Practical Stereo FM Servicing Guide

A handy "how to" booklet for electronic technicians describing...

- ☐ Circuit operation and characteristics
- ☐ Troubleshooting techniques
- ☐ System measurements

Edited by the publishers of **Electronic Servicing** *magazine*

Practical Stereo FM Servicing Guide

by Robert G. Middleton

The material in this servicing guide has been adapted from a series of articles published in **Electronic Servicing** magazine.

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Part 1

Frequency response tests

Although a stereo-FM or AM-FM stereo chassis is not a simple system, it can be broken down into functional sections as depicted in Fig. 1. Troubleshooting is comparatively easier when a section is completely dead, than when the system "still works but doesn't sound right."

Preliminary clues often can be observed by turning the function switch and listening carefully to the sound output on each function. For example let us consider the following situations:

1. If the AM radio operates satisfactorily, but FM mono is distorted, the FM front end and the FM detector are checked. There is also the possibility of trouble in the FM function of the IF amplifier.

2. If FM-stereo and phono-stereo functions are distorted, the audio sections are suspected first.

3. If the phono-stereo function is normal, but one channel is distorted on FM-stereo, we turn our attention to the multiplex section.

4. If the FM-stereo function is normal, but one channel is distorted on phono-stereo, check the phono cartridge first.

Tube vs. Transistor Circuits

Section malfunction in tube-type or hybrid systems is approached in the same manner as a TV receiver with the tubes being the immediate suspects. However, in an all-transistor system, capacitor defects are suspected at the outset as transistors have a much lower failure rate than tubes. A shorted or leaky capacitor will usually show up during DC voltage measurements, though an open capacitor usually must be pinpointed by scope waveform checks.

With this brief introduction to trouble localization we will now discuss specific trouble symptoms and their causes and cures.

Frequency-Response Characteristics

One of the more elusive causes of a "doesn't-sound-right" complaint is poor frequency response in a defective section. There are several common causes of this symptom:

1. A capacitor or other com-

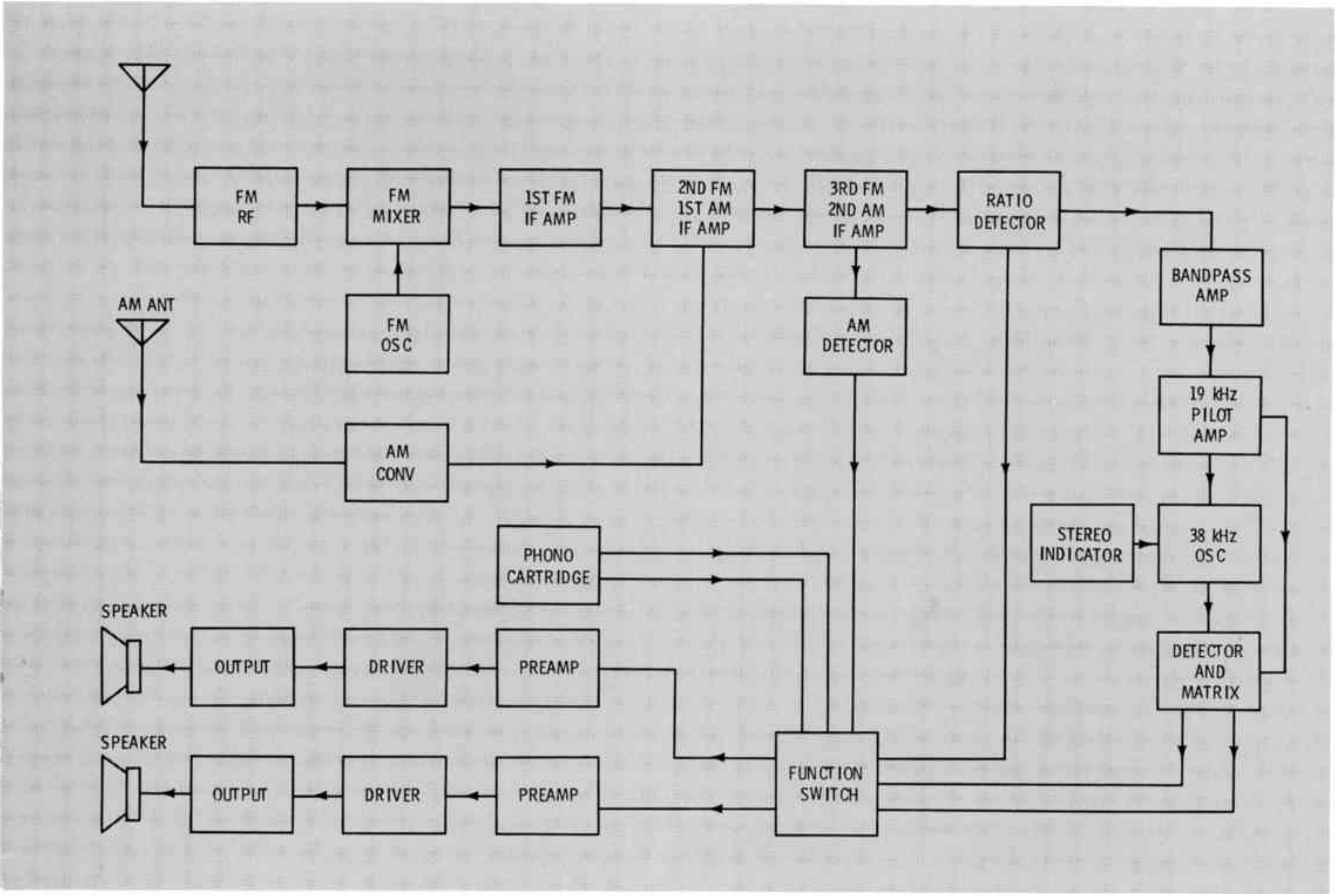


Fig. 1. Block diagram of a typical AM-FM stereo chassis.

ponent associated with a tuned circuit is defective.

2. Rough handling has caused tuned circuits to resonate off-frequency.

3. Expansion and contraction of components due to thermal cycling has caused frequency shift in a tuned circuit.

4. A do-it-yourselfer has "fiddled" with the alignment adjustments prior to the service call.

5. Previous tube or transistor replacements have changed the total capacitance in one or more tuned circuits, thereby causing tuned circuits to resonate off-frequency.

6. A tuned-circuit component has been replaced previously, and the stage aligned by ear.

Let us consider the normal frequency response for the front end of an FM tuner. Fig. 2 shows the circuit diagram for a typical tuner. The front end consists of the RF amplifier, oscillator, AFC circuit, and the mixer. Each channel has a bandwidth of 150 KHz. Therefore, an ideal frequency response for an FM tuner would have a rectangular shape, as indicated in Fig. 3. The bandwidth indicated for station A would be 150 KHz. However, it is impossible to obtain this ideal frequency response with tuned circuits. Accordingly, we settle for a frequency response similar to the actual response curve shown in Fig. 3.

It might seem that a full 150-KHz bandwidth would not be obtained from the actual curve in Fig. 3; however, the top portion of the curve is effectively sliced off by subsequent limiter action. Therefore, a full 150-KHz bandwidth is obtained, although the peak frequencies span a lesser bandwidth.

It is undesirable to align a front end for excessive bandwidth, because selectivity and gain are sacrificed. On the other hand, it is undesirable to align for decreased bandwidth, because fidelity is sacrificed due to sideband cutting, i.e., if the higher sideband frequencies are attenuated, the higher audio frequencies will be attenuated proportionally.

Note that the AFC and oscillator sections in Fig. 2 have no effect on the shape of the front-end response curve. The front-end response is developed between the antenna-input terminals and the mixer test-point (TP5) by the circuits that include L5, L3 and L4. The oscillator merely injects a beat signal that is 10.7 MHz higher than the center frequency of the front end. Since we are concerned only with the frequency-response characteristics of the front end at this time, details of alignment procedure will be reserved for subsequent discussion.

The IF amplifier includes the circuits from the plate of the mixer tube to the grid of the second limiter tube in Fig. 2. Normally, the IF amplifier has the same frequency-response curve as the front-end except that the skirts are much steeper due to the cascaded tuned circuits consisting of T4, T5, T6 and T7.

The bandwidth of a normal FM IF curve is measured as shown in Fig. 4. The frequency span between the 50 percent maximum amplitude points is normally 200 KHz. Thus, the FM signal with a bandwidth of 150 KHz can be passed by the flat-topped portion of the response curve without sideband cutting.

Note that the flat top of the response curve is due, in part, to limiter action. Therefore, minor peaks and valleys in the IF response prior to the limiter are sliced off, and a flat response is fed to the discriminator. Serious peaks and valleys cannot be removed by limiter action, though, so it is advisable to check the IF frequency response with a signal level that just produces silencing. This will be explained in greater detail later.

The normal frequency response of the FM detector, or discriminator, is shown in Fig. 5. The important characteristics of this S curve are adequate bandwidth (200 KHz in the example shown) and a linear interval between the band limits. If the bandwidth is inadequate, the high-frequency sidebands will be cut. If the frequency response is nonlinear, some audio frequencies will be attenuated and others will be enhanced. The end result of a good alignment job is flat frequency response over the complete audio-frequency range.

Most readers are familiar with pre-emphasis and de-emphasis in FM systems. For the benefit of less experienced technicians, let us note that the high frequencies are emphasized in an FM broadcast signal, but are not emphasized in a test signal obtained from a signal or sweep generator. Therefore, generator tests of frequency response are made between the grid of V9 and test point D in Fig. 2. The de-emphasis network consists of R24, C39 and C40. This is a low-pass filter with a normal frequency response as shown in Fig. 6. The response of the de-emphasis network can be checked with an audio oscillator and an AC VTVM.

FM stereo multiplex configurations, such as shown in Fig. 7, operate from low audio frequencies up to 53 KHz. Therefore, the discriminator must drive the multiplex sec-

Table 1. FM Stereo-Multiplex Adapter Alignment Procedure

FM stereo multiplex alignment using FM stereo signal generator (±.0001% accuracy)

Connect high side of generator to point C, low side to ground.

Generator Frequency	Indicator	Adjust	Remarks
67 KHz	Vert. amp. of scope through a 1 meg to point D; low side to ground.	A19, A20	Adjust for minimum. If whistle or interference is present, readjust A19, A20 to eliminate this condition.
19 KHz	Vert. amp. through 47K to point E; low side to ground.	A21, A22	Adjust for maximum.
19 KHz	Vert. amp. through 47K to point D; low side to ground.	A23	Adjust maximum for 38 KHz response.
Modulated Left Channel	Vert. amp. to point F; low side to ground.	A21, A22, A23	Adjust for minimum.
Modulated Right Channel	Vert. amp. to point G; low side to ground.		Check for minimum. Make compromise adjustments of A21, A22, A23 if necessary.

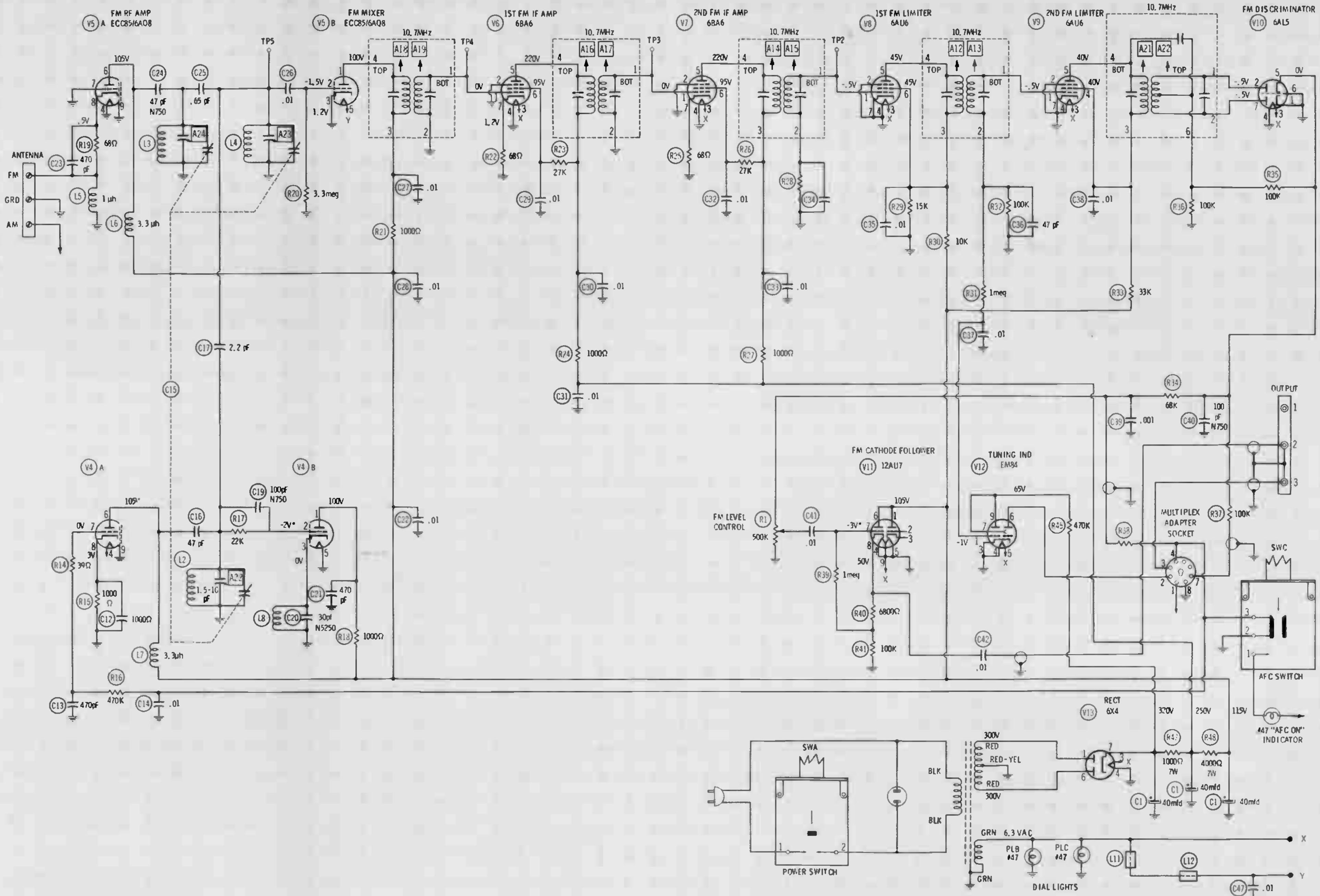


Fig. 2. Complete schematic diagram of a typical tube-type FM tuner.

