventional marker generator that provides a single marker is adequate, provided that it has good accuracy. The scale should also be reasonably expanded, so that it is easy to

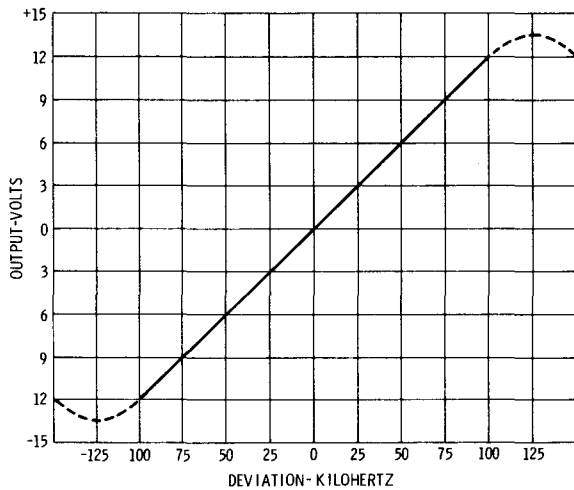


Fig. 2-10. S curve response produced by a normally operating ratio detector circuit.

read frequencies such as 10.6, 10.7, and 10.8 MHz (see Fig. 2-5).

The basic characteristics of an fm sweep signal are shown in Fig. 2-9. A normally operating ratio detector produces an S curve such as that shown in Fig. 2-10. An ideal response curve is perfectly linear for 100 kHz on either side of 10.7 MHz. Fig. 2-11 illustrates a typical rf or i-f response curve for an fm receiver. Although the top of the curve is not entirely flat, the end result of limiting through the ratio detector is to effectively "slice off" the top of the i-f curve, thereby changing the double-humped top into a flat top. The dotted lines indicate the limits of the fm signal swing, or deviation. Technician apprentices should observe that a single stage will always have a greater bandwidth than when it is followed by additional stages in cascade. It is the overall frequency

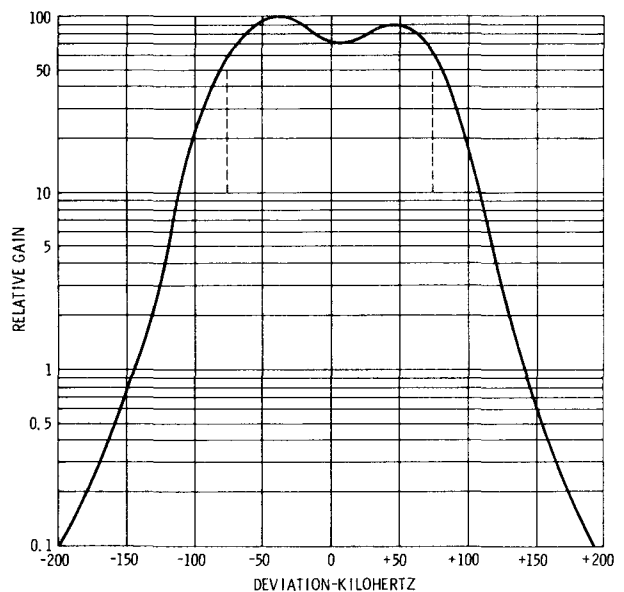


Fig. 2-11. Typical fm rf or i-f response curve.

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response that determines the fidelity of the fm system.

### Distortion Checks

After an fm tuner has been repaired and/or realigned, the final proof of performance consists in a measurement of the percentage of distortion. This

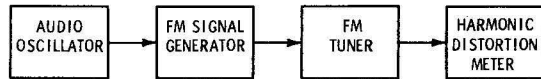


Fig. 2-12. Test setup for measuring percentage of harmonic distortion of a tuner.

test requires an accurate fm signal generator and a harmonic-distortion meter. If you wish to make distortion measurements at high and low audio frequencies (at audio frequencies other than 1 kHz or 400 Hz), an accurate audio oscillator will also be required. That is, fm signal generators provide modulation at 400 Hz or 1 kHz, but do not provide for tests over the entire hi-fi range from 20 Hz to 20 kHz. Fig. 2-12 shows the test setup that is used. The audio oscillator externally modulates the fm signal generator, which in turn drives the fm tuner—usually at a carrier frequency of 100 MHz. The output from the fm tuner is fed to the harmonic-distortion meter, which indicates the percentage of distortion that is present. As noted previously, the audio oscillator and the signal generator should have less than 1 percent distortion, in order to make a valid test of a hi-fi tuner. Fig. 2-13 illustrates a typical fm signal generator.

If the percentage of IM distortion is to be measured, the fm signal generator is externally modulated by the two-tone signal output from the IM analyzer. In turn, the output from the fm tuner is

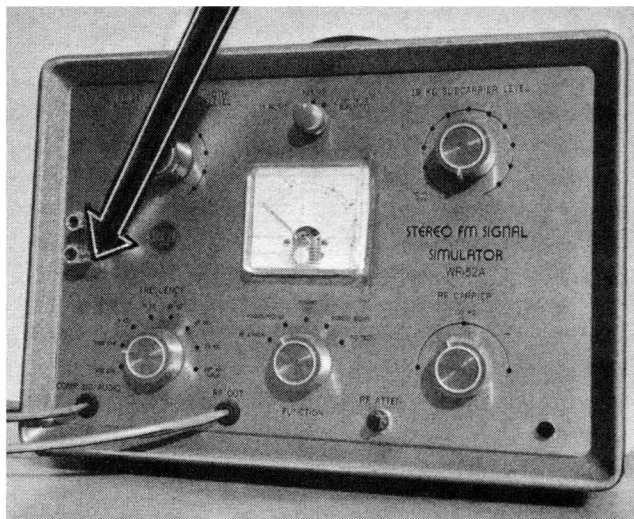


Fig. 2-13. Arrow points to external-modulation terminals on fm signal generator.

fed to the input of the IM analyzer. Of course, the harmonic distortion or the intermodulation distortion of the hi-fi system can be measured, if desired; if the total distortion percentage is tolerable, there is no need to measure the distortion percentage for the fm tuner by itself. On the other hand, if the total distortion percentage is out of acceptable limits, it becomes necessary to track down the major source of the distortion. In this situation, we need to make a measurement of the distortion percentage produced by the fm tuner.

### Signal Tracing, Signal Injection, and Gain Checks

Signal-tracing procedures are basically the same in troubleshooting either fm or a-m tuners. Beginners should note that an fm generator cannot be used to make signal-tracing tests in an fm tuner; the reason is that service-type indicating instruments such as VOMs, VTVMs, scopes, or audio amplifiers with demodulator probes cannot respond to an fm signal. Therefore, we are obliged to use an a-m generator and check the progress of an amplitude-modulated rf signal through the circuits of the fm tuner. The indicator may be an ac voltmeter, an audio amplifier used with a demodulator probe, or a scope with demodulator probe. Note that the ratio-detector section normally has no output when an am signal is applied; however, the dc voltage across the load resistance and stabilizing capacitor rises. Alternatively, the ratio-detector section can be checked by applying an fm signal to the tuner.

On the other hand, signal-injection tests start at the ratio detector, and proceed back step by step toward the rf amplifier. Accordingly, an fm signal generator is used for signal-injection tests. Although an ac VTVM or scope can be used as an indicator, the tuning meter (or similar tuning indicator) will serve the purpose adequately. In signal-injection tests, the output from the generator is reduced as the injection point is changed to include an additional stage of amplification. It is helpful to employ a generator that has sufficient output to drive the ratio detector normally when the signal is injected at the input of the ratio-detector section. Some service-type fm generators have rf output only, and do not provide an i-f output. In such a case, we must fall back on our a-m signal generator for signal-injection tests.

Stage-gain measurements are closely related to signal-injection tests. That is, we employ a generator signal and inject the signal at the bases of successive transistors, and proceed back step by step toward the rf amplifier. However, there are differences in procedural detail. Since we are concerned with comparative signal levels, a voltmeter must be used as

an indicator; the voltmeter is connected across the load resistance (stabilizing capacitor) in the ratio-detector output circuit. Note also that the *same* signal level is used from the generator in checking a given stage. In other words, we are measuring how many times the stage steps up the reference signal level. Of course, the generator output is reduced as required in the next step, to avoid overload. In case of doubt concerning gain adequacy, it is often possible to make a comparative check with another fm tuner of similar type which is in good operating condition.

**Oscillator Frequency and Stability Checks**

Some types of defects, such as open capacitors, can cause the oscillator to operate far off-frequency. In this situation, the trouble symptom is "dead tuner." The tuner will operate normally if a generator signal is injected or coupled into the mixer circuit at the required frequency. To determine whether the oscillator is dead or if it is operating at an incorrect frequency, check the collector circuit with the rf probe of a VTVM. In case an rf signal voltage is present, we know that the oscillator signal is not being coupled into the mixer circuit, or that the oscillator signal is far off-frequency. An open coupling capacitor can stop the signal, for example. To measure the oscillator frequency, a heterodyne frequency meter is

both accurate and convenient. Some TV marker generators are designed to operate as heterodyne frequency meters. We place the "rf in" lead of the frequency meter near the oscillator circuit, and tune the meter for zero beat. Then, the meter dial indicates the oscillator frequency.

When an oscillator has poor stability, its operating frequency drifts, with the result that the tuning dial must be reset at intervals to keep a station tuned in. Most fm tuners employ automatic frequency control, and this feature provides satisfactory oscillator stability under normal tolerances of drift. However, when the frequency drift exceeds the normal design tolerance, an afc circuit becomes ineffective and the station to which the tuner was set will begin to sound distorted and then suddenly disappear. Then, the tuning dial must be readjusted to tune in the station once again. In this situation, the prime suspects are a defective oscillator transistor, or a deteriorating capacitor in the oscillator circuit. Collector leakage or capacitor leakage tends to fluctuate, and the oscillator frequency is varied in turn. A systematic component check is generally required to pinpoint the offender.

**Mechanical Defects and Repair**

The most common mechanical defect encountered in fm tuners is a worn or otherwise defective dial

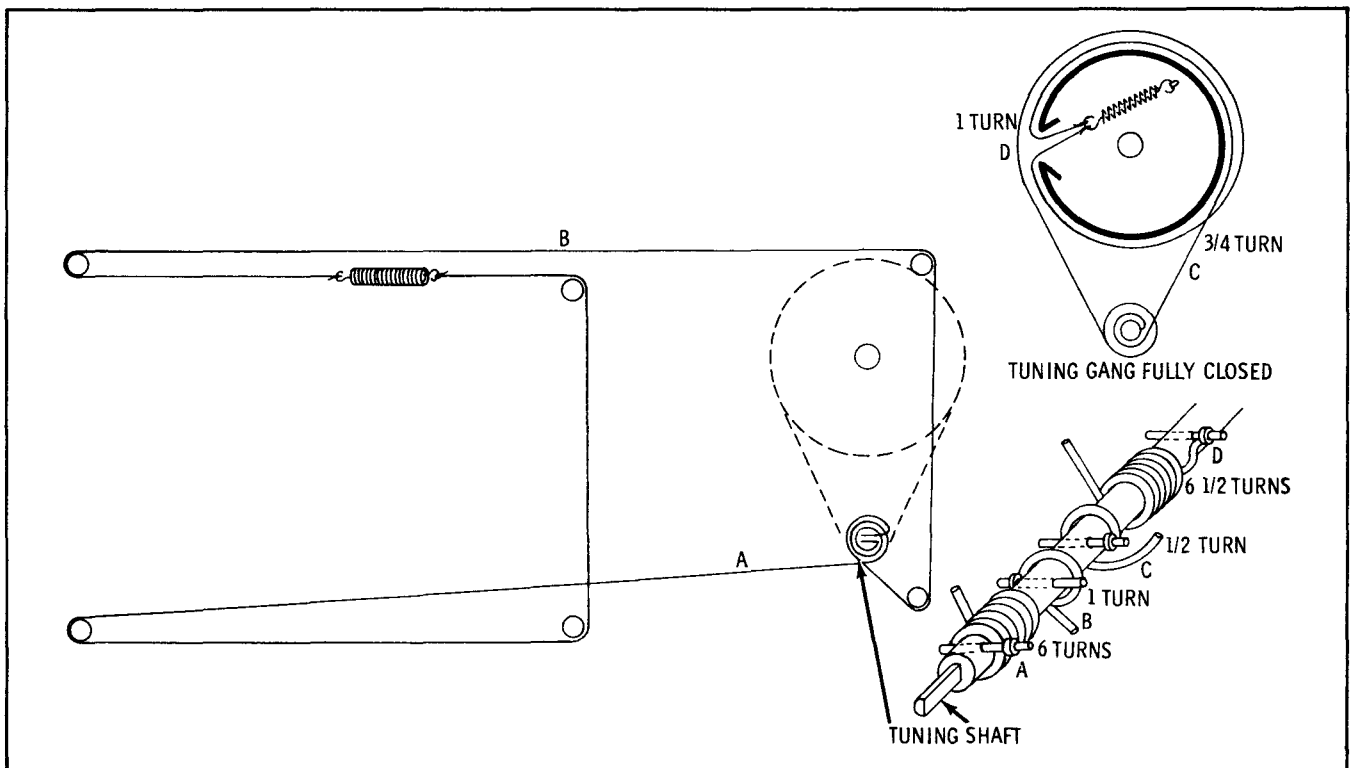


Fig. 2-14. Typical dial-cord stringing arrangement.

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cord. Dial-cord stringing instructions will be found in the receiver service data, as exemplified in Fig. 2-14. When the dial cord is not functioning properly, the tuning-scale indication will be incorrect. In permeability-tuned designs (see Fig. 2-15), a spring-loaded gear drive is often employed. In case of binding, erratic operation, or complete failure, look for damaged teeth, a broken tension spring, or foreign matter such as stray pieces of dial cord between the gears. There is also a possibility of stiffness in operation due to dry gear bearings; a drop of light machine oil in each bearing will cure the trouble in this situation. Stiffness in operation is accompanied by a greater torque requirement, with the result that the dial cord may tend to slip on the shaft. On the other hand, cord slippage in normal operation is usually the result of a polished and glazed surface on the cord; if you do not wish to replace the cord, try applying a small amount of belt dressing (no-slip compound).

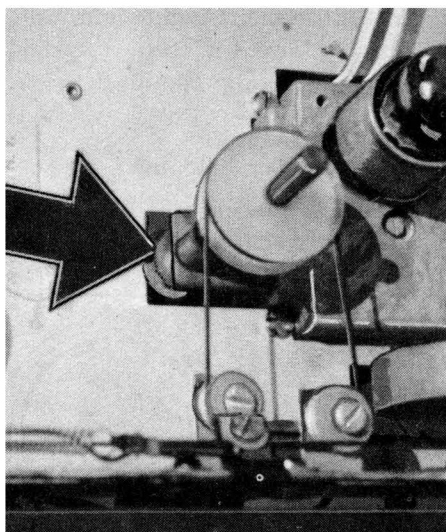


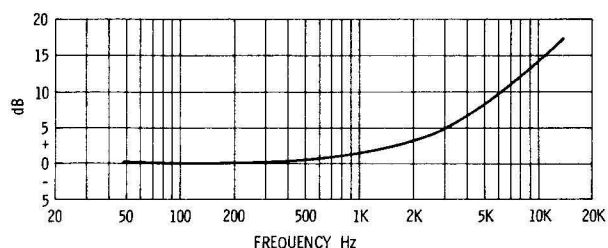
Fig. 2-15. Arrow points to gear drive in permeability tuner.

Other common mechanical defects include loose shields, partially inserted plugs, insecure PC boards, weak springs, broken resistors and damaged coils (generally due to careless use of tools), stray bits of solder and wire "whiskers," cracked PC boards, and leakage between PC conductors due to contamination with foreign agents such as cleaning preparations or beverages. In fm tuners that have had considerable service, we should also be on guard for broken conductors inside insulated leads. This can be an elusive trouble-maker if the possibility is not kept in mind. Control knobs occasionally become loose on their shafts, and marginal conditions can cause false indexing. Most scale plates are firmly mounted in place; however, if you are unable to

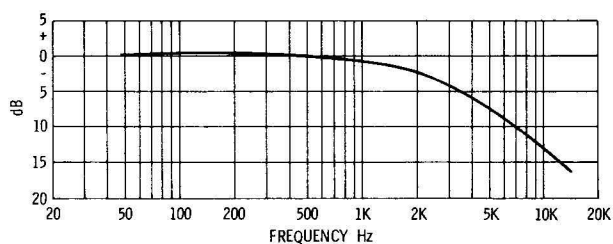
make the scale track correctly, check to see whether the clamps may have loosened.

### De-emphasis Considerations

To optimize the signal-to-noise ratio in the transmitted signal, fm stations pre-emphasize the high audio frequencies, as shown in Fig. 2-16A. In turn, a complementary de-emphasis network is provided in the fm tuner, in order to give a flat frequency response with respect to an fm broadcast signal. Fig. 2-16B shows the standard de-emphasis frequency characteristic. It is basically an integrating circuit, or low-pass filter. The standard time constant is  $75 \mu\text{s}$ ; the circuit in Fig. 2-17 has a  $66\text{-}\mu\text{s}$  time constant,



(A) Standard pre-emphasis curve at transmitter.



(B) Complementary de-emphasis curve used in fm tuner.

Fig. 2-16. Fm emphasis and de-emphasis characteristics.

because there is an effective  $9 \mu\text{s}$  time constant in the preceding circuitry. It is evident from Fig. 2-16 that if the de-emphasis capacitor (C86 in Fig. 2-17) should open up, the corresponding trouble symptom will be excessive high-frequency response (tinny sound) at the normal setting of the treble tone control.

When an fm tuner is designed as a separate package for use either with or without a multiplex adapter, the de-emphasis network is always included in the tuner, following the ratio detector. However, two output connectors are provided, one of which bypasses the de-emphasis network. Thus, when the tuner is used with a multiplex adapter, the de-

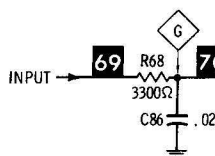


Fig. 2-17. Typical de-emphasis network.

emphasis network is bypassed, but when the tuner is used without a multiplex adapter, the ratio detector drives the audio amplifier through the de-emphasis network. A multiplex adapter contains a built-in de-emphasis network in its output section, so that a flat frequency response is provided. If the multiplex adapter were driven through the de-emphasis network of an fm tuner, the treble level would be very low, and there would be a serious unbalance between the two audio channels.

It is evident that the de-emphasis network in an fm tuner must be bypassed when making a harmonic-distortion test, as shown in Fig. 2-12. That is, if we should make the mistake of passing the output from the fm tuner through the de-emphasis network into the harmonic-distortion meter, an abnormally low reading would result because the higher harmonics would be greatly attenuated. In other words, an fm signal generator or an audio oscillator does not supply a pre-emphasized signal. Generators have an essentially flat frequency characteristic; therefore, any test that involves two or more audio frequencies must use a direct connection between the ratio-detector output and the indicating instrument. This requirement must be observed in harmonic-distortion and intermodulation-distortion tests, and in an audio-frequency response test such as is shown in Fig. 2-18.

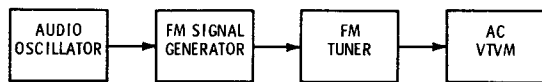


Fig. 2-18. Audio-frequency response test setup for an fm tuner.

The frequency-response test shown in Fig. 2-18 is not customarily made in service shops, because it can be assumed that the audio-frequency response will be

satisfactory if the tuned-circuit bandwidths are satisfactory (Figs. 2-10 and 2-11). Although nearly all shops have fm sweep-alignment equipment available, only a very few service benches are equipped with suitable instruments to run the audio-frequency response test shown in Fig. 2-18. An accurate test requires an rf deviation indicator in the fm signal generator, and only lab-type generators are provided with precision deviation meters.

**TROUBLESHOOTING PROCEDURE**

Troubleshooting starts with a review of the customer's complaint; useful preliminary data can often be obtained by listening to the customer's account of the difficulty. Tubes are tested as a matter of routine in the case of a tube-type tuner. Operating controls are checked—sometimes the trouble is caused merely by a loose control setscrew. The next step in most cases will be a thorough visual inspection and cleaning. This is an important procedure, which turns up common defects such as broken or disconnected leads, frayed insulation and short-circuits, heat damage such as burned resistors, loose parts, jammed gears, or poor connections. For example, function switches (Fig. 2-19) are often worn by extensive use to a point that their operation becomes erratic. If repair is impractical, the function switch must be replaced.

Evaluation of the noise level will often provide useful localization clues, as noted previously. For example, if the fm reception is dead or very weak, with a comparatively loud hissing sound across the band, the fm rf amplifier and oscillator (Fig. 2-19) are logical suspects. In a typical case history, this trouble symptom was tracked down to an open-circuited rf

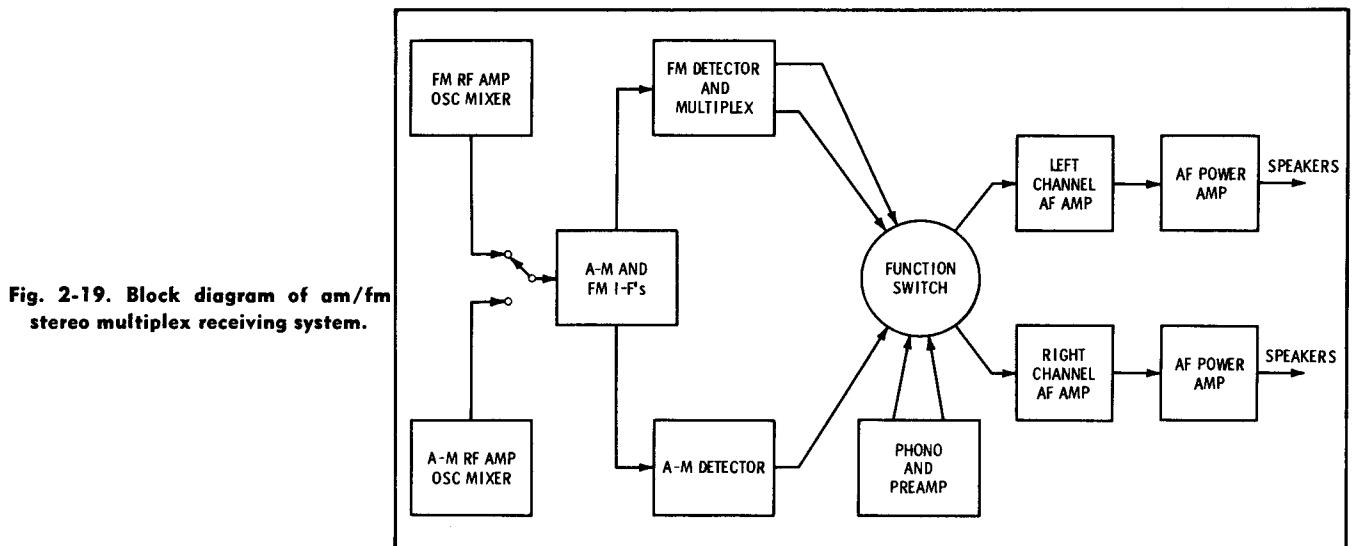


Fig. 2-19. Block diagram of am/fm stereo multiplex receiving system.

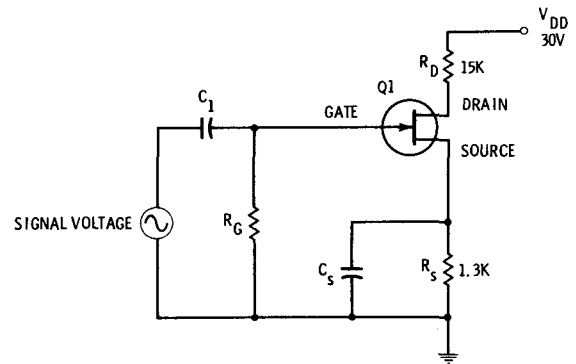


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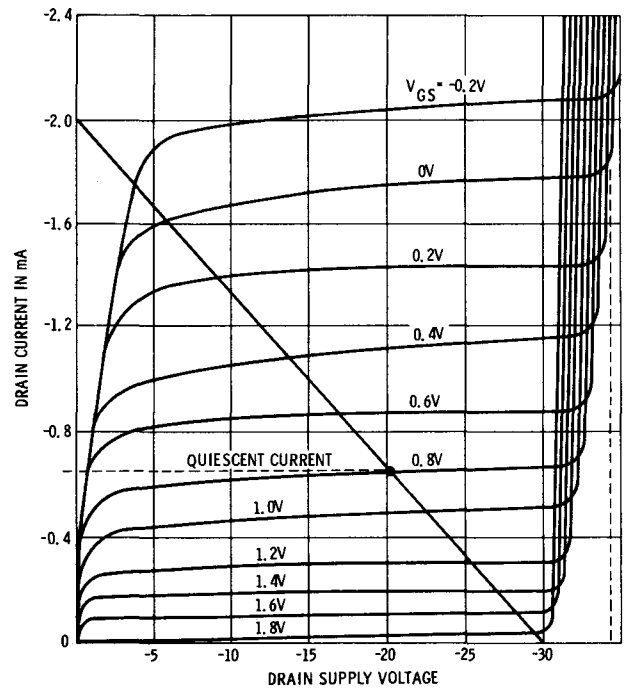
transistor. As noted in the previous chapter, part of the basic evaluation procedure consists in observing which functions of a combo are affected by the trouble. For example, a common situation involves dead fm and a-m functions, although the phono function is normal. With reference to Fig. 2-19, we would suspect the i-f section because it is common to both the fm and a-m channels. Signal injection with a noise generator or harmonic generator is a useful approach in this situation. In a typical case history, the trouble was tracked down to the second i-f transistor; an in-circuit transistor tester showed that the transistor was short-circuited.

To repeat an important point, defective transistors can usually be pinpointed by means of dc voltage measurements with a VOM, VTVM, or TVM. Control-action tests are also very useful for confirming suspicious dc voltage readings. Solid-state servicing requires a voltmeter that can indicate 0.1 volt or less with reasonable accuracy. If the instrument also provides a low-range ohms scale, checks of low-resistance tuner components will be facilitated. The base-emitter voltages of germanium transistors normally range from 0.1 to 0.2 volt; silicon transistors normally employ base-emitter voltages from 0.4 to 0.6 volt. A field-effect transistor (FET) differs from a junction transistor in that the control electrode (gate) does not draw current and is not forward biased. In this respect, an FET is similar to a triode tube. There are three basic types, which can be tested with an ohmmeter as shown in Fig. 2-20. Note the basic FET circuit in Fig. 2-21; the source electrode is self-biased at 0.8 volt, in the same manner as a vacuum tube.

When a frequency-influencing component such as a transistor is replaced, particularly in a high-fre-



(A) Schematic.



(B) Output characteristics with load line drawn.

Fig. 2-21. Basic FET voltage amplifier.

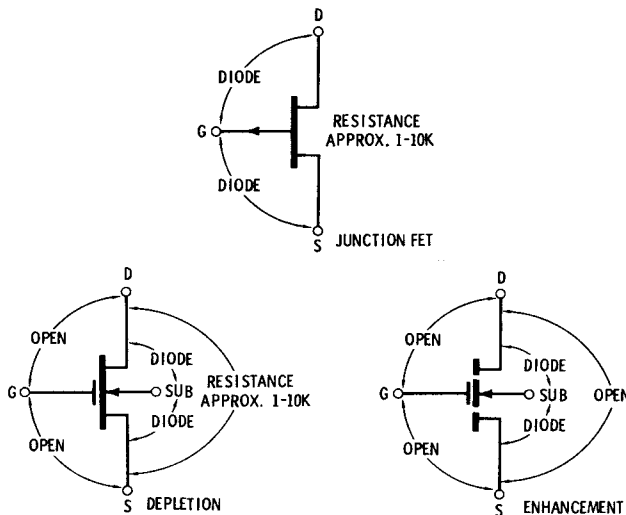


Fig. 2-20. FET resistance checks.

quency circuit such as the rf amplifier or oscillator, realignment is often required to restore hi-fi performance and/or proper tracking. Although alignment of the fm i-f amplifier is less likely to be affected by transistor replacement, because of its lower operating frequency, it is sometimes necessary to touch up the alignment adjustments in the associated stage. The effect of replacement transistors on tuned-circuit alignment is a consequence of the tolerance on junction capacitances. Replacement coupling capacitors in the rf and oscillator circuits also have a noticeable effect on alignment and tracking, because of tolerances on capacitance value. A 5-pF capacitor has a reactance of approximately 300 ohms at 100 MHz, and a 10 percent tolerance represents a range of reactance variation from 270 ohms to 330 ohms. In turn, this tolerance corresponds to a frequency varia-

tion of approximately 10 percent above or below the normal operating frequency.

## **ANALYSIS OF COMMON SYMPTOMS**

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

### **1. No FM Reception**

When there is no fm reception, but the other functions of the combination are normal, the logical conclusion is that the signal is being stopped in the fm tuner. In the case of a tube-type tuner, we check the tubes first.

Possible causes of a dead fm tuner are as follows:

- a. Defective or erratic function switch (see Fig. 2-19).
- b. Oscillator drop-out. Make a quick check by feeding a cw signal from a generator into the antenna-input terminals at the proper frequency.
- c. Shorted capacitor, such as C8 in Fig. 2-7.
- d. Open capacitor, such as C34 in Fig. 2-7.
- e. Leaky capacitor, such as C41 in Fig. 2-7.
- f. Defective transistor; make control-action test, measure dc terminal voltages, or use in-circuit transistor checker.
- g. Break in PC wiring; make voltage measurements and continuity checks of suspected conductors.

In case signal-injection or signal-tracing tests show that the signal is stopped by an IC unit (see Fig. 2-7), first check the capacitors associated with the IC terminals, to avoid the possibility of false conclusions. Most defects in IC packages are accompanied by incorrect terminal voltages, and often by abnormal or subnormal current drain.

### **2. Weak and/or Noisy Output**

When weak or noisy output, or both, are present on the fm function, with normal operation on the a-m function, the i-f transistors that are common to the fm and a-m functions are eliminated from suspicion. In an arrangement such as is shown in Fig. 2-7, the sections that can cause the foregoing symptoms are the fm rf, converter, third fm i-f, or ratio-detector. Stage-gain tests are the most useful approach, although signal-injection tests will often serve to localize a weak stage. A noisy stage is usually associated with low gain, although there are some marginal defects that produce noise without impairing gain.

A useful quick check consists of shunting a large bypass capacitor from the base of each transistor to ground, starting at the last i-f stage and proceeding toward the front end. When the noisy stage is passed, the bypass capacitor will not "kill" the noise.

Possible causes of weak and/or noisy output are:

- a. Defective transistor; check oscillator-mixer first.
- b. Unstable capacitor (C8 in Fig. 2-7 is a typical example).
- c. Open or shorted ratio-detector diode.
- d. Unsoldered or cold-soldered connection to PC board.
- e. Misaligned rf, i-f, or oscillator tuned circuits (weak output only).
- f. Defective i-f transformer (tuned circuits cannot be aligned properly).
- g. Reflected trouble from fm agc section. (Check tuner operation using over-ride agc bias).

An fm i-f transformer can develop more than one type of defect. For example, an internal fixed capacitor can become open, shorted, or leaky. A winding can be burned out if excessive battery or power-supply voltage is accidentally applied. Windings that have been overheated (although not burned out) can develop shorts or leakage between various turns or between layers. Leakage can be caused in humid weather by absorption of moisture. We occasionally find a leakage condition between primary and secondary; this shows up as abnormal transistor terminal voltages, along with the weak-output symptom. Whenever a tuned transformer is replaced, be sure that the shield is grounded properly, and align the stage carefully.

### **3. Poor Sound Quality**

Poor sound quality will be tracked down to a defective component unless there is an operating fault such as low supply voltage. If poor sound quality is caused by inadequate frequency response, the defect will show up on a frequency-response test (Fig. 2-18), but will not show up on a harmonic-distortion test (Fig. 2-12). The only exception is a misaligned ratio-detector circuit, particularly a misaligned secondary; this misalignment condition will show up on a harmonic-distortion test. If poor sound quality is caused by nonlinear operation, the defect will show up on a harmonic-distortion test, but not on a frequency-response test. The sure way to localize either type of defect is to use signal injection tests and proceed step-by-step back from the ratio detector, checking frequency response and harmonic distortion.

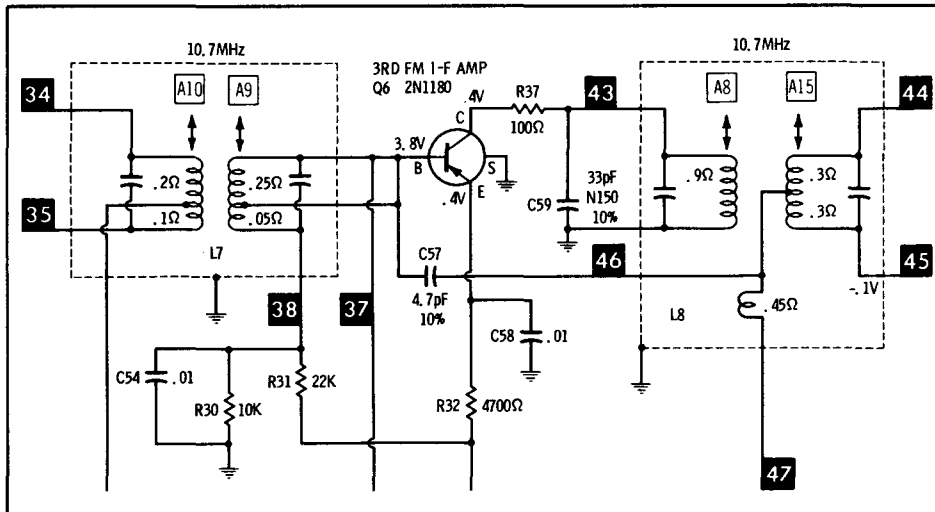


Fig. 2-22. Schematic showing neutralizing capacitor C57.

Possible causes of poor sound quality are:

- Defective neutralizing capacitor (see C57, Fig. 2-22).
- Resistor greatly increased in value, such as R104 in Fig. 2-7.
- Defective or unbalanced components in secondary of ratio-detector circuit.
- Failing transistor; check terminal voltages.
- Rf, i-f, or oscillator misalignment.
- Incorrect replacement transistor.

#### 4. AFC Failure

The basic trouble symptom in case of afc failure is lack of control over local-oscillator frequency drift. However, there are two ways in which the defect may affect the signal when the afc switch is turned on and off. Thus, the signal may disappear when the afc switch is set to its "on" position. In most cases, the signal will reappear when the receiver is retuned, but the basic lack of frequency-drift control will be observed. Again, the signal may be unchanged when the afc switch is turned on, but with the basic lack of frequency-drift control. Since the afc section is actuated by the ratio detector, substantial defects in ratio-detector operation will also affect afc action.

Possible causes of afc failure are:

- Defective afc diode, such as X3 in Fig. 2-7.
- Leaky capacitor, such as C42 or C43 in Fig. 2-7.
- Worn or erratic afc switch.
- Incorrect type of replacement afc diode.
- Resistor greatly increased in value, such as R39 in Fig. 2-7.

When the afc system is operating properly, it will "pull" the oscillator on-frequency when mistuned to either side of a signal. That is, if the afc switch is

turned off, and an fm station is tuned in so that it sounds distorted, the distortion normally clears up when the afc switch is turned on. Various fm tuners have different pull-in ranges; therefore, it is helpful to become familiar with current models so that a practical evaluation can be made of the pull-in action.

#### 5. Co-Channel and Adjacent-Channel Interference

Co-channel interference is not always the fault of the tuner; for example, if a receiver is located approximately halfway between two fm stations of equal power and operating on the same channel, co-channel interference can be anticipated. In this situation, a directional antenna with a rotor-motor is the only practical solution to the interference problem. Similarly, adjacent-channel interference is not always the fault of the tuner—if an omnidirectional high-gain antenna is used, it may be impossible to prevent powerful adjacent-channel fm stations from "feeding through" when the receiver is tuned to a weak station. In this situation, the practical solution is a directional antenna with a rotor-motor.

Possible causes of co-channel and adjacent-channel interference from tuner defects are:

- Unbalanced or defective components in ratio-detector circuit.
- Fm agc trouble, causing rf stage to operate "wide open" continuously.
- Previous sloppy alignment job, or component replacement in tuned circuit without follow-up alignment check.
- Defective rf or i-f transformer (cannot be aligned properly).
- Incorrect type of replacement rf transformer.
- Incorrect replacement type of built-in antenna.
- Defective local-distance switch.

**6. Incorrect Dial Calibration**

Since fm tuners employ the superheterodyne principle, the same basic causes of incorrect dial calibration apply that were noted previously for a-m tuners:

- a. Dial-cord stringing incorrect, slack, or loose. See receiver service data.
- b. Dial plate incorrectly mounted or shifting in position.
- c. Local oscillator incorrectly aligned. Check tracking per receiver service data.
- d. Defective capacitor in oscillator section.
- e. Shorted turns or other defect in oscillator transformer.
- f. Incorrect type of replacement transistor in oscillator circuit.
- g. Incorrect type of replacement tuning capacitor.

**7. Intermittent Operation**

Whenever a puzzling intermittent occurs in an fm tuner, the preliminary tests should be supplemented by monitoring the various circuit sections for the necessary length of time. As in the case of an a-m tuner, specialized monitoring instruments can be utilized, or the available meters and scopes can be put to use. It is helpful to energize the fm tuner with a steady signal from an fm generator. The basic monitoring points are at the rf amplifier output, i-f amplifier input, i-f amplifier output, and ratio-detec-

tor output. When the intermittent occurs, the instruments provide localization data without disturbing the circuits. If necessary, the monitoring instruments can be connected next at various points in the defective section, to close in on the defective component.

Possible causes of intermittent operation in an fm tuner are:

- a. Local-oscillator drop-out, often due to a marginal defect in the transistor.
- b. Mechanical intermittent such as a poor connection; this type of intermittent can often be localized by tapping.
- c. Microscopic break in PC wiring; can usually be confirmed by flexing the PC board slightly up and down.
- d. Thermally intermittent resistor or transistor; localize by using a heat lamp, followed by a spray coolant.
- e. Intermittent capacitor; use monitoring procedures to pinpoint.
- f. Marginal defect in tuned coil or transformer; tap with a tuning rod or pencil; press assembly gently from side to side; press terminals while observing output from the tuner.
- g. Broken conductor inside of an insulated lead; flex lead while monitoring.
- h. Frayed insulation, permitting intermittent short-circuit.
- i. Plug not fully inserted into receptacle.

## Chapter 3

# Stereo Multiplex Troubleshooting

Basically, an fm-stereo signal consists of two different audio-frequency signals that occupy the same fm channel. These separate audio signals provide stereophonic sound reproduction. The individual audio signals are identified as “left” (L) and “right” (R). In conventional programming, this pair of audio signals originates from a pair of microphones at a sound studio, as shown in Fig. 3-1. The audio signal from the L microphone differs from that of the R microphone. Therefore, the stereo signal consists of two audio waveforms that vary independently in frequency and amplitude. At the receiver, the L and R signals are fed to separate speakers. These speakers are spaced a suitable distance apart to simulate the placement of the transmitting microphones. Fig. 3-2 shows a block diagram of a typical am/fm stereo chassis. The multiplex section in this example comprises the bandpass amplifier, 19-kHz pilot amplifier, 38-kHz oscillator, stereo indicator, detector, and matrix sections.

Common trouble symptoms caused by defects in the multiplex sections are:

1. No output.
2. One channel dead.
3. Weak output.
4. Poor separation.
5. Distorted output.
6. Separation-control setting drifts.
7. Stereo indicator failure.

### GENERAL DISCUSSION

We know that in monophonic fm transmission, a swing of  $\pm 75$  kHz represents full modulation and produces sidebands that occupy the entire channel. For

this reason, the question arises how one channel can be used to transmit two signals without mutual interference. If high fidelity were not required, an fm channel could be divided into two equal parts for transmission of the L and R signals on individual carriers. However, since hi-fi transmission is a basic requirement, a modulation method must be employed that permits each signal to occupy the entire channel bandwidth.

To transmit both the L and R signals with high fidelity in a single channel requires multiplex transmission. This is a technique that permits satisfactory separation of the L and R signals at the receiver. Standard multiplex transmissions are also compatible, which means that to a conventional mono receiver, the fm multiplex signal “looks like” a mono transmission. But to a stereo-multiplex receiver, the fm multiplex signal “looks like” separate L and R signals. A stereo-multiplex system starts with the conventional mono audio signal, which is produced as the sum of L and R signals. That is, two microphones are employed as the equivalent of a single microphone. When the L and R signals are mixed, as shown in Fig. 3-3, the mono (L + R) signal is produced.

The L + R signal is frequency modulated on the rf carrier, and the result is the same as if a single microphone were used. Furthermore, to an ordinary fm receiver, only a mono signal is being transmitted—actually, as explained next, additional information to which an ordinary fm receiver is unresponsive is also being transmitted. To explain this additional transmitted information, let us consider the effect of adding a 38-kHz carrier as shown in Fig. 3-4. Both the L + R signal and the 38-kHz carrier are frequency-modulated on the rf carrier. However, only

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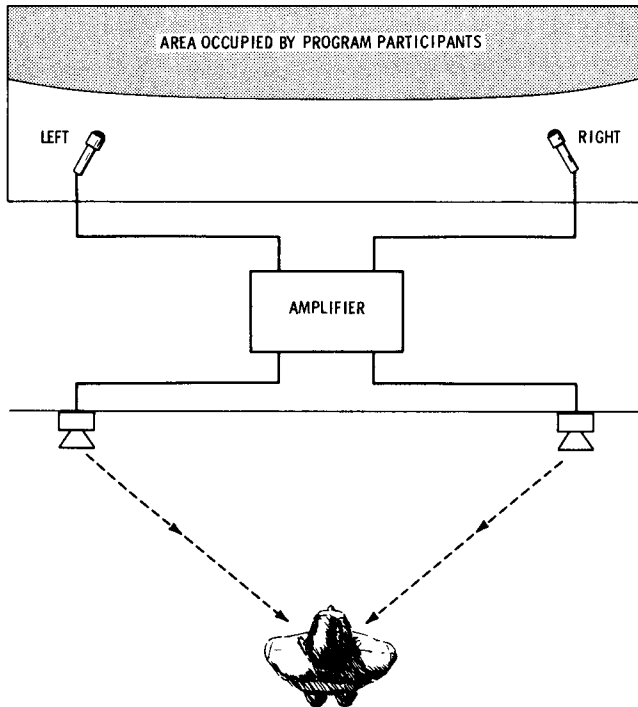


Fig. 3-1. Basic principle of stereophonic sound reproduction.

the L + R signal can be reproduced at the fm receiver. That is, the 38-kHz carrier is out of the range of audibility. This is the fundamental multiplex transmitting principle.

Next, if amplitude modulation is impressed on the 38-kHz carrier, this modulated signal will be inaudible on an ordinary fm receiver. With reference to Fig. 3-5, the L + R signal is frequency-modulated on the rf carrier, as before. In addition, an audio signal, A2, is amplitude-modulated on the 38-kHz carrier; this amplitude modulation produces sidebands, and the rf carrier is modulated by these sidebands along

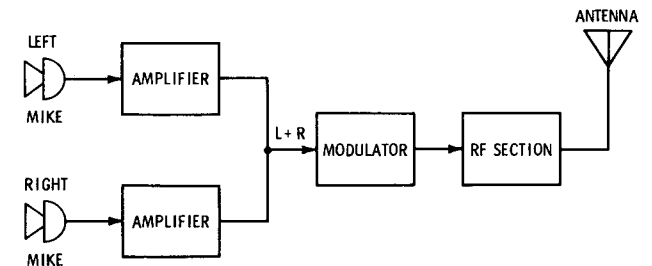


Fig. 3-3. Mixing of L and R signals produces a conventional monaural signal.

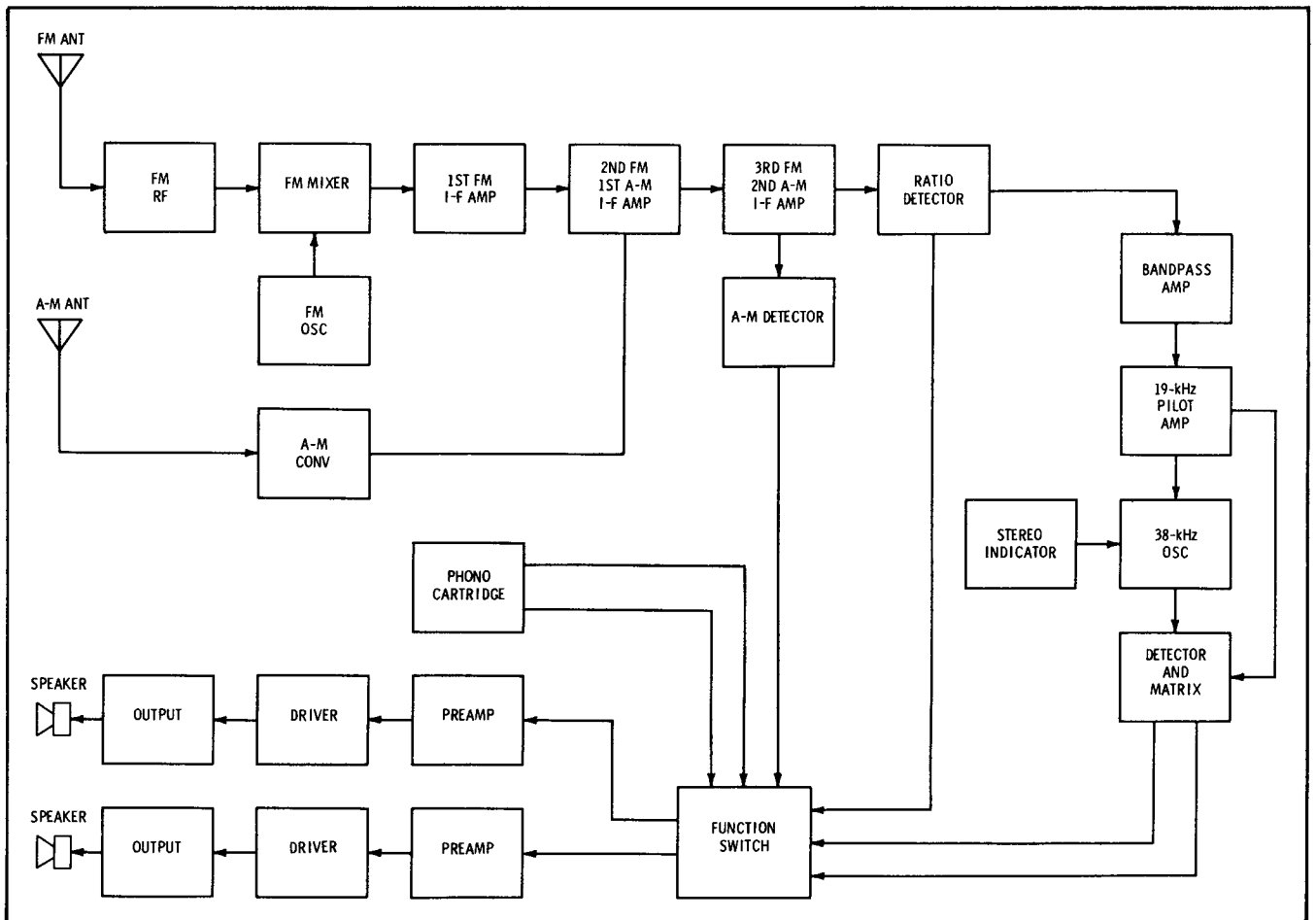
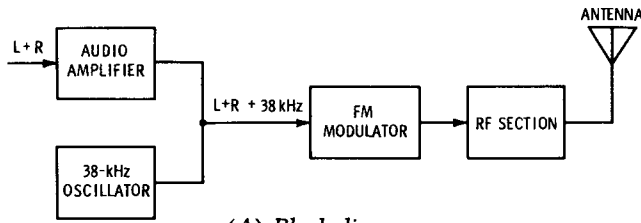
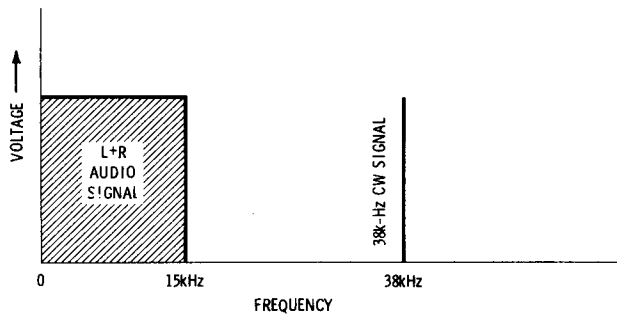


Fig. 3-2. Block diagram of a typical am/fm stereo chassis.



(A) Block diagram.



(B) Transmitted frequency spectrum.

**Fig. 3-4. L + R and 38-kHz signals modulated on an rf carrier.**

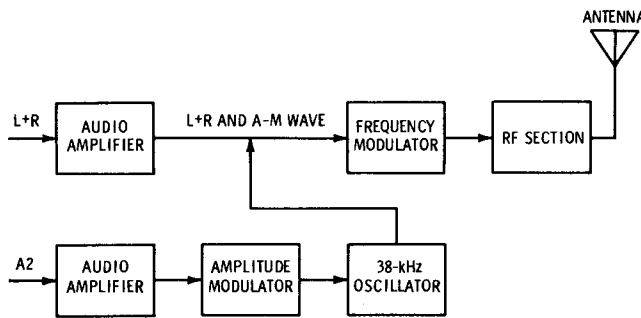
with the L + R signal. Fig. 3-5B shows the frequency spectrum of the modulating waveforms. The L + R signal has frequencies up to 15 kHz. On the other hand, the amplitude-modulated 38-kHz carrier has frequencies in the range from 23 kHz to 53 kHz. This represents an audio-frequency range up to 15 kHz.

Note that after the foregoing frequency spectrum is frequency-modulated on the rf carrier and then processed through the ratio detector of an ordinary

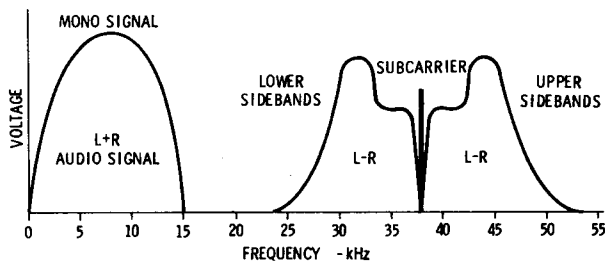
fm receiver, the frequency spectrum is recovered in its original form, as shown in Fig. 3-5B. Of course, only the L + R signal is reproduced by an ordinary fm receiver. In other words, the frequencies from 23 kHz to 53 kHz will be rejected by the audio system in an ordinary fm receiver; even if an extended-range audio amplifier were used, the reproduced tones would be inaudible to the ear. In multiplex terminology the frequencies from 23 kHz to 53 kHz are said to be "encoded" in the radiated fm signal.

Now, let us see how the encoded signal is recovered. Fig. 3-6A shows a bandpass filter with response from 23 kHz to 53 kHz, driven by the ratio detector, and followed by an a-m detector and a speaker. The bandpass filter picks out the a-m signal, which is then demodulated by the a-m detector. In turn, the demodulated (audio-frequency) wave envelope is fed to a speaker. This is the A<sub>2</sub> signal shown in Fig. 3-5. Note that the output from the ratio detector in Fig. 3-6B is a mixture of the L + R and the A<sub>2</sub> a-m signal. Thus, the bandpass filter provides separation of the signals in this example. These basic functions are elaborated somewhat in actual practice so that the L signal will be reproduced by one speaker, and the R signal reproduced by another speaker. Basically, an L - R signal is generated by means of a phase inverter at the transmitter, as shown in Fig. 3-7.

Observe that by inverting the polarity of the R signal and then adding it to the L signal, we obtain an L - R signal. In turn, both L + R and L - R sig-

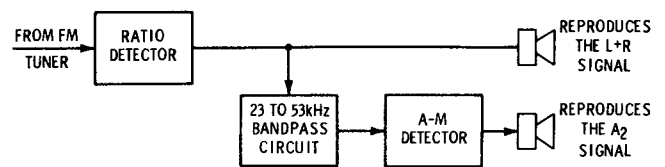


(A) Block diagram.

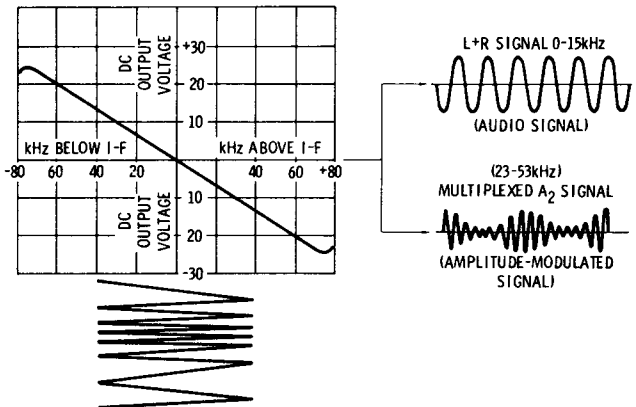


(B) Frequency spectrum.

**Fig. 3-5. Amplitude-modulated 38-kHz subcarrier combined with L and R signals.**



(A) Basic decoding system.



(B) Ratio-detector action.

**Fig. 3-6. Method for reproducing the multiplexed signal.**

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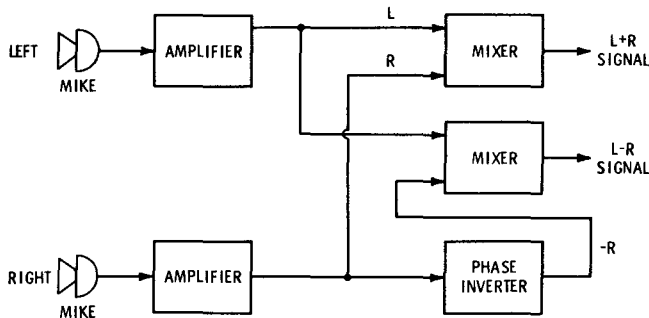


Fig. 3-7. Developing the L + R and L - R signals.

nals are available for processing. If we add  $L + R$  to  $L - R$ , we will obtain  $2L$  (the  $L$  signal with the  $R$  signal cancelled or separated). Again, if we subtract  $L - R$  from  $L + R$ , we will obtain  $2R$  (the  $R$  signal with the  $L$  signal cancelled or separated). Signal addition or subtraction is accomplished in mixers, with associated inverters. After the  $L$  and  $R$  signals have been separated from the  $L + R$  and  $L - R$  signals, we can feed the  $R$  signal to one speaker, and feed the  $L$  signal to the other speaker, thus obtaining stereo reproduction.

At the transmitter, the  $L + R$  and  $L - R$  signals are employed to frequency-modulate the rf carrier, as shown in Fig. 3-8. The  $L - R$  signal is amplitude-modulated on a 38-kHz carrier, and the resulting signal is mixed with the  $L + R$  signal. In turn, the mixed signals frequency-modulate the rf carrier. Fig. 3-5B shows the frequency spectrum of the modulating signal, wherein the upper and lower sidebands flanking the 38-kHz carrier are produced by the  $L - R$  signal. At the receiver, this same frequency spectrum appears at the output of the ratio detector.

To obtain stereo reproduction, the  $L - R$  signal is separated by means of a 23- to 53-kHz bandpass filter, amplitude-demodulated, and further processed in a phase inverter and a pair of mixers, as shown in Fig. 3-9. The addition of  $L + R$  and  $L - R$  produces a  $2L$  signal. Subtraction of  $L - R$  from  $L + R$  (the

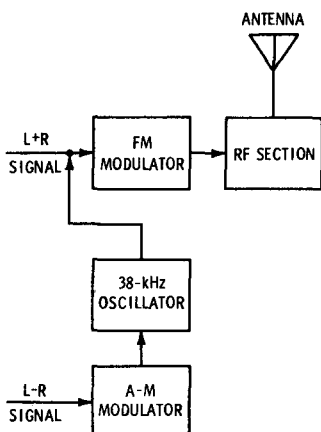


Fig. 3-8. Generation of the composite stereo signal at the transmitter.

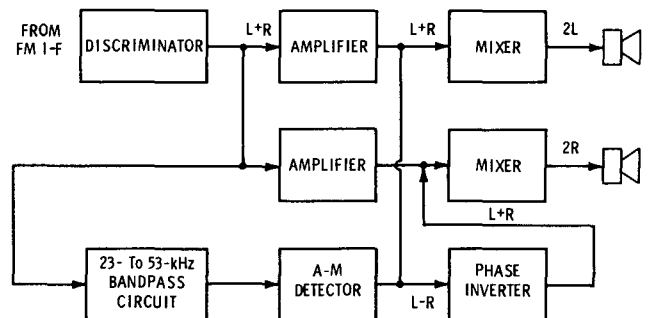


Fig. 3-9. Block diagram of multiplex adapter for stereo reproduction.

same as adding  $L + R$  and  $-L + R$ ) produces a  $2R$  signal. Thereby, stereo reproduction is obtained from the speakers. Hi-fi reproduction results if this processing is normal.

In theory, the 38-kHz carrier (technically termed the subcarrier) could be transmitted. However, the subcarrier is suppressed in practice, thereby permitting the  $L - R$  sidebands to be transmitted at a higher level, which improves the signal-to-noise ratio. In turn, the 38-kHz subcarrier must be reinserted at the receiver. This is accomplished by mixing a locally generated subcarrier with the  $L - R$  sidebands. This mixing process must be done at exactly the correct frequency and the exactly correct phase, in order to provide hi-fi reproduction. To permit precise reinsertion of the missing subcarrier at the receiver, a 19-kHz pilot subcarrier is transmitted. Fig. 3-10 shows how the 19-kHz pilot subcarrier is transmitted in an empty portion of the spectrum between the  $L + R$  signal and the lower sideband of the  $L - R$  signal.

The pilot subcarrier is transmitted at relatively low power, and because it is separated from the  $L + R$  and  $L - R$  signals, it can be easily trapped at the receiver free from interference. Thus, both a 38-kHz and the related 19-kHz subcarrier are generated at the transmitter, but the 38-kHz subcarrier is trapped out prior to transmission, so that the  $L - R$  signal is radiated as a suppressed-carrier signal. At the receiver, a tuned trap picks out the pilot sub-

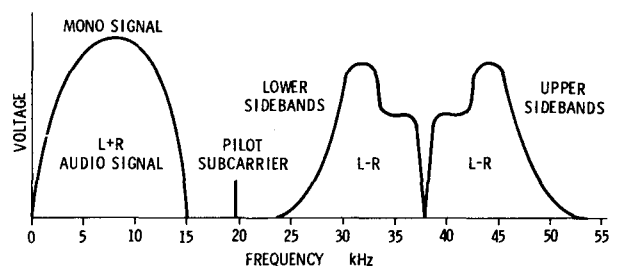


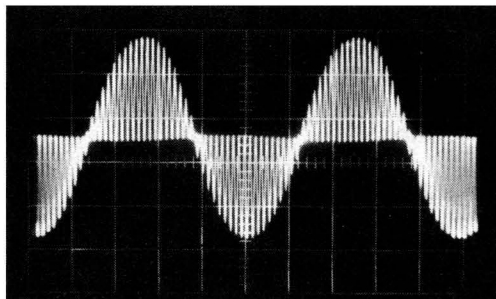
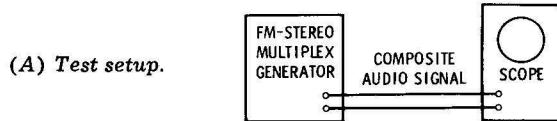
Fig. 3-10. Composite stereo signal showing 19-kHz pilot subcarrier in empty portion of spectrum.



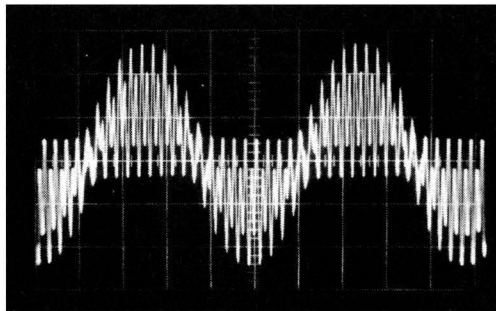
carrier and feeds it to a doubler, thereby developing the missing 38-kHz subcarrier. This 38-kHz subcarrier is then mixed with the L - R sidebands, and the end result is the same as if the complete L - R signal had been transmitted.

**Stereo-Multiplex Generators**

Stereo-multiplex troubleshooting requires the use of a suitable generator. A basic generator supplies an L + R signal, an L - R signal, and a 19-kHz pilot subcarrier. The L + R and L - R signals have a



(B) Normal 1-kHz composite audio signal.



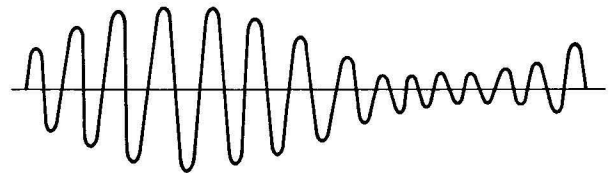
(C) Normal composite signal with 19-kHz pilot subcarrier.

**Fig. 3-11. Checking the output from a multiplex generator.**

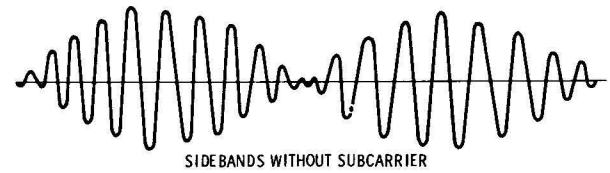
1-kHz audio frequency in the basic generator; the more elaborate generators may also provide 400-Hz and 5-kHz tones. Most generators include a 100-MHz rf oscillator, which may be frequency-modulated by the composite audio signal (Fig. 3-11). The composite audio signal is fed directly into a multiplex adapter (or combo section) for tests. On the other hand, an fm rf signal is required when the tests or measurements are made through the rf tuner. We will find that an L + R signal or an L - R signal seem to have the same waveform; this is because the only difference between them is their phase with respect to the pilot subcarrier. Let us see how these waveforms are produced by the generator.

With reference to Fig. 3-12, we observe:

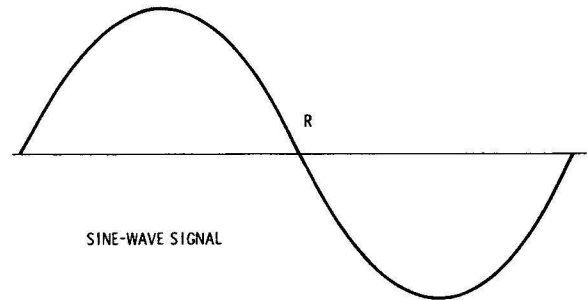
1. The waveshape of the subcarrier and both sidebands is the same as that of an ordinary amplitude-modulated signal.
2. When the subcarrier is suppressed, the sidebands without the subcarrier form an envelope that has twice the frequency of the original envelope.
3. In the generator, an L + R (or L - R) signal



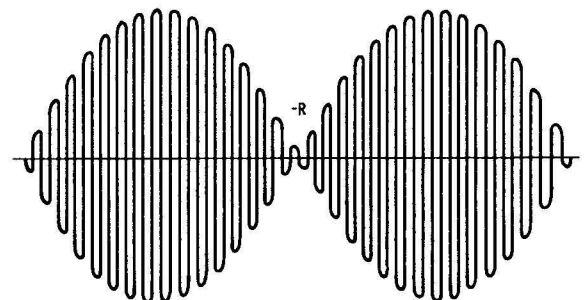
SUBCARRIER AND BOTH SIDEBANDS



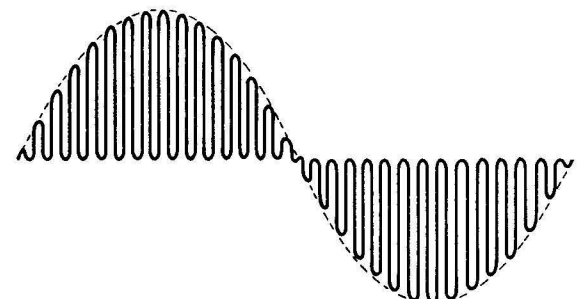
SIDEBANDS WITHOUT SUBCARRIER



SINE-WAVE SIGNAL



SIDEBANDS WITHOUT SUBCARRIER



SINE-WAVE SIGNAL PLUS UPPER AND LOWER SIDEBANDS

**Fig. 3-12. Development of the composite audio signal.**

## Hi-Fi Stereo Servicing Guide

is produced by modulating a 38-kHz sine wave by a 1-kHz sine wave, and then suppressing (removing) the 38-kHz frequency.

- Next, to develop an R signal, we may consider the result of energizing the R microphone only in Fig. 3-7. In turn, with reference to Fig. 3-12, we see that the R sine-wave signal is mixed with the  $-R$  signal which consists of sidebands without subcarrier. This mixture produces the sine-wave signal plus upper and lower sidebands.
- If an L signal is to be developed, the end result is the same, except for a phase difference, since the output from the L microphone in Fig. 3-7 does not pass through a phase inverter.

In a stereo-multiplex generator, of course, the L and R signals are produced by audio oscillators instead of microphones. The most basic test of a multiplex adapter is its ability to separate  $L + R$  and  $L - R$  signals. Fig. 3-13 shows a typical test setup.

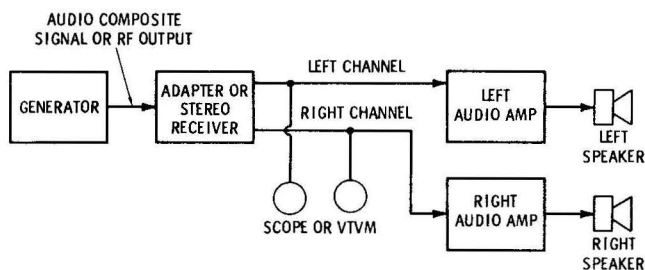


Fig. 3-13. Setup for testing separation in a stereo system.

If a multiplex adapter is being checked without the fm tuner, we apply the composite audio signal to the adapter input terminals. Either a scope or an ac VTVM can be used as an indicator. In theory, when we apply an R-channel signal to the adapter, we would observe full output from the R section and zero output from the L section, as shown in Fig. 3-14A and B. In practice, however, separation cannot be perfect, and the L channel output (undesired output) might appear as seen in Fig. 3-14C. Similarly, when an L-channel signal is applied to the adapter, it is impractical to reduce the undesired output from the R channel to zero.

A stereo-multiplex generator provides a test signal that has virtually complete separation, unless a defect occurs in the generator. In turn, we are concerned with how many decibels of separation should be normally provided by a multiplex adapter. A high-quality multiplex adapter (or multiplex section in a combo) will provide approximately 30dB of separation. It is generally considered that a separation of 10 dB is barely tolerable. Note that 30 dB

corresponds to a voltage ratio of more than 30 between the L and R channels. On the other hand, 10 dB corresponds to a voltage ratio of approximately 3 between the L and R channels. In the example of Fig. 3-14, we have a voltage ratio of 7 to 1 between (a) and (c); this is equal to about 17 dB, and represents average multiplex-adapter performance. A VTVM with a dB scale is illustrated in Fig. 3-15; this provides convenience in making separation tests. Note that a multiplex adapter or section is basically a signal processor and provides practically no gain from input to output.

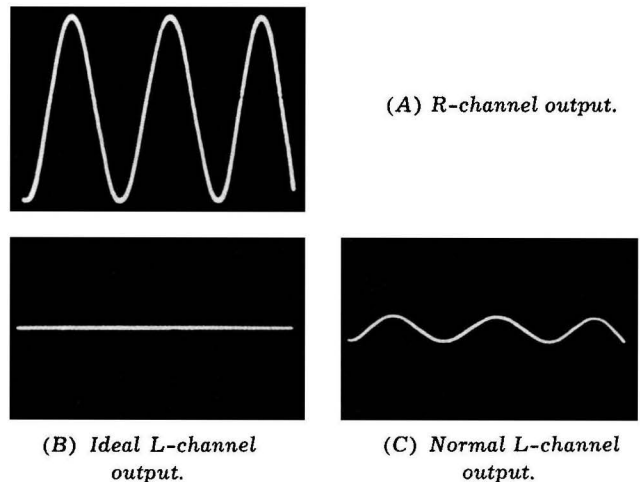


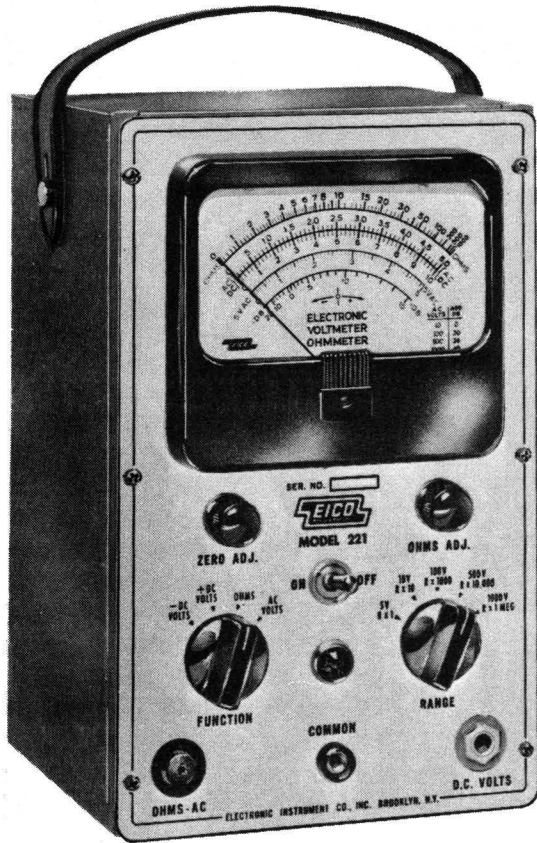
Fig. 3-14. Waveforms observed during separation test.

### Defects That Cause Poor Separation

Although poor separation is usually caused by a component defect, drift in tuned-circuit adjustments can also impair separation. Four tuned circuits are utilized in a typical multiplex adapter, as shown in Fig. 3-16; the corresponding circuit is seen in Fig. 3-17. The composite stereo signal is amplified by V1A and then fed to the grid of phase-inverter V2A. V2B operates as a mixer for the two outputs from V2A after sampling by the ring demodulator consisting of X3 through X6. Always check tubes first. If V2 is not defective, check the other tubes also. Next to tubes, capacitors are prime suspects. A systematic approach in this sample entails checking C8, C11, C9, C10, and C15.

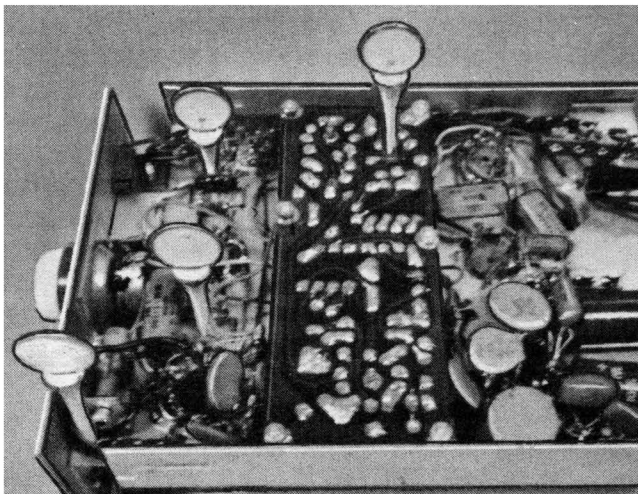
In the event that no capacitor defects are found, the next step is to check (or replace) the diodes in the ring demodulator. Diodes X3, X4, X5, and X6 must have good front-to-back ratios, and must also be reasonably well matched to obtain optimum separation. Another possibility of poor separation is a faulty frequency-doubling diode, such as X1 or X2; these diodes change the 19-kHz pilot subcarrier into a 38-kHz subcarrier. Resistors are less common

## Stereo Multiplex Troubleshooting



Courtesy EICO Electronic Instrument Co., Inc.  
**Fig. 3-15. VTVM with a dB scale.**

troublemakers, but become ready suspects in case a shorted tube has been replaced. For example, R33 and R34 can be damaged by overloads. If R34 is increased in value, V2B becomes incorrectly biased



**Fig. 3-16. Markers show tuned-circuit locations in a typical multiplex adapter.**

and does not operate as a linear mixer; in turn, separation is impaired. The setting of R31 determines the amount of  $L + R$  signal that is applied to mixer V2B, and thereby affects separation. In case R31 checks out satisfactorily, we would check R35 as a possible troublemaker.

Alignment of a multiplex adapter or section is checked last, because this is statistically the least likely cause of poor separation (unless a do-it-yourselfer has been tampering with the alignment adjustments). Alignment of the adapter in Fig. 3-17 is comparatively simple; T2 and T4 are the most critical circuits, and are peaked at 19 kHz and 38 kHz, respectively. To align T2, we inject an unmodulated 19-kHz signal at the grid of V1, and connect an ac VTVM at the plate of V4. The slug in T2 is then adjusted for maximum output. To align T3, inject a 19-kHz signal at the grid of V4 and connect an ac VTVM at the plate of V5A. The slug in T3 is then adjusted for maximum output. To align T4, we inject a 38-kHz signal at the grid of V5A, and connect an ac VTVM at the junction of X5 and X6. Then, the slug in T4 is adjusted for maximum output. Sometimes a slight compromise adjustment of T4 will give optimum separation.

Note that T5 in Fig. 3-17 is a 67-kHz trap. It is tuned by a slug (not shown in the diagram). This circuit does not affect separation; instead, its purpose is to suppress "birdies" in case an SCA signal is being transmitted. With reference to Fig. 3-18, an SCA signal employs a 67-kHz subcarrier. A subsidiary-carrier-assignment (SCA or storecasting) signal cannot be heard on a conventional stereo multiplex receiver, unless a special SCA adapter is used. However, an SCA signal can sometimes cause "birdies" in conventional fm stereo reception unless the SCA signal is trapped out. To check the alignment of T5 in Fig. 3-17, we feed a 67-kHz signal into the adapter, and connect an ac VTVM at the plate of V1A. Then, the slug in T5 is adjusted for minimum output.

Although the end result is the same, the configuration in Fig. 3-17 processes the multiplex waveform in a different manner from the arrangement in Fig. 3-9. That is, the multiplex signal is *sampled* in Fig. 3-17. It is important that this process be understood. The waveforms shown in Fig. 3-19 show how the envelopes above and below the zero axis provide the L and R signals after the 38-kHz subcarrier is inserted. Thus, the positive and negative excursions of the subcarrier serve to develop the individual L and R signals in the bridge circuit of Fig. 3-17. We will find another processing method called the envelope-detector configuration; this method employs

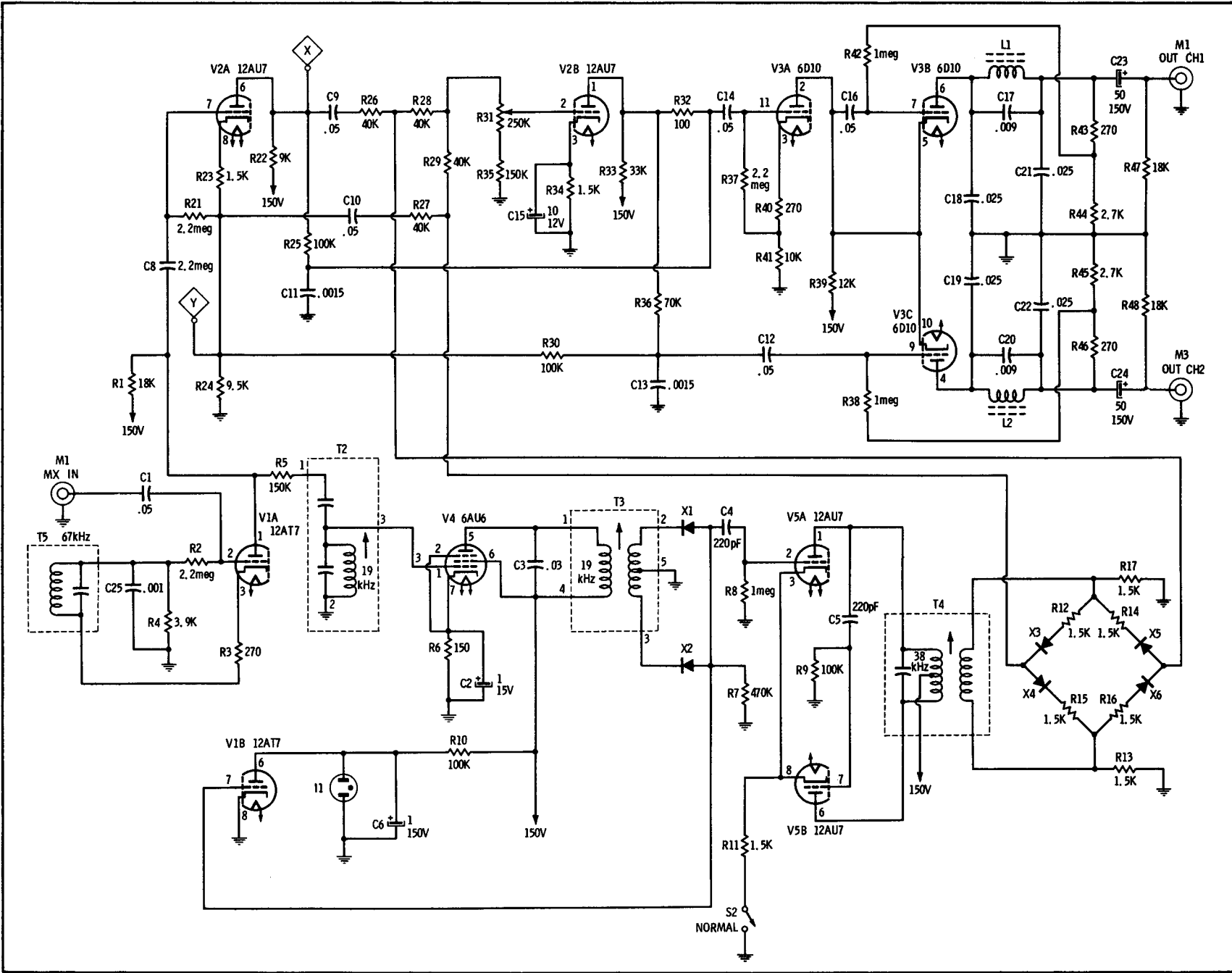
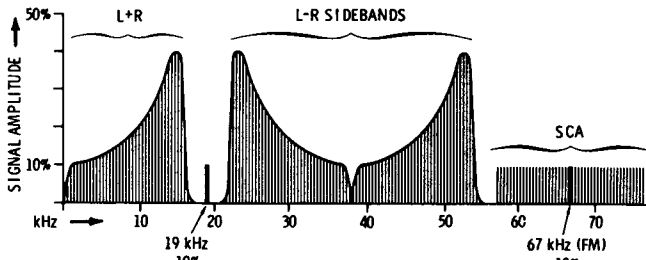
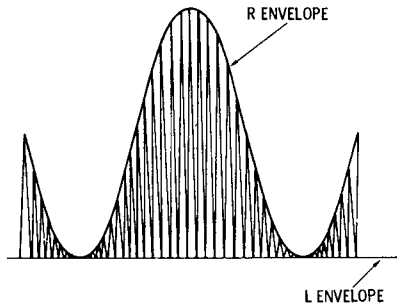


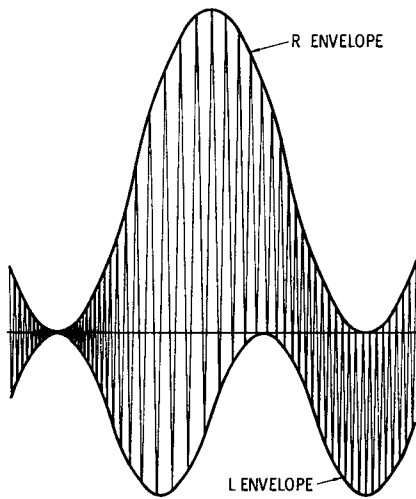
Fig. 3-17. Schematic of a typical tube-type stereo adapter.



**Fig. 3-18.** Frequency spectrum of signals applied to modulator in fm transmitter.



(A) Right microphone energized; composite signal at receiver with subcarrier inserted.



(B) Both microphones energized; composite signal at receiver with subcarrier inserted.

**Fig. 3-19.** Representative multiplex waveforms.

a pair of oppositely polarized amplitude detectors. It is evident that when a multiplex waveform passes through a positively polarized detector, the output waveform will be the R signal. Similarly, the output from a negatively polarized detector will be the L signal.

**Stereo-Indicator Circuits**

Most receivers have lamps that indicate when a stereo broadcast signal is being received. Typical arrangements are shown in Figs. 3-20 and 3-21. The first section, in Fig. 3-20A, is actuated by the pilot-

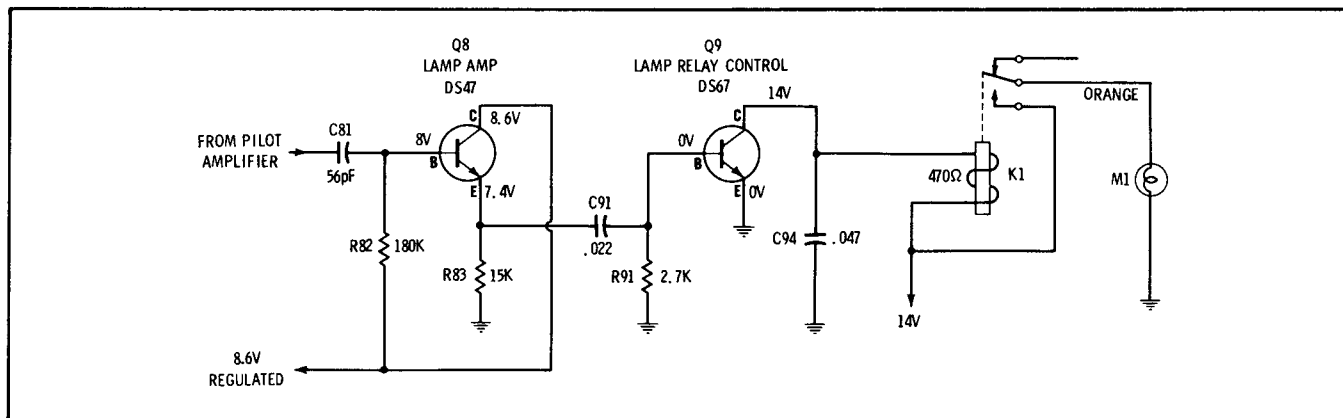
amplifier signal. The input circuit couples 19-kHz energy to the lamp-amplifier stage, whenever a pilot carrier is present. This amplifier is a conventional emitter follower. The output load of the control transistor is a relay coil, which is returned to the collector-supply voltage. When a drive signal is applied, the bias on the control transistor changes, and the collector current increases. In turn, the relay closes and a potential of 12 volts is applied to the indicator lamp.

Next, with reference to Fig. 3-20B, the three stages operate as dc amplifiers. Actuating energy is obtained from the pilot oscillator. In turn, the 38-kHz signal is rectified by twin diodes X14 and X15, which operate in the sensing stage. The negative-going change in dc voltage at the base of Q11 produces an opposite change at its collector. Thus, the base of Q12, connected directly to the Q11 collector, is driven positive. In turn, the voltages at the collector of Q12 and the base of Q13 go negative. This voltage change causes Q13 to conduct heavily. Since there is collector current through the stereo-indicator lamps, their filaments glow as long as the base of Q13 remains negative. Note that the input circuit for Q11 is a filter that smooths the pulsating outputs from the input diodes. Additional filtering is provided by C121.

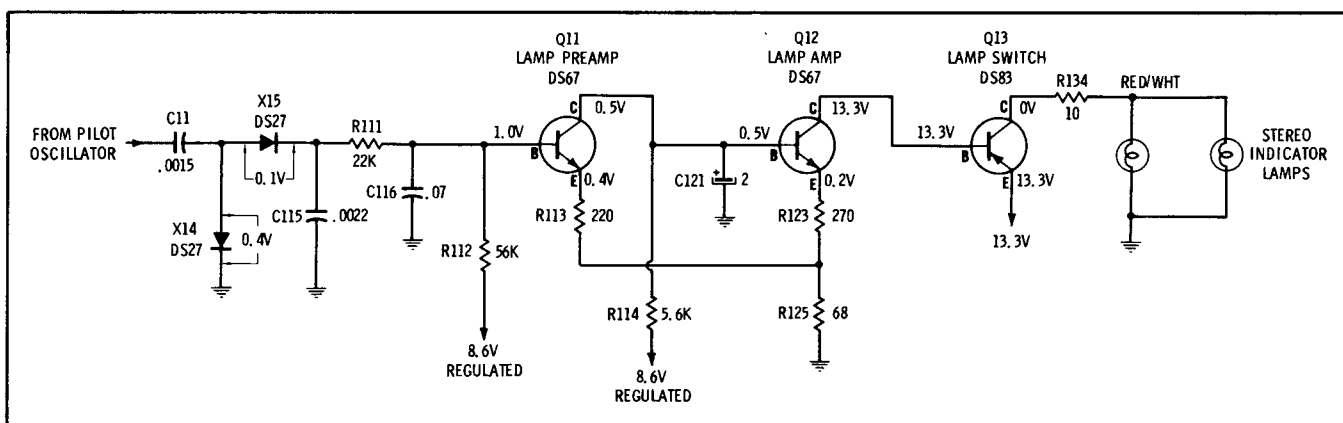
The stereo-indicator arrangements in Fig. 3-21 are comparatively simple. Both of them operate by means of applied dc voltages at the bases of the transistors. In Fig. 3-21A the pulsating-dc output from the pilot-subcarrier doubler circuit is filtered by means of a 5- $\mu$ F capacitor, and drives the base positive, thereby turning the transistor on. The arrangement in Fig. 3-21B is basically similar, except that two stages are provided. Note also that the input is actuated by the output from a balanced stereo detector; therefore, less filtering is required at the base of the input transistor.

**TROUBLESHOOTING PROCEDURE**

Bench work on an operating complaint starts with localization of the defective section. With reference to Fig. 3-2, trouble in the bandpass or matrix section could be confused with trouble in the ratio-detector circuit, or with a defect in one of the audio channels. Therefore, it is advisable to drive the multiplex section directly with the audio composite signal from a multiplex generator. For example, the signal would be applied at M1 in Fig. 3-17. A scope can be connected at M2 for preliminary analysis of the Ch-1 output; next, the scope can be connected at M3 for analysis of the Ch-2 output. This approach will iden-



(A) Actuated by pilot-amplifier signals.