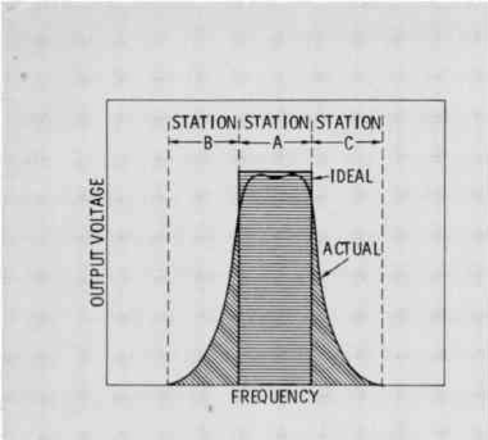


Fig. 3. Frequency-response curve.

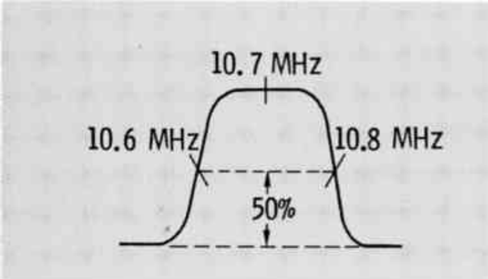


Fig. 4. IF amplifier frequency response.

tion directly, instead of through the de-emphasis network, As in Fig. 2, the output from the discriminator branches into two paths. One path leads through the de-emphasis network; the other through R37 to the multiplex adapter socket.

Fig. 8 shows the frequency spectrum processed by the multiplex adapter. To check the frequency response of the multiplex adapter, a suitable generator such as an FM

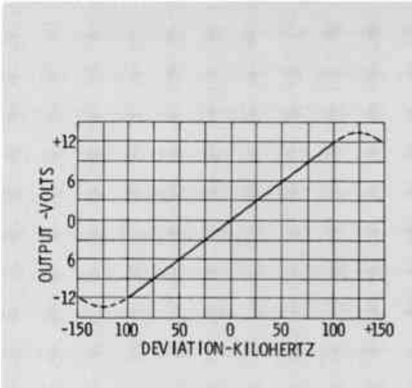


Fig. 5. Discriminator frequency response.

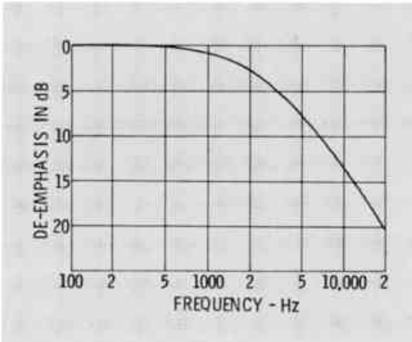


Fig. 6. Standard de-emphasize curve.

stereo multiplex generator or FM stereo signal simulator must be used.

The audio amplifiers are conventional hi-fi units, with an upper frequency response of approximately 20 KHz. Fig. 9 shows the flat audio-frequency response of which the amplifiers are normally capable along with the frequency characteristics of the bass and treble controls. Most shops check audio-frequency response with an audio oscillator and VTVM, though a few employ an audio sweep generator and scope in this test. The chief advantage of an audio sweep test is that the frequency-response pattern can be "sized up" almost at a glance.

Testing Frequency Response

It is a common practice to check the frequency response of stereo-FM systems by sections. Thus, the RF IF sections are checked before the discriminator, etc. An FM test signal is provided by a sweep-frequency generator (or simply sweep generator) and a scope is used as the indicator. As shown in Fig. 10, the FM signal is applied to the input of the tuned circuits, a scope is connected at the output of the tuned circuits, and a frequency-response curve is displayed on the scope screen. For example, the FM RF test signal might be applied at the antenna-input terminals of the FM tuner, and the scope connected at the grid of the limiter tube prior to the discriminator.

We might set the front-end to 100 MHz. In this example the center frequency of the sweep generator would also be set to 100 MHz. The sweep width, or deviation of the sweep generator, is set to a comparatively high value so that the entire response curve and portions of the base line are displayed on the scope screen. The exact bandwidth of the correct response curve should be noted in the receiver service data. It is just as important to align for correct bandwidth as for a properly-

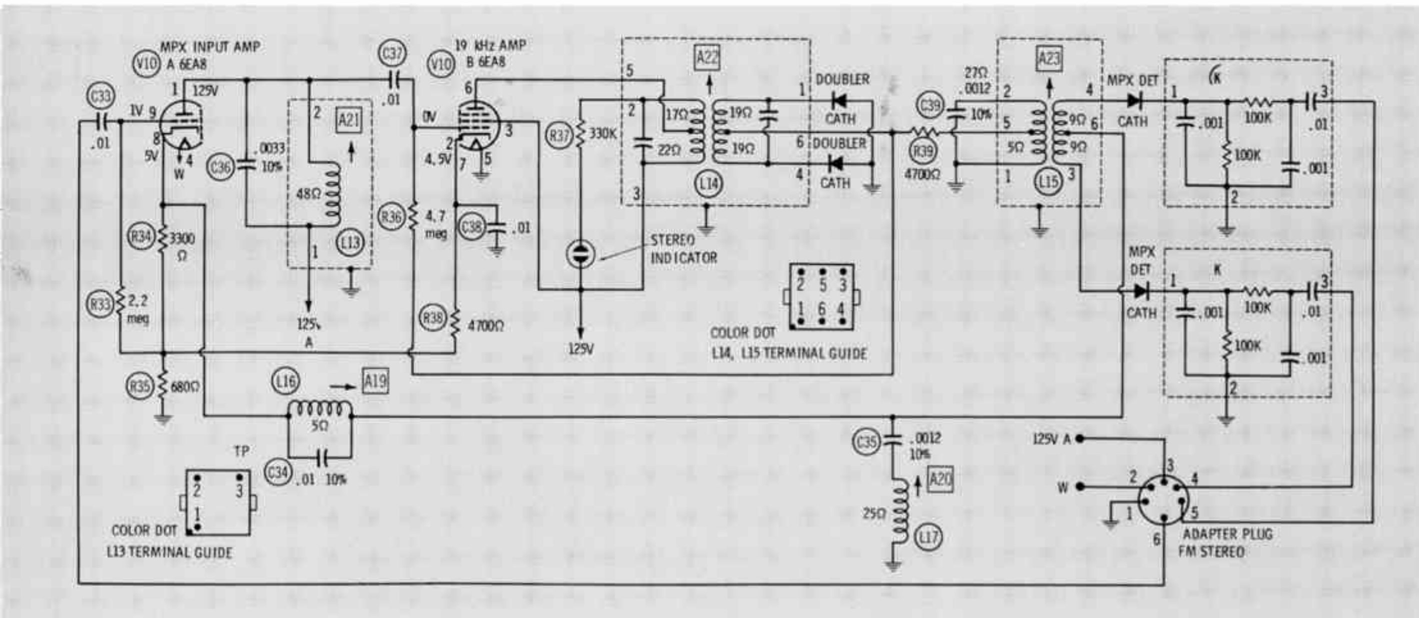


Fig. 7. Envelope-detection multiplex circuit.

shaped curve. A sweep width of approximately 450 KHz is generally used. Fig. 11 illustrates a distorted IF response curve. Compare it with the normal curve shown in Fig. 4.

With reference to Fig. 11, it is necessary to identify the center-frequency point and the frequencies at the 50 percent of maximum points on the curve. This necessitates the use of an accurate marker generator in combination with the sweep generator. (A marker signal is merely a CW signal with a precise frequency. It is marked with the sweep signal and produces a beat frequency that is displayed as a "pip" on the response curve at the marked frequency.) Some sweep generators have a built-in marker generator, if not, a separate marker generator must be used.

Some sweep generators are provided with a crystal holder into which can be plugged a 10.7-MHz (or other frequency) quartz crystal for precise marker indication. For example, if we use quartz crystals to mark the curve shown in Fig. 4, we will require at least two crystals, with frequencies of 10.6 and 10.8 MHz. A crystal with a frequency of 10.7 MHz is also desirable for IF alignment. The foregoing example assumes that the sweep signal is applied at the grid of the mixer tube (V5 in Fig. 2). If the sweep signal is applied at the antenna-input terminals, we then must use RF marker signals. For example, if the center frequency of the sweep is 100 MHz, the bandwidth must be checked with frequencies of 99.9 and 100.1 MHz.

Next, let us consider a test of the discriminator frequency response. An IF sweep signal is applied at the grid of V9 in Fig. 2, and the scope is connected at test point D. A pattern similar to that shown in Fig. 12 is normally displayed on the scope screen (compare it with Fig. 5). A beat marker is easily displayed at either of the band limits (10.6 and 10.8 MHz). However, it is difficult to display a marker at the center frequency (10.7 MHz), particularly when a ratio detector is being aligned. Therefore, many technicians employ a trick of the trade that involves using an amplitude-modulated marker. This causes the response-curve frequency to beat with the modulating frequency, although the marker pip remains invisible. If the S curve (Fig. 12) is centered on 10.7 MHz, the beat amplitude will pass through minimum as the marker generator is tuned through 10.8 MHz.

Two basic steps are used to test the frequency response of the FM stereo-multiplex adapter. First, the peak and trap frequencies are checked in accordance with the instructions in Table 1. Note that A19 and A20 in Fig. 7 are SCA (subsidiary carrier assignment) traps that are tuned to 67 KHz. L15 is aligned first by peaking it to the subcarrier frequency (38 KHz). However, the optimum adjustments of L15, A21 and A22 are made on the basis of a separation test with stereo signals. This is a specialized test procedure that will be explained in detail in the next article of this series. The procedure is finally verified by checking the frequency-response over the audio-frequency range. Since this verification is tied in with separation characteristics, it also will be covered in the next article.

Frequency-response tests of the audio-amplifier channels in Fig. 1 are commonly made as shown in Fig. 13. The important points to be observed are: (1) provision of a suitable value of power resistor across the output terminals so that the amplifier output circuit is correctly loaded; (2) setting of tone controls for flat response; (3) driving of the amplifier to its maximum rated power output; and (4) spot-checking of the audio-oscillator output to make certain that a constant level of drive is applied over the entire audio-frequency range.

FM Tuner Alignment

Either a signal generator and VTVM may be used, or a sweep-and-marker generator and scope. Let us consider the use of a signal generator. Referring to Fig. 2, proceed as follows:

1. The AFC switch is turned off. If this is not done, the oscillator in the front end will tend to "follow" the test signal, and the bandwidth will seem to be substantially greater than is actually the case.
2. Alignment of the IF amplifier is usually checked first. Turn the FM level control (R1) to maximum and connect the VTVM across the grid-leak resistance of the second limiter tube (V9). This is test point B.
3. A low-level 10.7-MHz CW signal from the signal generator is applied to the grid of the first limiter tube (V8). By setting the generator output below the limiting level, a much sharper peak indication is observed on the VTVM.
4. Slugs A12 and A13 are adjusted to obtain maximum response at 10.7 MHz. Note that the bandwidth is not checked at this time.
5. Align T6 in the same manner. The generator signal is applied to the grid of the second IF tube (V7), with the VTVM connected as before. Thus, T6 is aligned in combination with the response of T7. If the peak indication is broad, reduce the output from the signal generator.
6. Transformers T5 and T4 are peak-aligned in the same way, with the VTVM connected at the grid of V9, as before.
7. The primary of T8 is then peak-aligned. Leave the generator output connected to the grid of the mixer (V5), but move the VTVM connection to the center tap on the secondary of T8. With the generator set for an output below the limiting level, slug A10 is adjusted for maximum indication at 10.7 MHz.
8. To align the secondary of T8, the VTVM connection is moved to test point D, and the VTVM is set for zero-center scale indication. Slug

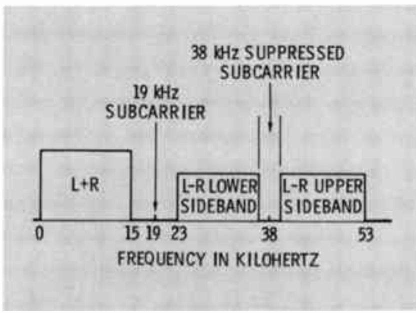


Fig. 8. Frequency spectrum processed by multiplex adapter.

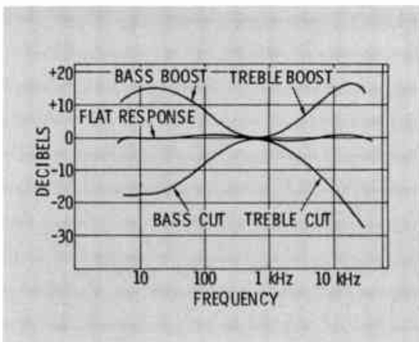


Fig. 9. Response curves for bass and treble tone controls.

A21 is then adjusted for zero indication on the meter. Since A20 and A21 tend to interact, it is advisable to repeat steps 7 and 8.

9. Peak alignment of the IF section is now complete, and we turn next to the RF and oscillator sections in the front end. The signal generator is set up to 108 MHz and its output is applied through a 270-ohm resistor to the FM antenna-input terminals. The VTVM is connected at test point B.

10. With the receiver tuned to 108 MHz, A22 is adjusted for maximum response. This puts the oscillator in rough alignment.

11. Tracking is checked by setting the generator and receiver tuning to 88 MHz. If A22 requires resetting to obtain maximum indication, a compromise adjustment is made.

12. The final peak-alignment procedure concerns the RF section. With the generator and VTVM connected as before, trimmers A23 and A24 are adjusted for maximum indication with the receiver and generator set to 106 MHz. If appreciable adjustment of A23 and A24 is required, it is good practice to repeat steps 10, 11 and 12 because the response is being checked through the IF section.

Peak alignment is ordinarily sufficient. However, when hi-fi response is sought, we must conclude by checking the bandpass of the FM tuner. This is done by leaving the generator and VTVM connected as in Step 12. The generator output is advanced to obtain limiting action (very broad peak indication on the VTVM). Then, the bandwidth is checked with a scope, as shown in Fig. 4. Insufficient bandwidth will cause attenuation of the high audio frequencies and impaired stereo-multiplex action, as will be explained in detail in the next article. Excessive bandwidth is undesirable because selectivity and gain are then sacrificed.

Abnormal Bandwidth

When the over-all response curve indicates incorrect bandwidth even after peak alignment has been attempted, there is probably a defective component in the system. The first step is to localize the defective section. Therefore, we recheck the peak-alignment procedure outlined previously, but we now measure the bandwidth from stage to stage. As we check back from the second limiter, a stage will be encountered at which the incorrect bandwidth suddenly appears. We direct our attention to this stage. Defects in early sections are most troublesome because the inaccuracy is amplified by the following stages.

A common cause of insufficient bandwidth is regeneration. Look for defective screen bypass capacitors, poorly grounded transformer shields and defective plate-decoupling capacitors. Regeneration shows up not only as insufficient bandwidth but also as a change in shape of the response curve at various signal levels. Peaks and valleys in the response curve that are not removed by normal limiter action are usually caused by regeneration. If you place your hand near a tube in a regenerative circuit, the meter indication will change. Stable stages (without regeneration) are practically immune to the effects of hand capacitance.

Excessive bandwidth is commonly caused by leaky capacitors connected across tuned coils. The leakage reduces the circuit Q which, in turn, reduces the stage gain and increases the bandwidth. Suspected capacitors must be disconnected for test since the shunting coil resistance is very low. Shorted turns in a coil have the same effect on circuit action, but this defect is rare.

When transistors are used in an FM tuner, low gain and excessive bandwidth point to collector-junction leakage. It is easy to make in-circuit checks of transistor control action. However, this topic is reserved for treatment in the next issue. ▲

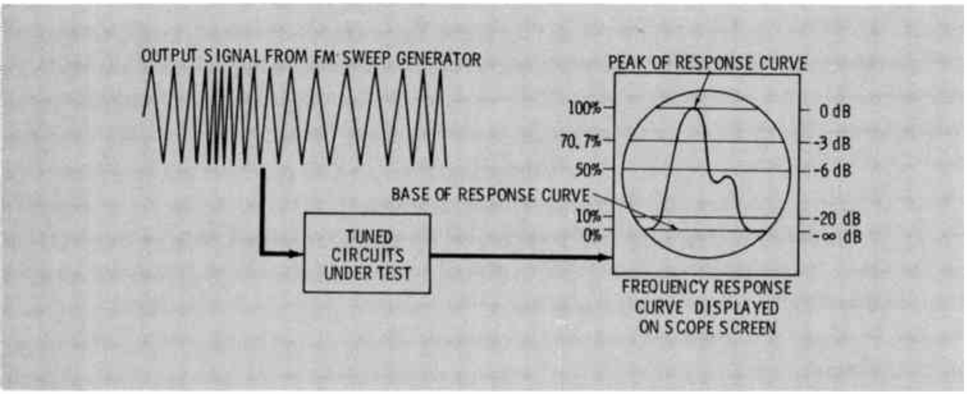


Fig. 10. Basic principle of sweep alignment.

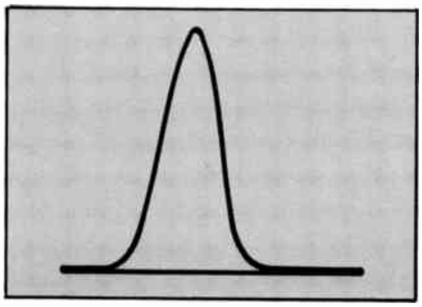


Fig. 11. Distorted IF response curve.

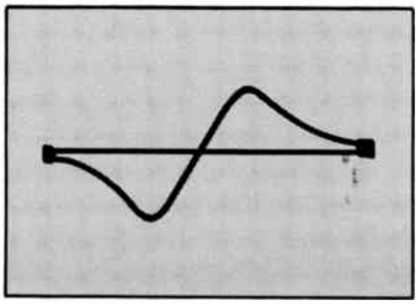


Fig. 12. Discriminator "S" curve.

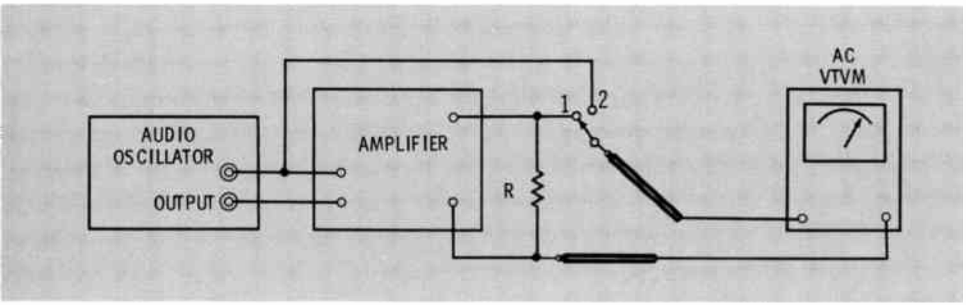


Fig. 13. Equipment setup for testing frequency response of audio amplifier.

Solving stereo separation problems

by Robert G. Middleton

A block diagram of an FM stereo multiplex receiving system is shown in Fig. 1. In part 1 of this series we discussed the circuitry up to the FM detector. In this article we will cover dynamic testing of multiplex adapters and defects causing poor separation. However, to provide a clearer understanding of stereo channel separation we first will review the generation of the FM stereo signal.

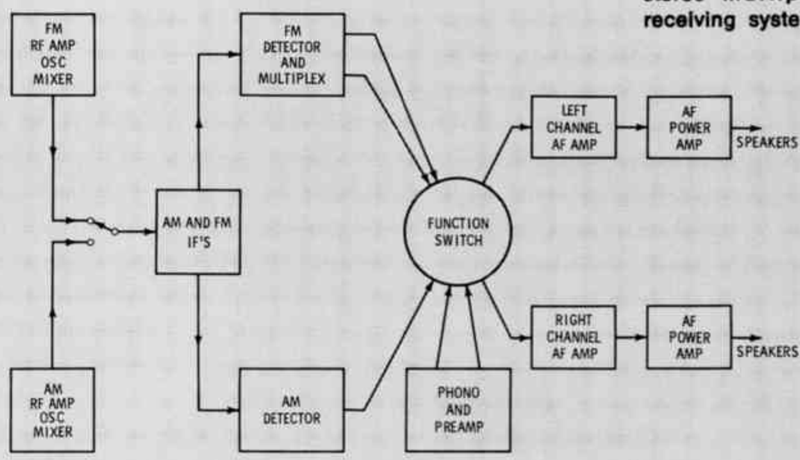
Producing the FM Stereo Signal

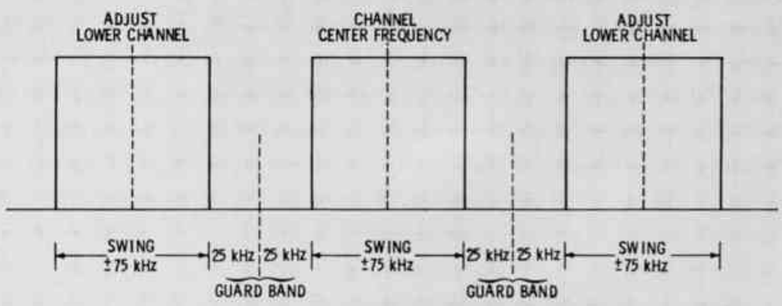
Basically, an FM-stereo signal consists of two different audio-frequency signals that occupy the same FM channel. These separate audio signals provide stereophonic sound reproduction. The individual audio signals are identified as "left" (L) and "right" (R). In conventional programming, this pair of audio signals originates from a pair of microphones at a sound studio, as depicted in Fig. 2A. 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 separated to simulate the placement of the transmitting microphones.

Fig. 2B shows the FM channel allocations. In monophonic transmission, the carrier (center frequency of a channel) is frequency-modulated as depicted in Fig. 2C. A swing of ±75 KHz represents FM sidebands that occupy the entire channel. In turn, the question arises concerning 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 high-fidelity transmission is a basic requirement, a method must be employed that permits each signal to occupy the entire channel bandwidth.

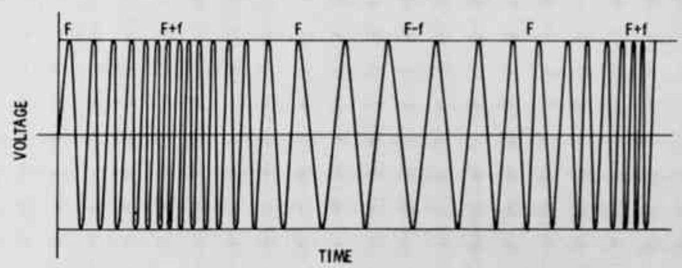
To transmit both the L and R signals at high fidelity in a single channel requires multiplex transmission. This is a technique which permits satisfactory separation of

Fig. 1 Block diagram of AM/FM stereo multiplex receiving system.





(B) Frequency spectrum

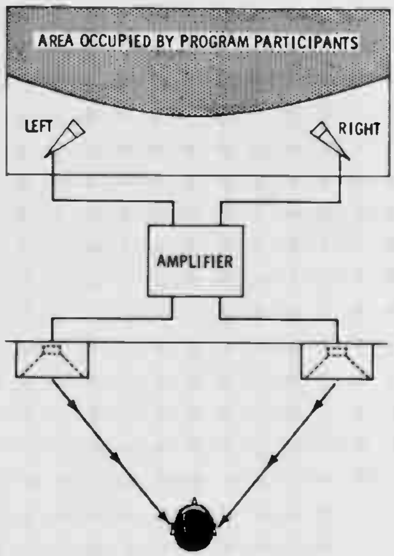


(C) Frequency-modulated carrier

the L and R signals at the receiver. Note also that the system specified by the FCC is a compatible system. In other words, to a conventional FM receiver (monophonic receiver) the multiplex signal "looks like" an ordinary FM signal. But to a stereo-multiplex receiver, the multiplex signal "looks like" separate L and R signals.

A stereo-multiplex system starts with the conventional monaural 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, the mono (L+R) signal is produced. This 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 on 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 depicted in Fig. 4. Both the L+R signal and the 38-KHz carrier are frequency-modulated on the RF carrier. However, only the L+R signal can be reproduced at the FM receiver. That is, the 38-KHz carrier is out of the range of audibility. 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. 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 AM carrier subsequently modulates the RF carrier in combination with the L+R signal.



(A) Block diagram

Fig. 2 Basic concept and characteristics of FM stereo system.

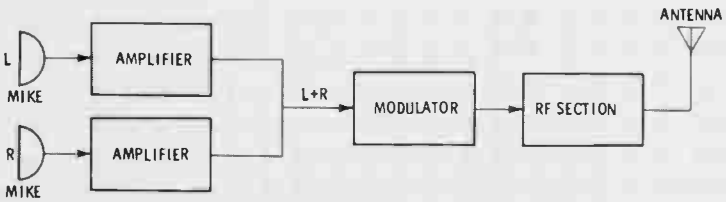
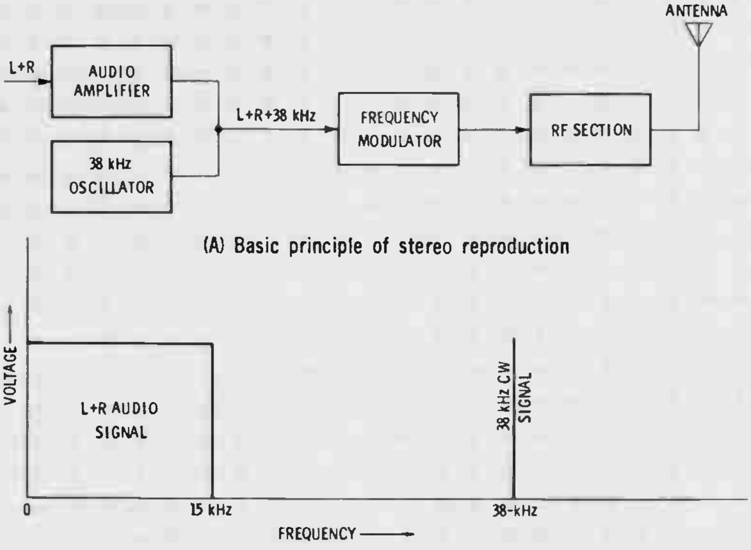


Fig. 3 L and R signals are mixed to produce monaural signal.



(A) Basic principle of stereo reproduction

(B) FM channel specifications

Fig. 4 L+R and 38-KHz signals frequency modulated on RF carrier.