(B) Actuated by pilot oscillator.

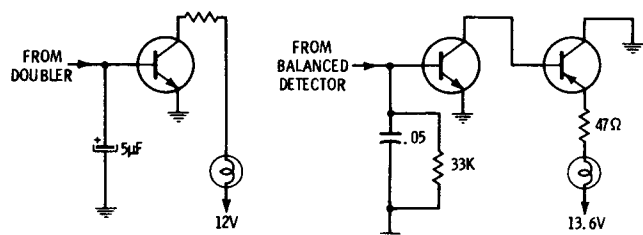
Fig. 3-20. Stereo-indicator light circuits.

tify prominent malfunctions, such as lack of output on one or both channels, poor separation, high hum level, or serious waveform distortion.

On the other hand, less prominent malfunctions such as moderate distortion or poor frequency response require additional tests for definite identification. A harmonic-distortion meter must be connected in place of the scope at the output of the multiplex adapter to measure percentage of distortion. Fig. 3-22 shows the test setup that is used. If it is suspected that the adapter has poor frequency response, the

multiplex generator is externally modulated by means of an audio oscillator, and the output from the adapter is indicated by an ac VTVM as shown in Fig. 3-23. The adapter should provide a reasonably flat frequency response from 20 Hz to 15 kHz, prior to the de-emphasis network.

In the case of a "dead" adapter, or lack of output on one channel, conventional signal-tracing procedures with a scope are useful to localize the defective stage or circuit section. Then, dc voltage and/or resistance measurements will usually serve to close in on the defective component. Most operating troubles in multiplex adapters are caused by leaky or open capacitors. However, a worn and erratic separation control occasionally causes trouble, and may be overlooked by the technician apprentice. Some adapters are designed to operate with a certain value of input-cable capacitance; if a cable with excessive capacitance is substituted, trouble symptoms may be introduced. Note also in Fig. 3-17 that C25 operates in combination with T5 to produce optimum SCA rejection. However, when using some types of fm tuners, C25 may affect the range of the separation



(A) Single-stage design from a Tenna multiplex adapter.

(B) Two-stage design by Motorola.

Fig. 3-21. Two simple stereo-light systems.

## Stereo Multiplex Troubleshooting

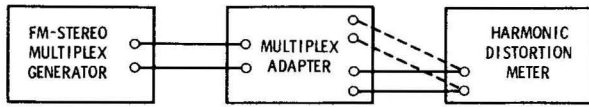


Fig. 3-22. Harmonic-distortion test setup for a multiplex adapter.

control; that is, the separation control might require setting near one end or the other, or might be out of range. In such a case, we vary the value of C25 from 2700 pF to zero, choosing the value that provides midrange setting of the separation control. Finally, T5 should be realigned for optimum 67-kHz rejection.

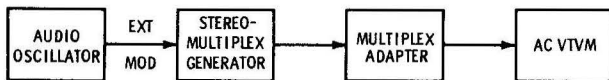


Fig. 3-23. Frequency-response check of a multiplex adapter.

The troubleshooting procedure for a transistor multiplex unit is much the same as for a tube-type unit, since designs are based on a transistor-for-tube basis, as seen in Fig. 3-24. A typical adapter is illustrated in Fig. 3-25. With reference to Fig. 3-24, Q1

is a high-impedance input transistor, with a 19-kHz resonant trap in its emitter circuit. Thus, the 19-kHz pilot subcarrier is dropped across the trap, and does not appear in the collector circuit. SCA signals are rejected by T1. Q2 operates as an emitter-follower

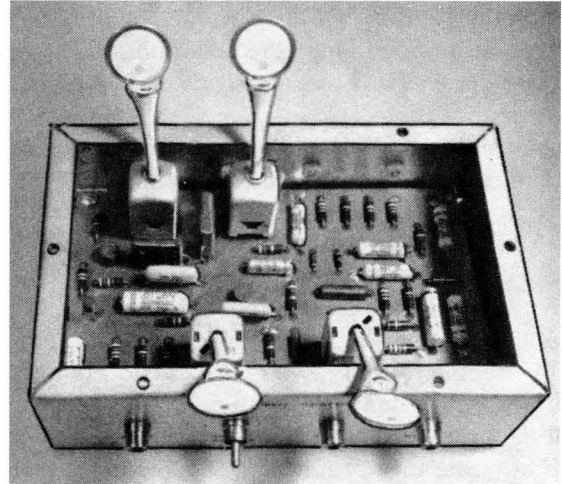


Fig. 3-25. Markers show tuned-circuit locations in a transistor multiplex adapter.

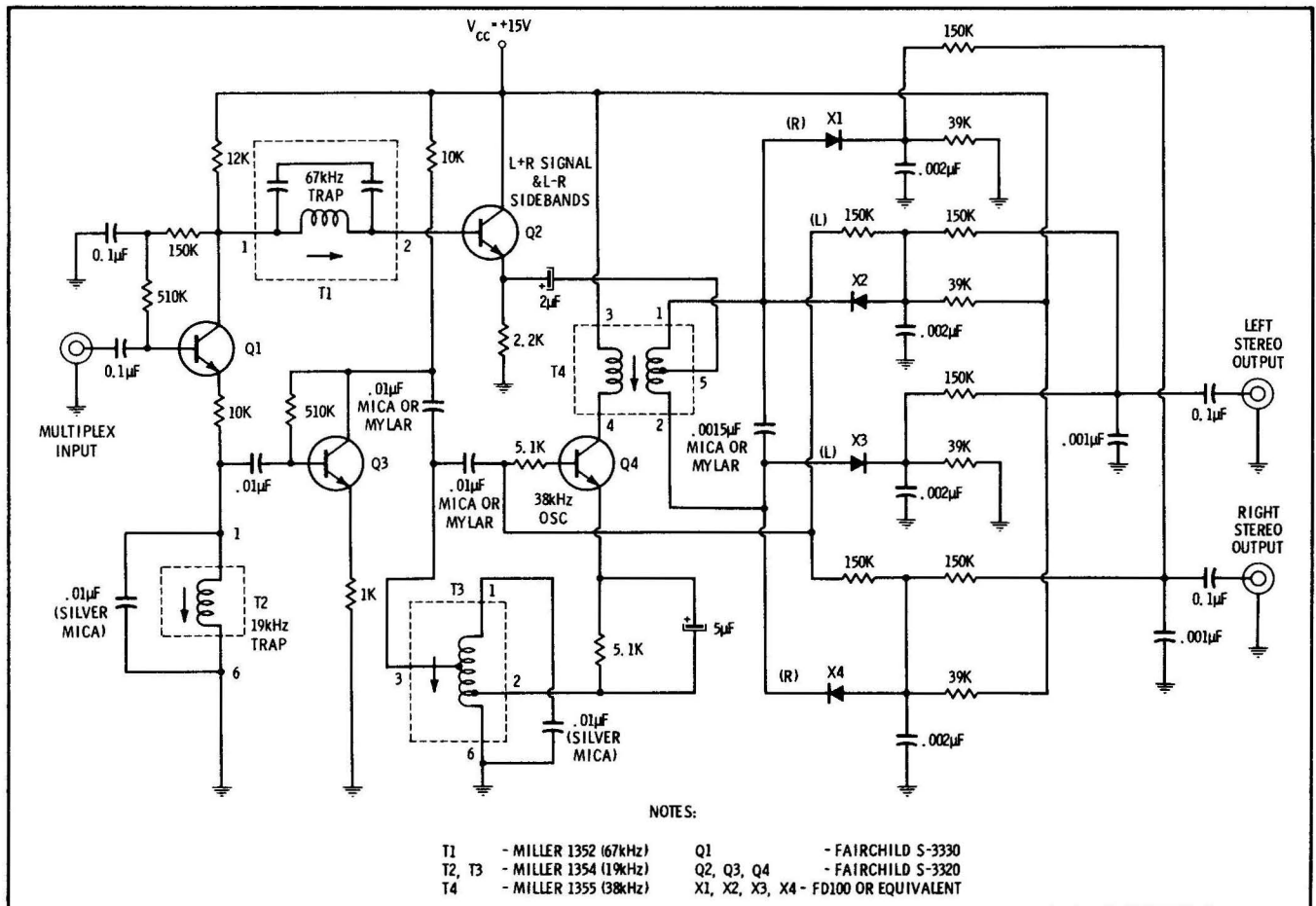


Fig. 3-24. A transistor multiplex adapter.

buffer stage for the multiplex signal. Q3 amplifies the 19-kHz pilot subcarrier, and Q4 operates as a 38-kHz locked oscillator, synchronized by the output from Q3. In turn, the complete multiplex signal appears at the secondary of T4. Diodes X1 through X4 operate in a switching bridge; X1 and X4 provide the R signal output, while X2 and X3 provide the L signal output. Note that the de-emphasis networks comprise series 150-k resistors with 0.001- $\mu$ F shunt capacitors.

In normal operation, we will find a separation of approximately 25 dB between the L and R channels. An input signal level of at least 0.5 volt rms is required to maintain tight locking of the 38-kHz subcarrier oscillator. At lower values of input signal, the oscillator locks erratically, causing serious distortion and noise in the output. This is a comparatively simple design, which does not employ a stereo-indicator light. We will find that there is a small amount of 19-kHz and 38-kHz signal in the L and R outputs; the normal 19-kHz rejection is greater than 20 dB, and the 38-kHz rejection is greater than 30 dB below 1 volt. To obtain an accurate harmonic-distortion test, the 19- and 38-kHz residual signals can be effectively suppressed by passing the adapter output through a 15-kHz audio amplifier.

By way of comparison, we will observe in Fig. 3-17 that all residual signals above 15 kHz are greatly attenuated by means of pi filters inserted in the L and R output circuits. One of these pi filters comprises L1, C18, C21, and C17; the other filter comprises L2, C19, C22, and C20. Basically, filtering is not provided for convenience of testing, but to avoid interference in tape-recording procedures. That is, tape recorders employ ultrasonic ac bias, which can form audible beats with ultrasonic residual signals that are produced in the multiplex decoding process. By providing filter action, the residual signals are reduced to an imperceptible level.

**ANALYSIS OF COMMON SYMPTOMS**

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

**1. No Output**

When there is no output from either the L or the R channel, we know that the defect is common to both channels. For example, there might be an interruption of supply voltage to the multiplex adapter. Again, the input plug might not be fully inserted. In case the trouble is not readily apparent, it is helpful to analyze the circuit diagram of the unit, to determine the signal-flow paths. If we avoid a "shot-gun" approach, we will usually save considerable time in the long run. After identifying the signal test points in the circuit, a scope can be used to trace up to the defective component or branch.

Possible causes of signal stoppage in a multiplex configuration are:

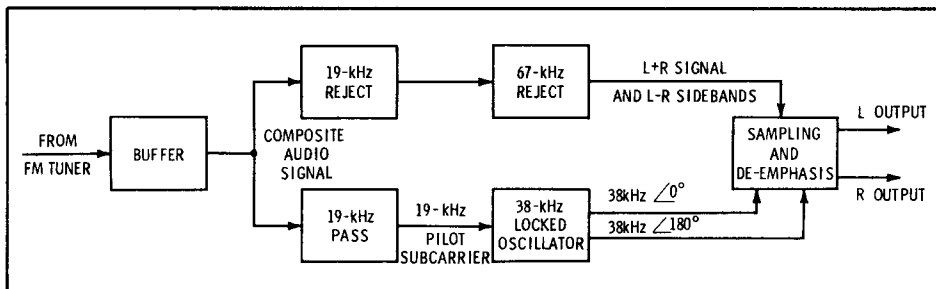
- a. Open coupling capacitor to base of transistor, such as Q1 in Fig. 3-24.
- b. Leaky or shorted capacitor, causing cutoff or saturation of transistor.
- c. Transistor failure.
- d. Cracked PC board, or damaged printed wiring.
- e. Unsoldered or cold-soldered connection.
- f. Defective transistor (not likely, but possible).

**2. One Channel Dead**

If only one channel is dead in a multiplex unit, the trouble symptom provides preliminary localization. The signal-flow paths can be determined by analysis of the circuit diagram, and signal-tracing with a scope will lead to the defective branch circuit or component. For example, with reference to Fig. 3-24, a block diagram that denotes signal-flow paths can be drawn as exemplified in Fig. 3-26.

Possible defects causing one channel to fail are as follows:

- a. Open capacitor, such as a series capacitor in output lead of a channel.
- b. Shorted capacitor, such as a de-emphasis capacitor in output lead of a channel.
- c. Cracked PC board, or damaged printed wiring.
- d. Poor connection or otherwise open secondary



**Fig. 3-26. Block diagram for circuit of Fig. 3-24.**



winding in 38-kHz transformer (such as terminal 1 of T4 in Fig. 3-24).

e. Output connector not fully inserted.

### 3. Weak Output

A defect that causes weak output may affect one or both channels. In case both channels have subnormal output, we know that the defect is in a circuit that is common to both the L and R channels. Normal waveform amplitudes are seldom specified in the service data. Therefore, it is helpful to compare the observed amplitudes with those in another adapter of the same type. When the defective circuit has been localized, dc voltage and resistance measurements will usually serve to pinpoint the defective component.

Possible defects that cause weak output on one or both channels are:

- Leaky coupling capacitor.
- Open bypass capacitor.
- Transistor with collector leakage.
- Off-value bias resistor (less likely, but possible).
- Low  $V_{CC}$  supply voltage.
- Replacement transistor incorrectly wired into circuit.

### 4. Poor Separation

Poor separation can be caused either by component defects or by misalignment of a multiplex

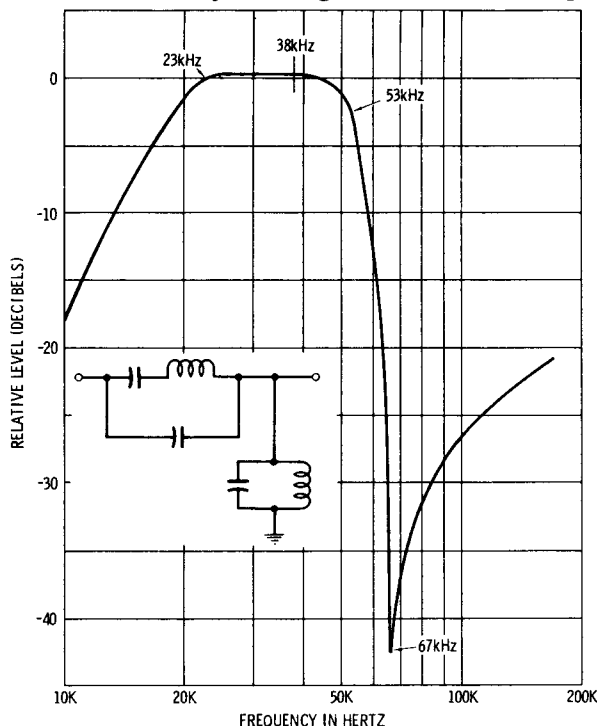


Fig. 3-27. Response of bandpass filter.

adapter. Alignment is particularly critical in the matrix-type decoder; with reference to Fig. 3-9, the bandpass has an optimum frequency response shown in Fig. 3-27. Although not shown, the bandpass circuit has a phase characteristic that affects the relative timing of the upper and lower sidebands in the L-R signal. In theory, the phase characteristic should be linear; however, in practice, linearity can only be approximated. A bandpass circuit has best linearity when the frequency characteristic has the shape shown in Fig. 3-27.

Possible defects that can cause poor separation are as follows:

- Poor front-to-back ratios, or unmatched diodes in sampling bridge.
- Leaky capacitor in sampling-bridge circuit.
- Weak output from 38-kHz oscillator (check for transistor collector leakage).
- Misalignment of bandpass circuit.
- Off-value resistor in matrix or bridge circuit (less likely, but possible).

### 5. Distorted Output

Two basic types of distortion can occur in multiplex adapters; these are nonlinear distortion and frequency distortion, as explained previously. Distortion can also be caused by interference, such as SCA squeals or birdies due to trap misalignment or defects. Erratic locking of the 38-kHz subcarrier oscillator can also cause distortion. For example, subnormal supply voltage can result in audio distortion.

Possible defects that can cause distorted output are as follows:

- Leaky capacitor in sampling bridge.
- Defective diode in sampling bridge.
- Transistor with collector leakage.
- Marginal electrolytic capacitor in subcarrier-oscillator section.
- Defective or mistuned SCA trap.
- Off-value resistor (less likely than preceding defects).

### 6. Separation-Control Setting Drifts

When the separation-control setting drifts, there is an unstable component present that affects the separation process. Separation controls are provided only in matrix-type multiplex units; for example, a separation control (R31) is included in the configuration of Fig. 3-17, whereas no separation control is present in Fig. 3-24. An elusive troublemaker consists of a worn and erratic separation control, because the possibility of resistance drift in this com-

## Hi-Fi Stereo Servicing Guide

ponent is often overlooked. Therefore, we should always check the control first.

Possible defects that can cause the separation control setting to drift are as follows:

- a. Deteriorating bypass capacitor.
- b. Coupling capacitor with unstable leakage condition.
- c. Marginal diode or transistor.
- d. Poor connection with fluctuating contact resistance.
- e. Inconstant supply voltage to adapter unit.
- f. Burned or otherwise defective resistor.

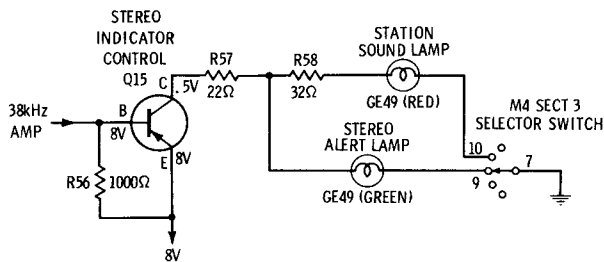


Fig. 3-28. Poor connection to pin 9 on M4 caused failure of stereo indicator.

## 7. Stereo-Indicator Failure

In most cases, stereo-indicator failure is obvious, and requires only the replacement of a burned-out lamp. However, if the lamp is not burned out, broken, or otherwise defective, we turn our attention to the indicator circuit and its components. A stereo fm indicator bulb will light only if a stereo-multiplex signal is applied to the adapter section. If the 19-kHz or 38-kHz tuned circuits are badly out of alignment for any reason, the stereo-indicator circuit cannot be energized. However, we check alignment last, unless there is reason to believe that the adjustments have been tampered with.

Possible defects that can cause stereo-indicator failure are as follows:

- a. Defective switch contact, or poor connection. (See Fig. 3-28.)
- b. Stereo-indicator light plug not fully inserted.
- c. Open coupling capacitor, such as C91 in Fig. 3-20A.
- d. Shorted capacitor, such as C121 in Fig. 3-20B.
- e. Defective transistor in indicator section.
- f. Diode with poor front-to-back ratio, such as X15 in Fig. 3-20B.

## Chapter 4

# Introduction to Audio Amplification

The dividing line between utility-type and hi-fi audio amplifiers is not sharply drawn, since no industry standards have been established. However, an amplifier in a low-priced record player or tape recorder will not be classified as a hi-fi unit by any technician or audio buff. On the other hand, almost any amplifier in an elaborate combo will be classified as a hi-fi unit. These examples leave an extensive “gray” area, in which classification is chiefly a matter of personal opinion. It is helpful to start our analysis of audio-amplifier servicing with a coverage of this intermediate type of equipment, because its comparative simplicity facilitates analysis and detailing of troubleshooting procedures. We are concerned with two broad categories of design, viz. solid-state and tube-type amplifiers. That is, the analytical approaches to trouble localization are not necessarily the same in these basic categories, inasmuch as design principles may depart from a transistor-for-tube plan.

Common trouble symptoms caused by defects in the intermediate types of audio amplifiers are as follows:

1. No audio output.
2. Subnormal output level.
3. Excessive distortion at rated output level.
4. Excessive distortion at all output levels.
5. Noise and/or hum interference in sound.
6. Intermittent operation.

### GENERAL DISCUSSION

Tube-type utility amplifiers may employ the “cordwood” assembly plan exemplified in Fig. 4-1. This is a traditional and conventional component layout that is comparatively easy to troubleshoot, although it

lacks the sophistication of more modern physical designs. The more elaborate designs of utility amplifiers in the conventional group feature ordered layout of smaller components on terminal boards, as seen in Fig. 4-2. Although this arrangement is slightly more difficult to trace, each component will be identified in the amplifier service data for ready reference. The next step in sophistication entails packaged printed circuits, as shown in Fig. 4-3. For example, an RC coupling circuit may be fabricated in this type of package. Since individual components are not always externally accessible, analysis must be made on the basis of input-output relationships.

Some tube-type utility amplifiers employ terminal boards and plated (or printed) circuits, as shown in Fig. 4-4. This type of circuitry is somewhat more difficult to trace; however, component identification is easy if we make reference to the amplifier service data. Only a few hybrid types of audio amplifiers (using transistor drivers and tube output stages) have been marketed. Solid-state utility amplifiers employ transistors, and semiconductor diodes may also be included. Printed-circuit boards are universally employed, and we will occasionally find integrated circuits in low-level stages. An integrated circuit differs from a packaged printed circuit in that an IC contains devices (specially fabricated transistors). Since the individual components are externally inaccessible, troubleshooting must be done on the basis of terminal-voltage measurements, and input-output relationships.

### Amplifier Classification

Bench work involves various categories of audio amplifiers, and many subclassifications. The chief basic arrangements include:

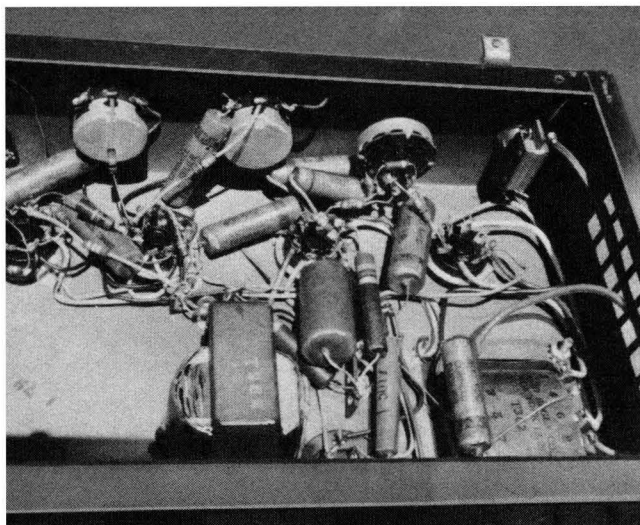


Fig. 4-1. Utility amplifier employing the "cordwood" type of component layout.

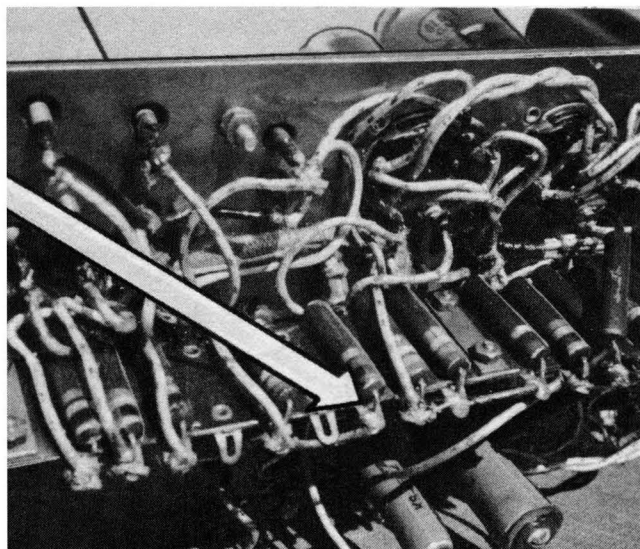


Fig. 4-2. Resistor terminal board in a tube-type amplifier.

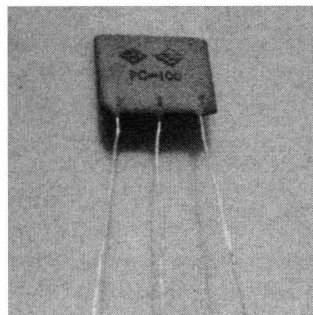
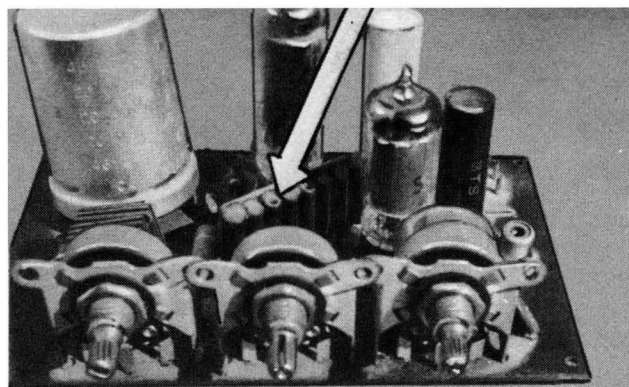
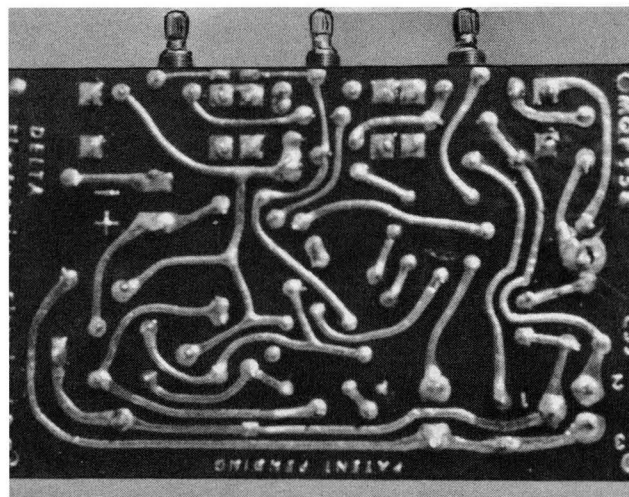


Fig. 4-3. Typical packaged printed circuit.

1. Resistance-capacitance coupled stages.
2. Transformer-coupled stages.
3. Hybrid RC and transformer coupling.
4. Direct-coupled stages.
5. Output-transformerless (OTL) arrangements.



(A) Capacitor terminal board indicated by arrow.



(B) Bottom view of plated circuit.

Fig. 4-4. Boards used in amplifiers.

Any of the foregoing types of coupling may be used with transistors or with tubes, and may be employed in single-ended or double-ended (push-pull) stages. Parallel connection of tubes or transistors may be used in output stages to provide a required power level. Triode, tetrode, pentode, or beam-power tubes are utilized. Plate output is most commonly used, although we will also find cathode output employed when substantial driving power is required, or when a very low impedance load must be accommodated as in OTL amplifiers. In solid-state amplifiers, the common-emitter configuration is in very wide use, although the common-collector (emitter-follower) configuration is used when substantial driving power is required, or when the low load impedance of an OTL amplifier is to be driven. The common-base configuration is occasionally utilized when an input device with very low impedance must drive a high-impedance load.

It is helpful to briefly consider the circuitry for the basic amplifier arrangements. Fig. 4-5 shows the

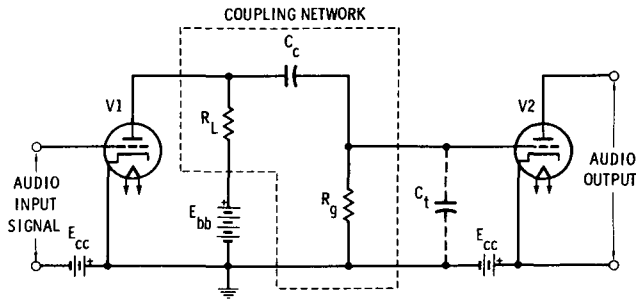


Fig. 4-5. RC coupling network between two triode amplifiers.

simplest form of RC-coupled triode amplifier. The most common source of trouble (apart from defective tubes) is leakage in the coupling capacitor  $C_c$ , which upsets the grid bias on V2. In practice, cathode self-bias is generally used, and pentode tubes are often employed because of their comparatively high gain (see Fig. 4-6). This type of circuitry utilizes more capacitors, and hence is more likely to develop trouble symptoms due to capacitor defects. For example, if  $C_k$  is open, the associated tube becomes degenerative, and the stage gain is impaired. Or, if  $C_k$  shorts, the grid-cathode bias is disturbed. In case  $C_{sg}$  opens up, we will find audio signal on the screen grid; the stage gain is greatly reduced, and the circuit develops poor frequency response. On the other hand, if  $C_{sg}$  becomes shorted,  $R_{sg}$  is likely to burn out; in any event, the stage gain becomes greatly subnormal.

Trouble in a given circuit is sometimes caused by a defect in the preceding circuit. For example, in Fig. 4-6, a short-circuit or heavy-leakage condition in  $C_c$  will cause seriously abnormal screen and plate current in V2. It is quite likely that the plate-load, screen-dropping, and cathode-bias resistors for V2 will be badly overheated or burned out during the process of destruction of V2. By the same token, this type of trouble can change the value of a circuit resistor, even if it is not burned out. In such a case, the amplifier starts working again when  $C_c$  is re-

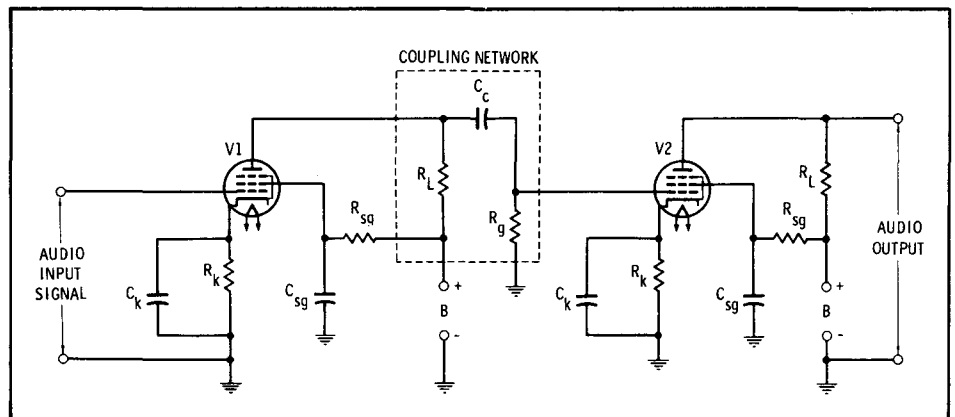
placed, but normal operation is not resumed. Let us briefly consider the factors that are involved:

1. When a cathode resistor is much too high in value, the grid-cathode bias is excessive; the stage gain is reduced, and distortion will occur on high-level signals if a sharp-cutoff pentode is utilized.
2. When a plate-load resistor is much too high in value, the stage gain is increased to some extent; however, the dynamic range of the stage is also reduced, and overload distortion occurs on high-level signals. High-frequency response is also impaired.
3. When a screen-dropping resistor is much too high in value, the stage gain is reduced and overload distortion occurs on high-level signals.

A simple transformer-coupled configuration is shown in Fig. 4-7. We are finding fewer amplifiers with transformer interstage coupling than in former years, simply because transformers are comparatively costly devices. Although low-priced coupling transformers are available, their frequency range is restricted and they tend to develop appreciable distortion. The response of a transformer depends considerably on the preceding plate-output impedance and the succeeding grid-input impedance. Therefore, exact replacements are advisable to avoid impaired response. Common causes of trouble symptoms in this type of amplifier are:

1. Primary winding of transformer burned out, due to preceding circuit fault that produces an excessive current demand.
2. Leakage to core or between windings due to absorption of moisture or because of serious overheating.
3. Response will sometimes be impaired if the primary or secondary leads are reversed accidentally.

Fig. 4-6. Typical RC-coupled amplifier using pentodes.



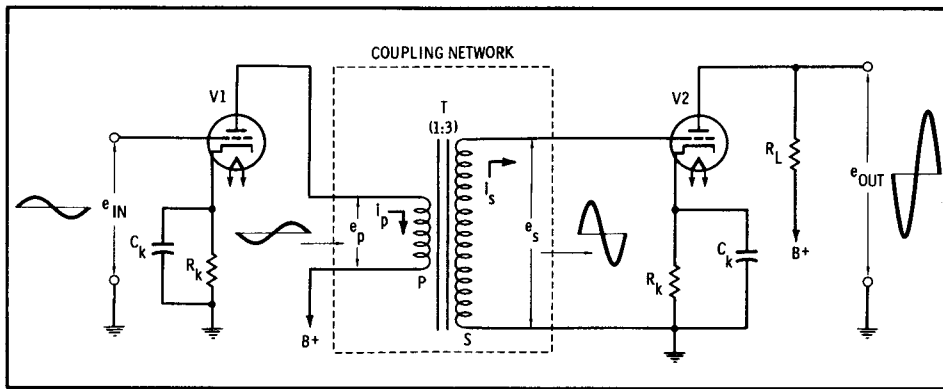


Fig. 4-7. Transformer-coupled amplifier using triodes.

4. Distortion due to shorted cathode-bypass capacitor.
5. Low gain due to open cathode-bypass capacitor.

In older types of amplifiers, such as those used in public-address (PA) systems, we will find hybrid RC and transformer coupling, as exemplified in Fig. 4-8. The arrangement is characterized by single-ended stages throughout, and provision of adjustable negative feedback for the second and third stages.

We will consider the operation and troubleshooting of negative-feedback circuits in greater detail later. The basic troubleshooting approach in Fig. 4-8 can be outlined as follows:

1. Check tubes first, or try substituting new tubes.
2. If there is no audio output, use a scope to trace the signal through the amplifier.

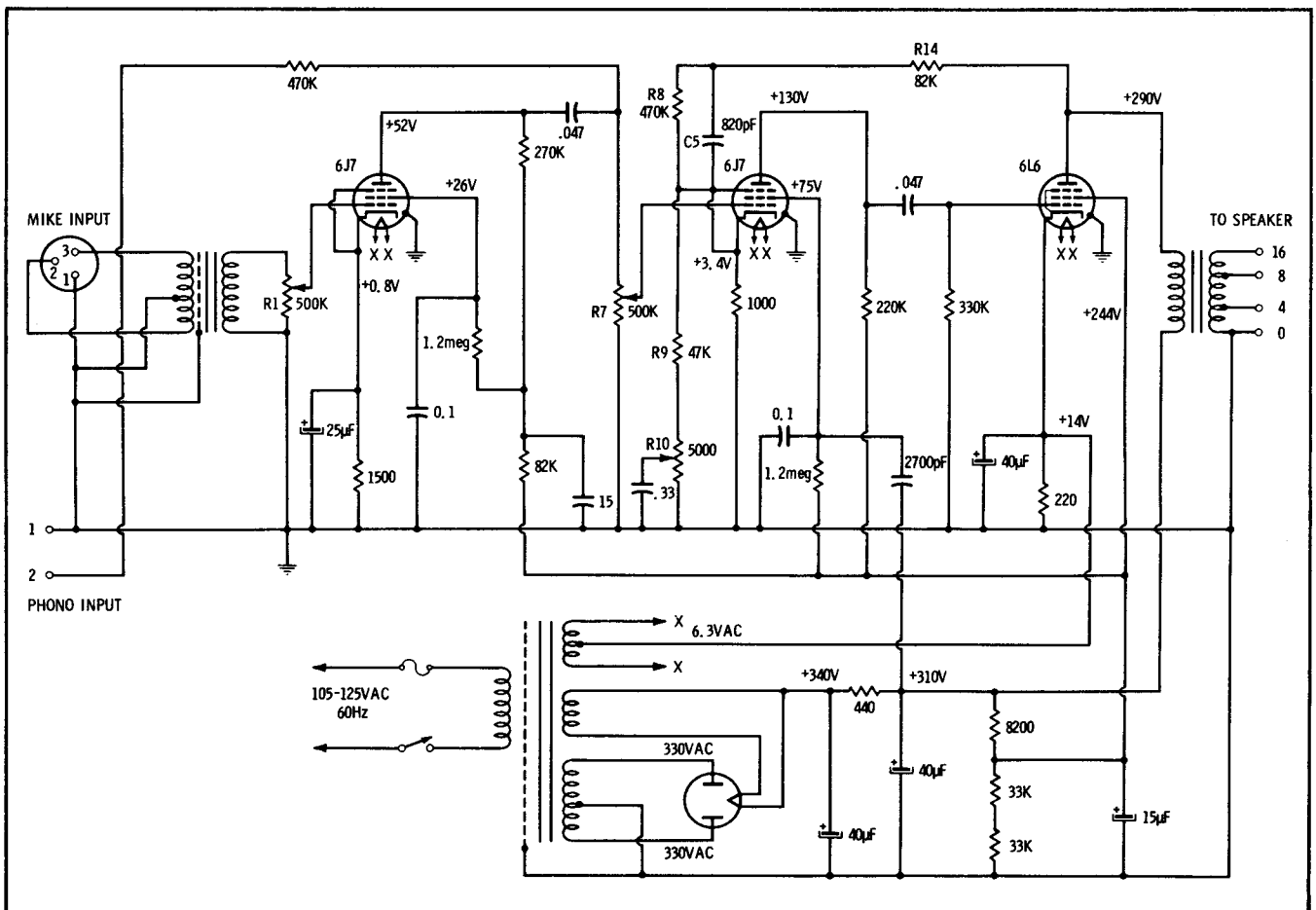


Fig. 4-8. An older model 6-watt PA amplifier of the hybrid RC- and transformer-coupled type.

3. After a defective stage is localized, measure dc voltages and compare the measured values with those specified.
4. In case of doubt, disconnect one end of a suspected capacitor and test it with a capacitor checker, or make a substitution test.
5. Make supplementary resistance measurements of controls, fixed resistors, and transformer windings to confirm these possibilities of trouble.

Direct-coupled amplifier configurations have been used to a limited extent in older designs, but are widely used in the newer models. Therefore, we will encounter many more dc-coupled circuits in solid-state amplifiers than in tube-type amplifiers. A basic example of a dc-coupled amplifier using triodes is shown in Fig. 4-9. The advantage of this design is that it can amplify very low bass signals with negligible distortion and with comparatively simple circuitry. A voltage divider is generally used to supply the necessary dc operating voltages for the amplifier from a regulated power supply. The B+ supply voltage is applied across the voltage-divider resistor  $R_d$ , which is tapped at suitable points. Capacitor  $C_d$  is used to bypass any ac voltage variations that might appear across  $R_d$ . The audio-input voltage is amplified by V1 and V2, and appears as the audio-output voltage at the plate of V2 in normal operation. The plate-load resistor  $R_{L1}$  also serves as the grid resistor for V2, inasmuch as the voltage drop across it is applied to the grid of V2.  $R_{L2}$  serves as a plate-load resistor for V2.

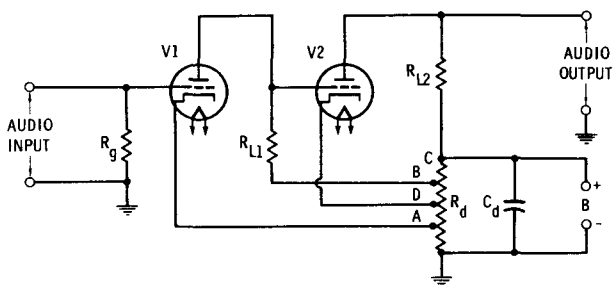


Fig. 4-9. Typical direct-coupled amplifier using triodes.

Note that the cathode of V1 in Fig. 4-9 is connected to point A, and the plate is connected to point B on the voltage divider. Point B is positive with respect to point A, which makes the plate positive with respect to its cathode. This voltage relationship permits V1 to conduct. The voltage developed from point A to ground serves as bias voltage for V1. The plate of V2 must be positive with respect to its cathode for conduction to take place. Moreover, the grid voltage of V2 must not be positive with respect to its cathode.

The plate current of V1, which flows through  $R_{L1}$ , produces a considerable voltage drop across this resistor. In turn, the voltage at the plate of V1 and at the grid of V2 is less positive than at point B on the voltage divider. Tap D is located at a point on resistor  $R_d$  such that the magnitude of the positive voltage on the grid of V2 is lower than that of the positive voltage on the cathode of V2. Therefore, the grid of V2 is actually less positive (or is effectively negative) with respect to the cathode of V2 in normal operation. The voltage between points C and D is the plate voltage for V2.

When trouble symptoms are encountered in dc-coupled stages of a tube-type amplifier, the most likely causes are as follows:

1. A tube may have developed grid current because of grid emission or traces of gas. A substitution test is preferred.
2. Regulation of the power supply may be unsatisfactory; a common cause is aging of a voltage-regulator tube.
3. The bypass capacitor for the voltage divider may be leaky, or may have opened up.
4. A fixed resistor (particularly of the composition type) will occasionally develop thermal instability.
5. Corroded contacts or imperfect connections are especially troublesome in dc-coupled stages.

Prior to the advent of transistors, output-transformerless (OTL) amplifiers were in the minority, due to the extensive trade-off that was involved between output power and efficiency. However, some of these early OTL amplifiers are still in use, and will be briefly considered. One of the most familiar types is the Futterman amplifier shown in Fig. 4-10. Its chief advantage is that it dispenses with an output transformer, which is a costly component if it is designed to provide reasonably good audio quality. In this example, three Type-6082 dual-triode power tubes are utilized in the output stage. A double-ended power supply is employed to provide both B+ and B- voltages. The 6SN7 is driven by a paraphase inverter stage, and in turn drives the 6082s, which operate in a push-pull/parallel configuration. Since the speaker is connected at the midpoint of a series tube arrangement, a basic cascode configuration is employed in which the tubes serve as plate and cathode loads for each other. Note that 40 dB of negative feedback is provided from the voice coil to the cathode of the phase-splitter driver tube. With 1 volt rms input, this amplifier normally provides 25 watts of audio power into a 16-ohm voice coil with only 0.3 percent harmonic distortion.

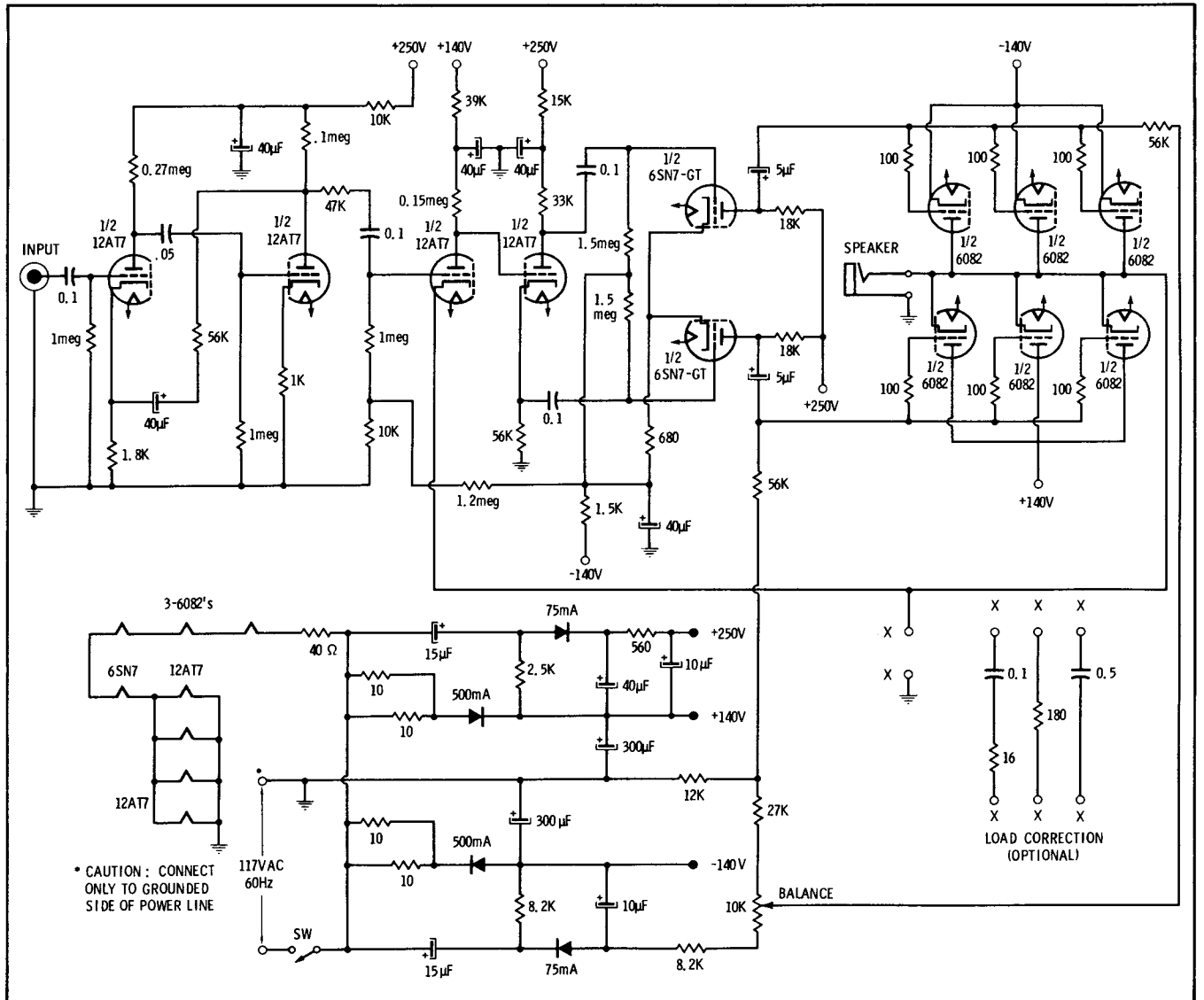


Fig. 4-10. The Futterman OTL amplifier.

The chief troubles encountered in OTL amplifiers are caused by the same component defects that were noted previously for RC-coupled amplifiers. It is helpful at this point to briefly note the various classes of operation employed in audio amplifiers. They are:

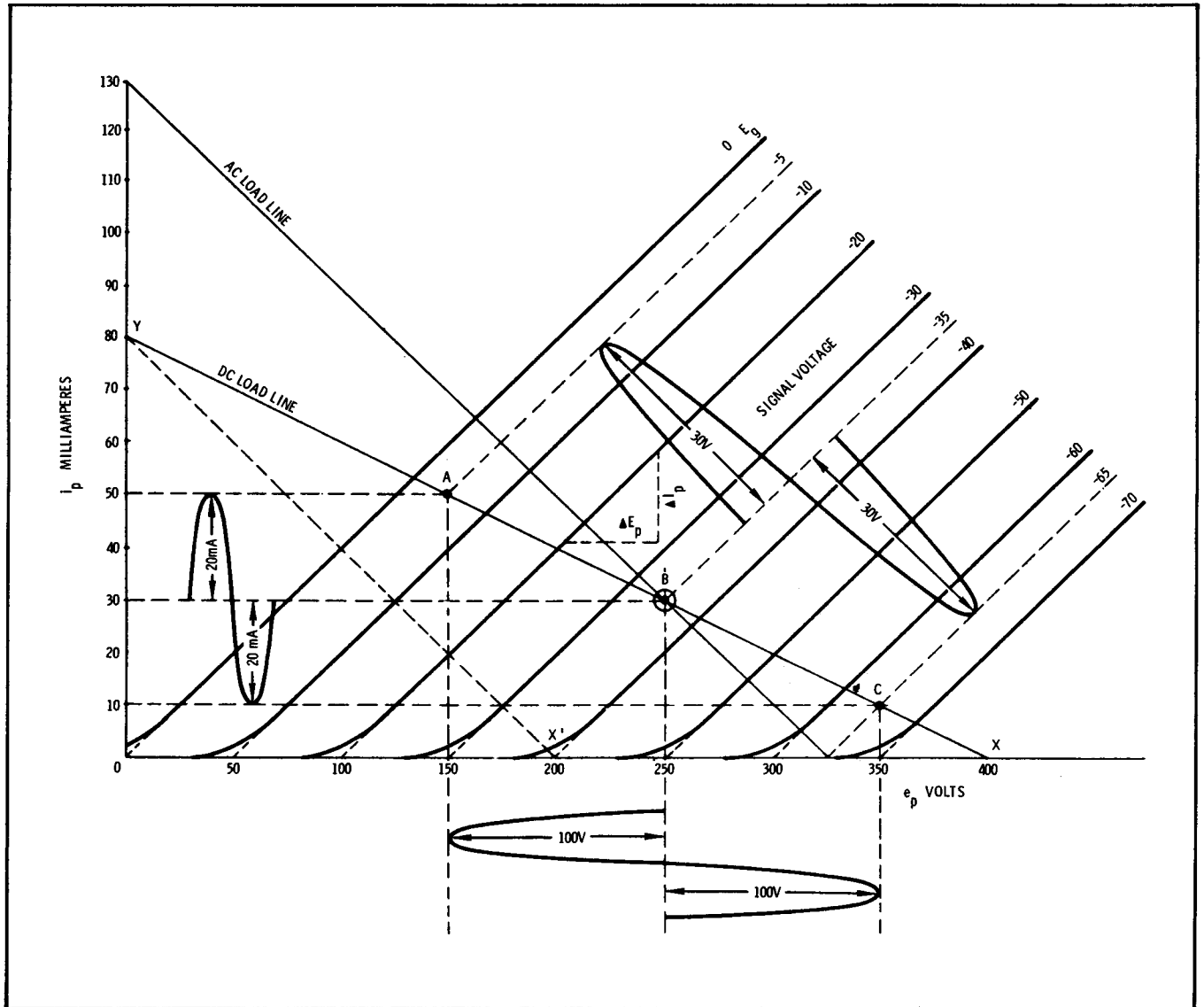
1. Class-A operation.
2. Class-B operation.
3. Class-AB operation.
4. Class-AB<sub>1</sub> operation.
5. Class-AB<sub>2</sub> operation.

Class-A operation is shown in Fig. 4-11. The tube is normally biased so that there is current at all times. Point B is called the operating point, and is determined by the value of grid voltage (dc bias voltage). The output plate waveform is practically undistorted in normal operation. Of course, over-

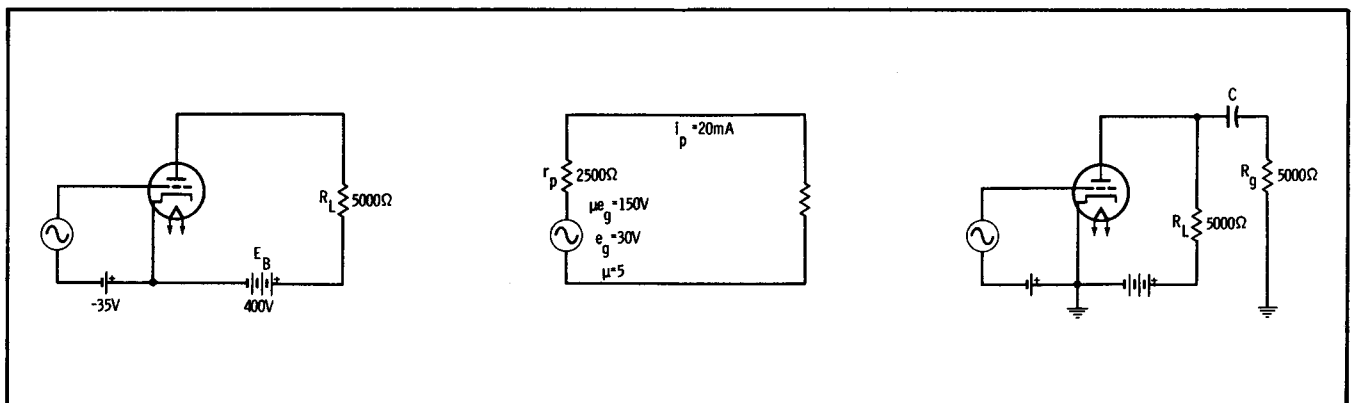
drive will cause a departure from linear operation and will result in distortion. Note that when the coupling circuit in Fig. 4-11D is included, the effective plate load is decreased with respect to an audio signal. Therefore, the signal development occurs on the ac load line shown in Fig. 4-11A. That is, the load for an ac signal is only half as great as the load for the dc plate circuit in this example.

Class-B operation is shown in Fig. 4-12. The tube is normally biased at or near cutoff. Thus, plate current flows during the positive half of the input grid signal, but stops flowing during the negative half cycle. In turn, the output plate-current waveform is seriously distorted. If the amplitude of the grid-driving signal is increased so that it extends beyond point F, power output is increased; but if the drive extends beyond point G, grid current will flow and





(A) Load line.



(B) Circuit.

(C) Equivalent circuit.

(D) Coupling circuit CR<sub>g</sub> added.

Fig. 4-11. Visualization of Class-A operation.

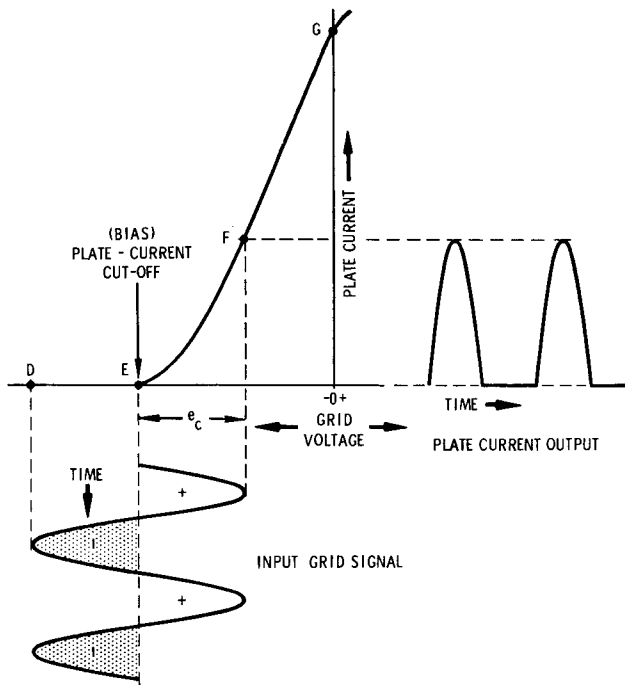


Fig. 4-12. Class-B operation on dynamic characteristic.

the plate-current waveform will be clipped, resulting in increased distortion. To minimize distortion in Class-B amplifiers, a stage commonly employs two tubes operating in push-pull as shown in Fig. 4-13. Thereby, the distortion produced by one tube is largely nullified by the other tube. Class-B amplifiers are occasionally designed to draw a little grid current at maximum rated power output. In such a case, the driver stage is designed to supply the required driving power.

Class-AB operation is pictured in Fig. 4-14. Note that the subclasses of this mode are designated AB<sub>1</sub> and AB<sub>2</sub>. This type of amplifier operates in the region between Class A and Class B. Plate current flows more than one-half, but for less than the entire cycle in normal operation. A subscript 1 denotes that grid current does not flow at maximum rated power out-

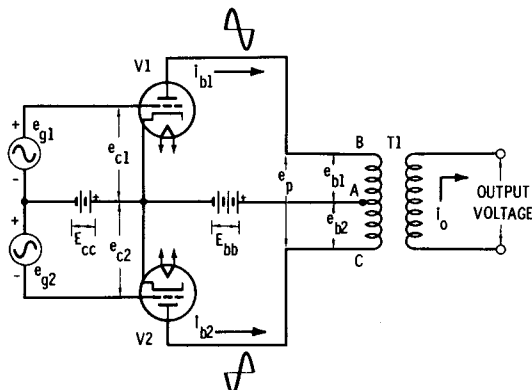
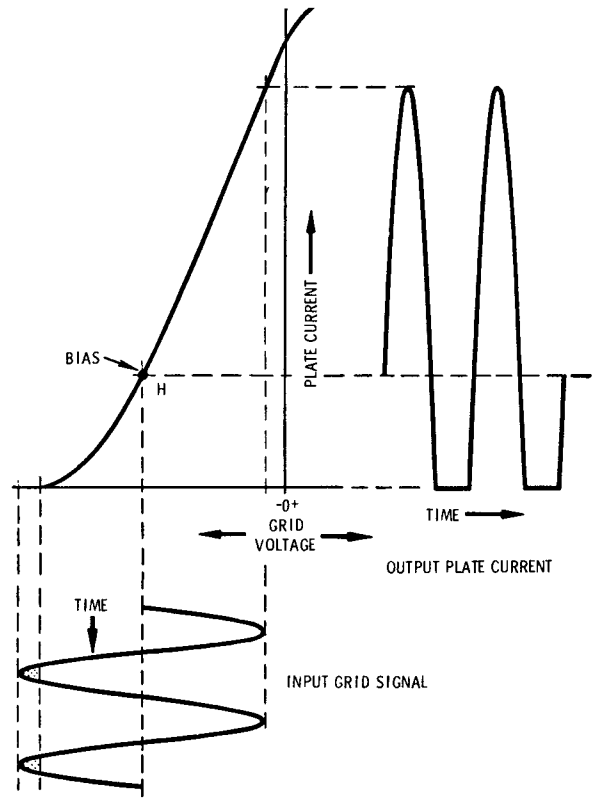
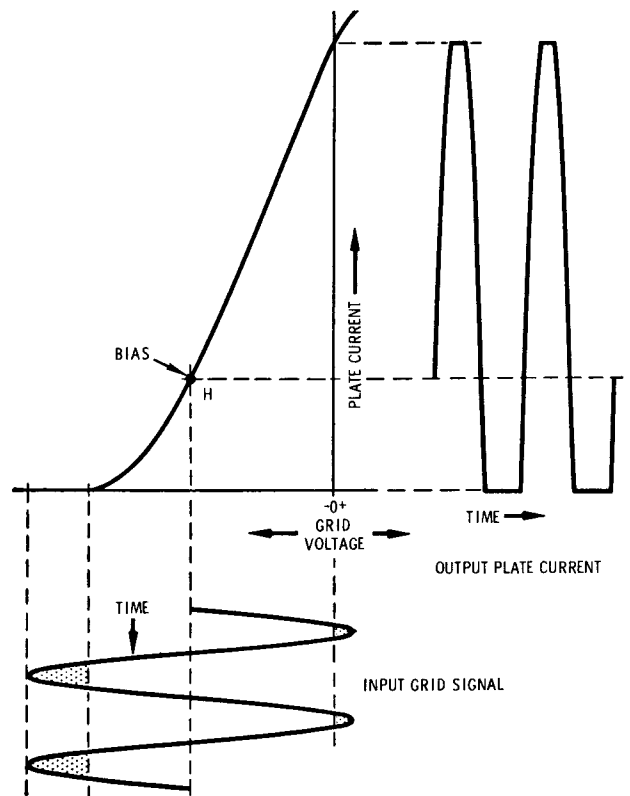


Fig. 4-13. Push-pull triode amplifier circuit.



(A) Class-AB<sub>1</sub> operation.



(B) Class-AB<sub>2</sub> operation.

Fig. 4-14. Class-AB operation on dynamic characteristic.

put. A subscript 2 denotes that grid current flows at maximum rated power output. Since the output waveform from a Class-AB tube is necessarily distorted, an audio stage commonly employs a pair of tubes operating in push-pull, as in a Class-B amplifier. The advantages of a Class-AB amplifier are its greater power output, compared with a Class-A amplifier, and its lower distortion, compared with a Class-B amplifier.

It should not be supposed that Class-A operation is basically distortionless. A triode provides less distortion than a pentode, and a high value of plate-load resistance minimizes residual distortion. However, as seen in Fig. 4-15, the dynamic transfer char-

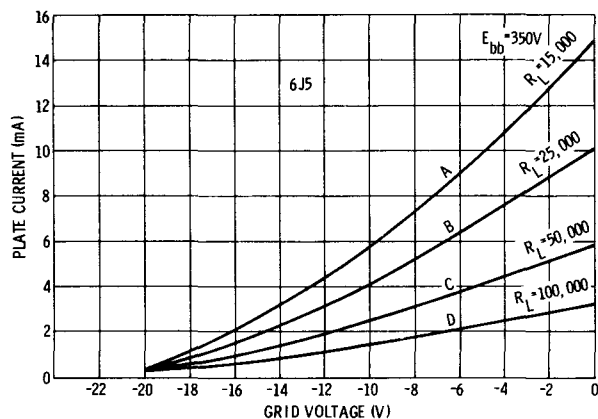


Fig. 4-15. Dynamic transfer characteristic curves for different plate loads of a triode.

acteristic of a triode is never absolutely linear. Therefore, even triodes in the Class-A mode must be operated in push-pull to obtain minimum distortion, particularly at high audio levels (see Fig. 4-16). Next, in Class-B operation, we will obtain a certain amount of crossover distortion when the tubes are biased exactly to plate-cutoff, as shown in Fig. 4-17. Crossover distortion is eliminated, of course, by operating the tubes in Class AB. The amount of distortion that is produced by Class-AB<sub>2</sub> operation depends chiefly on the ability of the preceding driver stage to supply the grid-current demand. Beginners will find the data tabulated in Table 4-1 informative.

We will find various types of phase inverters in commercial audio amplifiers. One of the simplest forms was shown in Fig. 4-10; this configuration exploits plate output and cathode output from the same tube. The inverter stage provides negligible gain, and operates as an electronic transformer to develop a double-ended output from a single-ended input. Phase inversion is often accomplished in the output stage; one widely used arrangement is shown in Fig. 4-18. The single-ended input is amplified by V1 and appears in opposite phase at the plate. A portion

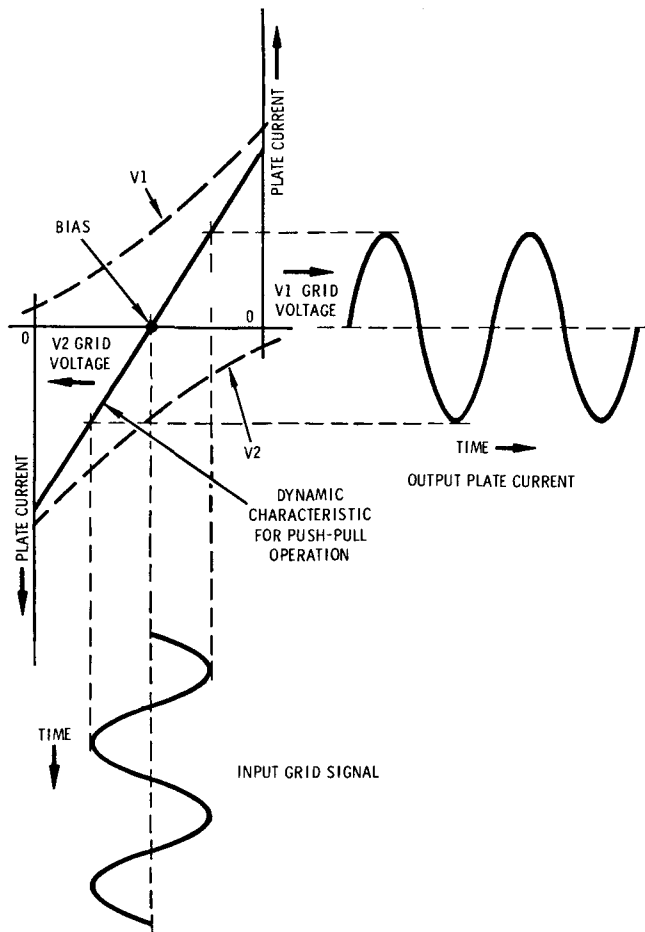


Fig. 4-16. Dynamic characteristic of two tubes in push-pull Class-A operation.

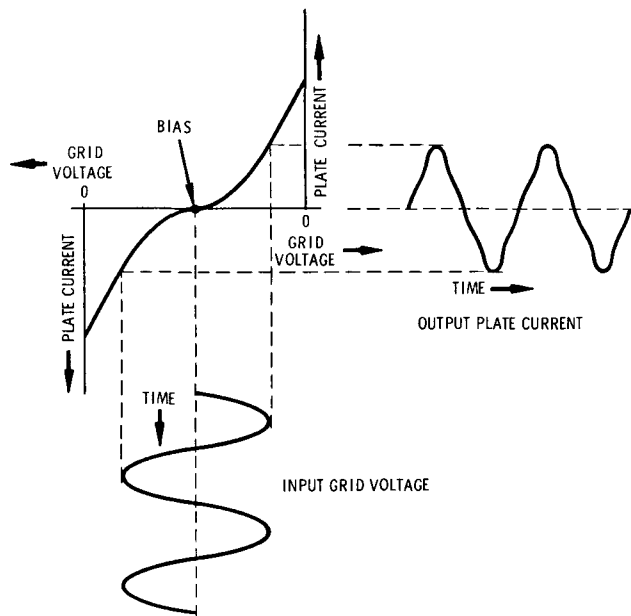


Fig. 4-17. Operation of push-pull, Class-B amplifier biased exactly at plate-current cutoff.

Table 4-1. Amplifier Characteristics

Class	Location of Operating Point on Dynamic Characteristic	Relative Distortion	Relative Power Output	Approximate Percentage of Plate Efficiency
A single-tube	On linear portion	Low	Low	Under 20%
A push-pull		Very low	Moderate	20 to 30%
AB single-tube	Between linear portion and plate-current cutoff	Moderate	Moderate	40%
AB push-pull		Low	High	50 to 55%
B single-tube	At vicinity of cutoff	High	High	40 to 60%
B push-pull		Low	Very high	60 to 65%

of this output signal is dropped across R4, and is coupled to the grid of V2; thereby V2 is driven in opposite phase to V1. Resistors R5 and R6 are parasitic-suppression resistors. These resistors are required to avoid parasitic oscillation at a very high frequency corresponding to lead resonances in the grid and plate circuits. Parasitic oscillation reduces the output of an amplifier, causes distortion, and sometimes damages the tubes.

The same basic method of phase inversion is often provided in the driver section for a push-pull output stage, as exemplified in Fig. 4-19. This arrangement is less troublesome with regard to parasitics. The configuration provides for either microphone or phono input, and the inputs may be mixed, if desired. Note that the bass tone control varies the proportions of bass and treble amplitudes applied to the grid of V2. The treble tone control operates on a similar principle, except that the setting of R1 determines the proportion of treble tones that are bypassed to ground. Negative feedback is employed from the voice coil to the cathode and suppressor grid of V2 to provide improved frequency response.

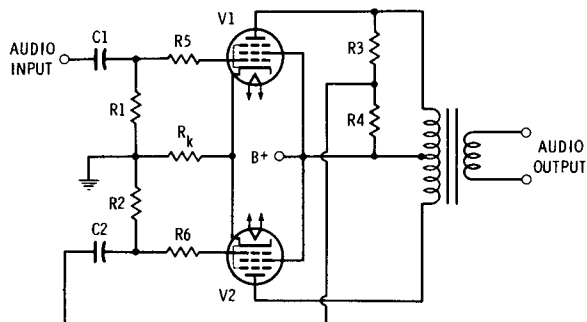


Fig. 4-18. Output stage with phase-inverter action.

It is desirable to have a reasonable impedance match from the output stage to the voice coil, both to obtain maximum power transfer, and also to optimize the frequency response.

Most solid-state audio amplifiers in the utility category are designed on a transistor-for-tube basis, and are thus comparatively easy to service by analogy with tube-type amplifiers. Fig. 4-20 shows the basic comparative transistor and tube configurations. One of the common trouble symptoms, particularly in a solid-state power stage, is thermal drift, which results in distortion and reduced power output, and can cause transistor damage in serious cases. A basic method of overcoming thermal drift is to use thermistor control of the emitter bias voltage, as shown in Fig. 4-21. The thermistor is usually a specified type of semiconductor diode. Proper bias control depends on the condition of the thermistor. Exact replacement is required in case of thermistor failure. In normal operation, greatly increased thermal stability is provided, as shown in Fig. 4-22.

We will also find amplifier stages that employ two junction diodes for compensation of temperature variation, as shown in Fig. 4-23. One of the diodes compensates for temperature variations of the emitter-base junction resistance; the other diode compensates for temperature variation of saturation current. Note that diode X2 is reverse-biased, whereas diode X1 is forward-biased. The improved thermal stability obtained in normal operation is shown in Fig. 4-24. As noted previously, direct coupling is in wide use—an example is shown in Fig. 4-25. This two-stage amplifier is arranged so that an increase in collector current caused by a temperature rise in transistor Q1 will reduce the

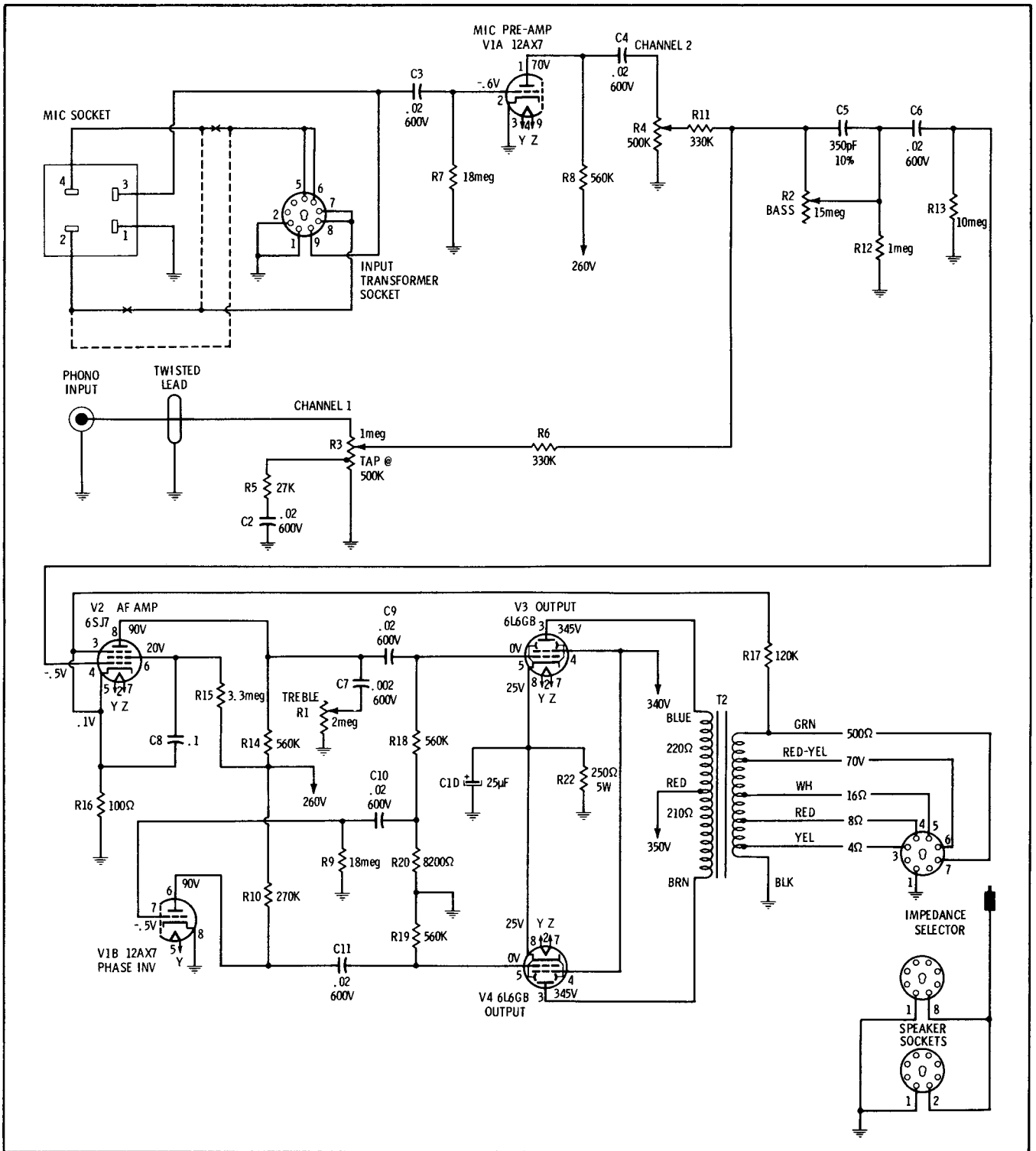


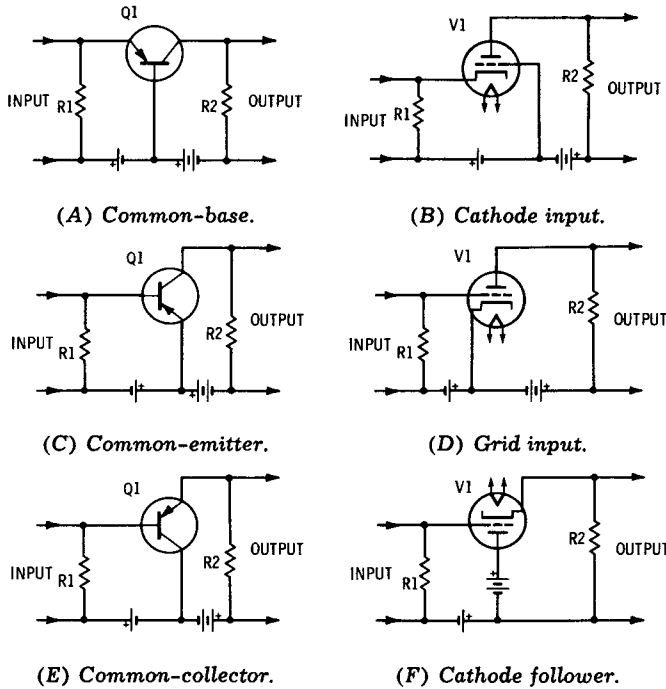
Fig. 4-19. Public-address amplifier with phase-inverting driver stage.

forward bias on transistor Q2. Circuit operation is as follows:

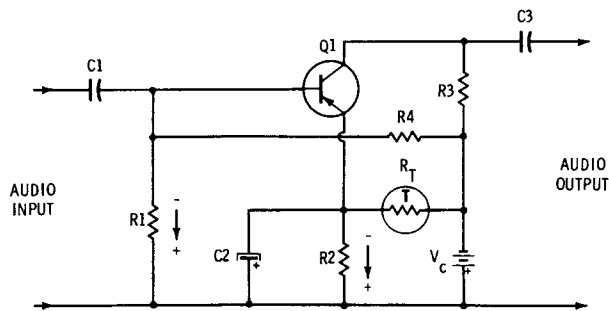
1. Transistor Q1 (Fig. 4-25) is connected in the common-base configuration, and is inherently

highly stable. However, there will be some variation in collector current with temperature change. If the temperature increases, the collector current increases in the direction of the arrow. A portion of this increment flows

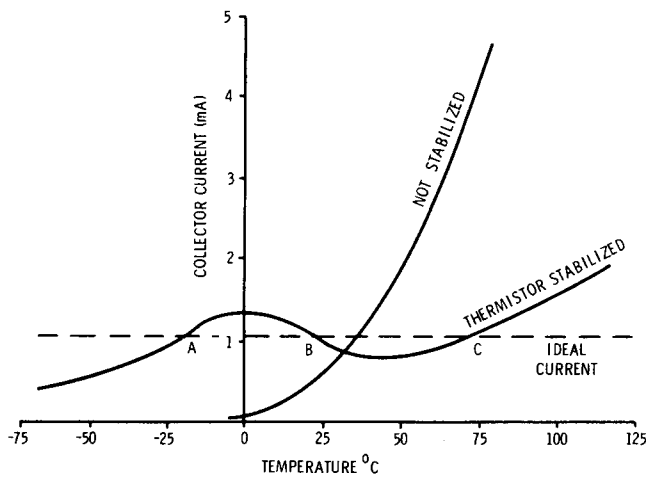
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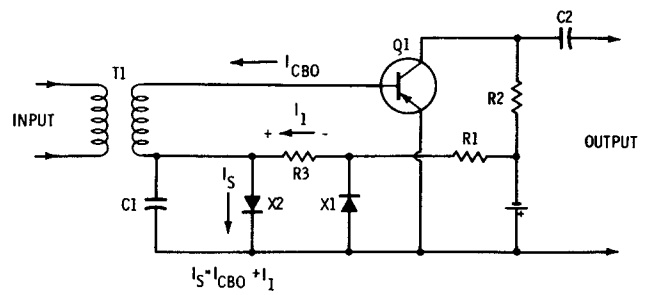
**Fig. 4-20. Basic amplifier configurations using transistors or triode vacuum tubes.**



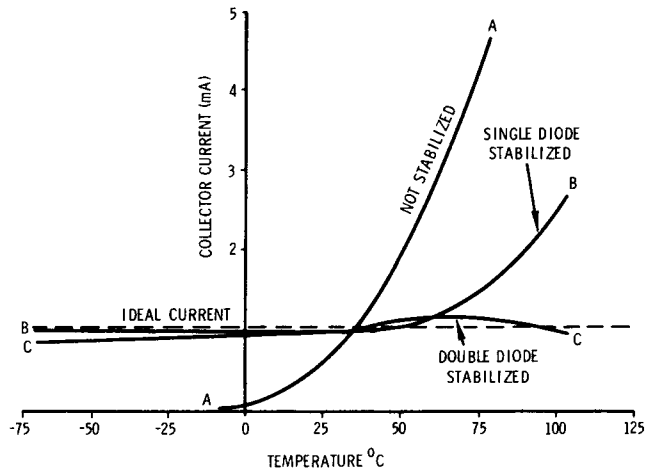
**Fig. 4-21. Transistor amplifier with thermistor control of emitter bias voltage.**



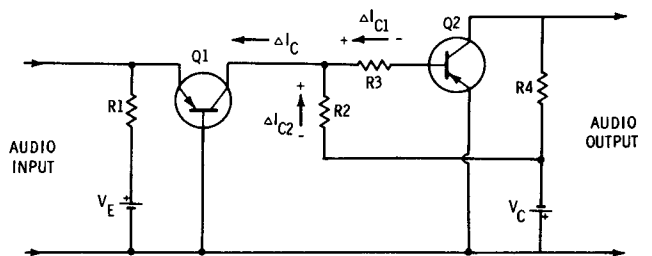
**Fig. 4-22. Collector current in stabilized and nonstabilized transistor circuit.**



**Fig. 4-23. Diode compensation for effects of temperature variation in a transistor circuit.**



**Fig. 4-24. Collector current with and without diode stabilization.**



**Fig. 4-25. Two-stage temperature-stabilized dc amplifier.**

through R3 and produces a voltage drop with the polarity indicated. Another portion of the incremental current is through R2, and develops a voltage drop with the indicated polarity.

2. Now consider the base-emitter bias circuit of Q2. This bias is the sum of the voltages across R3, R2, and battery  $V_C$ . The voltage indicated across R3 aids the forward bias; the voltage across R2 opposes the forward bias. By proportioning the values of R2 and R3 so that more voltage drops across R2, the resultant forward bias will be decreased. This action limits the tendency of the collector current in Q2 to increase in response to an increase in temperature.

Trouble can be caused if an excessive emitter-collector voltage should be applied when the normally forward-biased base-emitter circuit is reverse biased, due to internal oscillation that can destroy the transistor. This condition can occur in transformer-coupled amplifiers such as the example in Fig. 4-26. If the signal from the previous stage is suddenly stopped, or in case an excessively strong noise pulse occurs, the base-emitter circuit can be driven into a reverse-biased condition, and the collector current is thereby cut off rapidly. The magnetic field in transformer T2 collapses rapidly and produces a high emitter-collector voltage during the time that the base-emitter circuit is reverse biased. Because of this action, strong oscillations are set up, and this power must be dissipated in the transistor, which is likely to be destroyed.

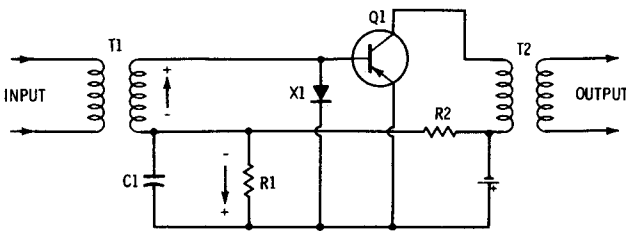


Fig. 4-26. Junction diode used to prevent reverse bias in a base-emitter circuit.

To prevent the possibility of transistor damage in the circuit of Fig. 4-26, a junction diode is connected between the base and emitter which prevents the base-emitter junction of the transistor from becoming reverse biased. We call this a shunt-limiting action. The voltage divider consisting of resistors R1 and R2 provides a voltage at the base which forward biases the base-emitter junction, and simultaneously reverse biases junction diode X1. In normal operation, X1 is effectively an open circuit. Now, if a strong surge voltage is applied from the secondary of T1, and if it is greater than the voltage across R1, the junction diode X1 will then become forward biased and will start to conduct. With X1 conducting,

only a small voltage drop is produced across the diode, which prevents the base-emitter junction from becoming reverse biased. Note that capacitor C1 serves to bypass the ac signal around R1.

Fig. 4-27 shows a typical circuit for a two-stage direct-coupled amplifier. Note that C2 and R3 form a low-pass filter, which provides some attenuation of high audio frequencies. This action compensates for the lack of full low-frequency response by microphone M1. Transistor Q2 employs the common-emitter configuration with degeneration in the emitter circuit. This increases the input impedance of Q2 to approximately 30,000 ohms. The collector voltage and emitter current of Q1 have low values to optimize the signal-to-noise ratio. Since most of the noise is contributed by the first stage (being amplified by subsequent stages), the collector voltage and emitter current of Q2 can have substantially greater values without greatly impairing the signal-to-noise ratio. Resistor R1 provides bias stability, and is bypassed by C1. Resistor R5 provides bias stability and is bypassed by C3. If C1 or C3 should become open, the amplifier gain would be greatly reduced.

A transducer may have more or less high-frequency attenuation. In such a case, the amplifier will include a suitable equalizer network, like that shown in Fig. 4-28. The equalizer network (shown inside the dashed lines), attenuates the low frequencies more than it does the high frequencies. Capacitor C1 serves as a dc blocking capacitor, but it also tends to produce some low-frequency attenuation.

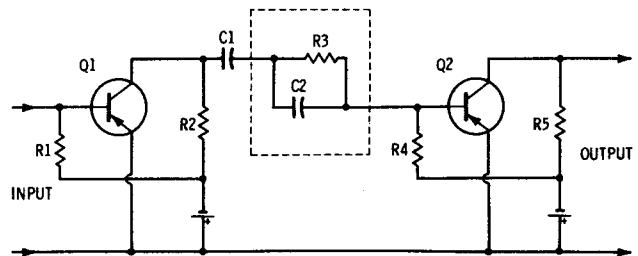
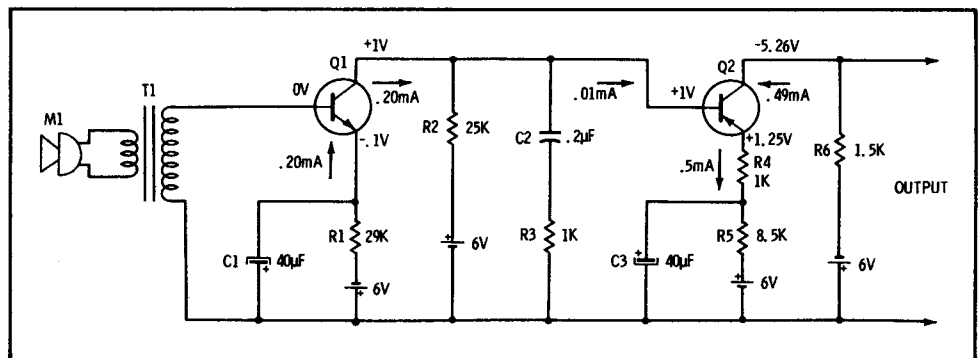


Fig. 4-28. Typical two-stage preamplifier, providing high-frequency compensation for transducer.

Fig. 4-27. Typical two-stage direct-coupled preamplifier, providing low-frequency compensation for transducer.



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However, this action is compensated and the desired overall frequency response is provided by C2 and R3. Because of its higher reactance at the lower frequencies, C2 bypasses the high frequencies around R3, and thus provides greater amplification of higher frequencies. At low frequencies, C2 can be regarded as an open circuit. Because the low frequencies must pass through R3, part of their voltage is dropped across the resistor.

Next, let us note the basic way in which transistors are used as phase inverters in audio driver stages. Fig. 4-29 shows the configuration for a split-load (paraphase) phase inverter employing transistor Q1 for driving a push-pull output stage (transistors Q2 and Q3). The path of output current from Q1 is through collector load resistor R3 and through emitter load resistor R2. Resistors R2 and R3 have equal values. R1 is used to establish the base-bias voltage of the inverter transistor. The collector and emitter outputs have equal amplitudes and are 180 degrees out of phase in normal operation. When high-level signals are present, however, noticeable distortion can occur because the collector output impedance of Q1 is higher than the emitter output impedance. In other words, the collector branch has less current capability.

To avoid this source of distortion, the arrangement shown in Fig. 4-30 may be employed. Note the addition of resistor R4, which eliminates the unbalance in output impedance that is present in Fig. 4-29. The values of resistors R2 and R4 in Fig. 4-30 are proportioned so that the signal source impedance for Q2 is the same as the signal source impedance for Q3. Thereby, both branches have the same current capability, and distortion is minimized at high levels of operation. The signal loss across R4 is compensated by making R2 higher in value than R3. Because of the large negative-feedback voltage developed across R2, a rather strong signal is required to drive a one-stage phase inverter. This disadvantage is overcome in practice by employing a two-stage phase inverter when appropriate.

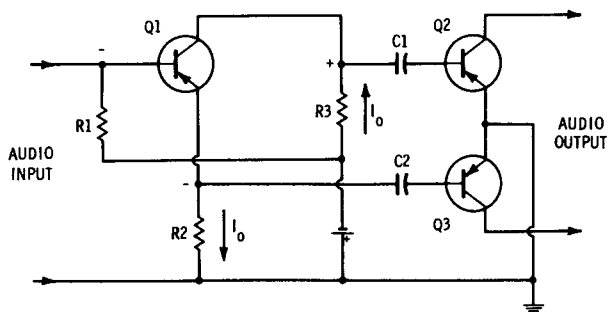


Fig. 4-29. One-stage phase inverter.

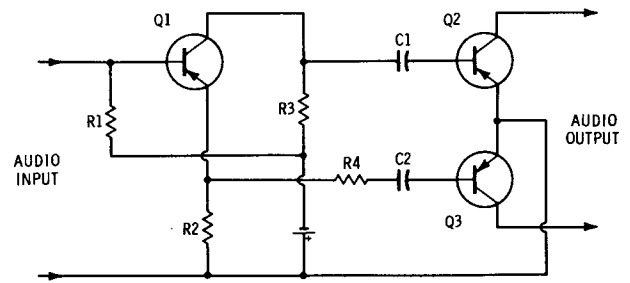


Fig. 4-30. One-stage phase inverter circuit with equalized output impedance.