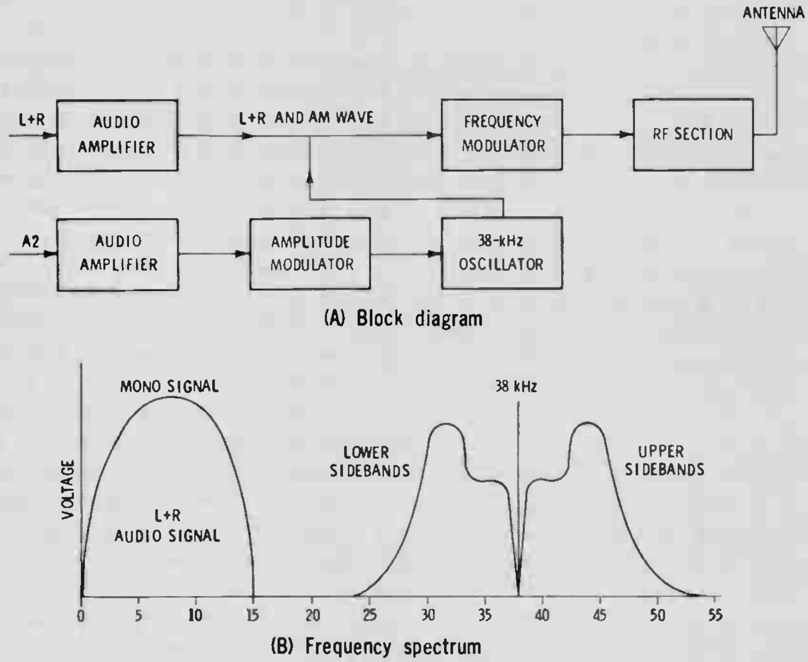


Fig. 5 Amplitude-modulated 38-KHz subcarrier combined with L and R signals.

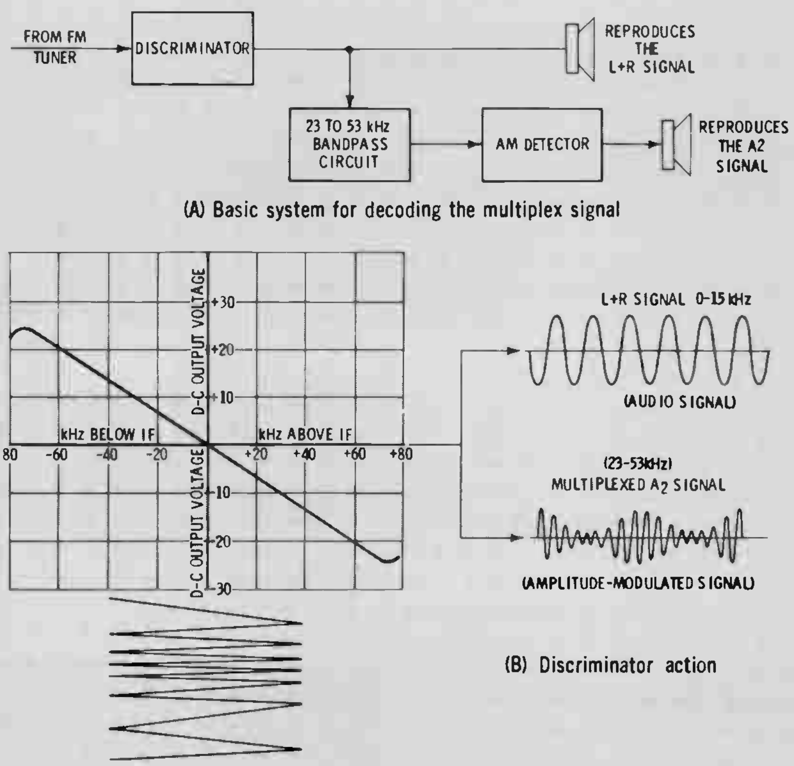


Fig. 6 Method for reproducing the multiplexed signal.

Fig. 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. Note that after this frequency spectrum is frequency-modulated on the RF carrier and then processed through the discriminator of an ordinary FM receiver, the frequency spectrum is recovered in its original form, as depicted in Fig. 5B. Of course, only the L+R signal can be made audible. That is, 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 can be recovered and made audible at the FM receiver. Fig. 6A shows a bandpass filter with response from 23 KHz, to 53 KHz, driven by the discriminator and followed by an AM detector and a speaker. It is evident that the bandpass filter picks out the AM signal, which is then demodulated by the AM detector. The demodulated (audio-frequency) wave envelope is then fed to a speaker which reproduces an audible signal. This is the A2 signal shown in Fig. 5. Note that the output from the discriminator in Fig. 5B is a mixture of the L+R and the A2-AM signal. Thus, the bandpass filter provides separation of the signals.

The foregoing example illustrates the basic principle of multiplex operation. However, the simple arrangement that we have analyzed must be elaborated somewhat to actually transmit R and L signals, and to obtain the R signal from one speaker and the L signal from the other. This requires the development of an L—R signal with a phase inverter at the transmitter, as shown in Fig. 7.

By inverting the polarity of the R signal and adding it to the L signal, we obtain an L—R signal. Thus we have both L+R and L—R signals available at this point. Note that if L+R is added to L—R, we can obtain 2L (the L signal with the R signal cancelled or separated). Again, if L—R is subtracted from L+R, we can obtain 2R (the R signal with the L signal cancelled or separated).

Addition or subtraction is accomplished in mixers, with associated in-

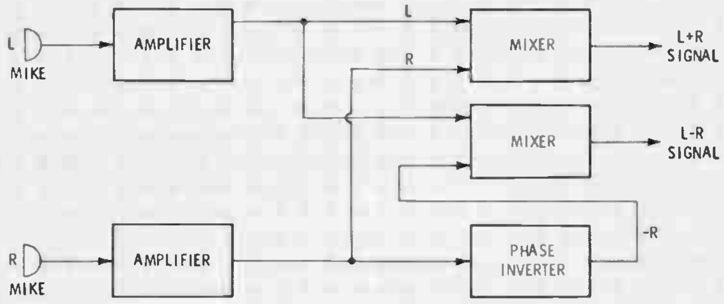


Fig. 7 Developing the L+R and L-R signals.

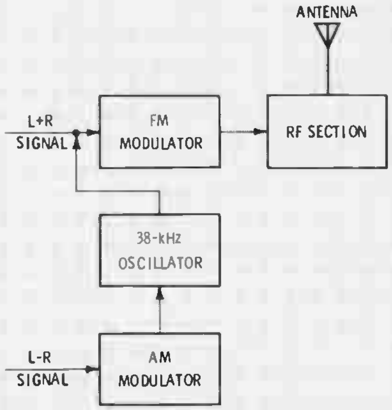


Fig. 8 Generation of the composite stereo signal at the transmitter.

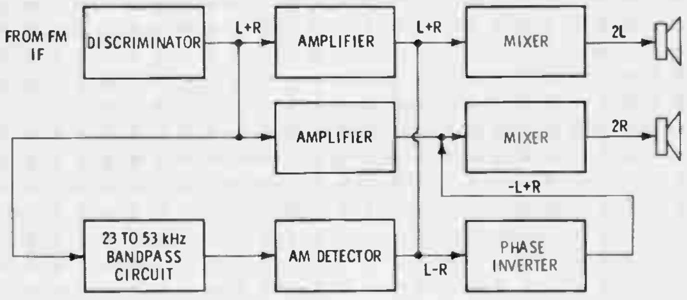


Fig. 9 Block diagram of multiplex adapter illustrates method used to obtain stereo reproduction.

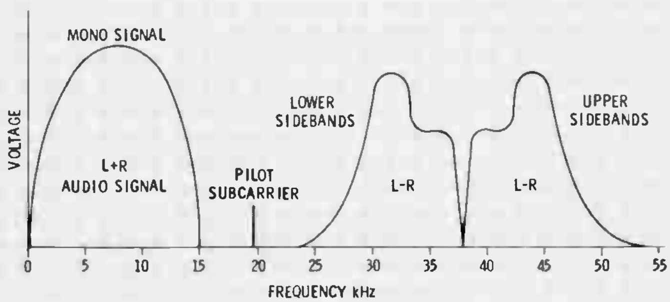


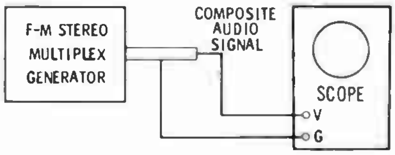
Fig. 10 Composite stereo signal showing 19-KHz pilot subcarrier in empty portion of spectrum.

verters. After the R and L 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.

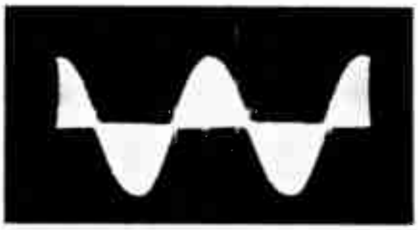
At the transmitter, the L+R and L—R signals are employed to frequency-modulate the RF carrier as shown in Fig. 8. The L—R signal is amplitude-modulated on a 38-KHz carrier, and the resulting signal is mixed with the L+R signal. In turn, the mixed signals frequency-modulate the RF carrier.

The frequency spectrum of the modulating signal is the same as shown in Fig. 5B, 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 discriminator.

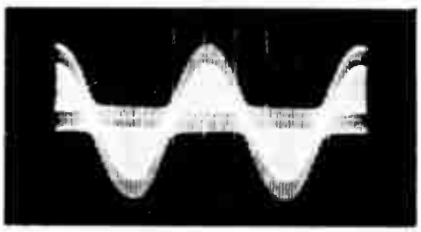
To obtain stereo reproduction, the L—R signal is separated by means of a 23- to 53-KHz bandpass filter, amplitude-demodulated, and further processed in a phase inverter and a pair of mixers, as shown in Fig. 9. The addition of L+R and L—R produces a 2L signal. Subtraction of L—R from L+R (the same as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



(A) Test setup



(B) Composite audio signal



(C) Composite audio with 19-KHz pilot subcarrier superimposed

Fig. 11 Output check of multiplex generator.

stereo reproduction is obtained from the speakers.

In theory, the 38-KHz carrier (technically termed the subcarrier) could be transmitted. However, in practice, it is preferable to suppress the subcarrier at the transmitter, to permit transmitting the upper and lower sidebands of the L—R signal at a higher level and, thereby, improve 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. Note that reinsertion must be accomplished not only at the exact frequency of 38 KHz, but also in correct phase, in order to obtain high-fidelity reproduction.

To permit precise reinsertion, a 19-KHz pilot subcarrier is transmitted for receiver synchronization. Fig. 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. Interference is thereby avoided and good separation is facilitated at the receiver. Note that the 38-KHz subcarrier is generated at the transmitter, but is trapped out before it is broadcast. A 19-KHz pilot-subcarrier oscillator is locked to the 38-KHz subcarrier signal, thereby maintaining precise phase and frequency relations.

At the receiver, a tuned circuit picks out the pilot subcarrier and feeds it to a doubler, thereby developing the 38-KHz subcarrier. This subcarrier is then reinserted by mixing it with the L—R sidebands. The end result is the same as if the 38-KHz subcarrier had been transmitted with the L—R sidebands.

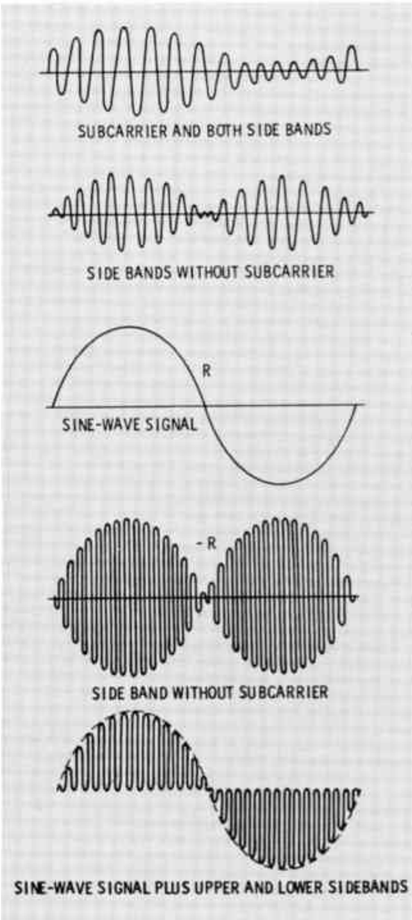


Fig. 12 Waveforms obtained at discriminator.

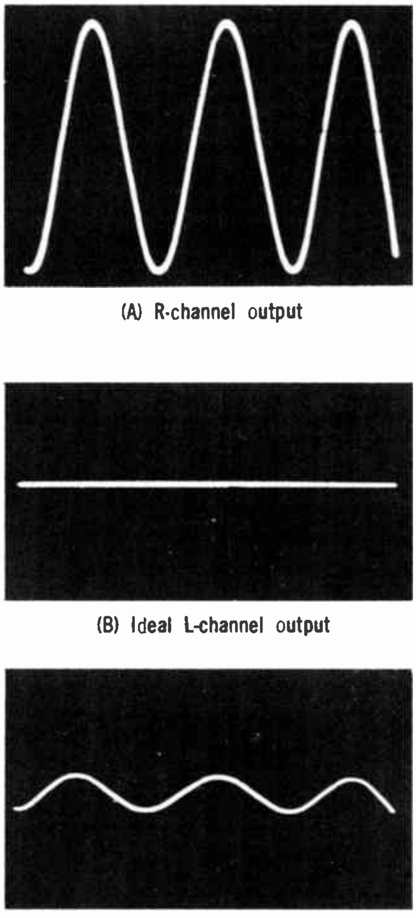


Fig. 14 Waveforms observed during separation test.