Fig. 4-31 shows the circuitry of a two-stage phase inverter consisting of two identical common-emitter configurations. Let us assume that an input signal drives the base of Q1 negative. Since a common-emitter configuration produces a 180-degree phase reversal, the collector of Q1 goes positive. One por-

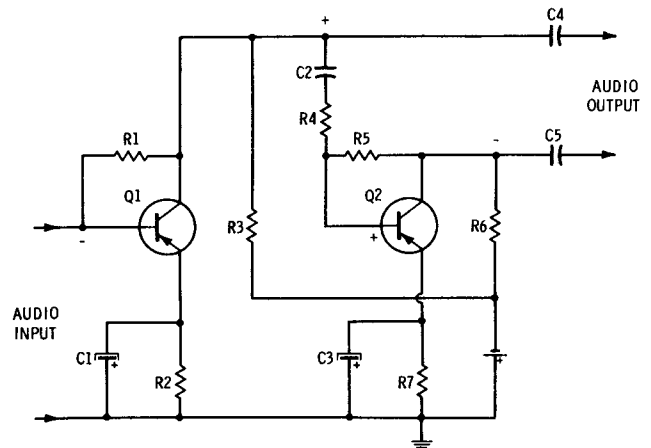


Fig. 4-31. Two-stage phase inverter circuit using two common-emitter configurations.

tion of this positive signal is coupled to the base of Q2 through C2 and attenuating resistor R4. The other portion of the positive signal is coupled through C4 to one input circuit of a push-pull output stage. The positive-going signal on the base of Q2 causes a negative-going signal at the collector of Q2. This negative signal is coupled through C5 to the other input circuit of the push-pull output stage. R1 provides base bias for Q1, and R5 provides base bias for Q2. Resistors R2 and R7 provide bias stabilization and are bypassed by C1 and C3, respectively.

## ANALYSIS OF COMMON TROUBLE SYMPTOMS

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.



## 1. No Audio Output

Possible causes of no audio output from an amplifier are as follows:

- a. Open coupling capacitor (such as  $C_C$  in Fig. 4-5).
- b. Shorted coupling capacitor (such as  $C_C$  in Fig. 4-6).
- c. Burned-out winding, such as primary P in Fig. 4-7.
- d. Open volume control, such as R1 in Fig. 4-8.
- e. Speaker plug not fully inserted in socket (see Fig. 4-19).
- f. Open or shorted transistor.
- g. Shorted overload diode, such as X1 in Fig. 4-26.
- h. Break in printed-circuit conductor.

To localize the defective stage, a scope may be used to trace the signal through the amplifier to the point of signal stoppage. Then, dc voltage measurements will usually permit the technician to pinpoint the defective component. Supplementary resistance measurements and continuity checks are also required in some situations.

## 2. Subnormal Output Level

When the audio output level is subnormal, the first step is to localize the weak stage in the amplifier. This requires a practical knowledge of stage gains in the various configurations. For example, an approximate gain of unity in a paraphase inverter is normal, whereas this figure would be greatly subnormal for a common-emitter stage. A scope is very useful to compare the amplitudes of input and output waveforms at each stage, in preliminary localization procedure. Subnormal output may or may not be accompanied by waveform distortion. If distortion is present, useful clues may be provided; for example, clipping distortion points to an incorrect operating point, whereas strong noise spikes point to a defect in the first audio section, such as a virtually open coupling capacitor.

Possible causes of subnormal output level from an audio amplifier are:

- a. Leaky coupling capacitor (such as  $C_C$  in Fig. 4-5).
- b. Open screen-bypass capacitor (such as  $C_{sg}$  in Fig. 4-6).
- c. Open cathode-bypass capacitor (such as  $C_k$  in Fig. 4-7).
- d. Shorted capacitor in negative-feedback loop (such as C5 in Fig. 4-8).

- e. Open coupling capacitor (such as C2 in Fig. 4-18).
- f. Shorted equalizing capacitor (such as C2 in Fig. 4-27).
- g. Open emitter-bypass capacitor (such as C1 or C3 in Fig. 4-31).
- h. Leaky transistor.

## 3. Excessive Distortion at Rated Output Level

Some types of amplifier defects will cause excessive distortion at the maximum rated output of the amplifier, whereas the distortion tends to disappear at lower output levels. This difference in response can provide clues concerning trouble possibilities when evaluated by an experienced technician.

Possible causes of excessive distortion at rated output level are:

- a. Subnormal supply voltage.
- b. Leaky cathode-bypass capacitor (such as  $C_k$  in Fig. 4-6).
- c. Leaky coupling capacitor (such as  $C_C$  in Fig. 4-6).
- d. Incorrect type of replacement transformer (such as T in Fig. 4-7).
- e. Open capacitor in negative-feedback loop (see C5 in Fig. 4-8).
- f. Incorrect load-correction network used with a particular speaker (see the optional load corrections in Fig. 4-10).
- g. Use of incorrect output-impedance tap (see T2 in Fig. 4-19).
- h. Marginal thermistor, such as  $R_T$  in Fig. 4-21.
- i. Leaky transistor.
- j. Leaky emitter-bypass capacitor, such as C1 or C3 in Fig. 4-27.

## 4. Excessive Distortion at All Output Levels

When excessive distortion occurs at all output levels, look for a major component defect that can produce serious waveform distortion regardless of the signal amplitude. With a practical knowledge of circuit action, the experienced technician can "weed-out" the possibilities by an analysis of the amplifier configuration. A scope is very useful to trace the signal and to determine the point at which distortion first occurs.

Possible causes of excessive distortion at all output levels are as follows:

- a. Component defect that "kills" the operation of one-half of a phase-inverter section; for example, Q2 might be open or shorted in Fig. 4-31.
- b. Component defect that "kills" the operation of

## Hi-Fi Stereo Servicing Guide

one-half of a push-pull amplifier section; for example, Q3 in Fig. 4-30 might be open or shorted.

- c. Open coupling capacitor in driver section; C1 in Fig. 4-29 is a typical example.
- d. Open equalizing capacitor, such as C2 in Fig. 4-28.
- e. Shorted equalizing capacitor, such as C2 in Fig. 4-27.
- f. Defective transducer; for example, the microphone in a PA system might be defective, or the speaker might be defective. A substitution test is advisable.

### 5. Noise and/or Hum Interference in Sound

Possible causes of noise and/or interference in the sound, due to defects in an audio amplifier are:

- a. Noisy resistor in the first stage (most of the noise is contributed by the first stage).
- b. Cold-soldered or unsoldered contact.
- c. Cable plug not fully inserted into connector.
- d. Noisy transistor.
- e. Defective filter capacitors produce hum, which should not be confused with heater-cathode leakage in a tube.
- f. Poorly grounded or ungrounded cable to the input of a high-gain amplifier will permit the entry and amplification of stray fields.
- g. Defective transducer.

### 6. Intermittent Operation

Intermittent conditions are unquestionably among the most vexing problems encountered at the service bench. Most "tough-dog" jobs fall into this category.

An intermittent may be mechanical, electrical, or thermal in nature. The time-honored procedure of jarring and tapping will sometimes turn up a mechanical intermittent. PC boards can be flexed gently, and the amplifier can be oriented in various positions to speed up the occurrence of certain types of mechanical intermittents. Electrical intermittents can occasionally be started by throwing the power switch on and off. Operation at undervoltage or overvoltage may also be helpful; however, in the case of solid-state receivers, never check with high supply voltage, as the transistors may be damaged. It is sometimes helpful also to vary the input signal level up and down from zero to maximum rated power output. Thermal intermittents can sometimes be turned up by heating component pigtailed with a soldering gun. However, use great caution when checking semiconductors for heat response. A coolant spray will sometimes cause an intermittent to start or stop. The chassis can be operated at elevated or reduced temperature—however, to avoid the possibility of transistor damage, solid-state receivers should not be operated at excessively high temperature.

Possible causes of intermittent operation in audio amplifiers are:

- a. Marginal open or short condition in a capacitor.
- b. Unstable composition resistor or operating control.
- c. Intermittent transistor.
- d. Poor connection, such as a cold-solder joint.
- e. Broken conductor inside an insulated lead.
- f. Microscopic break in a PC board.
- g. Defective function switch.
- h. Corroded plug or connector.

## Chapter 5

# Servicing Audio Amplifiers

A hi-fi mono or stereo audio system is basically divided into a preamplifier and a power-amplifier section. When stereo reproduction is provided, two separate audio channels are utilized, as exemplified in Fig. 5-1. If a mono program is being reproduced by the stereo system, both audio channels are driven by the same signal. The *power rating* or *output power* of the system in Fig. 5-1 is 20 watts. This means that the amplifier is normally capable of delivering 20 watts of audio-frequency power to the speakers without exceeding the manufacturer's distortion rating (such as 0.5 percent). Preamplifiers have two major functions: to increase a low-level audio signal to a value of about 1 volt rms, and to provide any equalization that may be required to obtain a flat frequency characteristic. Power amplifiers have one major function: to increase the power level of a 1-volt input signal to the rated power output of the amplifier without introducing objectionable distortion.

Common trouble symptoms caused by defects in the preamplifier or power-amplifier sections are:

1. Dead amplifier.
2. Weak output.
3. Distortion at all levels.
4. Distortion at high level only.
5. Noise and/or hum.
6. Poor balance.
7. Crosstalk.

### GENERAL DISCUSSION

Most hi-fi audio amplifiers use transistors, although many tube-type amplifiers are still in use. Solid-state amplifiers are often designed on a tran-

sistor-for-tube basis, although power amplifiers may exploit the unique characteristics of transistors to a greater degree than preamplifiers. Since an fm/am tuner or a multiplex adapter has a much higher output level than a phono cartridge or a tape head, less amplification is employed when the audio system is energized from the tuner, as seen in Fig. 5-2. Phono or tape-head signals are fed to the first stage. An equalization circuit is included for processing the output from a ceramic phono cartridge.

Negative feedback is widely used in audio systems; a feedback loop in Fig. 5-2 is included between the high-level driver and the first preamp to provide equalization for tape-head and magnetic-cartridge inputs. The other three equalization networks provide overall low-frequency compensation, high-frequency tape-head compensation, and phono-input compensation. Tuner and auxiliary-input signals are fed directly to the high-level driver. The high-level driver operates as an emitter-follower; its low-impedance output signal is fed to the tone-control amplifier through the volume control. Note that feedback-type tone-control networks are employed in this example.

Stereo-separation and balance controls are provided in Fig. 5-2. The separation control is in the emitter circuit of the high-level driver. As the separation control is advanced, progressively greater mixing of the signals takes place between the left and right channels, and this blending effect can be used to eliminate the "ping-pong" effect in some stereo program material. The balance potentiometer, in the output of the tone-control amplifier, is a linear control with equal resistances from its mid-position to ground—so no loss is inserted into either channel. As the control is rotated, more resistance is inserted

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into one channel and less into the other; in this way, right- and left-channel signal levels can be adjusted to obtain the proper volume relationship.

A typical solid-state preamplifier circuit is shown in Fig. 5-3. This configuration normally provides an output level of 1 volt rms with less than 0.3 percent harmonic distortion. Frequency-selective negative feedback is employed from the collector of Q2 to the emitter of Q1. The first two stages of this circuit

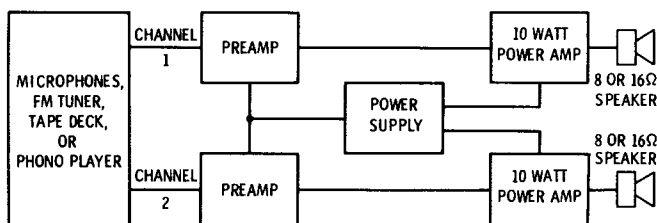


Fig. 5-1. Plan of hi-fi stereo audio system.

have a feedback bias arrangement with bias current supplied to the base of Q1 from the emitter branch of Q2. Thus, the bias on Q1 is directly proportional to the emitter current of Q2; if Q2 tends to draw more current, Q1 will draw less current in turn. Thus, the operating point of the direct-coupled arrangement is stabilized both with respect to component tolerances and temperature variations. Note that the 5-k level control in the emitter circuit of Q3 serves to stabilize this stage.

Flat frequency response on the phono position of the function switch requires a reproducing frequency-equalization curve as shown in Fig. 5-4. Distortion of an equalization curve is usually caused by an open, leaky, or shorted capacitor. The standard equalization curves for various tape speeds are shown in Fig. 5-5. Equalization responses are fixed. On the other hand, manual tone controls provide a choice of bass and treble response over a typical range of  $\pm 15$  dB, as exemplified in Fig. 5-6. A typical hi-fi stereo amplifier is illustrated in Fig. 5-7. This unit provides loudness compensation, a scratch filter, and a rumble filter, in addition to the bass and treble tone controls. Loudness compensation takes the form of about 11 dB bass boost, as shown in Fig. 5-8, and is employed at low settings of the volume control. That is, ear response to bass tones diminishes at low volume levels.

Fig. 5-9 shows the normal frequency-response curve for a scratch filter. This reduced high-frequency response is used when playing worn records that have become noisy. On the other hand, a rumble filter normally provides reduced low-frequency response, as shown in Fig. 5-10. A rumble filter is chiefly useful to attenuate interference caused by motor vibrations in economy-type record players. Some preamplifiers also provide a presence control that can be switched in or out. As shown in Fig. 5-11, a presence control provides approximately 5 dB of

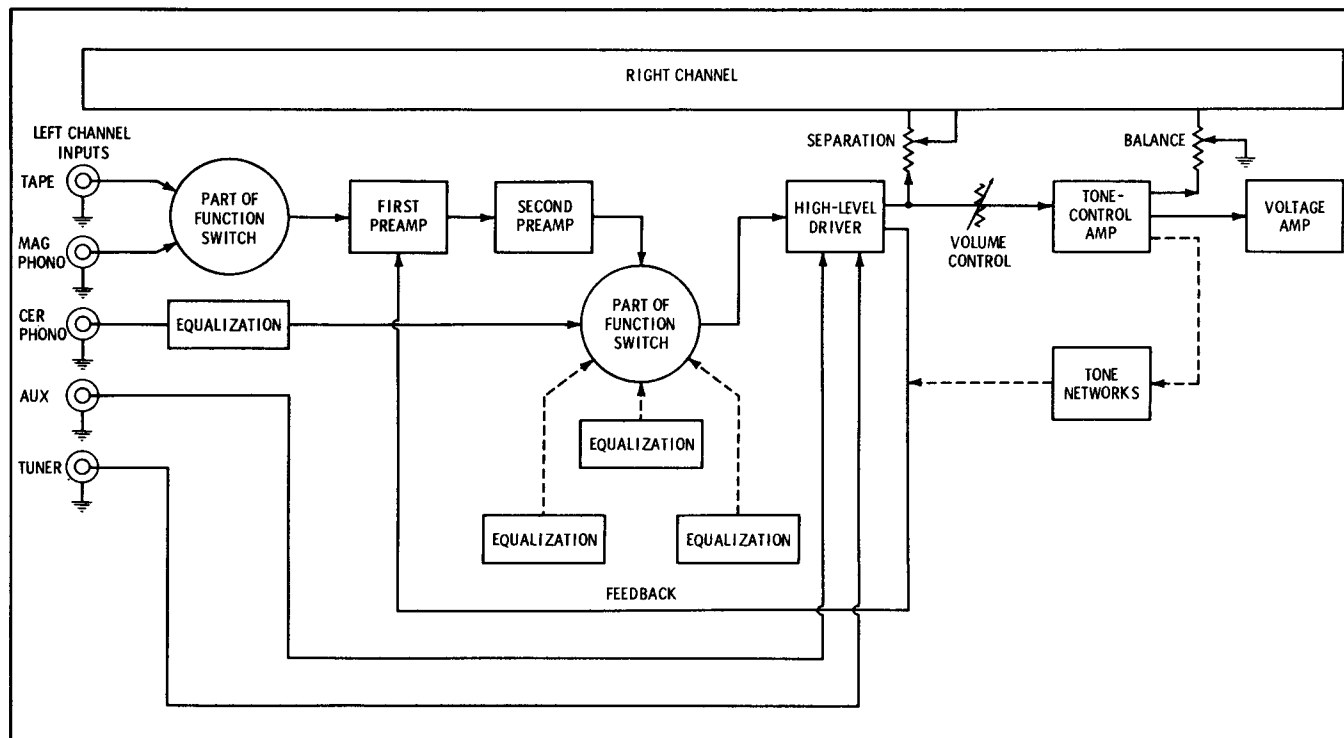


Fig. 5-2. Left channel of a transistorized stereo preamp with five inputs.

Fig. 5-3. Typical solid-state preamplifier circuit.

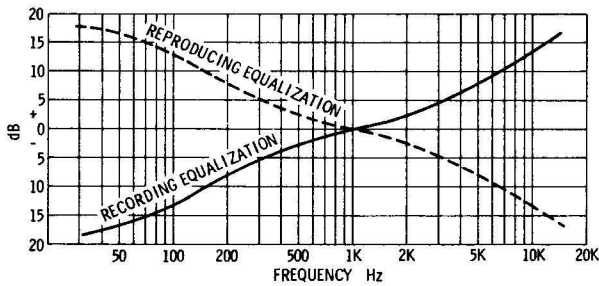
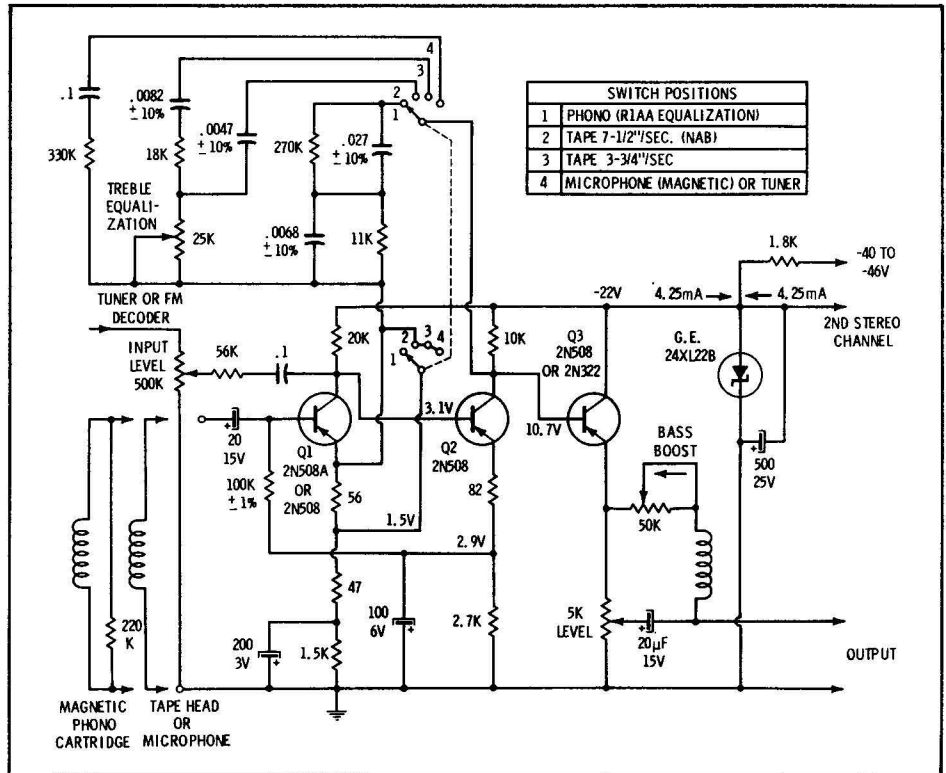


Fig. 5-4. Standard RIAA recording and playback equalization curves for disc recordings.

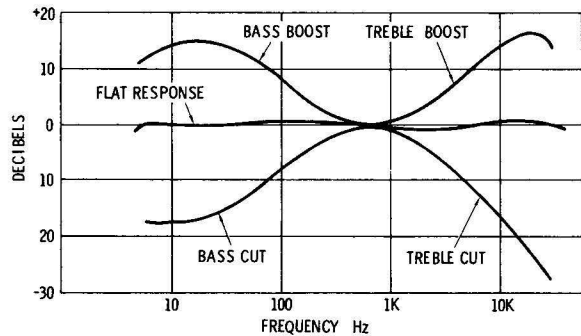


Fig. 5-6. Response curve for bass and treble tone controls.

peaking at about 2 kHz. Listeners sometimes prefer to use this mid-range boost when vocal selections are being reproduced.

Modern power amplifiers such as exemplified in Fig. 5-12 generally dispense with transformer coupling in order to obtain good square-wave response. Direct coupling is used, and the harmonic distortion

is normally less than 1 percent at a power output of 7 watts. Dc feedback is employed for temperature stabilization of all stages. It stabilizes the voltage division across Q4 and Q5, which operate in a

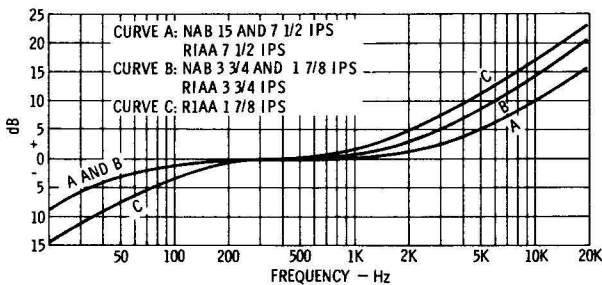


Fig. 5-5. Standard playback equalization curves in use for various tape speeds.



Courtesy Lafayette Radio Electronics

Fig. 5-7. Typical hi-fi stereo amplifier.

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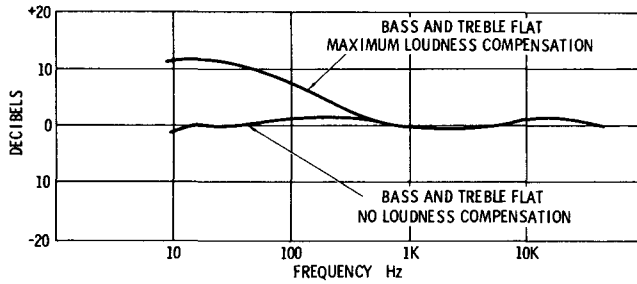


Fig. 5-8. Typical frequency response of a preamplifier.

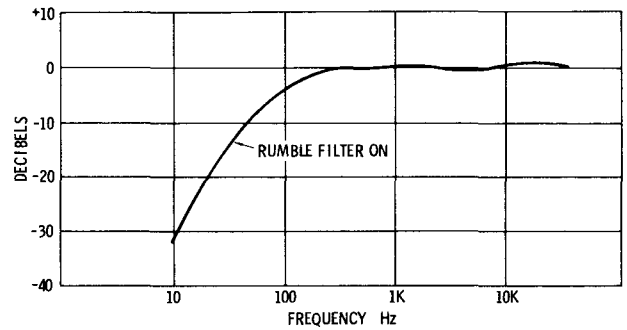


Fig. 5-10. Normal frequency response for a preamplifier with rumble filter operating.

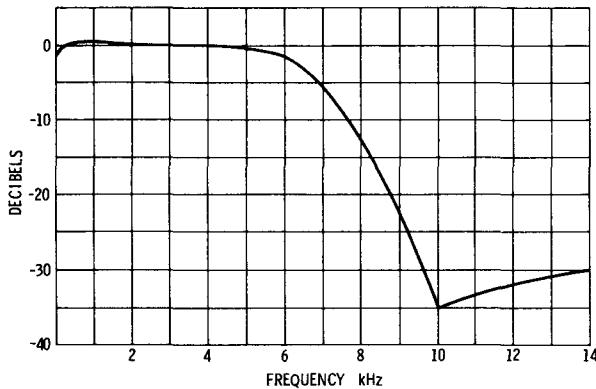


Fig. 5-9. Frequency response of a commercial scratch filter.

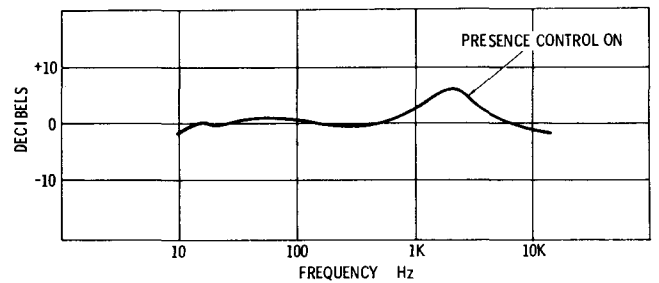


Fig. 5-11. Preamplifier response obtained with presence control on.

Class-B push-pull arrangement with single-ended output. Q2 and Q3 also operate in Class B with the Darlington configuration to obtain maximum current gain. An npn transistor is used in the Q3 socket, to obtain the required phase inversion for driving Q5. It also provides push-pull emitter-follower operation from the output of Q1 to the load. Since an emitter follower has 100 percent negative feedback (voltage

feedback), it provides low inherent distortion and low output impedance.

Transistors Q4 and Q5 in Fig. 5-12 employ a small forward bias of 10 to 20 mA to minimize crossover distortion. This bias is set by the voltage drop across the 390-ohm resistors that shunt the input to Q4 and Q5. Similarly, Q2 and Q3 are biased at about 1 mA to minimize crossover distortion. The voltage drop

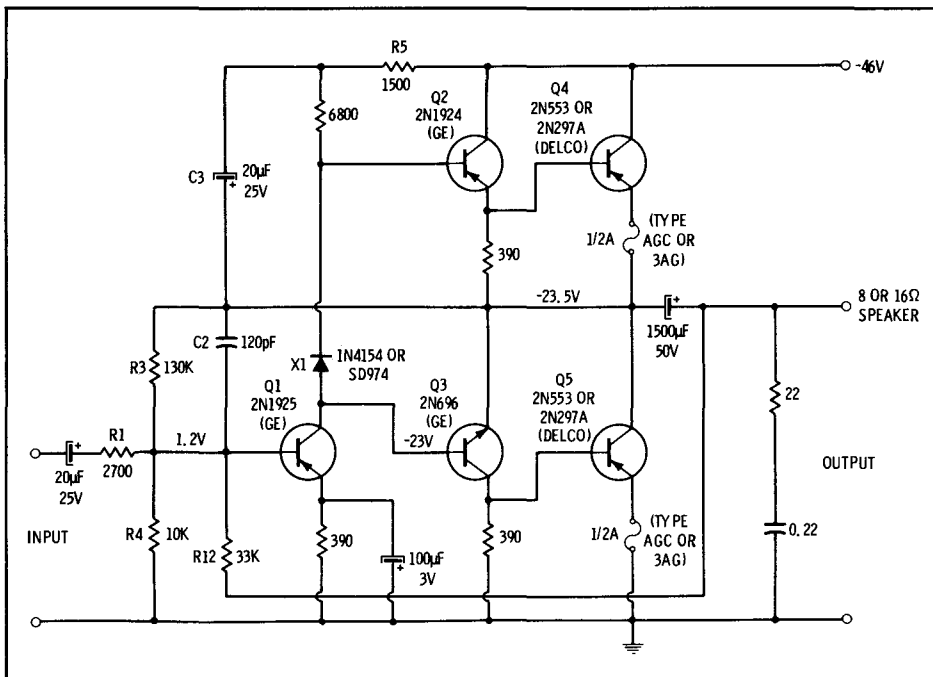


Fig. 5-12. Typical 10-watt amplifier circuit.

across silicon diode X1 supplies this bias. Junction diodes have a temperature characteristic similar to that of the emitter-base junction of a transistor. Therefore, X1 is a Class-A driver with a normal emitter current of about 3 mA. Negative feedback to the base of Q1 lowers its input impedance; resistor R1 limits the minimum value of source impedance. The value of R3 permits about half of the supply voltage to drop across Q5.

Note that positive feedback is employed in Fig. 5-12; about 11 dB of positive feedback is applied through C3 to R5. This is a bootstrap action which helps to compensate for the dissymmetry of the output circuit, so that the positive-peak signal voltage is essentially the same as the negative-peak voltage. Note that this positive feedback is offset in the system by about the same amount of negative feedback via R3 to the base of Q1. The net amount of negative feedback is approximately 14 dB, resulting from R12 connecting the output terminal to the input terminal. In addition, there is local feedback in the emitter-follower stages. A 1/2-ampere fuse is placed in the emitter lead of each output transistor for protective fusing of Q4 and Q5; the fuse resistance of about 1 ohm also provides some negative feedback. This local feedback improves bias stability and also holds up the frequency response at the high end of the audio range.

The speaker system in Fig. 5-12 is shunted by 22 ohms in series with 0.22  $\mu$ F to maintain a reasonably uniform speaker-circuit impedance and to minimize high-frequency phase shift. The output impedance is approximately 1 ohm, which provides good speaker damping, low distortion, and good frequency

response, whether an 8-ohm or 16-ohm speaker is used. A 1-volt rms input provides 8 watts output in normal operation. If the amplifier is overloaded, the protective fuses will blow. In music reproduction, a peak-power (music-power) output of 10 watts will not overload the amplifier, because the waveforms are such that a peak power of 10 watts is equivalent to a sine-wave power (rms power) of about 8 watts.

**Test Procedures**

Signal-tracing techniques with a scope are generally used to localize a dead stage in an audio system. An audio oscillator may be used to provide a constant-level signal for test. Signal-injection techniques may also be employed, using the speaker as an indicator, and moving the injection point back stage-by-stage. Some technicians use a simple noise generator, or a harmonic oscillator, instead of an audio oscillator. After the dead stage has been localized, dc voltage and resistance tests will usually serve to close in on the defective component. Since audio configurations are often dc-coupled, we need to know how interactions between cascaded stages can be analyzed. Basically, interactions are merely applications of Ohm's law. Let us consider the effects of component failures in a typical dc-coupled amplifier.

With reference to Fig. 5-13, if C3 short-circuits, the collector voltage on Q1 will increase because the bias voltage on Q1 is decreased, which lowers the collector current through R3 and thereby increases the collector voltage due to less drop across R3. Next, let us suppose that the voltage drop across R7 reads 4.1 volts; we would suspect that the bias on Q1 is

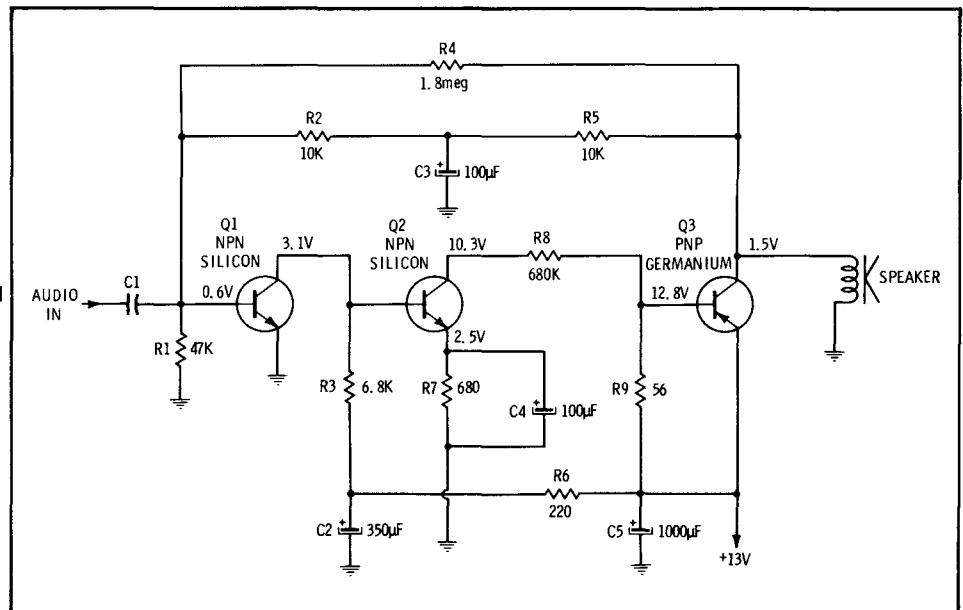


Fig. 5-13. Three-stage direct-coupled configuration.

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low. If the bias on Q1 is low, the bias on Q2 will increase, causing increased current demand and a greater drop across R7. Now, let us suppose that the collector voltage of Q2 reads lower than normal; we would suspect that the bias on Q3 will be higher than normal. That is, if the collector voltage on Q2 is low, it probably means increased current through R8 and increased current through R9, so that the bias on Q3 increases.

Next, let us consider the effect of open capacitors. In case C3 in Fig. 5-13 opens up, the audio signal will not be bypassed and about 30 percent of the audio from the collector of Q3 will feed back to the base of Q1. This is a negative-feedback situation, and the amplifier gain will be drastically reduced. Note that if C4 opens up, there is a considerable loss of gain in the Q2 stage, due to the negative feedback introduced by R7. In case C2 opens up, the frequency response becomes distorted, and in case C5 has partially deteriorated, the audio system will squeal due to the positive feedback that occurs. The easiest way to check a suspected open capacitor is to bridge it with a known good capacitor to see if the trouble symptom disappears.

Let us consider the symptoms of transistor failure in Fig. 5-13. In case Q1 develops a collector-emitter short, the collector becomes grounded and the collector voltage is zero. In turn, the base of Q2 becomes zero. This cuts off Q2 and its collector current falls to zero; therefore, there is no voltage drop across R8 or R9. In turn, Q3 is cut off. In summary, all three transistors are cut off in case of a collector-emitter short in Q1. Next, let us suppose than an open occurs in Q2; with zero collector current, there

is no current demand and no forward bias on Q3. Consequently, Q3 is cut off. Again, let us suppose that Q1 opens; its collector voltage will go more positive, and the base voltage on Q2 goes more positive. In turn, Q2 draws more current, which increases the voltage drop across R9 and causes Q3 to draw more current. Since Q1 is open, the feedback to the base of Q1 is defeated. However, R7 and R8 act to limit the current flow and thereby prevent transistor burnout. Note also that R7 and R8 limit the current flow in case Q2 becomes short-circuited.

Now, let us consider a very useful trick of the trade that is sometimes overlooked by the technician apprentice. Fig. 5-14 shows the plan of a basic stereo amplifier. In the event that a trouble symptom develops in Channel 2, for example, we can verify all of the Channel 2 waveforms and voltages by comparison with the corresponding waveforms and voltages in Channel 1. This can be of considerable assistance, especially when working with a modern and somewhat unconventional configuration such as that shown in Fig. 5-12. A common complaint in stereo reproduction is weak or low volume on one channel, even with the balance control set to one end of its range. Experienced technicians will first suspect a defective power transistor. Open or deteriorated electrolytic capacitors are also common offenders. A high-power transistor tends to have shorter life than a low-power transistor, because high-power operation requires corresponding heat dissipation and a peak output that approaches maximum rated output for the transistor.

Whenever an amplifier is tested on the bench, it must be properly loaded to obtain normal operation,

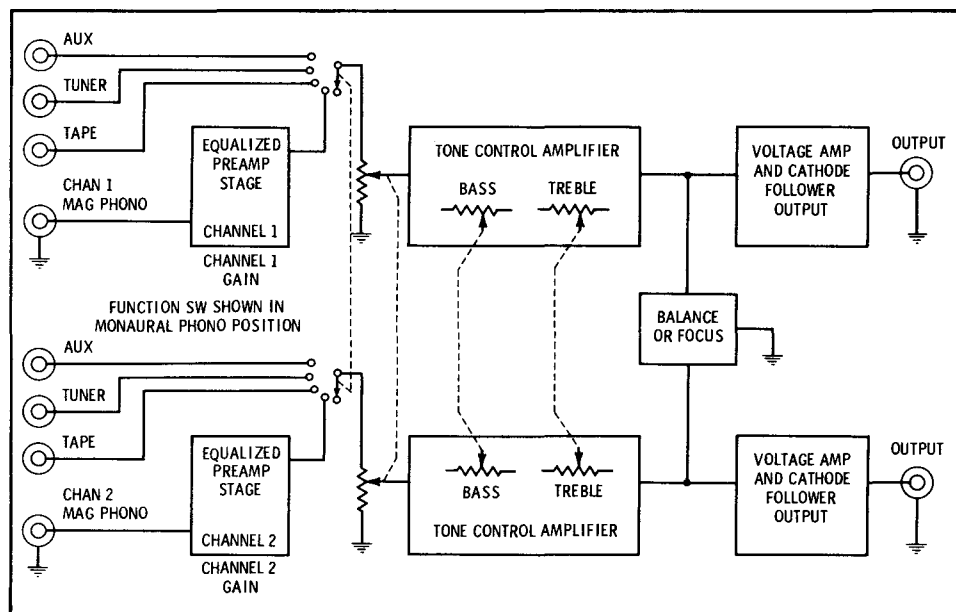


Fig. 5-14. Block diagram of a basic stereo amplifier.



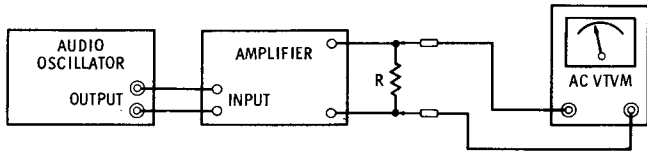


Fig. 5-15. Test setup for measuring power output.

and also to prevent possible damage to the output transistors. Fig. 5-15 shows the basic test setup. Resistor R is a suitable power type, and has a value equal to the rated load impedance for the amplifier. Since the audio oscillator provides a sine waveform, the power output is calculated by reading the rms value of output voltage and using the formula:

$$P = \frac{E^2}{R}$$

where,

- P is the power in watts,
- E is the voltage in rms values,
- R is the resistance in ohms.

Harmonic-distortion measurements are made with the test setup shown in Fig. 5-16. The basic test is made at maximum rated power output, and at a frequency of 1 kHz. However, a more thorough distortion analysis can be made by measuring the percentage of distortion at various other frequencies, such as 50 Hz and 10 kHz. A marginal defect in a feedback network, for example, may show up as excessive harmonic distortion at 50 Hz, whereas the distortion is within rated value at 10 kHz. In a well-equipped shop, harmonic-distortion measurements may be supplemented by intermodulation-distortion tests (see Fig. 5-17). An IM distortion meter usually has built-in load resistors, so that the rated load for the amplifier can be switched into the test circuit. The IM test signal is a two-tone wave, ordinarily 60 Hz and 60 kHz. It provides a somewhat more critical test than a simple harmonic-distortion measurement.

One of the advantages of an IM test is that certain types of distortion, such as clipping, show up more prominently than in an HD test. In any case, the percentage of IM distortion tends to run a bit higher than the percentage of HD, as exemplified in Fig. 5-18. These curves represent normal operation for a

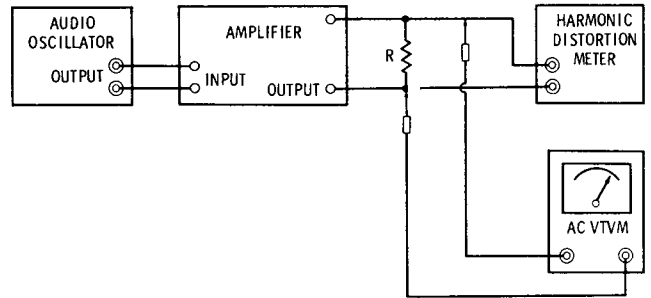


Fig. 5-16. Test setup for measuring harmonic distortion of an audio amplifier.

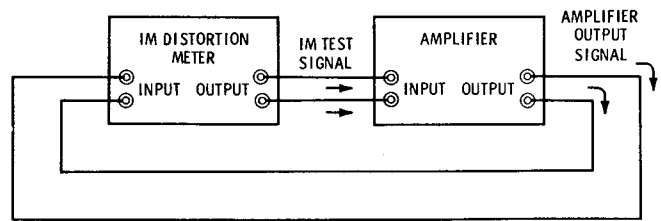


Fig. 5-17. Test setup for measuring intermodulation distortion.

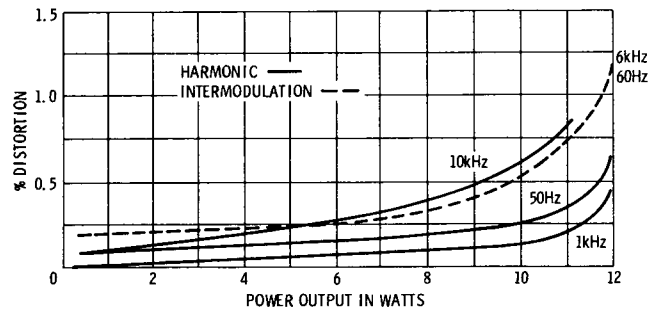
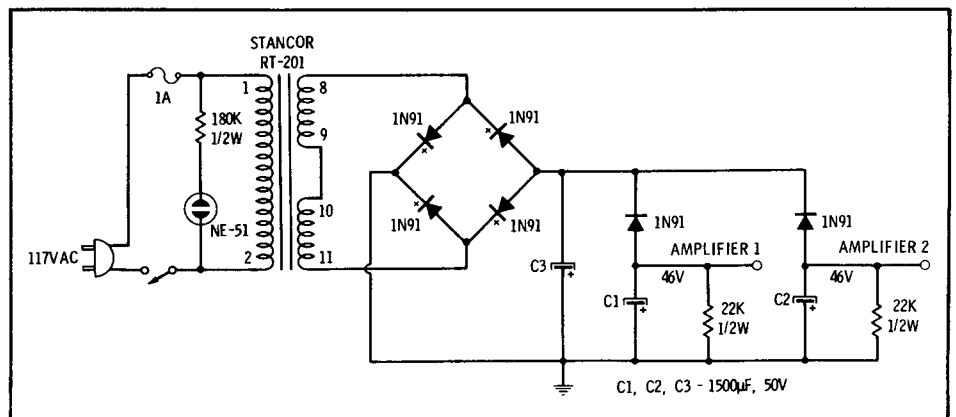


Fig. 5-18. Harmonic and intermodulation distortion percentages for a typical high-fidelity amplifier.

Fig. 5-19. Power supply for a stereo system.

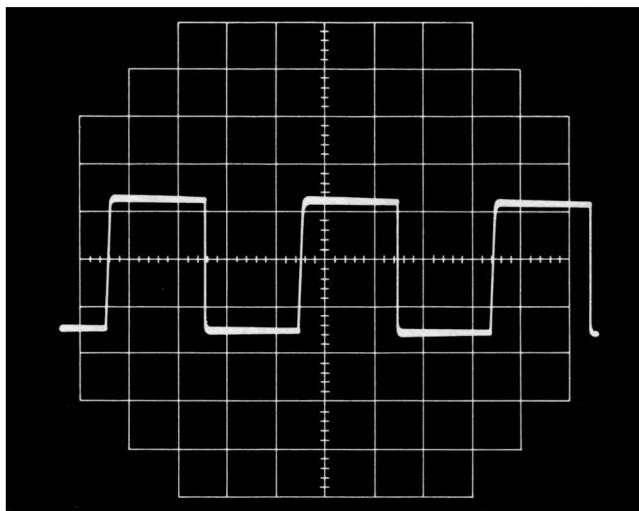


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typical hi-fi amplifier. We observe that the IM distortion measures consistently higher than the HD unless the HD test frequency is near the upper limit of the audio range. The IM curve crosses over the 10-kHz HD curve at about half rated power output, in this example. Note that both IM distortion and HD increase very rapidly near the maximum rated power output for the amplifier.

To avoid confusion between amplifier defects and power-supply defects, it is good practice to monitor the output of the power supply while making the foregoing tests. In a stereo system, the same power supply is employed for both channels. However, to minimize crosstalk between channels, elaborate designs are often used to obtain a very low internal impedance. For example, the power supply shown in Fig. 5-19 utilizes diode decoupling between the two outputs. This makes the outputs practically independent of each other, within rated limits of current demand. In normal operation, the arrangement in Fig. 5-19 provides about 80 dB of separation between the two audio channels when operating a pair of amplifiers such as those shown in Fig. 5-12. In other words, when one channel is driven to maximum rated power output, the crosstalk into the unenergized channel is normally 80 dB down.