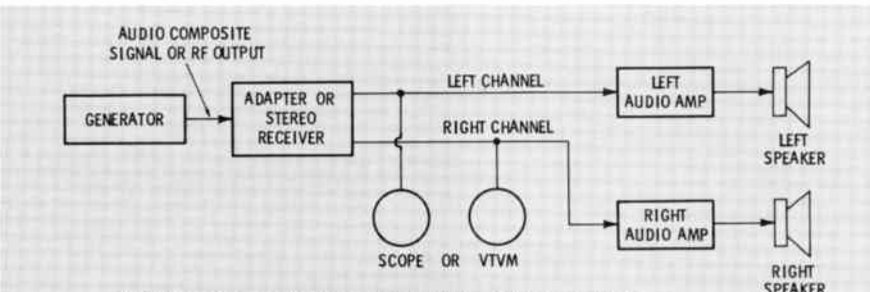


Fig. 13 Setup for testing separation of stereo system.

Dynamic Tests of Multiplex Adapters

A stereo-multiplex generator is required for dynamically testing of stereo-multiplex circuits. Such a generator supplies an L+R signal, an L—R signal, and a 19-KHz pilot subcarrier. A 1-KHz audio tone signal is employed to generate the L+R and L—R signals. Most generators provide a choice of modulated-RF (FM) signal output, or unmodulated signals. The unmodulated (composite audio) output is used when a multiplex adapter is tested directly.

A composite audio signal consisting of a multiplexed L+R sine wave and the 19-KHz pilot subcarrier appears on a scope screen as illustrated in Fig. 11. Technicians are often puzzled concerning the waveshape of the composite audio signal. Therefore, the following points should be observed with reference to Fig. 12:

1. The waveshape of the subcarrier and its sidebands are the same as an ordinary amplitude-modulated signal.
2. When the subcarrier is suppressed, the envelope of the waveform has twice its original frequency.
3. The generator L+R (or L—R) signal is produced by modulating a 38-KHz signal with a 1-KHz sine wave and then suppressing the 38-KHz signal.
4. The end result of the generator action is the waveform illustrated in Fig. 11B. Insertion of the 19-KHz pilot subcarrier results in the waveform illustrated in Fig. 11C.

The most basic test of a multiplex adapter is its ability to separate L+R and L—R signals. Fig. 13 shows the test setup. To test the multiplex adapter by itself, the audio composite signal is applied to the adapter input terminals. Either a scope or an AC VTVM can be used as the indicator. In theory, when an R-channel signal is applied, we would observe full output from the R terminal, and zero output from the L terminal, as shown in Fig. 14A and B. However, in practice, we usually observe more or less output from the L channel, as shown in Fig. 14C. Similarly, when an L-channel signal is applied, there is, ideally, zero output from the R terminal of the adapter.

A stereo multiplex generator provides a test signal with practically complete separation, unless the generator is defective. Our concern is with how many decibels of separation are provided by the multiplex adapter. A high-quality multiplex adapter will provide approximately 30 dB of separation in normal operation. It is generally considered that a separation of 10 dB is barely tolerable. 30 dB corresponds to a voltage ratio of more than 30 between L and R channels. On the other hand, 10 dB corresponds to a voltage ratio of approximately 3 between L and R channels.

Defects That Cause Poor Separation

The circuitry of a tube-type stereo adapter is shown in Fig. 15. 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.

If V2 is not defective, check the other tubes as a matter of course. Next to tubes, capacitors are the prime suspects. Therefore, a systematic approach requires checking C8, C11, C9, C10 and C15. If all capacitors check normal, the next step is to test (or replace) the diodes in the ring demodulator. Therefore, we turn our attention to X3, X4, X5 and X6. Note also that poor separation can be caused by a defective diode in the frequency-doubling section that changes the 19-KHz pilot subcarrier into a 38-KHz subcarrier. Thus, if the ring demodulator is not defective, check X1 and X2.

Resistors are not likely to cause trouble; however, off-value resistors are occasionally responsible for poor separation. R33 is vulnerable because a shorted tube can cause the resistor to overheat and change value. Similarly, R34 can change value due to overload. If this resistor increases substantially in value, V2B is biased incorrectly and does not operate as a linear mixer; in turn, separation is affected. The setting of R31 determines the amount of L—R signal that is applied to mixer V2B and, consequently, affects separation. If R31 is not defective, suspect R35 as the cause of poor separation.

Alignment of the multiplex adapter is checked last, because this is the least likely cause of poor separation (unless a do-it-yourselfer enters the picture). Alignment of the adapter in Fig. 15 is comparatively simple: T2 is peaked at 19 KHz, and T3 at 19KHz. T4 is peaked at 38 KHz.

To align T2, inject an unmodulated 19-KHz signal at the grid of the preceding tube, V1, and connect an AC VTVM at the plate of V4. Adjust the slug in T2 for maximum indication. The test signal can be obtained from a multiplex generator, or from an accurate, conventional signal generator.

To align T3, inject a 19-KHz signal at the grid of the preceding tube, V1, and connect an AC VTVM at the plate of V4. Adjust the slug in T2 for maximum indication. The test signal can be obtained from a multiplex generator, or from an accurate, conventional signal generator.

To align T3, inject a 19-KHz signal at the grid of V4 and connect an AC VTVM to the plate of V5A. Adjust the slug in T3 for maximum indication.

Finally, to align T4, inject a 38-KHz signal at the grid of V5A, and connect an AC VTVM at the junction of X5 and X6. Adjust the slug in T4 for maximum indication. If alignment corrects a poor-separation symptom, it is good practice to try to improve the separation by compromise adjustment of the slug in T4. If a slightly different setting of T4 is required on an L-signal test, compared with that of an R-signal test, use a compromise setting for optimum separation in both tests.

Conclusion

Poor stereo separation can be the result of poor alignment in the front end or in the IF amplifier. Poor discriminator alignment can also cause impaired separation. Therefore, if a multiplex adapter shows good separation on an audio composite signal, troubleshoot the preceding receiver sections to localize a symptom of poor system separation. The necessity for good alignmment in an FM stereo-multiplex system cannot be overemphasized.

After separation has been verified (or restored satisfactorily), check for good balance and negligible crosstalk. These tests are made on the audio amplifiers. Since this is a rather extensive topic, explanation of test procedures will be reserved for the next article. ▲

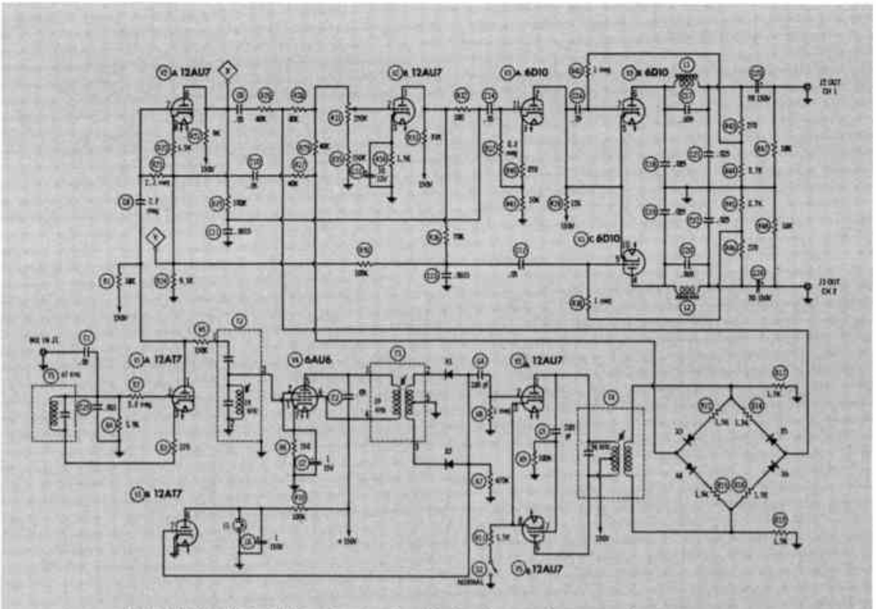


Fig. 15 Schematic of a typical tube-type stereo adapter.

Part 3

Balance, Crosstalk and Distortion

by Robert G. Middleton

An FM stereo multiplex system employs two audio-amplifier channels. High-fidelity amplifiers are provided in all of the better systems. Although there are no absolute standards for hi-fi reproduction, it is generally agreed that an amplifier should have essentially flat frequency response from 40 Hz to 15 KHz; flat response from 30 Hz to 20 KHz is preferred. "Essentially flat" means that the output should not vary more than ±1 dB; a variation of less than ±0.5 dB is preferred. Variations are measured with reference to the mid-band level. A flatness check is always made at maximum rated power output, and is usually rechecked at medium and low levels of power output.

DC Balance in Push-Pull Stages

Basic push-pull stage configurations are shown in Fig. 1. A fundamental requirement is good DC balance. This means equal emitter currents for Q1 and Q2 in Fig. 1A, or equal cathode currents for V3 and V4 in Fig. 1B. Various types of balance adjustments may be provided; for example, one or both of the emitter or cathode resistors might be adjustable. To facilitate checks of DC balance, jacks are often included in the circuits. A DC milliammeter is plugged into the jacks to measure the current values.

A cathode current of approximately 60 ma can be expected in a typical tube-type amplifier that is operating normally. The measurement should be made under no-signal, medium-signal, and full-rated output conditions; an audio oscillator is commonly used to provide a drive signal. Since objectionable

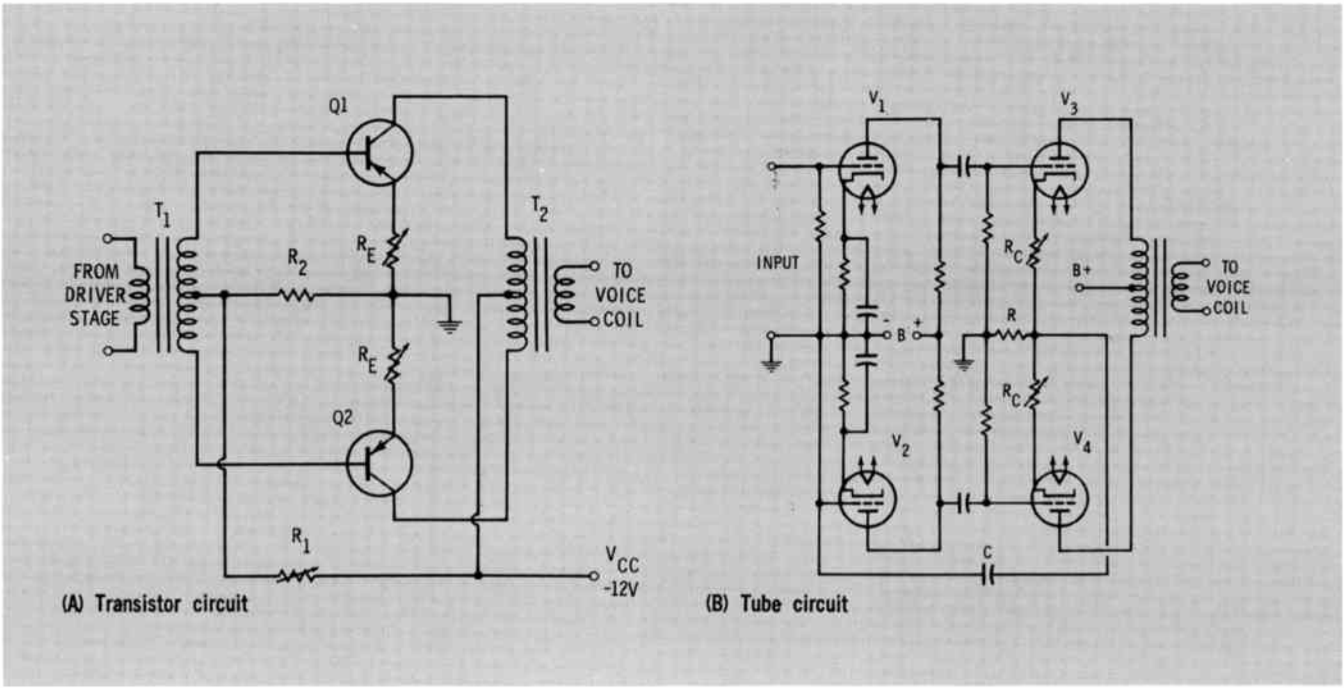


Fig. 1 Basic transistor—and tube-type push-pull circuits.

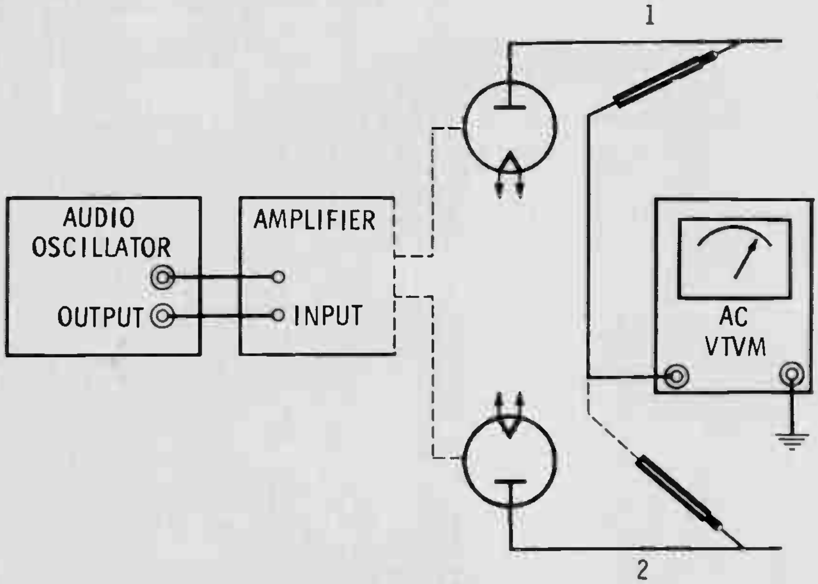


Fig. 2. Test setup for checking AC balance.

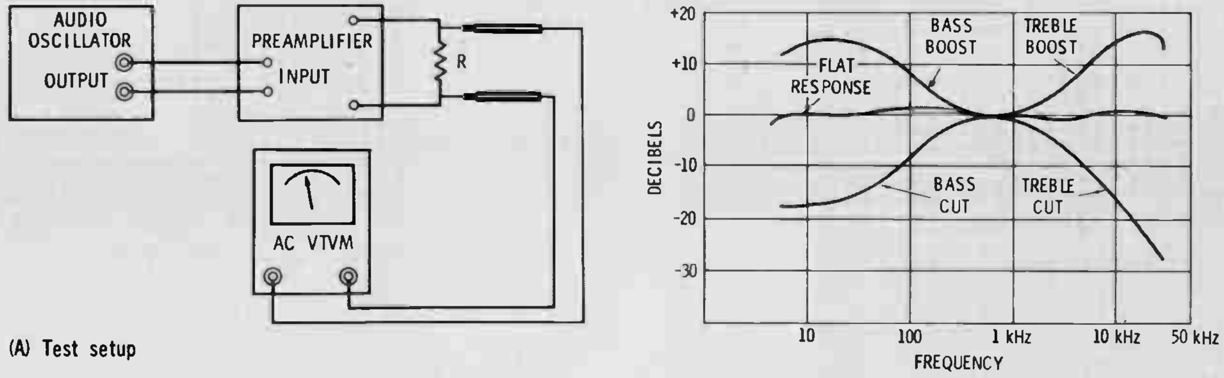
sound output results in a high-level test, the speaker is usually replaced by a power resistor of suitable value (resistance equal to the rated speaker impedance). If substantial DC imbalance is observed, first check the transistors or tubes.

The DC balance of a push-pull output stage should be adjustable to within 5%, and preferably within 2%. It is often necessary to choose a slight compromise adjustment to obtain best balance from the no-signal condition to the maximum rated output condition. When tubes or transistors are unbalanced, hum, harmonic distortion and intermodulation distortion tend to increase. If DC balance controls are not provided, it becomes necessary to select a matched pair of transistors or tubes. Distortion will be analyzed later; however, common sources of hum will be discussed at this point.

Sources of Hum Interference

Hum in the output of an audio amplifier is not always due to filter defects in the power supply, although a routine scope check of ripple voltage should be made. A common offender is heater-cathode leakage in tubes that have cathodes operating above ground. Tube selection is helpful, particularly in low-level input or driver stages. Some amplifiers have a hum-balance control in the heater circuit; if so, the control is adjusted for minimum hum output. Note also that poor ground connections or improperly ground components that cause circulating ground currents are sometimes responsible for objectionable hum output.

High-impedance, low-level grid circuits can pick up stray hum fields if defective shielding is present. To check this possibility, go through the amplifier stage-by-stage, and shunt each grid to ground in succession with a large bypass capacitor. This test will show whether a grid circuit is picking up stray field hum. Sometimes a low-level audio



(A) Test setup

(B) Flat response, with added curves showing the effect of bass and treble controls

Fig. 3 Test setup and desired results of frequency-response tests.

cable needs to be grounded at one end only; at other times, hum can be reduced by grounding the shield braid at both ends.

AC Balance in Push-Pull Stages

After the DC balance is verified (or adjusted), turn your attention to AC balance of the push-pull stage. This test requires an audio oscillator and an AC VTVM, as shown in Fig. 2. Adjust the audio oscillator to provide nearly maximum-rated power output from the amplifier. Note that power output is measured by checking the AC voltage across the output resistor (or voice coil) and applying the power law: $W = E^2/R$, where W is in watts, E is in rms volts, and R is the ohmic value of the output resistor or impedance value of the voice coil.

AC balance is usually checked at a standard test frequency; this is either 400 Hz, or 1 KHz, depending upon the particular standard that is preferred.

Apply the VTVM input lead, in turn, from plate 1 to ground and from plate 2 to ground, as illustrated in Fig. 4. The two readings should be nearly equal. If the readings are substantially different, it is possible that the push-pull tubes have unequal Gm values, or that the push-pull transistors have unequal beta values. Other possibilities are unequal plate voltages or unequal collector voltages. If the output stage has been cleared of suspicion, next measure the AC levels in the driver stage. For example, in Fig. 1B, defective tubes, faulty capacitors, or off-value resistors could cause AC imbalance. Note that although good AC balance can always be obtained at a mid-band frequency, such as 400 Hz or 1 KHz, more or less imbalance can be anticipated at very low and at very high audio frequencies. At the extreme ends of the frequency range, circuit reactances cause unequal drive signals to the grids of the push-pull tubes, or bases of the push-pull transistors.

Measurement of Frequency Response

We are concerned next with frequency response measurements. The frequency response of a preamplifier is checked as depicted in Fig. 3. Note that the bass and treble controls must be set for flat response. It also is necessary to use an audio

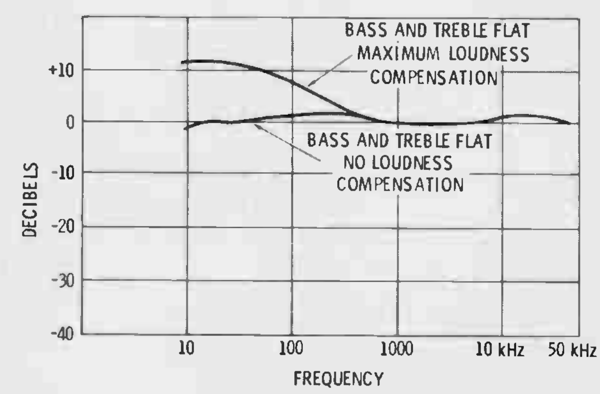
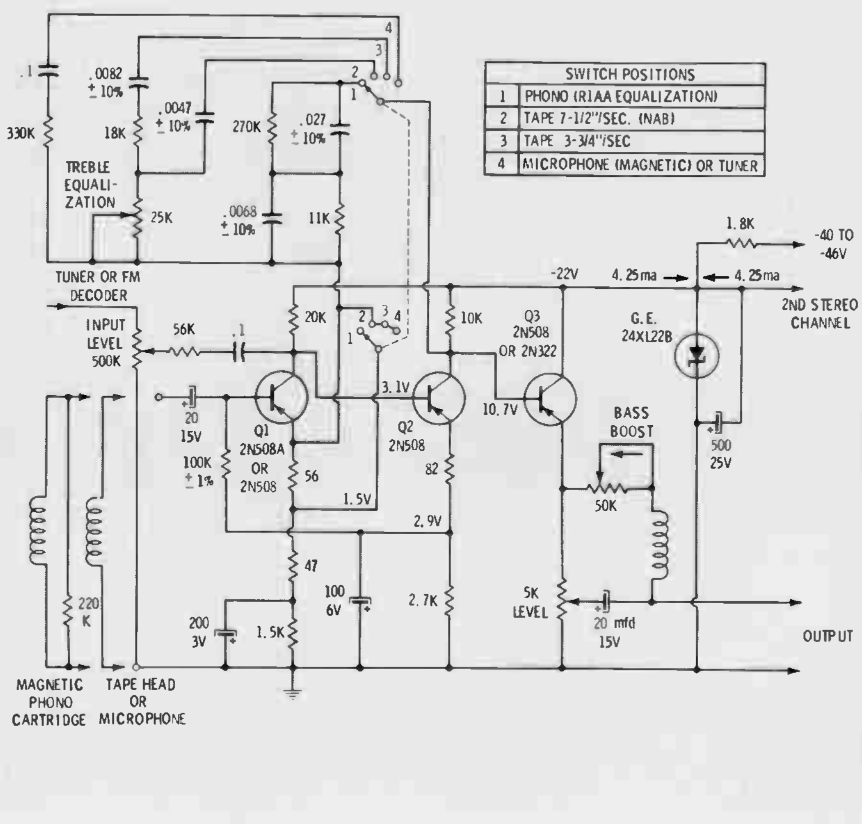


Fig. 4 Typical frequency response of a preamplifier, with tone controls set for flat response and the volume control at maximum.

Fig. 5 Preamplifier response obtained with presence control on.



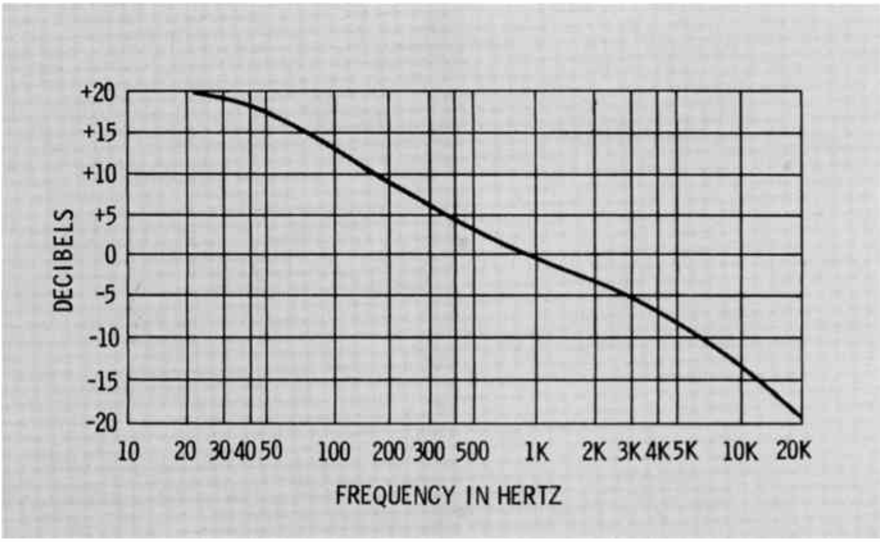


Fig. 7 Record Industry Association of America (RIAA) equalization curve for play-back of records.

oscillator that has a very flat output; otherwise, a VTVM must be connected across the audio-oscillator output terminals to monitor the output level as the operating frequency is changed. If the amplifier has a loudness control, this control must be set for zero loudness compensation. Otherwise, a low-frequency rise will be observed, as shown in Fig. 4.

Some hi-fi amplifiers have a presence control. This control must be turned off during a frequency-response check, otherwise, a mid-band frequency rise will be observed, as illustrated in Fig. 5. Be sure that you do not apply the

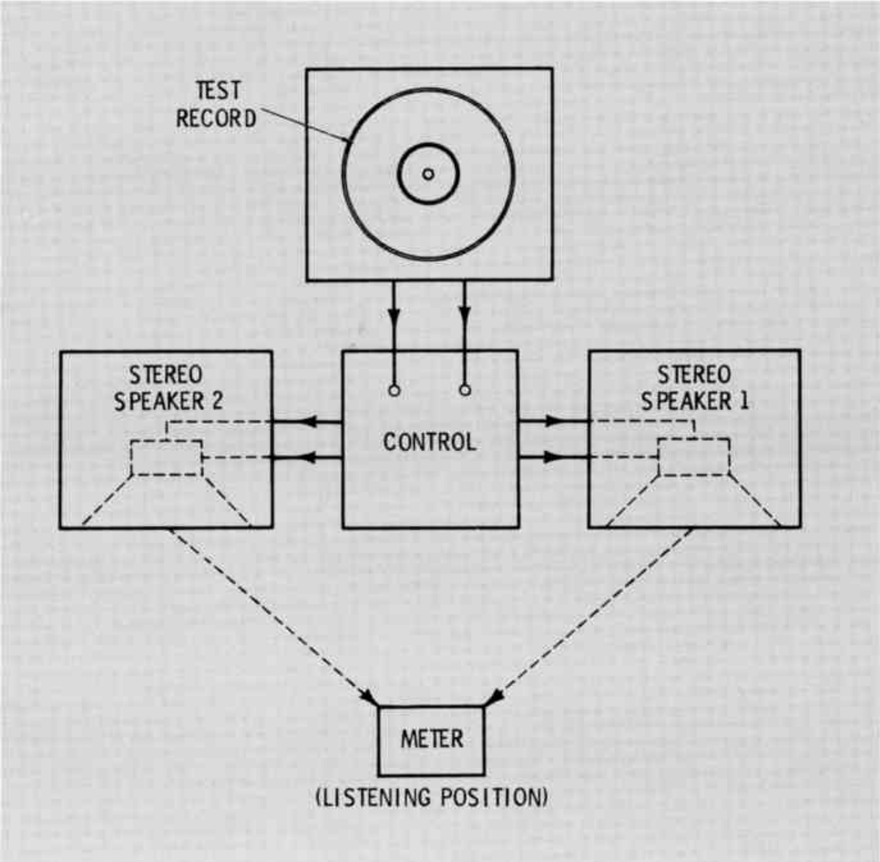


Fig. 8 Test setup for checking the balance of a stereo system.

audio-oscillator signal through the scratch filter. If this error is made, the amplifier will appear to have poor high-frequency response. The same precaution applies to a rumble filter. If the audio-oscillator signal is passed through the rumble filter, the amplifier will appear to have poor low-frequency response.

A typical solid-state preamplifier circuit is shown in Fig. 6. In the RIAA equalization switch position, high-frequency attention is provided, as depicted in Fig. 7.

When troubleshooting poor frequency response, first suspect defective tubes or transistors. The next most likely source of trouble is a defective capacitor. In a transistor preamplifier, capacitors are more likely to fail than transistors because of the low power level present in the circuit. On the other hand, transistors are prime suspects in output amplifiers because the higher power requirement operates the transistors at or near maximum design ratings.

Balance Check of a Stereo System

It is necessary that the gain of each channel in a stereo system be the same; otherwise, the full stereo effect is not obtained. To check for proper balance and precise adjustment of the stereo balance control, a stereo balance meter and test record may be utilized as shown in Fig. 8. A stereo balance meter is basically the same as a noise-level meter. The meter is placed in the position normally occupied by a listener, and the 400-Hz or 1-KHz tone from the test record is reproduced at normal level. Each channel is operated in turn, and the stereo-balance control is adjusted to obtain the same reading from each channel.