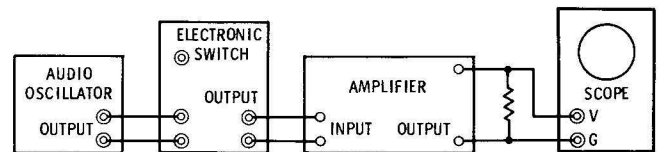
Square-wave testing is a useful analytical procedure, provided the normal square-wave response of the amplifier is known. Manufacturers of hi-fi amplifiers often specify this characteristic. For example, the amplifier shown in Fig. 5-12 normally has the 2-kHz square-wave response illustrated in Fig. 5-20. The rise time of this square wave is less than  $2 \mu\text{s}$ , corresponding to a bandwidth of approximately 160



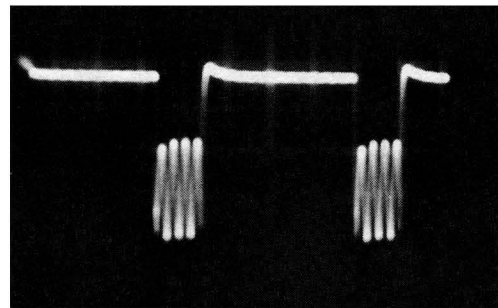
Courtesy General Electric Company

Fig. 5-20. Normal 2-kHz square-wave response for the amplifier in Fig. 5-12.

kHz. A related but somewhat more sophisticated transient-response test employs a tone-burst signal, such as that illustrated in Fig. 5-21. This is basically a rectangular waveform on which is superimposed an audio sine wave. Hi-fi manufacturers will often provide specifications for tone-burst response; otherwise, the technician may be able to make a comparative tone-burst test on another amplifier of the same type which is known to be in good operating condition. Similarly, the tone-burst response on one amplifier in a stereo system can be compared with that of the other amplifier.



(A) Test setup.



(B) Typical tone-burst signal.

Fig. 5-21. Tone-burst test of an amplifier.

Noise can be a problem, particularly in high-gain audio systems. To localize a noisy section or circuit, it is helpful to make step-by-step silencing tests, starting with the output stage. A non-polarized motor-starting capacitor is convenient, because polarity is then inconsequential. Proceed by shunting the base and emitter leads of each transistor in turn with the capacitor, while listening to the noise output from the amplifier. A large shunt capacitance will "kill" the noise, provided that the test is made between the noisy component and the speaker. As soon as you pass the noisy circuit, application of the capacitor has little or no effect. Therefore, you can eliminate all but a few components from suspicion by a simple test. Deteriorating transistors are prime suspects, although resistors can also become noisy unless they are wirewound.

There are two principal sources of hum; power-supply trouble is the most usual cause, and stray-field pickup is the other frequent offender. In the case of a tube-type amplifier, it is assumed that the

tubes have been checked previously for heater-cathode leakage. Stray fields are picked up either by capacitive or by inductive coupling. For example, in a transformer-type amplifier, the cores of the inductors (particularly in the preamp section) are generally oriented for minimum hum pickup. In case a universal replacement type is mounted in the same location, it may still be necessary to vary the orientation of the new transformer experimentally to find the position of minimum hum pickup. The original grounding points should also be strictly observed whenever a component is replaced, because circulating ground currents in low-level circuits can produce baffling hum problems.

Hum pickup due to capacitive coupling is encountered in high- or medium-impedance circuits when shielding is poorly grounded or omitted, or if a component defect has greatly increased the input impedance of the circuit. For example, suppose that the coupling capacitor opens up in an emitter-follower circuit. The result can often be much the same as an open coupling capacitor in a conventional vacuum-tube stage. That is, the input impedance increases, and in some circuits the increase is so great that even weak stray fields will drive the amplifier to full output. Localization is a straightforward procedure using conventional signal-tracing tests.

### ANALYSIS OF COMMON SYMPTOMS

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

#### 1. Dead Amplifier

Possible causes of signal stoppage in a preamp or power amplifier are:

- a. Defective function switch (see Fig. 5-2).
- b. Open coupling capacitor, such as C1 in Fig. 5-13.
- c. Blown fuse in emitter circuit of output transistor (see Fig. 5-12).
- d. Open or shorted transistor.
- e. Shorter zener diode (see Fig. 5-3).
- f. Cracked PC board.
- g. Shorted capacitor (such as the 500- $\mu$ F capacitor in Fig. 5-3).
- h. Defective input level control (see Fig. 5-3).

Most defects, with the exception of open capacitors, can be pinpointed by means of dc-voltage and resistance measurements; an open capacitor can be located with either signal-tracing or signal-injection tests. Voltage measurements at transistor terminals

can be supplemented by checking with an in-circuit transistor tester. "Turn-on" and "turn-off" transistor tests are also helpful; a "turn-off" test is most informative in doubtful situations.

#### 2. Weak Output

Possible causes of weak output in an audio amplifier are as follows:

- a. Open or dried-out electrolytic capacitor, such as C3 in Fig. 5-13.
- b. Transistor with substantial collector leakage.
- c. Low supply voltage.
- d. Leaky coupling capacitor that upsets the base-bias voltage.
- e. Off-value resistor (less likely than other defects).

The normal stage gains are generally unknown; however, the experienced technician can usually "size up" a circuit and make a reasonable estimate of the normal gain. In the case of a stereo amplifier, the signal amplitudes can be compared between corresponding points in the two channels. Comparison tests can also be made with a preamplifier, if a similar model in good working condition is available.

#### 3. Distortion at All Levels

Possible causes of distortion at both high and low levels are:

- a. Leaky coupling capacitor, as exemplified in Fig. 5-22.
- b. Defective transistor.
- c. Burned resistor in output stage, as exemplified in Fig. 5-23.
- d. Shorted capacitor, such as C3 in Fig. 5-13.

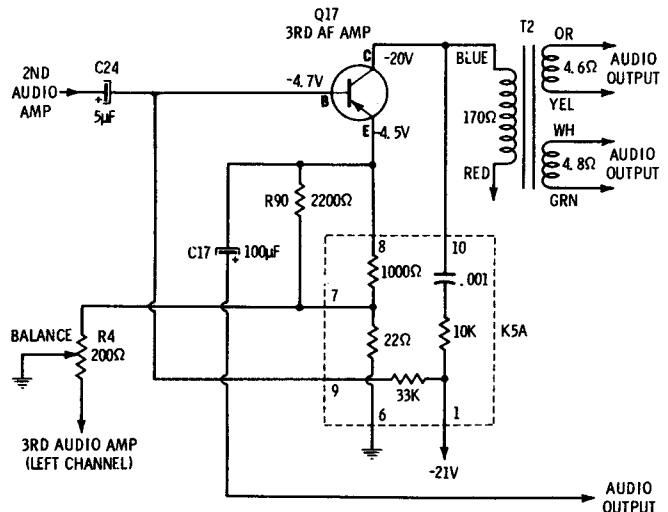


Fig. 5-22. Leaky coupling capacitor C24 causes distortion in the right channel.

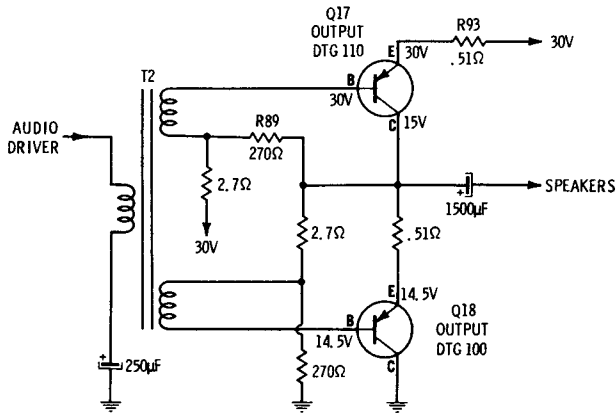


Fig. 5-23. Shorted transistor and burned resistors (R89 and R93) can cause extreme distortion and low volume.

- e. Blown emitter fuse in one side of a push-pull circuit.
- f. Cracked PC board causing an open feedback loop, or an open circuit in a push-pull stage.

**4. Distortion at High Levels Only**

Possible causes of distortion at high levels only are as follows:

- a. Low supply voltage.
- b. Leaky emitter bypass capacitor.
- c. Transistor with collector leakage.
- d. Marginal diode, such as X1 in Fig. 5-12.
- e. Off-value bias resistor (less likely than other defects).

When distortion occurs at high levels only, we know that some component defect is causing limited power-handling capability. Thus, a defective power supply can reduce the supply voltage and thereby cause compression or clipping at high power levels. Test methods are dependent on the amount of distortion. If the distortion is substantial, a scope can be used to localize the defective component. However, if an amplifier has only 1 or 2 percent more distortion than rated at maximum power output, a scope is almost useless, because it is virtually impossible to pick out small percentages of distortion in a sine-wave display. Therefore, we must use an HD meter instead of a scope to signal-trace small amounts of distortion.

**5. Noise and/or Hum**

Possible causes of noise and/or hum in an audio system are:

- a. Defective transistor.
- b. Intermittent capacitor, as exemplified in Fig. 5-24.

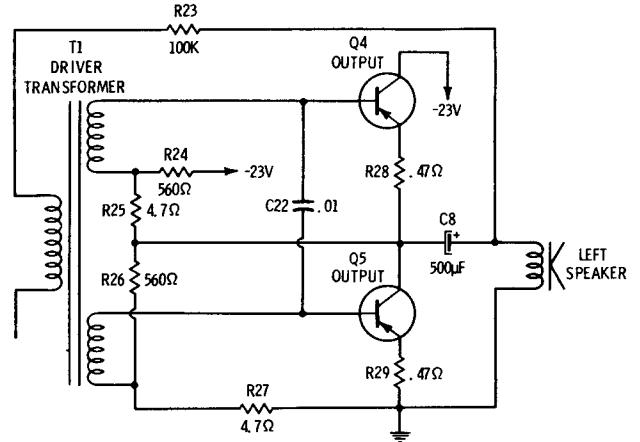


Fig. 5-24. Intermittent noise is caused by defective C8.

- c. Defective power supply (check ripple with scope).
- d. Ungrounded volume or tone control.
- e. Open base resistor, or cracked PC board.

**6. Poor Balance**

Possible causes of poor balance in a stereo amplifier are as follows:

- a. Defective transistor (balance control out of range).
- b. Capacitor with unstable leakage resistance.
- c. Worn and noisy balance control.
- d. Fluctuating voltage from one side of the power supply.
- e. Thermally unstable resistor (less likely than other defects).

An accurate test of balance and of the ability of a stereo amplifier to maintain good balance is shown

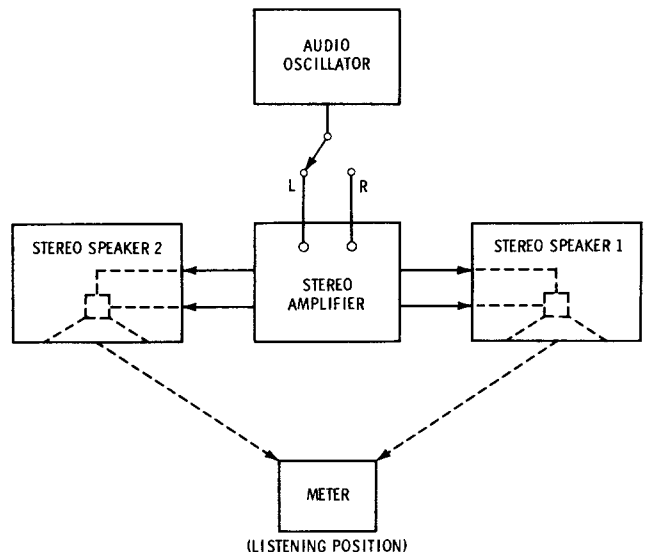


Fig. 5-25. Test setup for checking stereo-amplifier balance.

in Fig. 5-25. A sound-level meter is placed midway between the two speakers, and the output from an audio oscillator is applied in turn to the L and R inputs of the amplifier. The balance control is properly adjusted when the sound-level meter reads the same value for either speaker. If the amplifier has good stability it will retain correct balance after an extended period of operation.

### **7. Crosstalk**

Possible causes of crosstalk in a stereo amplifier are:

- a. Excessive output impedance in power supply.
- b. Defective decoupling diode (see Fig. 5-19).

- c. Leakage between contacts of function switch.
- d. Leakage between conductors of PC board.
- e. Microphonic tube in tube-type stereo amplifier.

In nearly all cases, crosstalk is most troublesome when one channel is being driven to maximum rated output, with no input to the other channel. It is advisable to make a scope check of the ripple on the supply lines to both amplifiers, inasmuch as crosstalk is very often caused by coupling through the power-supply leads. That is, the defect is usually in the power supply. If the trouble is elsewhere, as in the function switch, or in the PC board, waveform checks with a scope provide a straightforward approach to the trouble location.

## Chapter 6

# Installing Hi-Fi Speakers

Speaker installation is an essential consideration in any hi-fi system; a poor acoustic arrangement can defeat the characteristics of the finest electronic components. Similarly, a poor acoustic arrangement can defeat the potential of the finest speakers. Acoustics is the science of perceived sound, and its basic concern is the clarity with which sounds can be heard in a room or other listening space. That is, certain physical properties of a room affect the quality of sounds radiated by a speaker. Since the clarity of sound perception involves a subjective judgment, acoustics has both psychological and physical aspects. Many important characteristics of an acoustical situation can be measured; in other words, there are various quantitative approaches to evaluation of room "response." The experienced technician makes a clear distinction between poor acoustic arrangements and unsatisfactory speaker characteristics.

Common trouble symptoms caused by poor acoustics or speaker defects are:

1. Echoes
2. Resonances
3. Room distortion
4. Speaker distortion
5. Rattle
6. Weak output

### GENERAL DISCUSSION

There are four basic acoustic factors that need to be considered by the hi-fi installation technician. These factors are (1) room noise level, (2) reverberation characteristics, (3) optimum loudness, and (4) room sound distribution. The same sound-level meter used to check the balance of a stereo ampli-

fier is suitable for measurement of the room noise level. Noise tends to mask the sound radiated by a speaker. Since noise often has a dominant frequency, the masking effect is greatest at this frequency. For example, if room noise is dominated by the rumble of street traffic, the acoustic balance can be improved by judicious boost of the lower audio frequencies. On the other hand, room noise in rural areas is often dominated by medium- or high-pitched background noises. In this situation, the acoustic balance can be improved by appropriate boost of medium or high audio frequencies.

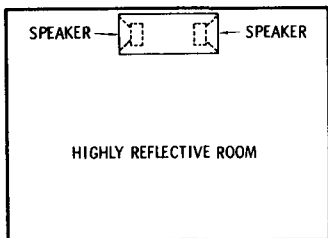
Reverberation is another important room characteristic. Sound must "hang on" long enough to allow musical notes or spoken words to sound natural, but not to the extent that one word tends to blur the next word in normal speech. It has been determined that it is often desirable to have low frequencies reverberate longer than high frequencies in a given room. Whether or not specific measurements are made, or whether evaluation of reverberation is made by ear, we are basically concerned with how long it takes sound to decay in the room, and how the decay time varies at different frequencies. Since commercial sound-analysis equipment is comparatively expensive, the hi-fi installation technician is usually dependent upon practical experience in this area.

Loudness and clarity are separate considerations. Optimum loudness occurs at a level that makes the listener unaware of this factor. Contrary to popular opinion, an abnormally loud reproduction of sound does not necessarily mean clearer sound. Echoes, an aspect of reverberation, will increase the total amount of sound, although often at the expense of intelligibility. Room size may be an essential factor

## Hi-Fi Stereo Servicing Guide

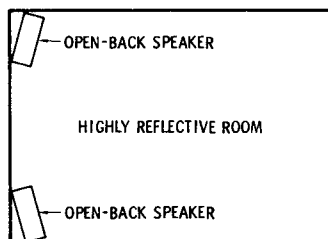
in planning the installation. That is, good acoustics at one location in a room should not be provided at the expense of marginal acoustics at other locations in the room. This aspect of acoustic planning involves primarily the types of speakers to be used, and their placement in the room.

It is obvious that various rooms have different acoustic characteristics, apart from size. In other words, methods of providing good acoustics in a living room that is furnished with wall-to-wall carpet and plush furniture, plus heavy drapes, differ considerably from the methods that are exploited in a recreation room with tiled floor and panelled or bare walls, plus wrought-iron furniture. We will find that any good hi-fi speaker system, with the L and R speakers properly spaced, will provide good acoustic characteristics in an acoustically damped room. On the other hand, the same hi-fi speaker system might be completely unsatisfactory in a highly reflective recreation room. One basic approach to acceptable stereo reproduction in a room with strong reflections is to install an oppositely directed speaker package, as shown in Fig. 6-1. In most cases, this is the optimum plan.

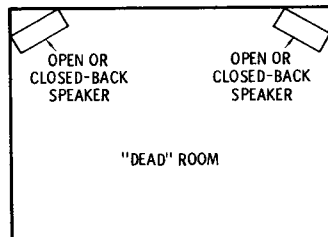


**Fig. 6-1. Speakers mounted at opposite ends of cabinet.**

Open-back speakers can also be used to advantage in a highly reflective room, if located as shown in Fig. 6-2. This is a good basic layout that can often be used in other situations; the speakers are placed in the corners at floor level. The corner walls and floor serve as sound directors. It is less desirable to mount the speakers half-way between the floor and ceiling. If the floor location is impractical, the best alternative is to mount the speakers at the ceiling in opposite corners. Next, if the installation is to be made in a "dead" room with little or no reverbera-

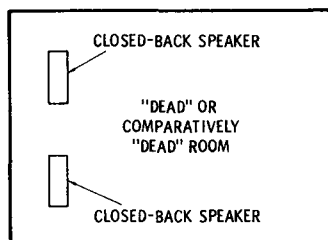


**Fig. 6-2. Open-back speakers in corners of a room.**



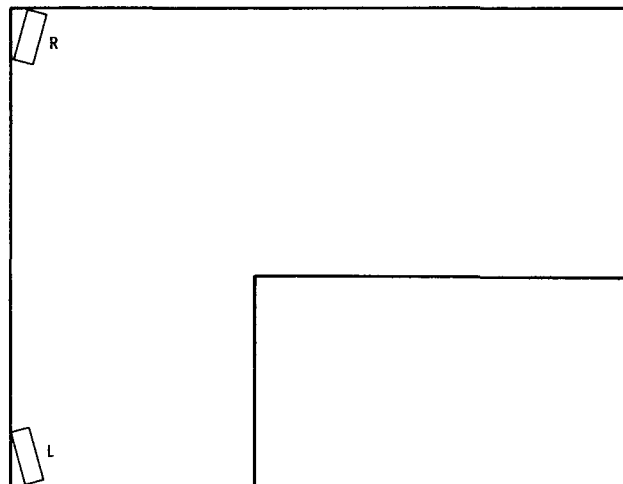
**Fig. 6-3. Preferred arrangement of speakers in a "dead" room.**

tion, better stereo reproduction will often be obtained with the speaker placement shown in Fig. 6-3. Either open- or closed-back speakers may be used, and some advantage is usually realized by placing the speakers at floor level. Fig. 6-4 shows the conventional arrangement using closed-back speakers; the disadvantage in this example is that good stereo reproduction is obtained only along the center line of the room. If the listening positions are in this line, there is no objection to the plan.



**Fig. 6-4. The conventional arrangement is less desirable.**

The basic considerations become modified in accordance with substantial variations in room shape. Aside from the rectangular room, the L-shaped room is most often encountered by the hi-fi installation technician. Although a do-it-yourselfer tends to install stereo speakers as pictured in Fig. 6-5, this is an example of poor practice, inasmuch as the radiation from the L speaker is not only impeded, but un-



**Fig. 6-5. Poor placement of speakers in an L-shaped room.**

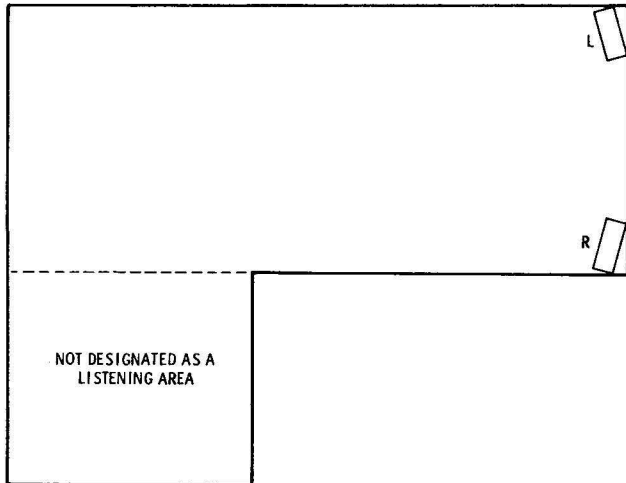


Fig. 6-6. Preferred placement of speakers in an L-shaped room.

dergoes many more reflections than the radiation from the R speaker. The preferred speaker arrangement in this situation is shown in Fig. 6-6. Although a portion of the room is excluded as a stereo listening area, the advantage of this arrangement is that the acoustical characteristics in the main area are much better than in the example of Fig. 6-5.

Next, let us consider the characteristics of the arrangement shown in Fig. 6-7. In this example, the speakers are too far apart; the result is poor distribution of sound and a serious "hole in the middle." Customers who have had little or no experience with stereo reproduction sometimes believe that they must hear the L and R separation prominently. A listener with this misconception will tend to place the L and R speakers so far apart that an obtrusive ping-pong effect results. The goal of realism is accordingly lost in this overemphasis on separation. An occasional situation is encountered in which the

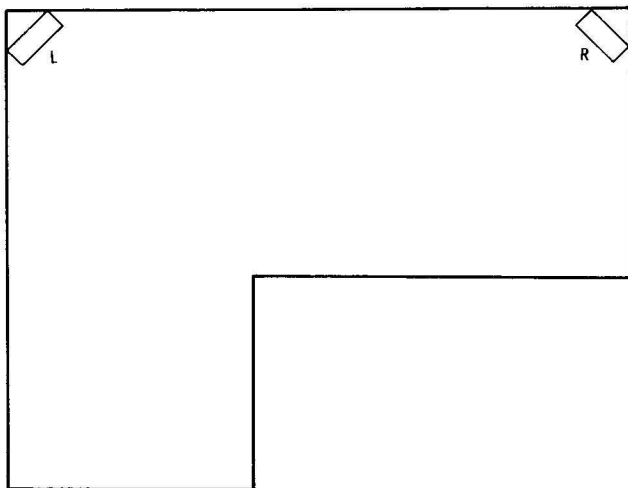


Fig. 6-7. This placement gives poor sound distribution and a serious "hole in the middle."

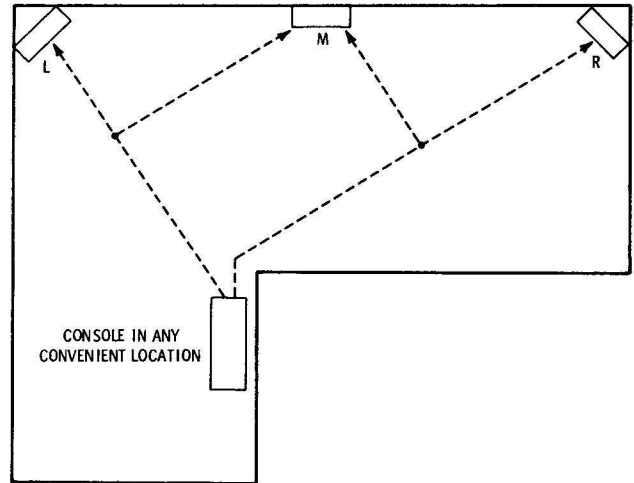


Fig. 6-8. A mono speaker in the center improves the acoustical characteristics.

arrangement of Fig. 6-6 is not feasible because of room layout, decor, etc. In such a case, the arrangement of Fig. 6-7 will have much better acoustic characteristics if a third speaker is added between the L and R speakers, as shown in Fig. 6-8. This speaker is essentially a mono speaker, inasmuch as it is driven by both the L and the R signals. A separate volume control must be included with the mono speaker, and adjusted for best acoustic balance.

Only an anechoic chamber such as that illustrated in Fig. 6-9 is completely free from reverberation or acoustic reflection. No domestic installation is completely "dead," and reflected wavefronts are promptly set up and trail close behind the main acoustic wavefront, as exemplified in Fig. 6-10. An acoustic wavefront travels at a speed of 1100 feet per second, so that it takes only 20 milliseconds for the wave to reach the opposite wall of a 22-foot room. Thereupon, the main wavefront becomes a

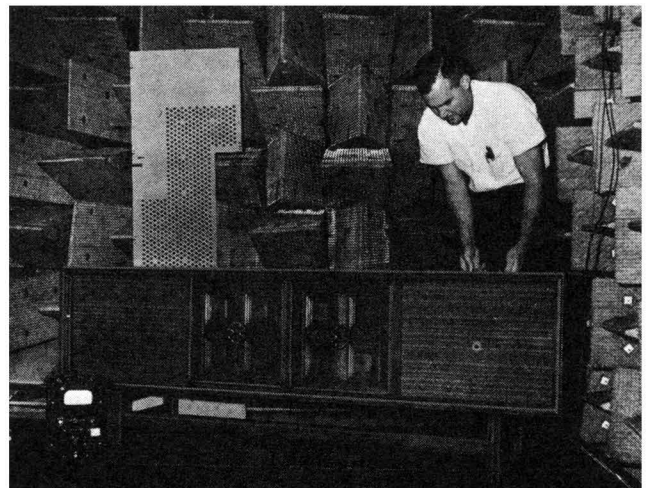
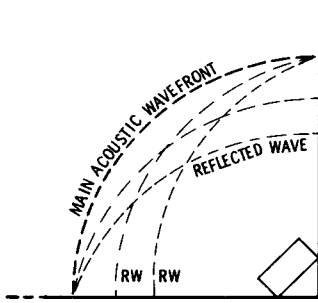


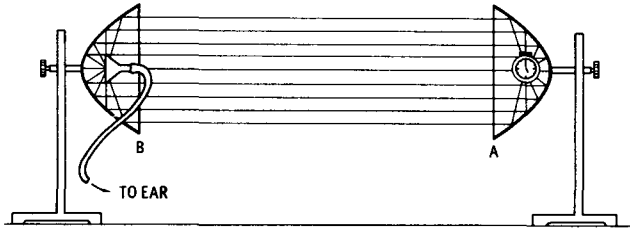
Fig. 6-9. An anechoic room.





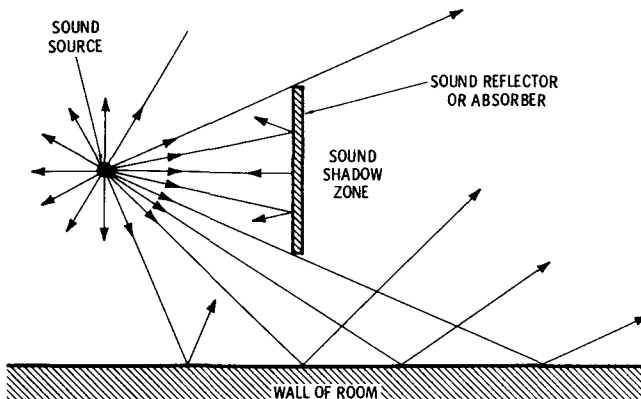
**Fig. 6-10. Development of initial reflected wavefronts.**

reflected wavefront, and the listening area is soon populated by a multitude of reflected wavefronts. However, each reflection entails a substantial energy loss, and re-reflections may fall below the threshold of audibility, except in special cases. An extreme example of reflection and acoustic focusing is a "whispering gallery," as exemplified by St. Paul's Cathedral in London. A whisper at one side of the cathedral is distinctly audible at the other side, a hundred feet distant.

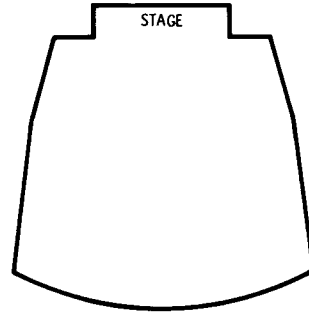


**Fig. 6-11. Principle of acoustic focusing.**

Acoustic focusing is provided by the walls and floor of a corner in a room, when a speaker is mounted in the corner. However, sharp focusing is obtained by means of curved surfaces (ideally parabolic contours) in architectural design. The principle of the whispering gallery is illustrated in Fig. 6-11. "Sound shadows" are produced by interfering surfaces in the path of an acoustic wavefront, as shown in Fig. 6-12. As in the case of an acoustic reflector, an object must have a comparatively large surface area to produce a significant sound shadow. Note also that a limited area which shadows high audio fre-

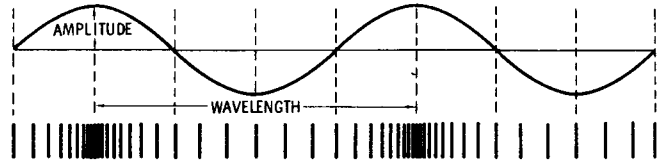


**Fig. 6-12. Large objects produce sound shadows.**



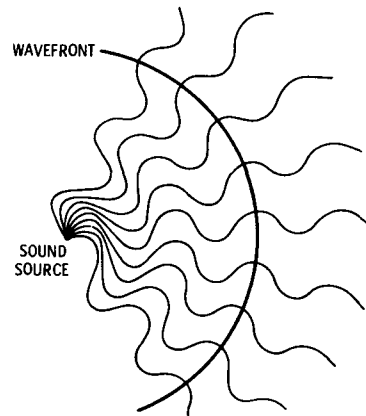
**Fig. 6-13. Shape of hall designed for optimum acoustic characteristics.**

quencies will have little effect on low frequencies. It follows from the basic principles of acoustics noted previously that there are many factors which must be taken into account if a room or hall is to be designed for optimum sound reproduction. Carnegie Hall is an example of outstanding acoustic design; a plan view is seen in Fig. 6-13. A basic acoustic rule of thumb states that the best shape for a listening room is a height/width/length ratio of 1:1.27:1.62.



**Fig. 6-14. Sine wave corresponding to compressions and rarefactions in a sound wave.**

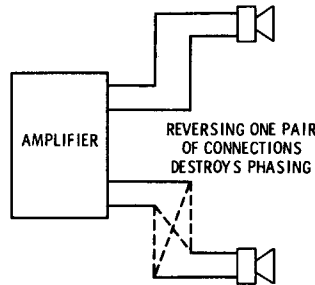
We know that sound waves are propagated as compressions and rarefactions in the distribution of air molecules; however, it is helpful to represent a sound wave in sine-wave form, as shown in Fig. 6-14. This representation clearly points out the phase characteristic. Phase relations in an acoustic wavefront may be visualized as shown in Fig. 6-15. It is evident that reflected wavefronts will aid or oppose, depending on their phase relation. This phase relation changes as the listener changes his position in the room. In most installations, this variation is not troublesome, because the reflected waves decrease



**Fig. 6-15. Phase representation in an acoustic wavefront.**

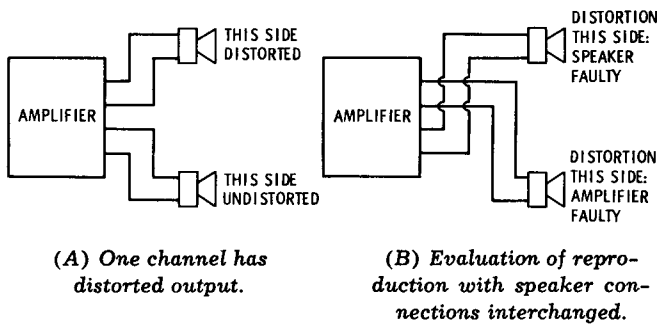
in amplitude rapidly. However, in highly reflective rooms, we may find certain locations that are virtually "silent zones" due to wavefront cancellation. The same principle applies to installation of stereo speakers in any location—that is, the speakers must be connected to operate in phase, as shown in Fig. 6-16.

**Fig. 6-16. Incorrect phasing of stereo speakers will produce cancellations in the sound wavefronts.**



### Analysis of Faulty Reproduction

To avoid confusion between amplifier defects and speaker defects, the preliminary check shown in Fig. 6-17 is helpful. If the connections to the two speakers are interchanged, the distortion will either remain in the same channel, or will appear in the other channel. If the distortion remains in the same channel, the defect is in the speaker. On the other hand, if the distortion appears in the other channel, the defect is in the amplifier. Fig. 6-18 shows the



(A) One channel has distorted output.

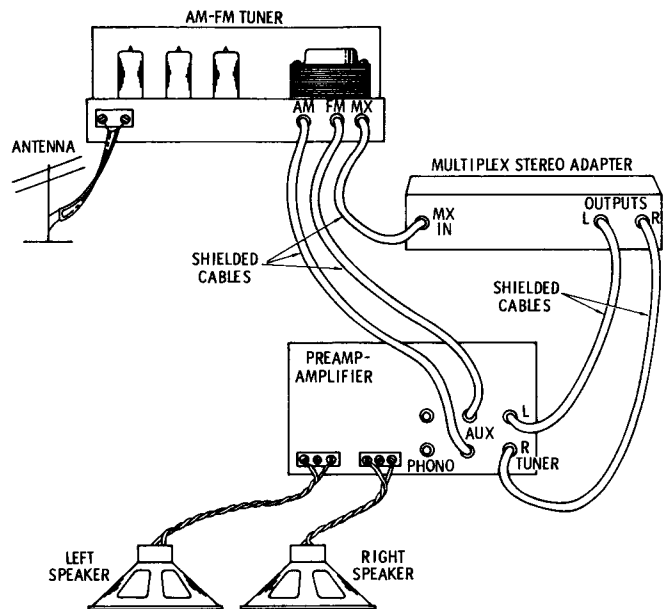
(B) Evaluation of reproduction with speaker connections interchanged.

**Fig. 6-17. Preliminary analysis of distorted reproduction.**

interconnections for a typical component-receiver system. To verify that the speakers are phased correctly, the tuner may be driven by an a-m generator, and the radiated sound may be analyzed as you walk through the line midway between the speakers. If the audio tone does not rise and fall in intensity, the speakers are phased correctly. On the other hand, if the audio tone rises and falls in intensity as you move your listening position, the speakers are out of phase. To correct the trouble, simply reverse the connections of one of the speakers.

Deteriorated stereo reproduction sometimes occurs in do-it-yourself installations because of excessive

cable capacitance from the tuner to the multiplex adapter. This trouble will not be confused with speaker defects if a-m and mono-fm reproduction is compared with stereo-fm reproduction. It is good practice to keep the shielded cable for the adapter input as short as possible. The capacitance of conventional cable ranges from 20 to 100 pF per foot, and the loading effect of a long cable can seriously reduce the amplitude and shift the phase of the higher-order frequencies in the composite stereo signal. For this reason, some manufacturers supply pre-cut lengths of high-quality, low-capacitance cable for this interconnection. If you do not use pre-cut cable, it is good practice to choose a low-capacitance type, and to mount the adapter as near the tuner as may be convenient.



**Fig. 6-18. Interconnections for a typical component-receiver system.**

Faulty reproduction due to speaker defects can be corrected by the hi-fi technician, provided that some major malfunction has not developed. For example, children sometimes push knitting needles, pencils, or other potentially damaging objects into speaker grilles. Cracks, holes, or tears in a cone will impair the fidelity of sound reproduction. Therefore, the entire cone should be inspected carefully (Fig. 6-19). Damaged cones should be replaced; however, at a customer's request, minor holes or tears can be repaired with all-purpose cement. Patching of comparatively large holes is poor practice, and is regarded as a last-resort measure. Any lack of complete bonding is a potential source of rattle. Remember

## Hi-Fi Stereo Servicing Guide

that when an enclosure is reassembled, all parts must be securely tightened; a loose screw or nut is very likely to cause trouble.

If we find that the moving coil is rubbing against a pole piece, replacement is generally advisable. Sometimes, if the design permits and if damage has not occurred, it may be practical to recenter the moving coil. A substantially scuffed or open voice coil should not be repaired unless a replacement is unobtainable. Experienced technicians know that a complaint of speaker rattle may not indicate a speaker defect, but may be tracked down to some

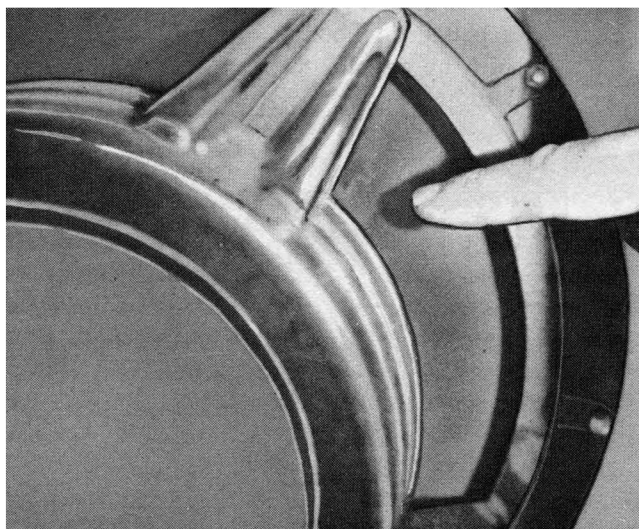


Fig. 6-19. Inspect speaker cone carefully for possible defects.

vibrating surface or object on or near the speaker. For example, knicknack display cabinets are frequent offenders. Such cabinets are acoustically self-resonant, and can "store" considerable acoustic energy. In turn, the contents and shelves tend to vibrate and produce sound interference. Cabinet doors should be securely latched. To avoid possible misunderstanding, a hi-fi installation technician may feel it advisable to discuss some basic acoustical principles with his new customers. This precaution is helpful in forestalling future accusations such as "Why wasn't I told about that?"

Crossover networks are part of a speaker system. A crossover network comprises filter circuits used in multiple-speaker system to separate the high and low audio frequencies into separate groups that are fed to individual speakers designed to reproduce the high and low frequencies, respectively. A typical crossover network is shown in Fig. 6-20. Capacitor defects are the most common sources of trouble. To check out a crossover network, connect the output of an audio oscillator to the input terminals of the amplifier. Connect suitable values of power resistors in place of the woofer and tweeter speakers. In the first test, connect a VOM or TVM across one terminating resistor. In the second test, connect the meter across the other terminating resistor. Note that a VTVM is not entirely suitable in this test, because of its ground capacitance. Rolloff rates for a crossover filter are chosen at widely different values by various designers. However, the important point is that the high-frequency and low-frequency rolloffs should be consistent, so that the overall frequency response will be flat. We will encounter rolloff rates from 6 dB to 18 or 24 dB per octave. In general, transient response is better with slow rolloff, although a greater demand is placed on the tweeter for power output.

In some amplifiers, the crossover network is connected to intermediate stages instead of directly to the speakers. Moreover, a few amplifiers have electronic crossovers instead of LC filter configurations as shown in Fig. 6-20. However, the same basic test procedure is appropriate. As noted previously, some systems employ a woofer, midrange (squawker), and tweeter assembly. In this situation, a test involves measurement of three frequency responses, instead of two. The essential consideration is that the combined responses should show an overall flat frequency characteristic.

### ANALYSIS OF COMMON SYMPTOMS

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

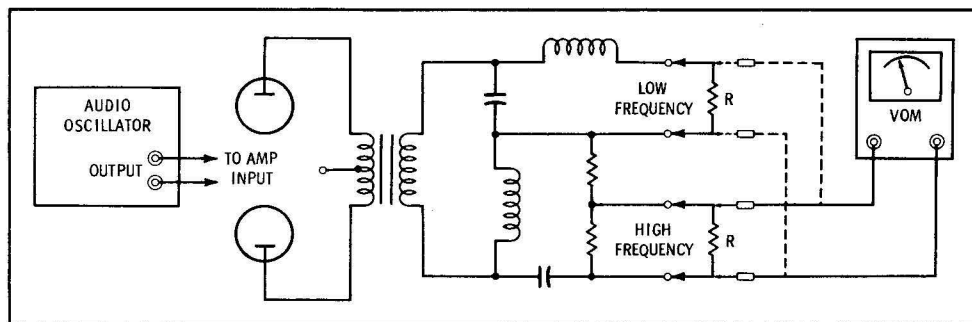


Fig. 6-20. A crossover network.

**1. Echoes**

When echoes are a problem in a speaker installation project, there are several possible remedial courses. In any given situation, the best method will depend both upon the physical properties of the acoustic area, and also upon the viewpoint and desires of the customer. A hi-fi buff will go to much greater lengths in the pursuit of good acoustical conditions than will an interior decorator whose chief concern is eye-appeal.

Factors contributing to objectionable echoes in a speaker installation are:

- a. Speakers may distribute sound energy to better advantage if located at the opposite end of the room.
- b. Draperies or tapestries may be needed on a wall to provide sonic absorption (see Table 6-1).
- c. Speakers may operate more satisfactorily if located in ceiling corners, instead of floor corners.
- d. Echoes may be less troublesome if the speakers are mounted in opposite ends of a cabinet (see Fig. 6-1).

**2. Resonances**

Acoustic resonance cannot develop in the absence of reverberation (sonic standing waves). Any enclosure that is acoustically coupled to the surrounding air will act as a sonic resonator at a particular frequency or frequencies. Fig. 6-21 shows a classic example of sonic standing waves. The reduction of acoustic resonances is accomplished by minimizing the pertinent reflections, by reducing or eliminating the coupling to the sonic resonator, or by a combination of both approaches.

Possible causes of perceptible resonances in a conventional hi-fi installation are:

- a. Display cabinets or hollow decorative structures may need to be closed off from the surrounding air, or lined with a special acoustic material.

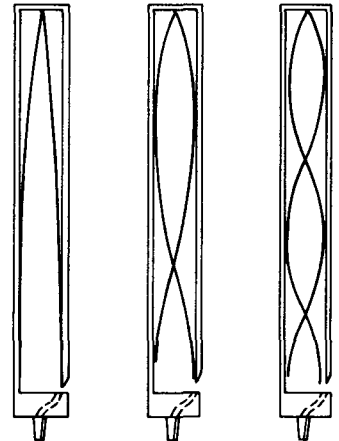
**Table 6-1. Typical Acoustic Absorption Coefficients**

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

- b. Speakers may have been mounted in cabinets of poor acoustic design by a do-it-yourselfer.
- c. Speakers may have been located under a table or in a nook that develops acoustic resonances.
- d. A thin wall behind a speaker may act as a sonic resonator; try relocating the speakers.

**3. Room Distortion**

Room distortion takes on various forms and often changes with the conditions of use. Some forms of distortion, such as echoes or resonances are comparatively simple to analyze; on the other hand, complaints such as "poor stereo effect," or "poor



**Fig. 6-21. Examples of sonic standing waves.**

sound quality in the daytime, and good sound quality at night," are more difficult to analyze. The customer sometimes fails to recognize basic acoustical situations; for example, a customer may be unaware of the fact that a highly reflective recreation room will have improved acoustical characteristics when a large number of people are present. A person has an acoustic absorption coefficient of 0.44, whereas smooth wood has a coefficient of only 0.04.

Possible causes of room distortion are as follows:

- a. Poor stereo effect can be caused by improper speaker placement (see Figs. 6-5 and 6-7).
- b. Changed sound quality at night compared with daytime is generally caused by drawing drapes over large picture windows.
- c. Changed sound quality in cold weather compared with hot weather is usually due to closing windows and doors in the listening area.
- d. Branched portions of an L- or T-shaped room have poor acoustic characteristics (Fig. 6-6), and are not regarded as listening areas.
- e. A large screen or similar object may have been added to the room decor, producing sound shadows (Fig. 6-12).