In some cases, a slight difference in setting of the stereo-balance control may be noted at low and high audio frequencies, due to system tolerances. In such a case, make a compromise setting based on tests at 100, 1000, and 8000 Hz. The technician should not confuse stereo balance with DC balance or AC balance. Stereo balance concerns equality of gain in the two audio channels, whereas DC and AC balance refer to conditions in the push-pull output stage of an individual channel. We normally check stereo balance after the DC and AC balance conditions are satisfied.

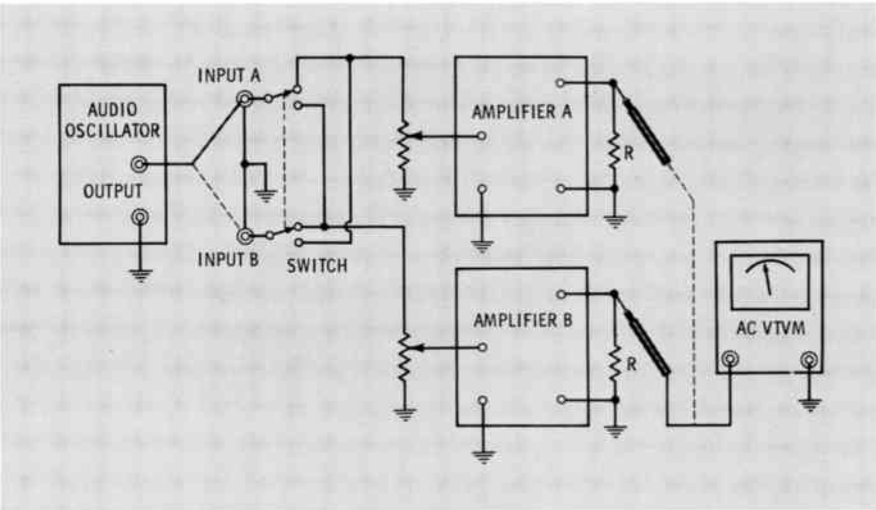


Fig. 9 Checking crosstalk between stereo amplifiers.

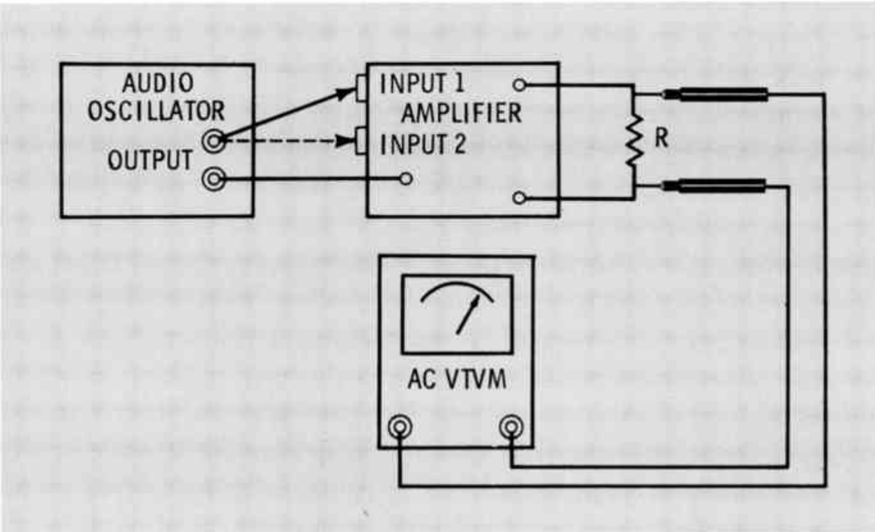


Fig. 10 Test setup for checking crosstalk between inputs of an amplifier.

Crosstalk

Crosstalk pertains to a stray coupling of signals between the inputs of an amplifier. For example, let us consider a check for crosstalk between stereo amplifiers, as shown in Fig. 9. With the audio oscillator connected to input A, measure the relative output from amplifiers A and B. Then, with the audio oscillator connected to input B, again measure the relative outputs from each amplifier. The crosstalk figure is the number of dB measured in these tests. Note that the crosstalk figure tends to change with the test frequency; therefore, we state the frequency of maximum crosstalk in evaluating a stereo amplifier.

Crosstalk can be caused by any form of stray coupling, such as capacitance between leads. However, an objectionable crosstalk figure is usually due to defective decoupling capacitors that result in common impedances in the power-supply leads. Crosstalk shows up most prominently when operating at maximum rated power output. A scope check across suspected capacitors is the best method of localizing the trouble. The output filter capacitor is also a ready suspect; if this capacitor is defective, the ripple amplitude will also be higher than normal.

Next, let us consider crosstalk between inputs of the same amplifier, as depicted in Fig. 10. The audio oscillator is connected to input 1, and an input level is applied which produces maximum rated power output. The test may be made

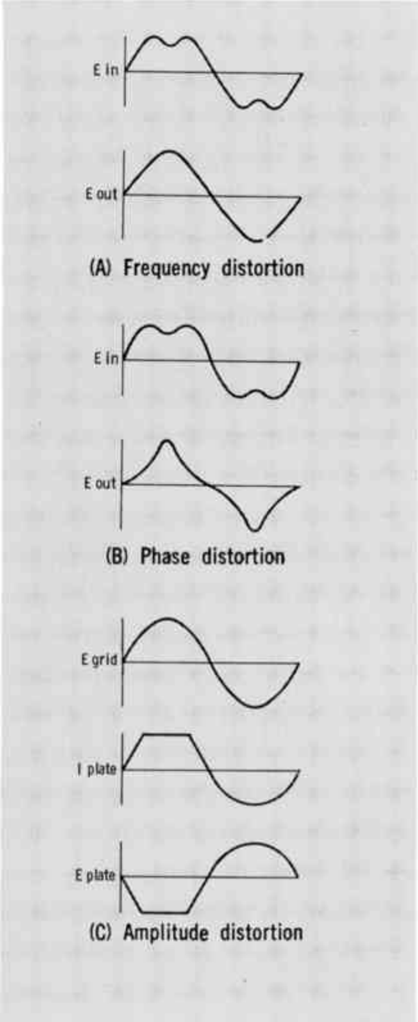


Fig. 11 Three basic types of amplifier distortion.

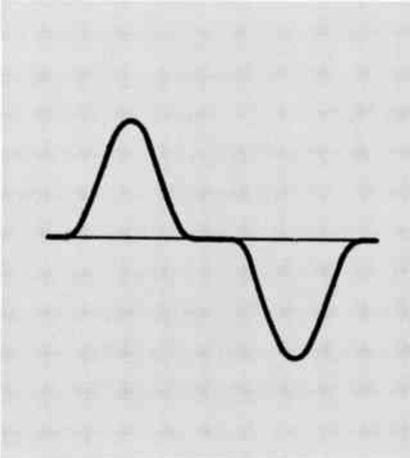


Fig. 12 An example of crossover distortion.

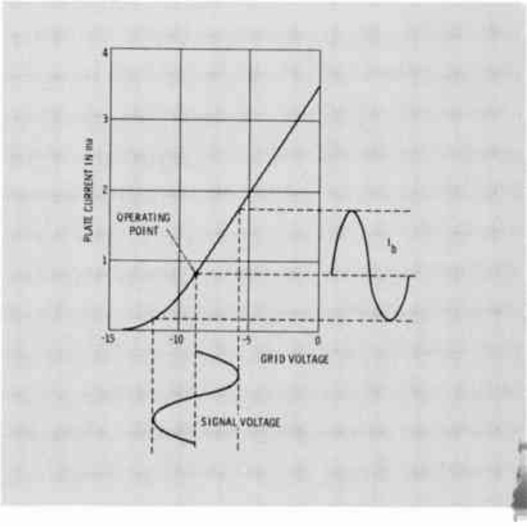


Fig. 13 Compression of the negative half cycle.

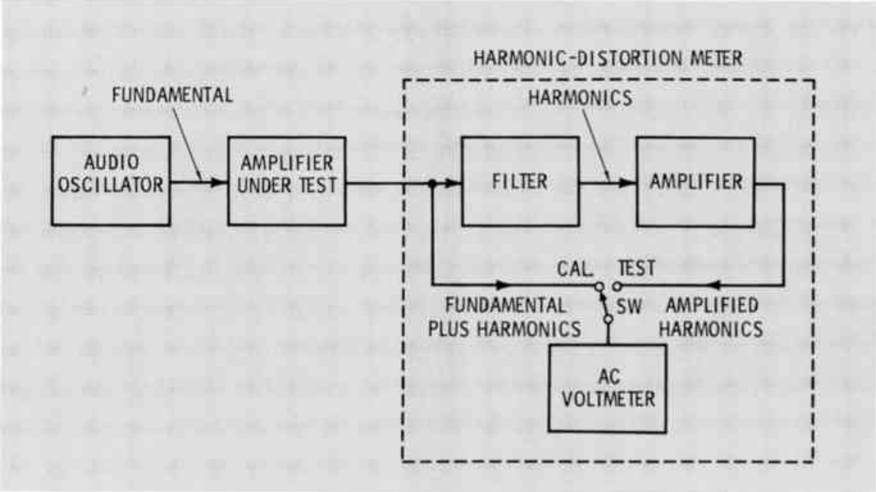


Fig. 14 Test setup for measuring harmonic distortion.

at 1 KHz. Next, the amplifier is switched to input 2, and the meter reading observed (if any). A similar test is made with the audio oscillator driving input 2. Objectionable crosstalk between inputs can be caused by poor grounds, disturbed lead dress, or similar defects that produce stray coupling between the input circuits.

Harmonic Distortion

Three basic types of distortion are shown in Fig. 11. First, consider frequency distortion. The input waveform, "Ein", has a low-frequency component and a high-frequency component. However, because of poor high-frequency response, the high-frequency component does not appear in the output waveform, "Eout."

In Fig. 11B frequency distortion does not occur; however, the high-frequency component has been shifted in phase. This type of distortion is called phase distortion; it is of minor importance in audio systems, although phase distortion is minimized in better equipment.

We are chiefly concerned with amplitude distortion, a basic form of which is illustrated in Fig. 11C. This type of amplitude distortion is termed clipping. Clipping may occur in the center of a waveform, as well as on a peak of the waveform. For example, Fig. 12 shows a waveform with crossover distortion. It is the result of overbias in a push-pull stage that clips the initial portion of the drive waveform. Another type of nonlinear distortion, called compression, is depicted in Fig. 13. Compression and clipping are merely various degrees of amplitude nonlinearity in amplifier operation.

The distinctive characteristic of any type of amplitude nonlinearity is the production of harmonics, which appear in the amplifier output when a single-frequency drive waveform is applied to the amplifier input. Therefore, this general class of distortion is called harmonic distortion. Although there are no absolute standards, it is generally agreed that a hi-fi amplifier should have less than 1% harmonic distortion, and preferably as little as 0.1% harmonic distortion. If an amplifier has 1% harmonic distortion, this means that the algebraic sum of the harmonic voltages is equal to 1% of the fundamental output voltage.

It is a basic law of electronics that harmonics are always integral multiples of the fundamental frequency. For example, the distorted waveforms shown in Figs. 11C, 12 and 13 will have harmonic frequencies which are 2 times, 3 times, 4 times, etc., higher than the fundamental frequency. Harmonic distortion is measured with a harmonic distortion meter; a block diagram of this instrument is shown in Fig. 14. The amplifier is driven by a sine waveform with a typical frequency of 1 KHz. If the amplifier develops harmonic distortion, frequencies of 1 KHz, 3 KHz, etc., will be fed to the harmonic distortion meter.

If the audio oscillator in Fig. 14 operates at 1 KHz, the filter in the harmonic distortion meter is tuned to eliminate frequencies below 2 KHz, but passes frequencies of 2 KHz, 3 KHz, etc. That is, a high-pass filter is employed. The harmonic voltages are stepped up through a calibrated meter amplifier, and the percentage of harmonic distortion is indicated by an AC voltmeter. The voltmeter scale is calibrated in percentage values in order to be direct-reading. It is obvious that the test equipment must have lower distortion than the amplifier under test. That is, the audio oscillator must have very low distortion, and the meter amplifier also must have very high fidelity.

Amplifier distortion is usually measured at maximum rated power output. Accordingly, a test setup includes an AC VTVM, as shown in Fig. 14. The output from the audio oscillator is advanced until the power in R is equal to the maximum rated value for the amplifier. To repeat an important point, the power in the resistor is calculated by the formula $W = E^2/R$, where W is in watts, E is in rms volts, and R is in ohms. The percentage of harmonic distortion is then indicated by the harmonic distortion meter for a condition of maximum power output.

When an objectionably high value of harmonic distortion is measured, look for non-linearity in the amplifier. The trouble, most likely, is caused by defective tubes or transistors. Nonlinearity is seldom caused directly by capacitor or resistor defects, since these are basically linear components. However, defective capacitors or resistors indirectly can cause nonlinear operation. For example, a leaky capacitor can shift the normal bias on a tube or transistor, causing operation in a nonlinear region. Similarly, an off-value resistor can cause bias shift. Although the incidence is rare, shorted turns in an audio transformer also can cause nonlinear operation.