### 4. Speaker Distortion

In some installations, the room acoustics are better suited to a particular type of speaker. That is, the trouble symptom may not be due to a speaker defect, but to the installation of an unsuitable type of speaker. An experienced hi-fi installation technician will quickly recognize this source of difficulty. Another type of difficulty occurs when a do-it-yourselfer has installed speakers that do not match the amplifier characteristics satisfactorily. Knowledge of components is required in this situation, and the experienced technician will suggest a comparative trial of conventional combinations when this trouble symptom occurs.

Possible causes of speaker distortion are as follows:

- a. Open-back speakers operating in wrong corners in a highly reflective room. Replace with oppositely directed closed-back speakers (see Fig. 6-1). Alternatively, try locating the open-back speakers as indicated in Fig. 6-2.
- b. Basic trouble due to "hole-in-the-middle" (see Fig. 6-7). Relocate speakers as shown in Fig. 6-6.
- c. Basic trouble due to a sound shadow zone, as shown in Fig. 6-12.
- d. Speaker phasing incorrect (see Fig. 6-16).
- e. Serious impedance mismatch between amplifier and speaker; use speaker with appropriate input impedance.
- f. Speaker defect such as shorted turns or high-resistance connection.

### 5. Rattle

As noted previously, a nontechnical person may confuse an environmental rattling with speaker rattle. That is, a rattle developed by a nearby cabinet, or by a decorative lamp placed on the speaker enclosure might be confused with a speaker defect. Therefore, the hi-fi technician should keep the possibility of customer confusion in mind when this trouble symptom is reported. Experienced technicians can recall case histories in which a deceptive

rattle was actually caused by an amplifier defect, instead of an environmental or a speaker defect.

Possible causes of rattle in a hi-fi installation are as follows:

- a. Object placed on speaker enclosure.
- b. Loose door or shelf in nearby cabinet.
- c. Loose connection to a speaker terminal.
- d. Assembly screws not tight in speaker enclosure or speaker assembly.
- e. Speaker damage, such as a pencil pushed through the grille.
- f. Amplifier defect that simulates speaker rattle (see Fig. 6-17).

### 6. Weak Output

Weak output from a speaker can be caused by various defects, most of which fall into the category of electrical troubles. Weak output may or may not be accompanied by distortion. For example, if a do-it-yourselfer has been tampering with a speaker, and the voice coil is obstructed by iron particles or other foreign matter, weak output will be accompanied by rasping noises and/or loud popping and crackling. On the other hand, an insulation defect that causes a low-resistance shunt path across the voice coil produces weak output without substantial distortion. If a speaker is located at a considerable distance from the console, it is good practice to use a correspondingly larger size of conductor, in order to keep the  $I^2R$  loss within reasonable limits.

Possible causes of weak output in a speaker installation are:

- a. Corroded or high-resistance connection in the speaker circuit.
- b. Deteriorated insulation causing a low-resistance shunt path.
- c. Plug not fully inserted into connector.
- d. Defective stranded lead in speaker circuit that introduces substantial resistance.
- e. Internal defect in speaker (less likely than external troubles).

## Chapter 7

# System Evaluation and Trouble Localization

The chief consideration in any hi-fi installation is an evaluation of how the components work together as a team. We are concerned with the overall frequency response, overall harmonic (or intermodulation) distortion, and the overall transient response. Distortion- and transient-response tests are usually made at maximum rated power output. In case the system performance is unsatisfactory in some respect, the technician must make more detailed tests to locate the component that is causing the difficulty. If a system has been in use for an extended period of time, it is possible that more than one component may have deteriorated past acceptable limits. A hi-fi system commonly includes an fm/am tuner, pre-amplifier, multiplex adapter, power amplifier, speakers, record player, and a tape recorder or tape deck. In this book, the coverage does not include record and tape equipment, since these are specialized units. Interested readers are referred to record-changer and tape-recorder servicing guides.

Common trouble symptoms associated with system operation are:

1. Stereo-reproduction failure or malfunctioning.
2. Poor bass or treble response.
3. Distorted reproduction.
4. High noise level.
5. Interference on fm or a-m function.
6. Weak output.
7. Microphonic ringing or howling.

### GENERAL DISCUSSION

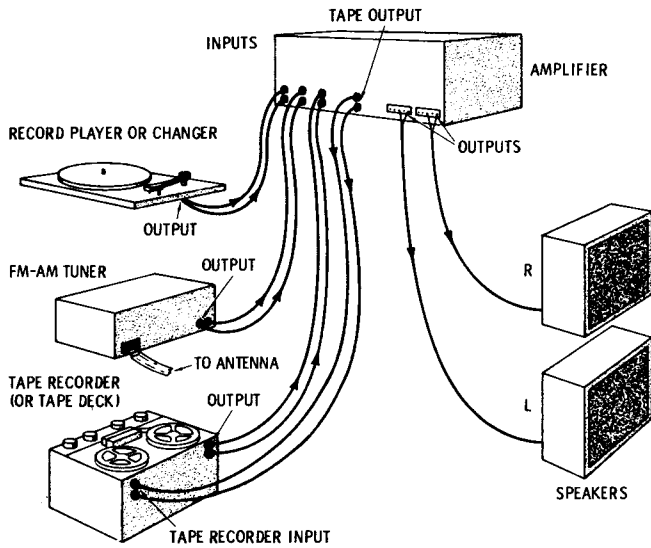
A typical hi-fi stereo system is illustrated in Fig. 7-1. Any system test requires the application of a

suitable test signal at the input, and connection of a suitable indicating instrument at the output. For example, the a-m tuner function would be energized by an a-m generator, and the amplifier output would be indicated by a VOM, VTVM, TVM, or scope. Since an a-m generator usually provides only one tone signal, a check of the system frequency response requires that the generator be externally modulated by an audio oscillator, as shown in Fig. 7-2. To measure the system harmonic distortion, a harmonic-distortion meter is used in place of the VTVM, as shown in Fig. 7-3. If it is desired to measure the percentage of harmonic distortion at a frequency other than the modulating frequency provided by the generator, an audio oscillator may be used to externally modulate the generator.

To measure the system intermodulation distortion, an IM distortion meter is employed, as shown in Fig. 7-4. In most cases, the IM signal source is built into the IM distortion meter. If a tone-burst system test is to be made, the test setup shown in Fig. 7-5 is employed. Again, a system square-wave test is made with the test setup shown in Fig. 7-6. The foregoing tests require an accurate a-m generator with low distortion. In addition, an audio oscillator used in the tests must have low distortion, and the external-modulation function must be linear. Otherwise, distortions introduced by the test instruments would be falsely charged to the hi-fi system. It is good practice to use lab-type instruments in hi-fi test work. If service-type generating equipment carries hi-fi ratings, it will be satisfactory.

Next, if system tests are to be made through the fm tuner, the same instrument arrangements are used, except that an fm signal generator is used in-

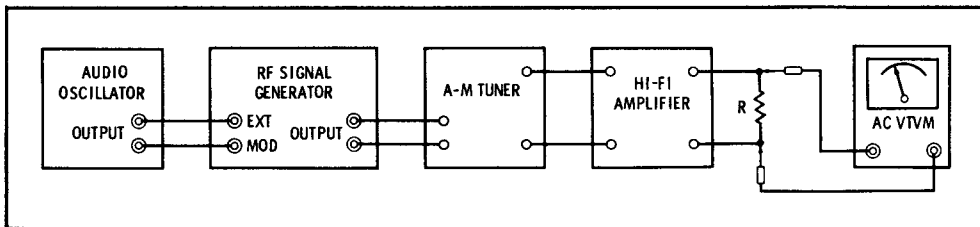
# Hi-Fi Stereo Servicing Guide



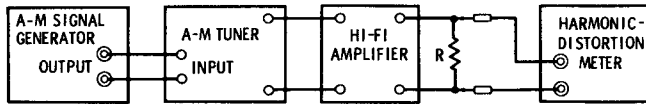
**Fig. 7-1. Typical stereo system arrangement.**

stead of an a-m signal generator. Since most hi-fi systems include a multiplex adapter, the generator needs to be of the stereo-fm signal simulator type. To make an overall frequency-response test, the signal simulator may be externally modulated by an audio oscillator, or by an audio sweep generator. A test setup with an audio sweep generator is shown in Fig. 7-7. Since audio sweep generators (Fig. 7-8) are lab-type instruments and are comparatively expensive, most hi-fi shops prefer to use conventional audio oscillators. A stereo-fm signal simulator used with either type of audio generator must be quite linear and rated for hi-fi application.

In the example of Fig. 7-7, the system frequency response will *seem* to be poor, even if we vary the setting of the treble control from its normal mid-range position. The reason for this is due to the effect of de-emphasis on the audio test signal, as shown in

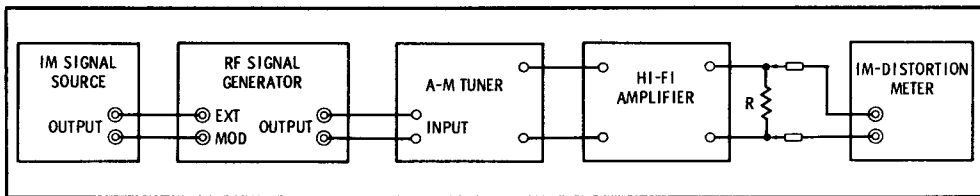


**Fig. 7-2. Test setup for checking system frequency response.**

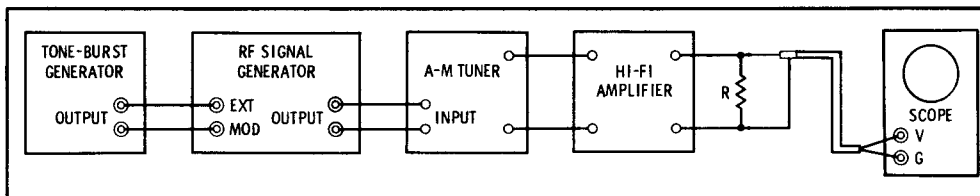


**Fig. 7-3. Test setup for checking system harmonic distortion.**

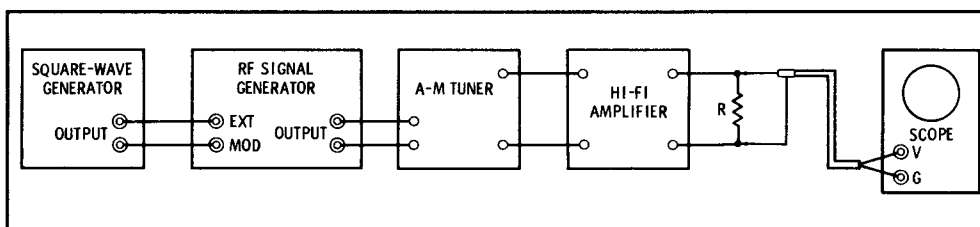
Fig. 7-9. In other words, an fm broadcast signal is pre-emphasized, whereas the output from an audio generator is flat. De-emphasis is provided in the output section of the multiplex adapter. Therefore, we



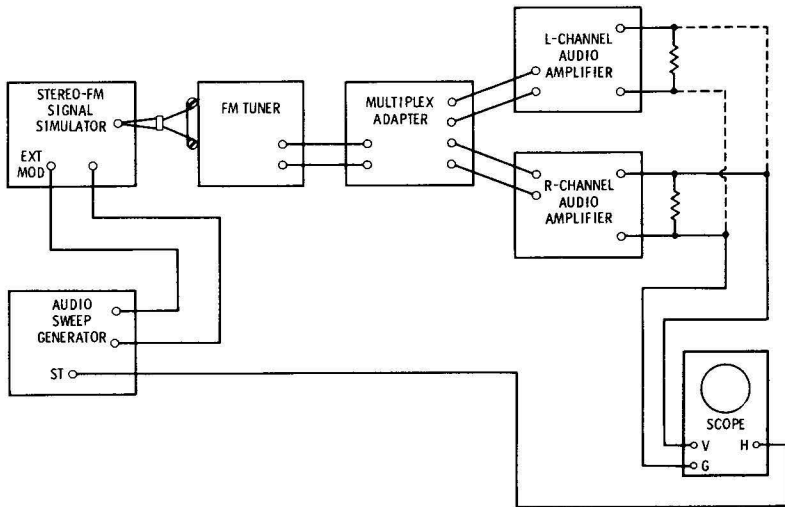
**Fig. 7-4. Test setup for checking system IM distortion.**



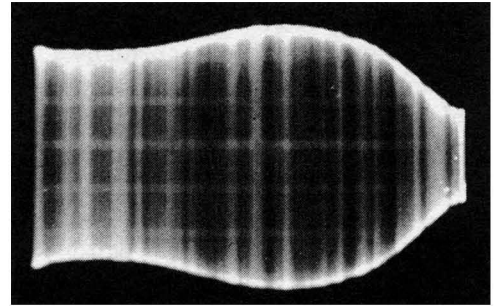
**Fig. 7-5. Test setup for making system tone-burst test.**



**Fig. 7-6. Test setup for making system square-wave test.**



(A) Test setup.



(B) Scope display of typical response pattern.

Fig. 7-7. Sweep-frequency test of multiplex system.

normally find a flat frequency response at the input of the adapter, whereas we will find a frequency response that is 20 dB down at 20 kHz when testing at the output of the adapter. In other words, the normal frequency response at the outputs of the L and R amplifiers will start to drop off at 1 kHz, and will be 20 dB down at 20 kHz. The response at 20 kHz is 10 percent of the peak-to-peak response at 1 kHz.

Let us suppose that the frequency response is abnormal at the output of the R amplifier in Fig. 7-7.

The next step is to check the frequency response at the output of the L amplifier. In case a normal frequency response is found at the output of the L amplifier, we conclude that there is a defect in the R amplifier; amplifier troubleshooting is explained in Chapter 5. If we find the same abnormal response at the output of the L amplifier, we know that the trouble is occurring in the fm tuner or in the multiplex adapter. To localize the trouble, we transfer the vertical-input leads of the scope to the output of the fm tuner. If we find a *flat* frequency response in this test, we know that the trouble source is in the adapter; on the other hand, an abnormal frequency response indicates that the trouble is in the fm tuner. Now, with the faulty unit known, further troubleshooting can be done; tuner troubleshooting is explained in Chapter 2, and multiplex-adapter troubleshooting is explained in Chapter 3.

When making a system harmonic-distortion test through an fm tuner, or through an fm tuner and multiplex adapter, it is advisable to employ a 400-Hz test frequency. Since the second and third harmonics

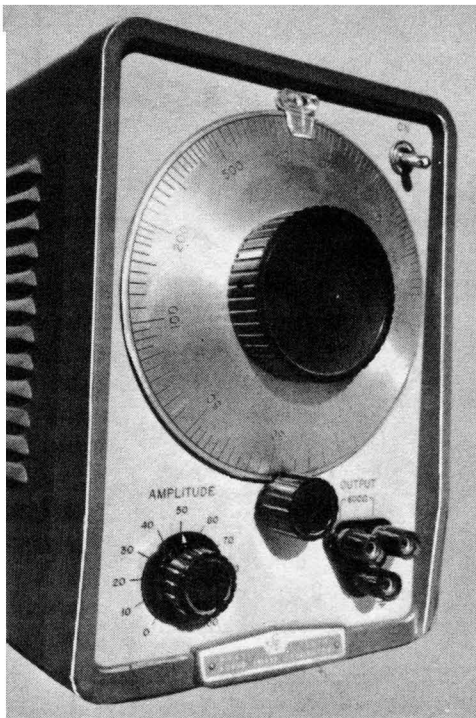


Fig. 7-8. A typical audio sweep generator.

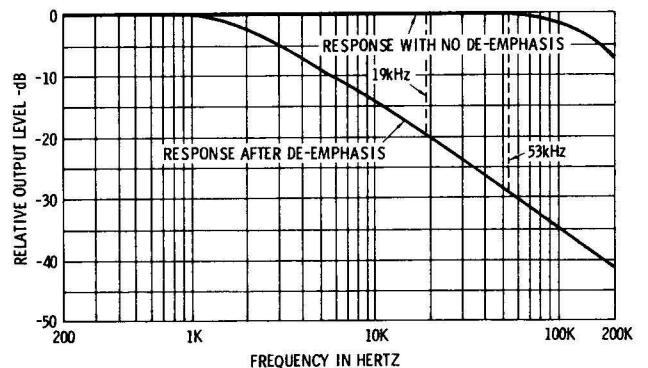


Fig. 7-9. Effect of de-emphasis on multiplex frequencies.



## Hi-Fi Stereo Servicing Guide

of this frequency are 800 and 1200 Hz, respectively, the de-emphasis network has little effect on the reading of the harmonic-distortion meter. On the other hand, if a higher test frequency were used, the distortion reading would be abnormally low. It is difficult to apply correction factors, because the proportion of second to third harmonics is unknown in a routine measurement. Therefore, the practical procedure is to confine a system test to a frequency of 400 Hz, or lower if desired. In case the system percentage of distortion is unduly high at the output of the L channel, check next at the output of the R channel. An acceptably low distortion figure at the output of the R channel indicates that the trouble will be found in the L amplifier. On the other hand, if the distortion readings are too high at the outputs of both amplifiers, we conclude that the trouble is occurring in the fm tuner, or in the multiplex adapter.

Distortion measurements should be made at maximum rated power output. Apprentice technicians should not confuse a music-power rating with a steady-power output rating. A music-power rating is always higher, and specifies the ability of an output amplifier to process a pulse-type waveform. If the output amplifier is driven by a sine-wave signal and is operated at the music-power rating, a transistor-type amplifier may be damaged. That is, a transistor amplifier should not be tested at an output greater than the steady-power rating when driving the system from an audio oscillator. A tube-type amplifier will not be damaged in a sine-wave test if operated at its music-power output rating, but the distortion will be excessive. That is, the sine-wave distortion increases rapidly as the steady-power output rating is exceeded.

Intermodulation-distortion tests of a hi-fi system through the fm tuner are not practical unless the de-emphasis network is bypassed. That is, typical IM analyzers employ test frequencies of 60 Hz and 6 kHz. Although the 60-Hz signal is passed without attenuation, the 6-kHz signal is normally attenuated about 10 dB. This action of the de-emphasis network makes the reading of an IM analyzer virtually meaningless. Depending upon the design of the multiplex section, it may or may not be practical to "jump" the de-emphasis circuit(s) for the purpose of making a system IM test. In any case, a percentage IM measurement can be made at the output of the fm tuner (input of the multiplex section). Next, the IM test signal can be applied at the input of the pre-amplifier to check the sections past the multiplex adapter. In a situation of this type, our check of distortion in the multiplex section is limited to a

harmonic-distortion measurement. As noted above, the test frequency should be 400 Hz, or a lower value, if desired.

System distortion tests are informative because we cannot necessarily predict the amount of system distortion from a knowledge of the component-distortion values alone. In other words, there are some situations in which the distortion produced by one component is partially compensated by the distortion produced by the following component. For example, if a preamplifier has 1 percent distortion, and an output amplifier has 1 percent distortion, we cannot assume that the combination will have a total distortion of 2 percent. Each component has a particular transfer (input/output) characteristic, and it is the system transfer characteristic that determines the amount of system distortion. If one amplifier has a concave transfer characteristic, and the following amplifier has a convex transfer characteristic, the system characteristic may be more nearly linear than either of the individual characteristics. Fig. 7-10 illustrates the principle that is involved.

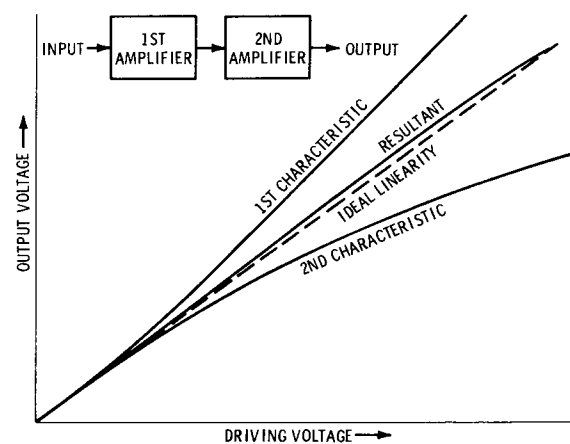


Fig. 7-10. How oppositely curved transfer characteristics tend to linearize a system.

Apprentice technicians sometimes assume that a test record or a test tape is equivalent to an audio oscillator for checking amplifier distortion. This signal source has valid uses, but gives higher distortion readings than if an amplifier system is driven by an audio oscillator. The reason for this disparity is seen in Chart 7-1; that is, conventional recording systems are associated with 2 to 4 percent distortion, and the same order of distortion is associated with playback. Since a good audio oscillator such as that used in a hi-fi shop has less than 0.5 percent distortion, it is the most accurate signal source for use in distortion measurements.

## System Evaluation and Trouble Localization

**Chart 7-1. Typical Harmonic-Distortion Values**

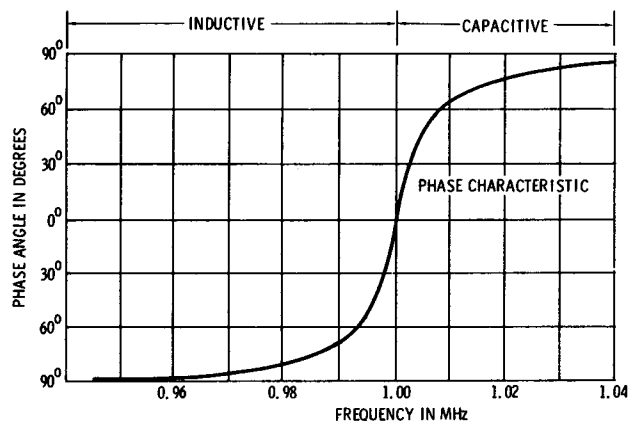
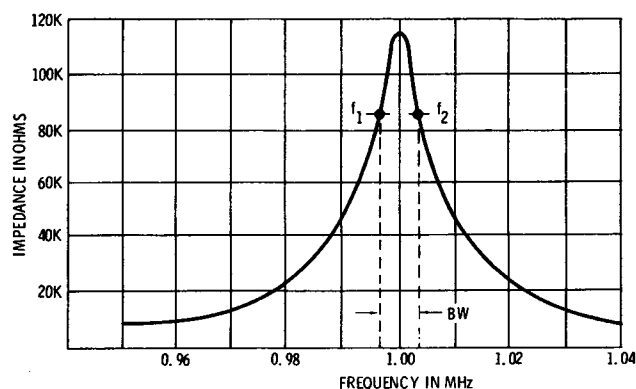
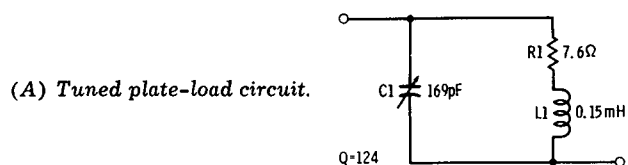
<b>Frequency-Modulated Transmitters</b>	
100% modulation	—5 to 7%
80% modulation	—2 to 3%
<b>Disc-Recording Systems</b>	
record/playback	—2.5 to 4%
<b>Audio Amplifiers</b>	
High quality	—0.10 to 1%
Medium quality	—2 to 5%
<b>Magnetic Recorders and Reproducers</b>	
record/playback	—2 to 4%

A system square-wave test such as that shown in Fig. 7-6 is straightforward so far as the a-m function is concerned. On the other hand, if the square-wave signal is used to modulate an fm generator for driving an fm tuner, we are confronted with the same basic de-emphasis problem that was noted previously in regard to IM tests. That is, a square-wave signal differs basically from a sine wave in that the square waveform is built up from a number of different frequencies. Therefore, system square-wave tests through the fm tuner are not practical unless the de-emphasis circuit(s) can be "jumped" during the test. In any case, a square-wave check can be made at the output of the fm tuner (input of the multiplex section). Next, the square-wave test signal can be applied at the input of the preamplifier to check the sections past the multiplex adapter. The same general procedure should be observed in making tone-burst tests through an fm tuner.

Square-wave tests differ from frequency-response tests in that a square-wave test takes the transient response of the system into account. The transient response is fundamentally dependent upon the phase characteristics of the various components. If each component has a linear phase characteristic, the output square-wave will be free from transient distortion. However, most components have phase characteristics that are more or less nonlinear. Fig. 7-11 shows the phase characteristic for a simple tuned circuit, as in an a-m tuner. Phase-shift charts for RC-coupled amplifiers are shown in Fig. 7-12. Experienced technicians know that phase characteristics combine in an entirely different manner from that of frequency characteristics. Let us consider the basic principles that are involved.

With respect to a system frequency response, we recall that the output of an amplifier is multiplied by the gain of the following amplifier at any given frequency. For example, if we apply 0.1 volt at 1 kHz to an amplifier that has a gain of 10 times at this frequency, the output will be 1 volt. Next, if this output signal is fed to a following amplifier that has

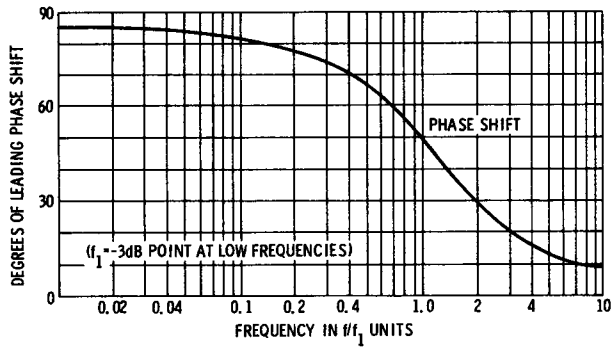
a gain of 6 times at 1 kHz, the output will be 6 volts. In other words, the system gain is  $6 \times 10$  or 60 times at 1 kHz. If we apply 0.1 volt at the input of the system, we will obtain  $60 \times 0.1$  volt, or 6 volts output at 1 kHz. However, phase characteristics do not combine in a simple or readily predictable manner.



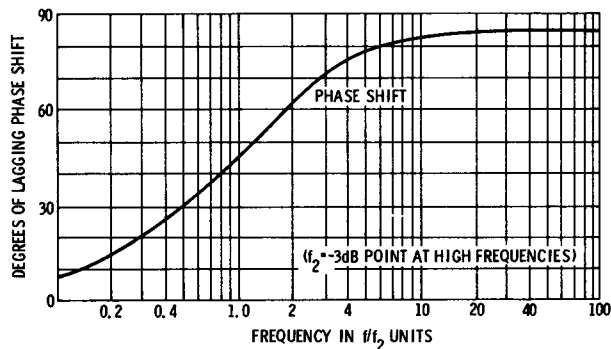
**Fig. 7-11. Frequency and phase characteristics of a tuned circuit, as in an a-m tuner.**

In general, it is much more difficult to compensate a nonlinear phase characteristic than to compensate a nonlinear amplitude characteristic. Rapid curvature in the phase characteristic of any component causes more or less overshoot, often accompanied by ringing. The basic square-wave distortions are shown in Fig. 7-13.

A square-wave test of a hi-fi system must take the tuner bandwidth into consideration, as this is normally the limiting factor. In the case of an a-m tuner,



(A) For low frequencies.



(B) For high frequencies.

Fig. 7-12. Universal phase-shift charts for RC-coupled amplifier at low and high frequencies.

the normal bandwidth is in the range from 5 kHz to 10 kHz. Since it is a rule of thumb that about 15 harmonics (odd harmonics) should be reproduced in a square-wave test, the repetition rate of the square wave will be chosen in the range from 20 to 200 Hz. In the case of an fm tuner, a square-wave repetition rate up to 400 Hz is suitable. The tone controls should be set for flat frequency response when making a square-wave test. If the system is unsatisfactory, the scope can be used as a signal tracer to localize the defective component. That is, the vertical-input leads can be transferred to the output of the preamplifier, and then to the output of the tuner.

### ANALYSIS OF COMMON SYMPTOMS

An analysis of the common trouble symptoms listed earlier in this chapter is presented in this section.

#### 1. Stereo Reproduction Failure or Malfunctioning

Poor stereo reproduction or stereo failure is not necessarily due to a component defect. For example, a broken antenna lead can reduce the strength of a stereo-broadcast signal to a sufficiently low level that the subcarrier oscillator in the multiplex section is not tightly locked. In case the system stereo reproduction is unsatisfactory when checked with a generator, the antenna is eliminated from suspicion. It is often helpful to carefully check the system operation on each function; this evaluation can provide preliminary localization of the trouble.

Possible causes of stereo reproduction failure or malfunctioning are:

- Weak output from fm tuner; troubleshooting procedures are discussed in Chapter 2.
- Defect in multiplex section; troubleshooting procedures are explained in Chapter 3.
- Subnormal output or no output from one of the audio channels; audio troubleshooting is covered in Chapter 5.
- One speaker defective; speaker operation is discussed in Chapter 6.
- Defective lead or plug to one speaker (see Fig. 7-1); less likely than other faults, but possible.

#### 2. Poor Bass or Treble Response

When a hi-fi system develops poor bass or treble response, the first step is to check the response on each function as an aid to preliminary localization. For example, if we find poor bass response on the phono function, but have normal response on the

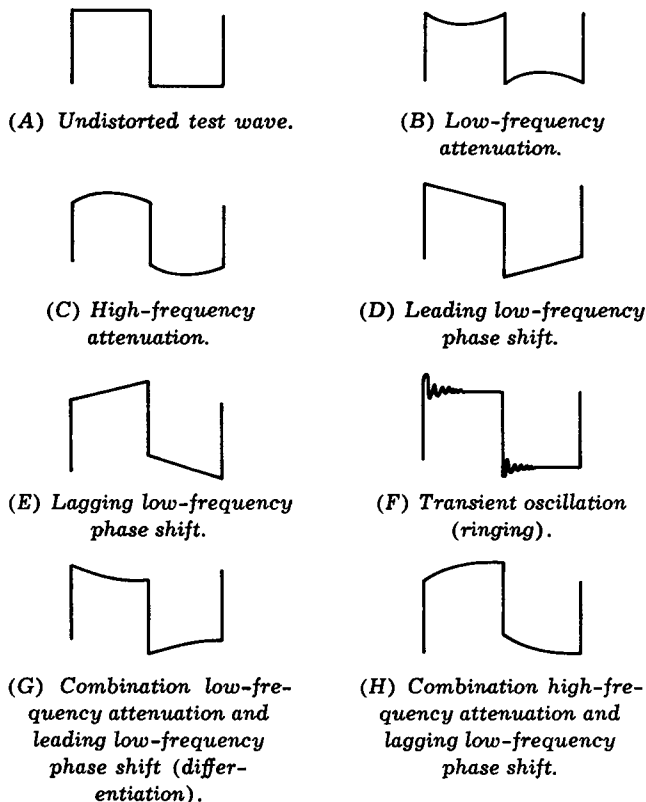


Fig. 7-13. Distortion of square wave by circuit under test.

a-m, fm, and tape functions, we will suspect that there is a defect in the equalization network of the phono-input section. On the other hand, if we find poor treble response on all functions, it is logical to conclude that the trouble will be found in the audio section. The speakers are included in the general audio section; thus, if the brilliance-control network in the speaker circuit has a defective component, poor bass or treble response can result. Although subnormal bass or treble response is the most common type of "tone" trouble symptom, the reverse situation occurs occasionally. That is, a defect in a negative-feedback loop that produces a midband "suck-out" is associated with abnormal bass and treble response—in other words, the mid-range response is poor when this malfunction occurs.

Possible causes of poor bass or treble response are as follows:

- a. Tuner misalignment (see Chapters 1 and 2).
- b. Component defect in an equalization network.
- c. Preamplifier malfunction.
- d. Output amplifier malfunction.
- e. Defective capacitor in crossover network (see Chapter 6).
- f. Defective component in brilliance-control network.

### 3. Distorted Reproduction

If a hi-fi system develops distorted reproduction, we start the trouble analysis by checking the system response on each function. In a stereo installation, distortion may develop in only one channel, or in both channels; this observation is also helpful in preliminary localization. Occasionally, we can find a localization clue in the type of distortion that is occurring. For example, if there is objectionable hum modulation or interference, we turn our attention to the power supply at the outset. Acoustic-feedback (microphonic) distortion can often be localized to a system component by rapping your knuckles against each component in turn. If a distortion symptom disappears when the volume control is turned down considerably, we know that the trouble will be found in a section following the volume control.

Possible causes of distorted reproduction in a hi-fi system are:

- a. Defective component in the output amplifier (see Chapter 5).
- b. Speaker malfunction (see Chapter 6).
- c. Component breakdown in preamplifier.
- d. Leaky capacitor, defective transistor, or misalignment in tuner section.

- e. Mismatch of output amplifier to a replacement speaker.

### 4. High Noise Level

A high noise level directs attention to the input sections of a hi-fi system because the noise voltage then becomes amplified by the following sections. If noise occurs only on the phono function, for example, most of the system circuitry is immediately cleared from suspicion. On the other hand, we often encounter situations in which the noise level is the same on all functions. In this case, the preamplifier becomes the most logical suspect. Although various components can develop noise voltages, case histories show that transistors are most likely to become defective in this manner, and that poor connections are a runner-up. Resistors of the composition type may become noisy, particularly in low-level stages.

Possible causes of a high noise level in a hi-fi system are as follows:

- a. Extraneous noise voltages feeding in from the power line; check the line filter in the power supply.
- b. Plug not fully inserted into connector.
- c. Cold-solder or high-resistance connection.
- d. Noisy component, such as a transistor, resistor, or capacitor.
- e. Broken antenna conductor (noise occurs only on a-m or fm functions).

### 5. Interference on FM or A-M Function

When investigating a complaint of interference on the fm or a-m function of a hi-fi system, it is essential to determine the reception conditions that prevail in the particular area. For example, if the installation is in the vicinity of a high-powered broadcast station, other receivers in the same neighborhood will also be subject to interference. In a typical case history, all receivers in the vicinity of a high-powered radar installation were subject to brute-force interference. Environmental interference can be reduced by means of antenna and line traps, but complete elimination may require operation in a screened room. In difficult situations of this kind, the customer should be encouraged to discuss interference problems with his neighbors. This procedure serves to eliminate customer suspicion that the technician is "trying to sell a bill of goods."

In the event that a complaint of interference on the fm or a-m function of a hi-fi system is legitimate, possible causes of the malfunction are as follows:

- a. Customer may be using the power line as an

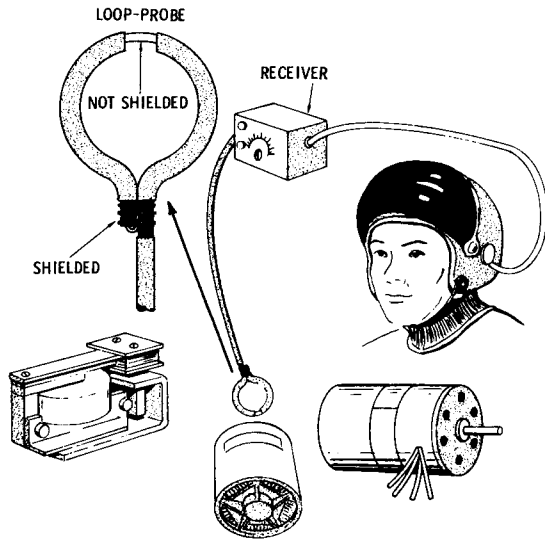


Fig. 7-14. Location of radio noise.

antenna via a capacitive-coupling device. This practice is conducive to noisy reception and various forms of interference.

- b. An excessively large or high antenna may be in use; most tuners are designed to operate normally with comparatively short and low antennas.
- c. Mobile installations are subject to ignition interference, etc., and should be installed by technicians with experience in this area.
- d. Marine installations have the same basic interference problems as mobile installations, plus that of interference produced by specialized electrical and electronic equipment; reference should be made to handbooks or texts on this subject.

- e. Interference due to tuner defects may be tracked down to component failure, or to misalignment (see Chapters 1 and 2).
- f. Telephone-line crosstalk has been found in installations employing unshielded lines to the speakers with the lines installed parallel to telephone service conductors.

Whether a difficult interference problem is encountered in a domestic, marine, or an airborne installation, the basic attack is the same. In the event that sophisticated professional equipment is available, we can proceed by tuning in radio noise on a field-intensity meter. Sometimes, the signal can be identified by means of earphones, as shown in Fig. 7-14. An essential part of professional equipment is an electrostatically shielded loop probe; it is used as an antenna to locate sources of noise in machinery. Moving the probe in the direction of the source (or some conductor radiating the noise energy) causes the signal strength to increase. A block diagram for a professional interference-locating unit is shown in Fig. 7-15.

6. Weak Output

Weak output in a hi-fi system may originate at any component in the chain. Some defects impair only one function, whereas other defects can impair more than one, or all functions. Therefore, the trouble analysis starts with a check of operation on each function. If the fm tuner is defective, only fm reception may be affected; however, some defects cause impairment of both fm and a-m reception. In normal operation, a typical fm tuner provides an output of 1 volt rms when driven by a 10- $\mu$ V input at 75 kHz

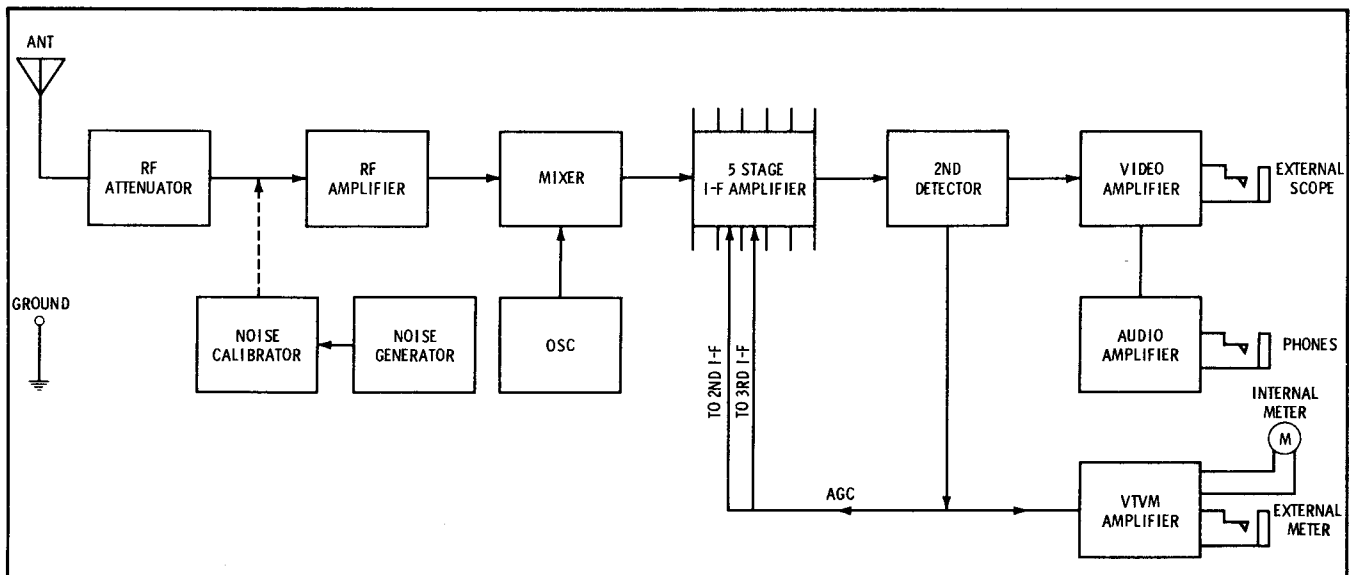


Fig. 7-15. Block diagram for a professional interference-locating unit.

deviation. A typical a-m tuner provides approximately the same output when driven by a  $10\text{-}\mu\text{V}$  input with 30 percent amplitude modulation. This check can be made only if a lab-type generator is available. However, in case of doubt, it may be possible to make a comparison check with another tuner, using an ordinary generator.

Possible causes of weak output in a hi-fi system are:

- a. Broken antenna conductor, or short-circuited lead-in.
- b. Defective function switch.
- c. Plug not fully inserted into connector.
- d. Subnormal power-supply voltage.
- e. Tuner defect or misalignment (see Chapters 1 and 2).
- f. Amplifier malfunction (see Chapter 5).
- g. Defective speaker (see Chapter 6).

### 7. Microphonic Ringing or Howling

Microphonic ringing or howling (acoustic feedback) is much more common in tube-type hi-fi systems than in transistorized designs. In most cases, a defective tube is the offender. However, other kinds of microphonic components are occasionally found both in tube-type and transistor equipment. For example, a phono cartridge sometimes causes this trouble symptom; in such a case, the ringing or

howling occurs on the phono function only. The two main sources of microphonics in an a-m tuner are the local-oscillator components and poor installation practices. Any loosely secured component in the local-oscillator section is a potential source of microphonics. For example, a local-oscillator coil replacement suspended by its leads or a long thin wire hung loosely between the tuning capacitor and its associated components is a typical troublemaker. Acoustic feedback is aggravated by poor installation practices, such as mounting the tuner in front or on top of a speaker. It is comparatively easy to localize a microphonic component by means of tapping tests.

Possible causes of microphonic ringing or howling in a hi-fi system are:

- a. Cold-soldered or unsoldered contact that develops varying resistance when subjected to slight mechanical vibration.
- b. Operation of system at excessive output level in a confined space.
- c. Deteriorated resistor that responds like a carbon microphone.
- d. Defective tube.
- e. Malfunctioning phono cartridge.
- f. Mechanically insecure component in local-oscillator section.
- g. Poor system-installation practices.

## Chapter 8

# Test-Equipment Performance Verification

Test-equipment performance verification is an essential shop responsibility, because instruments must be accurate and in good working condition if they are to be a help instead of a hindrance at the bench. The VOM and VTVM (or TVM) are unquestionably the most basic test instruments. However, since these are well-known to practically all technicians, we will begin our coverage with the audio oscillator, which is the basic audio test instrument. The oscilloscope is also among the more basic test instruments; like the VOM, it is comparatively well known to the majority of technicians. Therefore, we will continue our coverage with the harmonic distortion meter and the intermodulation analyzer, which are among the more advanced specialized audio instruments. The chapter concludes with a practical discussion of tone-burst generator characteristics and performance-verification procedures.

### GENERAL DISCUSSION

Service-type audio oscillators are often rated for less than 1 percent harmonic distortion. A few instruments are rated for less than 0.5 percent distortion; however, the lower-priced instruments are rated in the order of 5 percent distortion. In some cases, a distortion rating applies over less than the complete frequency range. For example, an audio oscillator that has a frequency range from 20 Hz to 1 MHz might have a distortion rating of less than 0.25 percent from 20 Hz to 20 kHz, with no rating from 20 kHz to 1 MHz. To measure the percentage of distortion in an audio-oscillator output signal, feed the signal directly into a harmonic-distortion

meter, as shown in Fig. 8-1. A scope is optional, but is useful to determine whether the distortion is chiefly second-harmonic or third-harmonic. A scope also helps in analyzing possible hum in the signal. The flatness of output over the complete range of the audio oscillator can also be checked with a scope.

Harmonic-distortion measurements should be made at several intervals; for example, measurements may be noted at 20, 200, 2000, and 20,000 hertz. The values indicated by the HD meter might not be accurate if there is a defect in the instrument. Therefore, it is good practice to cross-check with another HD meter, if possible. In case both HD meters are in agreement, it is generally safe to conclude that both meters are accurate. If the scope is deflected at a horizontal rate that displays one cycle when the HD meter is operating on its Set Level function (display of the fundamental output from the audio oscillator), harmonics are identified as follows when the HD meter is set to read percentage of distortion. If the distortion pattern displays two cycles on the scope screen, we know that second-harmonic distortion is present. Or, if the distortion pattern displays three cycles on the scope screen, we know that third-harmonic distortion is present.

In many cases, the distortion products from an audio oscillator will be a combination of second- and third-harmonic components. In such a case, the peaks in the waveform will have unequal heights; the number of cycles in the pattern will depend on whether the second harmonic or the third harmonic is predominant. Higher-order harmonics may be perceptible also but they are usually quite small in relative amplitude. The scope pattern will change as

## Hi-Fi Stereo Servicing Guide

the HD meter is tuned through the null point, as illustrated in Fig. 8-2. In this example, the second photo shows the true distortion-product pattern, with

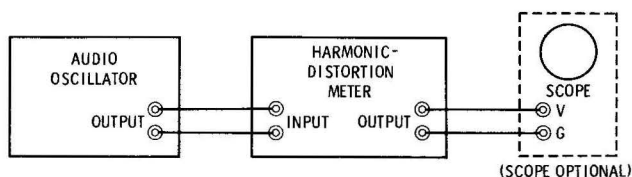


Fig. 8-1. Harmonic-distortion check of audio oscillator.

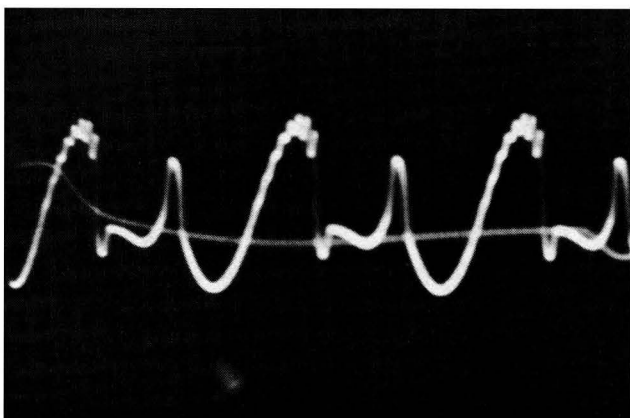
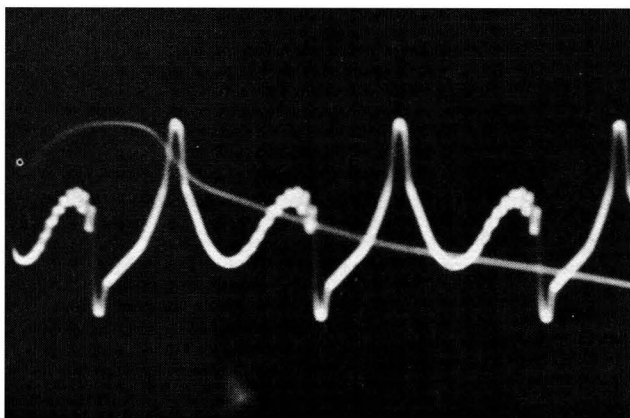
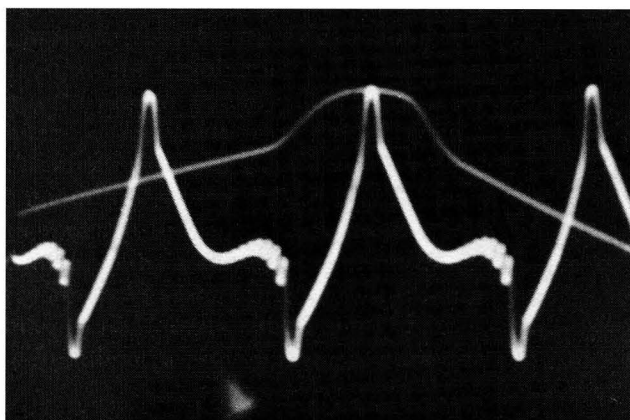


Fig. 8-2. Typical change in shape of a harmonic-distortion waveform as the harmonic-distortion meter is tuned through the null point.

the HD meter reading at the null point. Hum will show up as a blurring of the distortion pattern, except when the HD meter is operated at 60 Hz or 120 Hz. If tuned slightly to one side of 60 or 120 Hz, hum voltage causes a "writhing" of the distortion pattern; the needle will also be seen to "wobble" on the HD meter.

If the measured percentage of distortion is out of rating for the audio oscillator, the trouble may be due to excessive power-supply ripple, as noted above. The filter capacitors are usually the offenders in this case. On the other hand, if the B+ ripple is negligible, the tubes should be checked or replaced. In the majority of cases, these simple procedures will bring the percentage of distortion within its rated value. However, if the difficulty persists, the next step is to check the circuit capacitors for leakage or opens. Note that in very-high-resistance bridge circuits, such as employed in some audio oscillators, even a slight amount of capacitor leakage will seriously upset the bridge operation. For example, resistors with values up to 18 megohms may be used on the first range. Fig. 8-3 shows a simplified circuit diagram for a Wien-bridge oscillator.

Resistors are less likely to cause trouble in an audio oscillator than are capacitors. However, resistors will occasionally become defective. In tube-type instruments, resistors may become damaged by excessive current due to tubes that develop inter-electrode shorts. This source of trouble is much less common in transistorized audio oscillators, even if a transistor develops a "dead" collector-to-base short. That is, the supply voltages are much lower in a solid-state instrument, which generally eliminates the possibility of burned resistors. Composition resistors tend to drift in value with age, and the general tendency is for the resistance value to creep upward. However, there are exceptions, and most technicians can recall situations in which a resistor was found to have decreased in value.

Next, let us consider frequency-calibration requirements. The tuning dial of an audio oscillator is usually rated for calibration accuracy, and all instruments have trimmer capacitors or some equivalent provision for frequency adjustment. Fig. 8-4 shows a practical shop-test setup for low-frequency calibration of an audio oscillator. The 60-Hz power line is used as a frequency reference. Although a single frequency check might not have extremely high accuracy, the long-time accuracy of the 60-Hz power frequency is very good. Therefore, we can make several frequency checks at separated intervals, and take the average to maximize the accuracy of measurement. Zero beat is displayed as a Lissajous



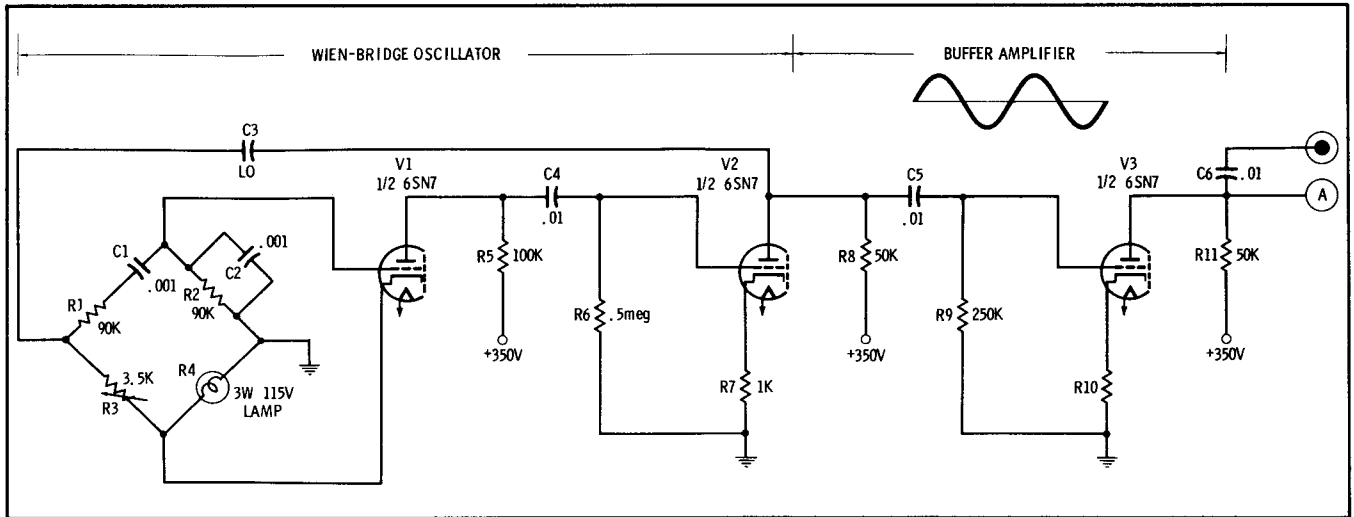


Fig. 8-3. Basic Wien-bridge oscillator configuration.

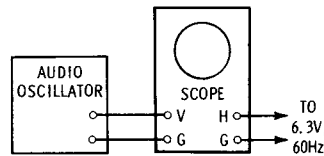
pattern; the 60-Hz Lissajous figure is illustrated in Fig. 8-5. Its aspect depends upon the phase relation of the two input voltages; if the audio oscillator is tuned slightly off zero beat, the pattern slowly progresses through its various aspects.

to the vertical input is 60 Hz, the frequency applied to the horizontal input is given by:

$$\frac{f_h}{f_v} = \frac{2}{3} \times 60 = 40 \text{ Hz}$$

The National Bureau of Standards maintains WWV transmissions on carrier frequencies from 2.5 to 25 MHz, with audio modulation frequencies as tabulated in Fig. 8-9. To check the calibration of an audio oscillator against the 440- or 600-Hz WWV modulation frequencies, simply tune in the signal on a radio receiver and feed the speaker voltage to the input terminals of a scope. Then, spot-checks of numerous frequencies can be made by means of Lissajous figures as has been explained. This procedure becomes more critical as we proceed to check higher audio frequencies, since a very slight movement of the tuning dial then changes the audio-oscillator frequency appreciably. A vernier tuning control is helpful in this regard.

Fig. 8-4. Low-frequency calibration test setup.



When the dial of the audio oscillator is tuned to 120 Hz, the Lissajous pattern has the form pictured in Fig. 8-6; its aspect depends on the phase relation of the two input voltages. A summary of basic patterns for various frequency ratios is seen in Fig. 8-7. We will observe that the ratio of the two input frequencies is given by the points of tangency in a Lissajous figure. Thus, we write:

$$\frac{f_h}{f_v} = \frac{n_v}{n_h}$$

where,

- $f_h$  is the frequency applied to the horizontal input,
- $f_v$  is the frequency applied to the vertical input,
- $n_v$  is the number of points at which the pattern is tangent to a vertical line,
- $n_h$  is the number of points at which the pattern is tangent to a horizontal line.

With reference to Fig. 8-8, observe that the Lissajous figure touches vertical line B-B at two points (4 and 16), and touches horizontal line A-A at three points (5, 13, and 21). The frequency ratio in this example is 2:3. For example, if the frequency applied

### FLATNESS OF OUTPUT

Most audio oscillators are rated for uniformity of output amplitude (flatness). For example, a typical instrument has a rating of  $\pm 1.5$  dB over its entire frequency range. It was previously explained how the uniformity of output can be checked with a scope; this procedure assumes that the scope has a frequency response that is quite flat. Alternatively, an ac VTVM can be used. Remember to check each band on the audio oscillator. If the VTVM has a dB scale, checking of flatness ratings is thereby facilitated. We are not concerned with the specific dB level, but with the number of dB of variation over the frequency range of the audio oscillator. Thus, it

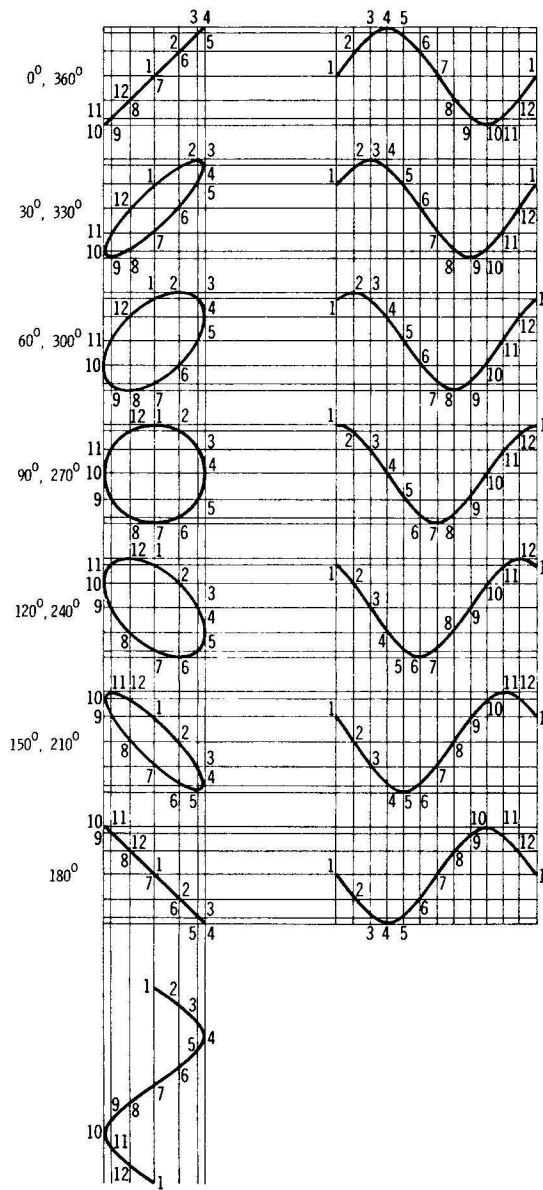


Fig. 8-5. Effect of phase difference on pattern aspect.

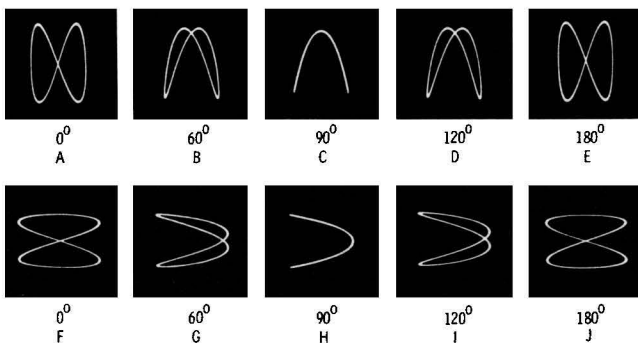


Fig. 8-6. Lissajous pattern for a 2:1 frequency ratio.

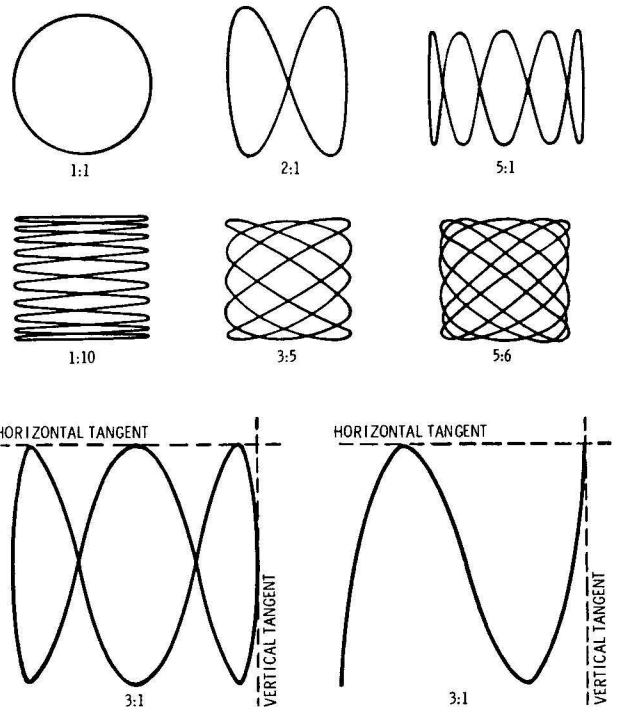


Fig. 8-7. Lissajous patterns for various frequency ratios.

is convenient to adjust the output level so that the dB meter indicates 0 dB at 1 kHz. Then, if the maximum observed variation is between the limits of +1 and -1.6 dB on the scale, the total variation is 2.6 dB, which can be expressed as  $\pm 1.3$  dB, or as +1 and -1.6 dB with respect to 1 kHz. It is good practice to check an audio oscillator under rated load, because the flatness may vary somewhat when the load is other than the rated value.

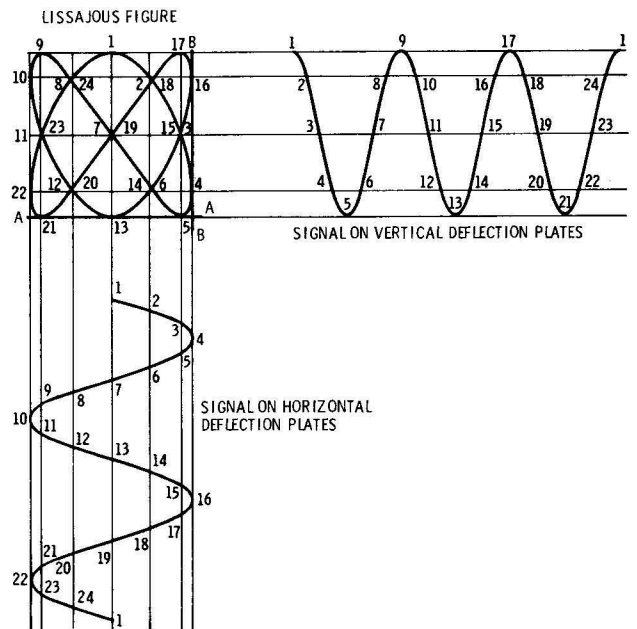


Fig. 8-8. In-phase Lissajous pattern for a 2:3 frequency ratio.

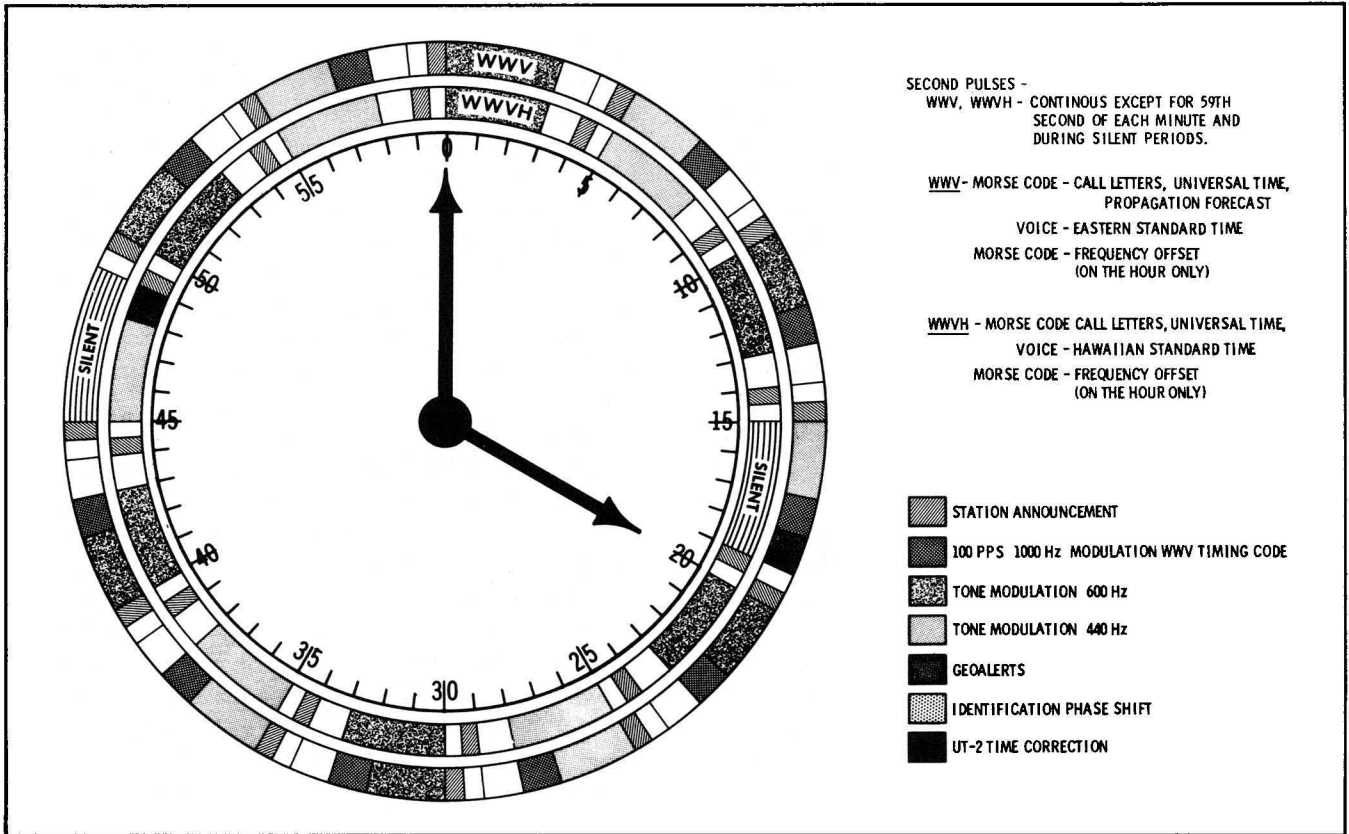


Fig. 8-9. WWV and WWVH transmissions.

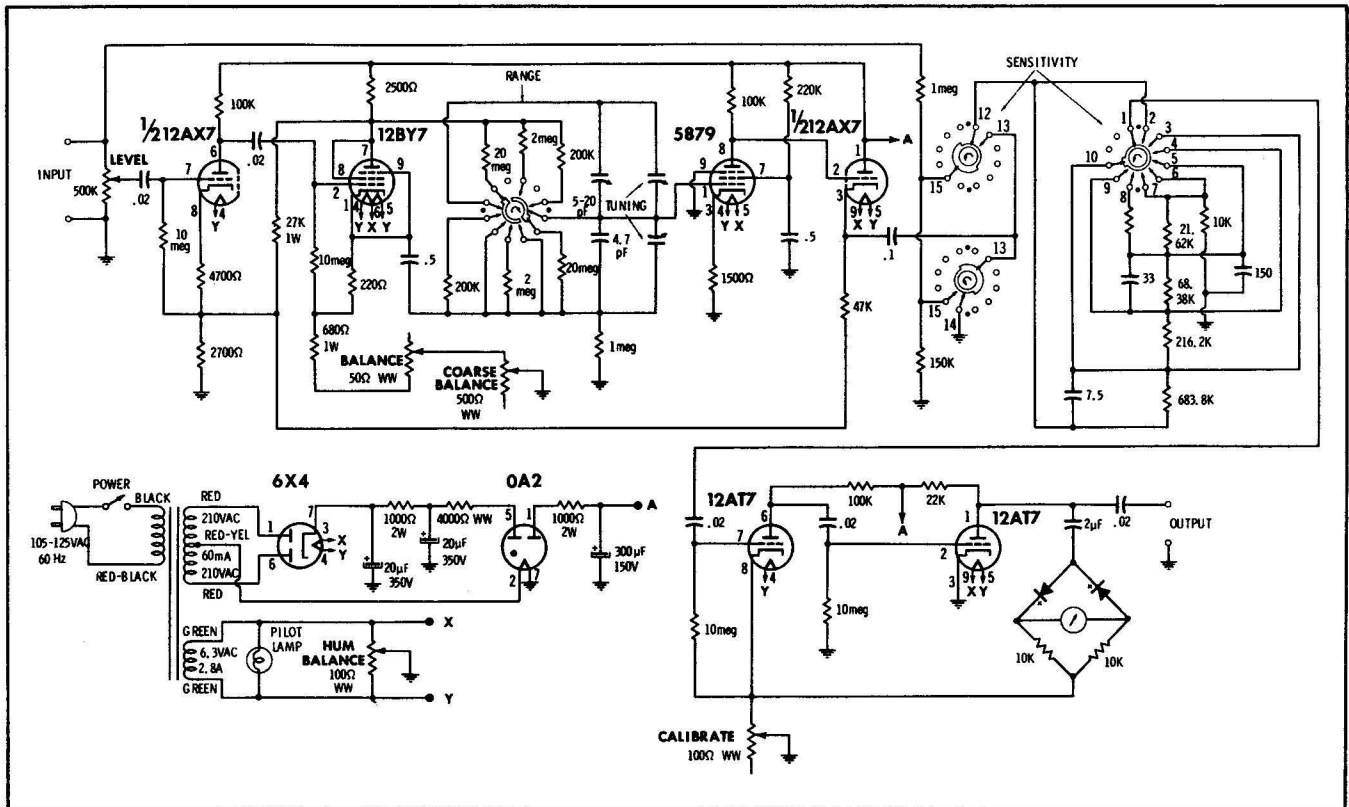


Fig. 8-10. Circuit diagram of a typical harmonic-distortion analyzer.

HARMONIC-DISTORTION METER

The configuration for a typical harmonic-distortion meter is shown in Fig. 8-10. This instrument comprises a high-gain VTVM with a tuned RC-type rejection filter. Switching facilities are provided so that the meter can be operated either with or without the rejection filter. In this example, the VTVM has full-scale ranges of 1, 3, 10, and 30 volts rms. The meter responds to the average value of a waveform because instrument rectifiers are employed—the meter scales are calibrated in rms values. A 500-k potentiometer serves as the input set-level control; this is an operating control. The coarse-balance control is a maintenance adjustment—it is set to a point that gives an average setting of mid-range for the balance control, which is an operating control.

Note that the sensitivity (range) switch in Fig. 8-10 is basically a VTVM multiplier, and is an operating control. Frequency-compensating capacitors are included in the multiplier network to provide flat response from 20 Hz to 20 kHz. The accuracy of full-scale VTVM indication is adjustable by means of the 100-ohm calibrate control. In practice, cali-

bration is usually made by feeding the output from an audio oscillator to the harmonic-distortion meter and to a VTVM that is known to be reasonably accurate. Then, the calibrate control in the HD meter is adjusted to provide the same scale indication as the VTVM. Note in Fig. 8-10 that a hum-balance control is also provided; this is a maintenance control. It is adjusted by switching in the rejection filter while the HD meter is being driven by an audio oscillator. The filter is tuned for minimum indication on the meter, and the hum-balance control is then adjusted to reduce the reading still farther, if possible.

Most trouble symptoms in harmonic-distortion meters are caused by defective tubes. As in the case of many other electronic instruments, capacitors are the next most likely suspects. Although semiconductor instrument rectifiers are normally long-lived, they will fall under scrutiny if the calibrate control is out of range. As in all quantitative-indicating instruments, the power supply must be well regulated; if the meter calibration tends to drift, monitor the B+ voltage to see if it varies from time to time. Remember that a voltage-regulator tube will con-

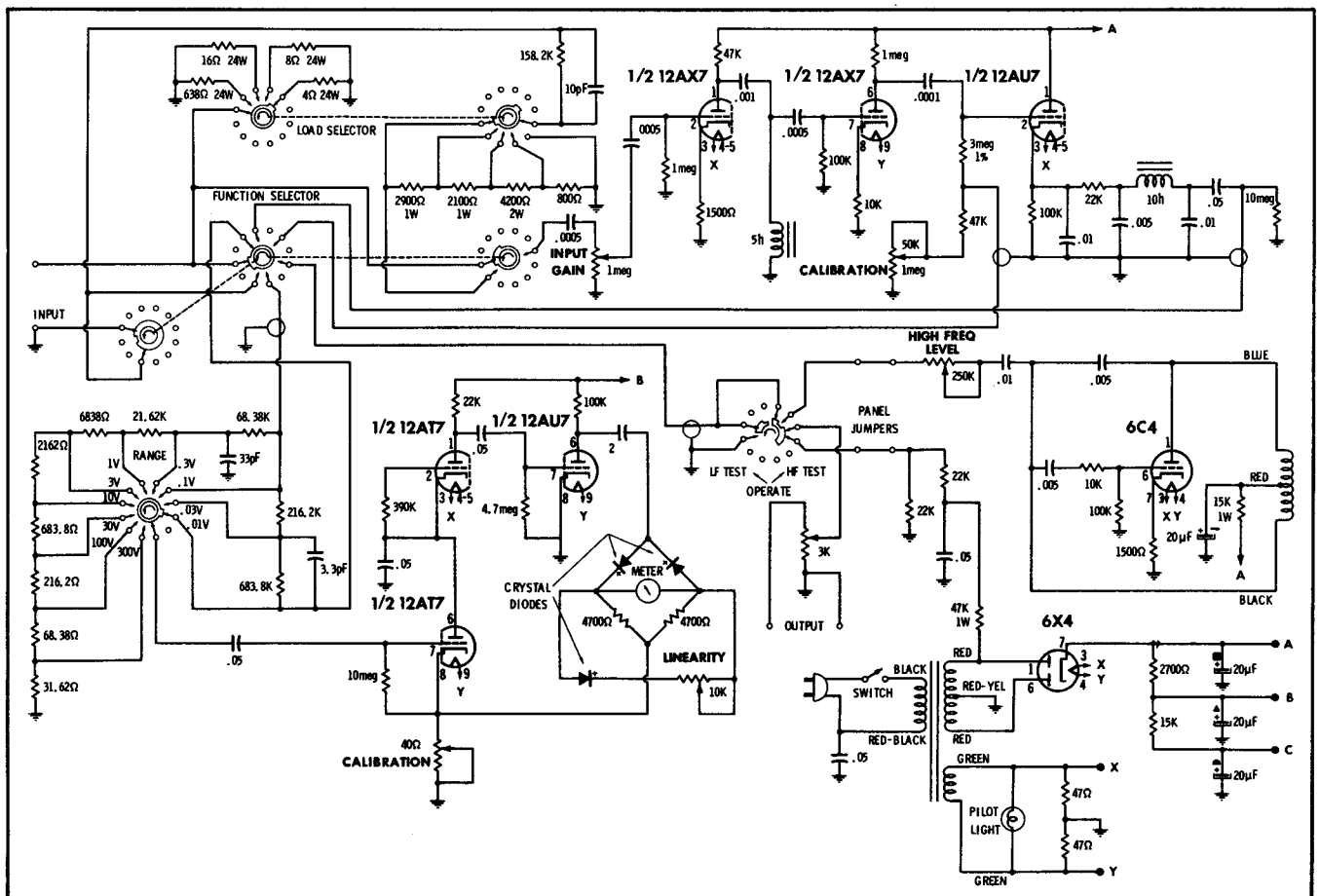


Fig. 8-11. Circuit diagram of a typical intermodulation-distortion analyzer.

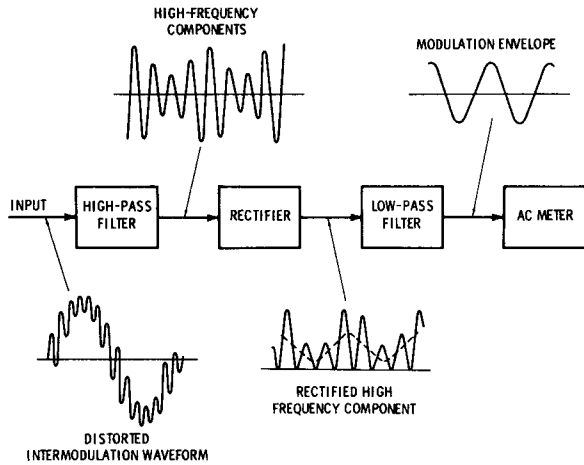


Fig. 8-12. Basic configuration and waveforms in an intermodulation-distortion analyzer.

tinue to glow, although it might be nearing the end of its useful life as its regulating action becomes progressively poorer.

### INTERMODULATION ANALYZER

Intermodulation analyzers are somewhat more complicated instruments than harmonic-distortion meters, and are less widely used in hi-fi service shops for this reason. However, IM analyzers are used in all professional and engineering laboratories. A circuit diagram for a service-type IM analyzer is shown in Fig. 8-11. The indicating meter operates in a basic VTVM configuration. The plan of the instrument is shown in Fig. 8-12. The signal channels employ two filters with high- and low-pass characteristics as shown in Fig. 8-13; these filters separate the IM distortion products from the input waveform. These distortion products are generated as the result of nonlinearity, as shown in Fig. 8-14.

It is helpful to review briefly the operation of an IM analyzer. A built-in 6-kHz oscillator generates the high-frequency component of the test signal, and

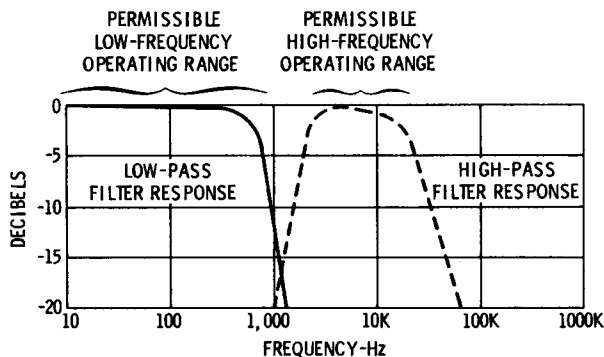
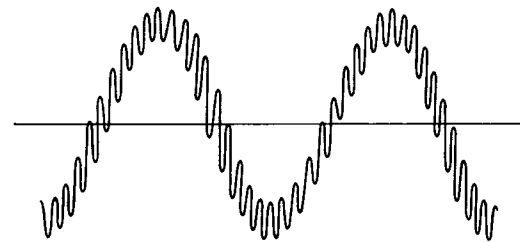


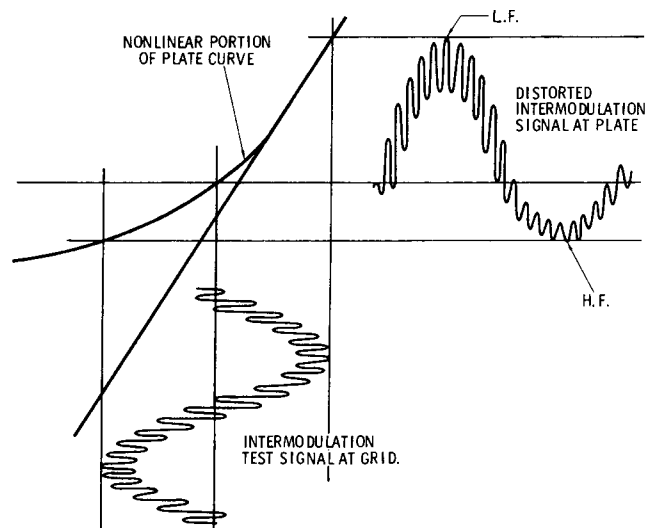
Fig. 8-13. Characteristics of the filters in a typical intermodulation-distortion analyzer.

a 60-Hz voltage from the power line serves as the low-frequency component. This is called a two-tone test signal, and has the normal waveform shown in Fig. 8-14A. The two-tone signal is available at the output terminals in Fig. 8-11, and it can be checked with a scope. Note that the low-frequency level is fixed, but the relative amplitude of the high-frequency component is adjustable by means of a maintenance control (High Freq. Level). An amplifier or device under test for intermodulation distortion is driven by this two-tone signal. If the transfer characteristic of the amplifier or device is perfectly linear, the two-tone signal is undistorted; in other words, no distortion products are formed.

Next, let us consider the situation in which the amplifier or device under test has a nonlinear characteristic. In this case, the two-tone signal will have one peak region compressed, as shown in Fig. 8-14B. This is an intermodulation process that results from a nonlinear transfer characteristic, and its effect is to modulate the low-frequency signal on the high-frequency signal, as shown in Fig. 8-12. This modulated waveform from the amplifier under test is applied to the input terminals of the IM analyzer in



(A) Standard intermodulation test signal.



(B) Intermodulation distortion resulting from nonlinear amplifier characteristics.

Fig. 8-14. Test signal and results of intermodulation-distortion test of a nonlinear amplifier.

## Hi-Fi Stereo Servicing Guide

Fig. 8-11. With reference to Fig. 8-12, the signal flows first through the high-pass filter, which removes the 60-Hz component and leaves the modulated signal. This modulated signal is next rectified in a detector circuit that develops the rectified high-frequency component. The demodulated signal then flows through a low-pass filter which develops the low-frequency modulation envelope. The amplitude of this envelope waveform is proportional to the percentage of intermodulation distortion, which is indicated by the VTVM section of the IM analyzer.

The indicating meter in a service-type IM analyzer is an average-indicating type employing semiconductor-diode instrument rectifiers. In the example of Fig. 8-11, a linearity diode is also provided. The VTVM calibration control is a maintenance adjustment, and is set in the same manner as previously explained for a harmonic-distortion meter. It interacts to some extent with the Linearity control, which is a maintenance adjustment. The Linearity control is set to a point that provides best scale-indication consistency when the VTVM multiplier is dropped down one step. We work back and forth between the two controls to obtain accurate full-scale indication and accurate  $\frac{1}{3}$ -scale indication.

Since the Input Gain potentiometer is an operating control, it requires no attention unless it becomes defective. The IM Calibration control is a maintenance control, and it requires due attention if the accuracy of the IM measurements is to be retained as tubes age and components drift. This IM analyzer calibration can be carried out in various ways. In any case, a signal that involves 10 percent IM distortion is applied to the analyzer section, and after the input control is adjusted to produce full-scale reading on the 1 percent IM range, the IM control is then set for full-scale reading on the Set Level positions of the Function and Range switches.

A satisfactory method of IM-analyzer calibration, using a minimum of equipment, requires an audio oscillator and a pair of resistors of approximately 10-k each. A signal that will produce 10 percent IM distortion can be formed by mixing the signal from the IM generator section in this example with the signal from the audio oscillator in a voltage ratio of 10:1, and with a frequency difference of 60 to 100 Hz. We make up the calibrating resistor network by connecting the 10-k resistors to a 4.7-k resistor, as shown in Fig. 8-15. The test switch on the IM analyzer is set to the HF Test position, so that the 60-Hz signal is removed from the two-tone output, leaving only the 6-kHz component. With reference to Fig. 8-15, some connections are shown in dotted lines, and other connections are shown in solid lines.

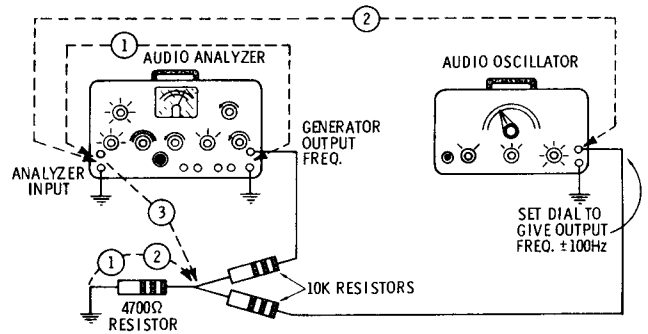


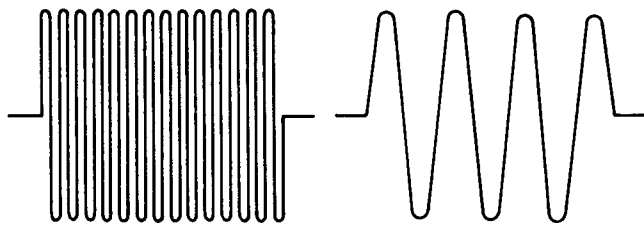
Fig. 8-15. Calibration resistor arrangement.

The solid-line connections are used throughout the calibration procedure, whereas the dotted-line connections are used only in certain portions of the procedure.

Next, we check the IM calibration as follows: Connect one of the matched resistors from the "hot" output terminal of the IM analyzer to the 4.7-k resistor, and connect the other end of the 4.7-k resistor to ground. The other matched resistor is connected from the audio-oscillator output terminal to the same 4.7-k resistor, and thence to ground. The IM analyzer is operated in the VTVM mode, with the load switch turned to the Hi-Z position. We set the Input Gain and the Generator Output controls to minimum; the range switch is set to 10 volts, and the input terminal of the IM analyzer is connected to the junction of the three resistors, as denoted by the dotted line No. 3 in Fig. 8-15.

Next, the audio oscillator is tuned for zero-beat with the output from the analyzer. This part of the procedure requires that the High-Frequency Level and Generator Output controls be set to maximum. Using maximum output from the audio oscillator, watch the pointer as the audio oscillator is tuned through 6 kHz. The dial is carefully adjusted for zero beat (motionless pointer). Then, remove connection No. 3 in Fig. 8-15, and set the audio oscillator to a frequency that is 60 to 100 Hz higher or lower than the zero-beat setting. Make the two No. 1 connections as follows: Short the 4.7-k resistor and connect the input terminal to the IM analyzer signal-output terminal. Adjust the High-Frequency Level and Generator Output controls to indicate a maximum value on the 30-volt range of the meter.

Remove the No. 1 connections in Fig. 8-15 and make the No. 2 connections, shifting the input-terminal lead from the output of the IM analyzer to the output of the audio oscillator. Then, adjust the output control on the audio oscillator to obtain the same meter deflection as noted above, except that now the meter is operated on its 3-volt range. In this manner, establish a voltage ratio of 10:1.



(A) High frequency. (B) Low frequency.

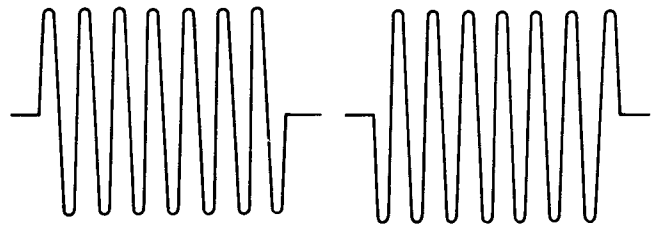
Fig. 8-16. Tone-burst waveforms.

Now remove the No. 2 connections and make the No. 3 connection. This takes the short-circuit away from the 4.7-k resistor and connects the input of the IM analyzer to the junction of the three resistors. Making these connections results in applying the desired mixed signal to the analyzer and will result in a carrier with 10 percent modulation at the detector in the analyzer. Set the range switch to its 10 percent IM position, and set the function switch for measurement of percentage of IM. Then, adjust the IM Input Gain control for full-scale meter deflection. The function switch is turned to its Set Level position, and the Range switch is set to the indicated Set Level position (3 percent IM). Finally, adjust the IM Calibrate control for full-scale meter deflection, and the procedure is completed.

### TONE-BURST GENERATOR

A tone-burst generator provides a sine-wave output that is keyed on and off at regular intervals. The frequency of the sine wave is ordinarily adjustable to any value in the audio range. Fig. 8-16 shows a pair of ideal tone-burst waveforms. Since tone-burst generators are comparatively expensive instruments, they are used chiefly in engineering labs. Although it is easy to connect an audio oscillator to an electronic switch in order to generate a tone-burst waveform, there is a basic disadvantage to this simple method. That is, the sine wave does not start at

### Test-Equipment Performance Verification



(A) Waveform starts at  $0^\circ$ . (B) Waveform starts at  $180^\circ$ .

Fig. 8-17. Noncoherent bursts.

the same point in successive bursts, as shown in Fig. 8-17. When the sine wave starts at random phases in successive bursts, the waveform is said to be noncoherent. To generate a coherent tone-burst waveform, the gating circuit (electronic switch) must be tightly locked to the sine-wave frequency.

It is desirable to employ a coherent tone-burst waveform, because the transient response of an amplifier or speaker depends on the polarity of the first excursion in the sine wave, and also on the exact phase of the starting point. If a coherent tone-burst generator is checked by feeding its output into a scope, the following waveform characteristics are observed in normal operation:

1. There is no blurring in the burst display, and there is negligible jitter.
2. The burst has a good sine waveform, with level top and bottom peaks.
3. No dc component is evident; that is, the burst is centered on the horizontal axis.
4. Stability is maintained over the entire audio-frequency range; the tone burst does not blur at certain critical frequencies.
5. Rise time is sufficiently fast that there is negligible distortion of the first half-cycle in the burst, even at high audio frequencies.
6. The tone burst ceases abruptly, without noticeable ringing.

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