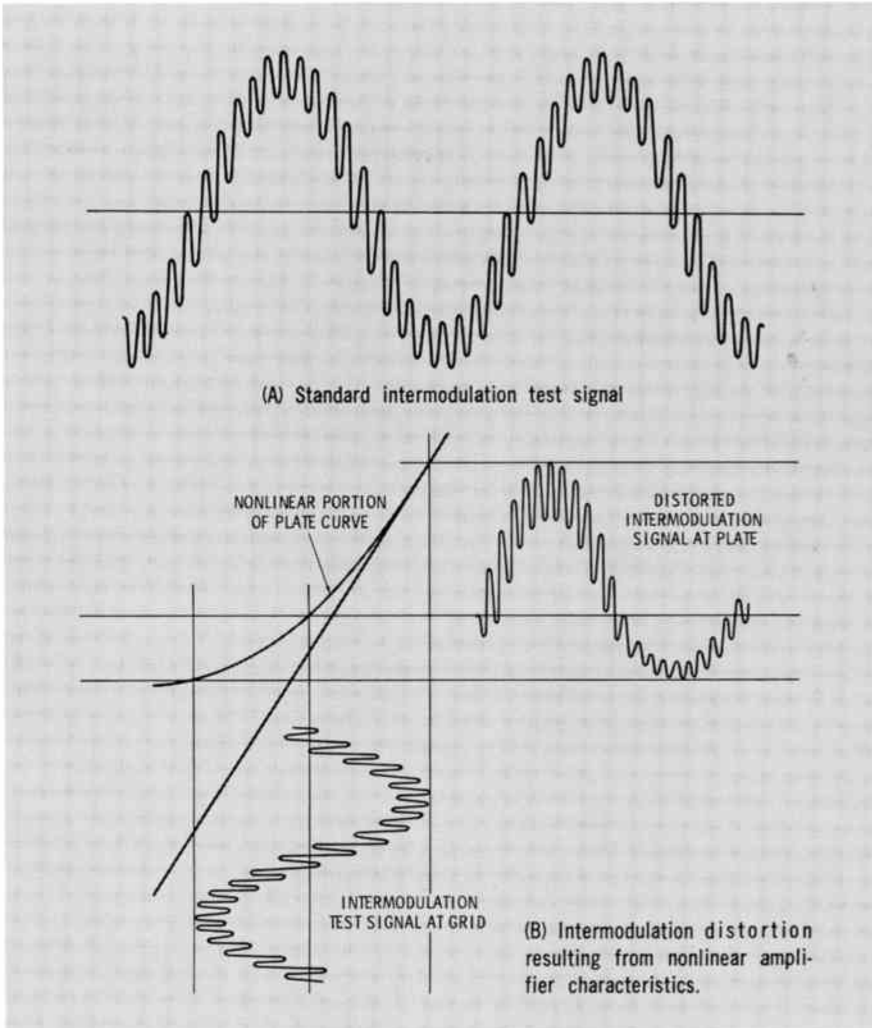


Fig. 15 Test signal and results of intermodulation distortion test.

Intermodulation Distortion

Intermodulation distortion is another characteristic of amplitude nonlinearity. It is basically a two-tone test, as depicted in Fig. 15. The two test frequencies are applied simultaneously, with typical frequencies of 60 Hz and 6 KHz. When amplitude nonlinearity is present in the amplifier under test, intermodulation of the test frequencies will result. In turn, sideband frequencies appear in the amplifier output waveform. These are sum and difference frequencies; thus, the sideband frequencies in the foregoing example are 5940 and 6060 Hz.

Fig. 16 shows the basic design of an intermodulation distortion analyzer. The high-pass filter picks out the high-frequency components in the 6-KHz region, and feeds this modulated waveform to a rectifier. In turn, the rectifier applies the demodulated waveform to a low-pass filter. This low-pass filter permits the frequency of the modulation envelope to energize an AC meter that is calibrated in percentage of intermodulation distortion. The frequency characteristics of filters utilized in an intermodulation distortion analyzer are shown in Fig. 17.

The two-tone signal used in an intermodulation distortion test is usually obtained from audio oscillators which are built into the instrument. Thus, a test setup as shown in Fig. 18 is employed. It is customary to drive the amplifier to maximum rated power output in the test, because intermodulation distortion tends to increase at high power levels.

Fig. 19 shows the results of both intermodulation and harmonic distortion tests on a hi-fi amplifier. We will find that there is no simple correlation between harmonic dis-

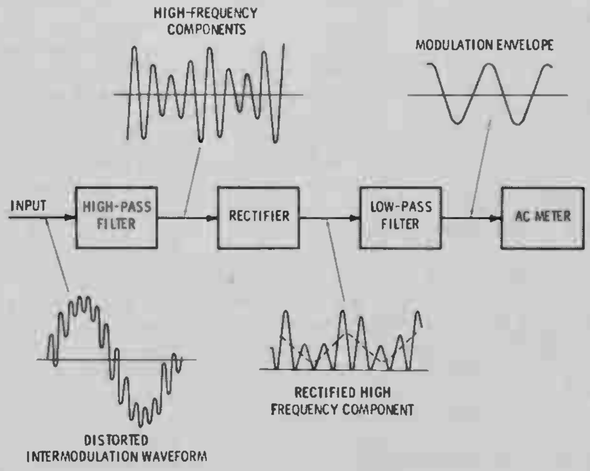


Fig. 16 Basic configuration of an intermodulation distortion analyzer.

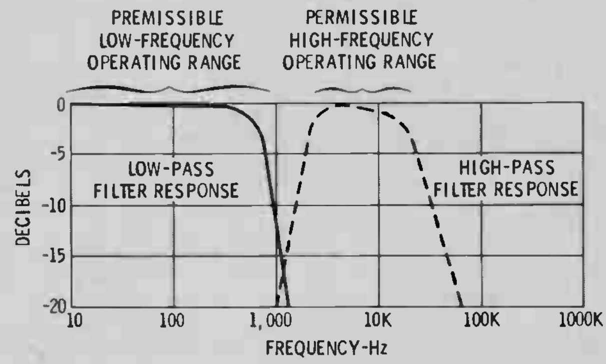


Fig. 17 Characteristics of the filters in a typical intermodulation distortion analyzer.

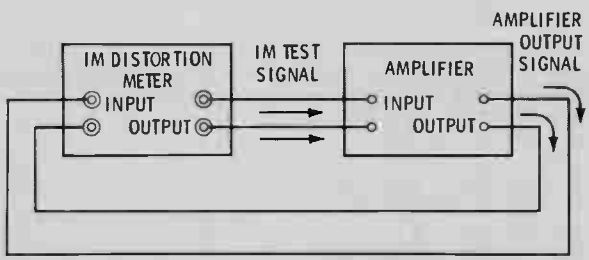


Fig. 18 Test setup for measurement of intermodulation distortion.

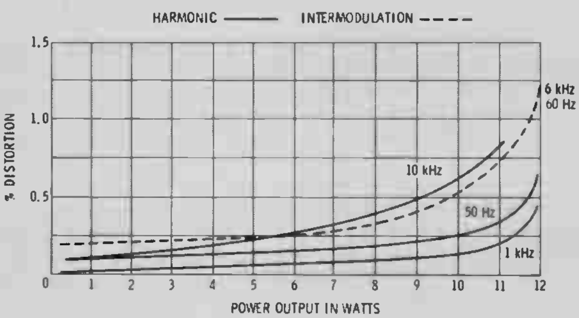


Fig. 19 Harmonic and intermodulation distortion percentages for a typical high-fidelity amplifier.

tortion and intermodulation distortion values. Therefore, it is good practice to check both values.

When an amplifier exhibits objectionable intermodulation distortion, the probable causes are the same as for harmonic distortion. To localize the trouble to a particular stage, it is advisable to leave the distortion meter connected at the output of the amplifier, and to inject the test signal step-by-step, working back from the output stage. That is, the test signal is injected through a blocking capacitor progressively at the bases of the transistors, or at the grids of the tubes. Signal tracing with a scope is not practical because it is very difficult to observe small percentages of distortion in a scope waveform.

Coming Next

Thus far, we have considered individual units of the FM stereo multiplex system. We are now in a position to consider system response and system tests. This is an important topic because it is quite difficult to predict what the system response will be on the basis of unit or section tests. System tests will be explained in the next article. We will also consider variations in design of commercial equipment, and step-by-step system alignment procedures. ▲

Overall System Evaluation

Overall system evaluation using alignment procedures and harmonic-distortion and intermodulation tests.

Operational checks of an FM stereo-multiplex receiver system that doesn't sound right may provide clues that indicate a step-by-step system alignment is needed. A preliminary test comparing the sound reproduction of an FM stereo broadcast with that of a stereo record may indicate much better sound quality with the record. This symptom points to a malfunctioning of the FM tuner and/or multiplex section, and a step-by-step alignment job may be in order. More detailed data can be obtained by making system harmonic-distortion or system intermodulation-distortion measurements.

System Distortion Measurements

The test setup for making a system harmonic-distortion measurement is shown in Fig. 1. The receiver is driven by a modulated RF signal from an FM stereo multiplex generator. Suitable values of power resistors are connected across the audio-amplifier output terminals. A harmonic-distortion meter is connected across each of the load resistors. The FM tuner must be set precisely to the RF frequency of the generator (typically 100 MHz). Sufficient generator output is used to insure that the limiters in the receiver are saturated. When testing the output from the R-channel audio amplifier, the generator must be set to "R-channel output." When testing the output from the L-channel amplifier, the generator must be set to "L-channel output."

Most technicians are familiar with the operation of an FM stereo-multiplex generator. However, it seems that few technicians are familiar with the harmonic distortion meter. Therefore, let us spend a few minutes reviewing the operation of the instrument.

Fig. 2 shows a schematic diagram for a typical instrument. The meter indicates the "remains" of a signal under test as a percentage of the signal under test after the fundamental frequency is eliminated. The "indicated remains" include all

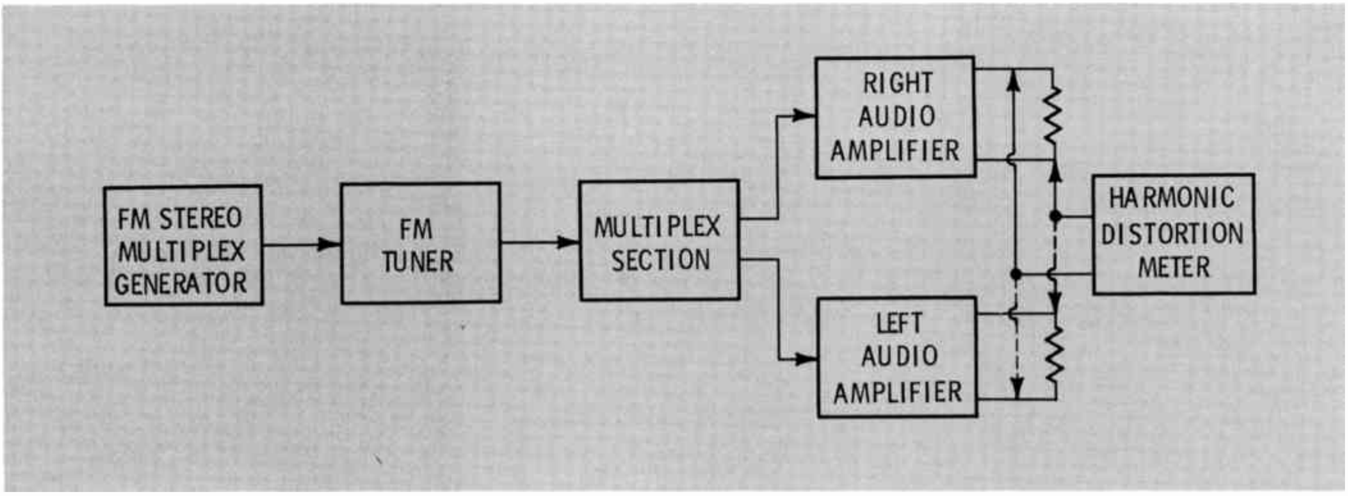


Fig. 1 Test setup for performing harmonic distortion measurement.

frequencies in the audio range (hum, harmonics and noise). Normally the "remains" are predominantly the harmonics produced by the receiver system.

The circuit depicted in Fig. 2 may be considered to have three parts: (1) the fundamental suppression circuit; (2) the voltmeter circuit; and, (3) the power supply. Note that the fundamental suppression circuit consists of a triode voltage amplifier (one-half of a 12AX7) driving a phase splitter (12BY7). This phase splitter feeds a Wien-bridge null network that suppresses the fundamental test frequency. In turn, the signal "remains" are stepped up by a voltage amplifier (5879), which drives a cathode follower (one-half of a 12AX7).

It is important to note that the Wien bridge is tunable, and has a rejection characteristic as shown in Fig. 3. When the tuning control of the harmonic-distortion meter is rotated, the null point moves along the frequency axis in Fig. 3. It is also important to note that the Wien bridge can be switched in or out of the meter circuit. When the function switch of the instrument is turned to the "Set Level" position, the Wien bridge is out of the meter circuit. This setting permits us to adjust the receiver output for a reference value of full-scale indication on the meter.

After the full-scale reference level has been set, adjust the frequency switch on the harmonic-distortion meter to a suitable range and carefully adjust the tuning control for minimum indication on the meter. For example, suppose that the generator provides a 1-KHz audio signal. In such a case, set the frequency switch on the harmonic-distortion meter to the 200 to 2000-KHz range, and adjust the tuning control precisely to 1 KHz. At this point, the Wien bridge is tuned exactly to reject the 1-KHz fundamental, and the meter has a minimum reading.

As the minimum meter indication is approached, the pointer usually falls so near to zero that it is difficult to determine the exact minimum adjustment. This can be overcome by employing a higher indication sensitivity (see Fig. 4). If the setting of the sensitivity switch is dropped down from the original 100% setting to 10%, the meter deflection is increased 10 times. Or, if the sensitivity setting is dropped down from the original 100% setting to 1%, the meter deflection is increased 100 times. If a normally-operating hi-fi system is being tested, it is advisable to operate on the 1% setting to obtain ample pointer deflection on the meter.

After the minimum setting has been made, we are ready to read the percentage of harmonic distortion. This reading is made in much the same way as we read a VTVM scale. The only difference is that the meter scale is calibrated in percentage instead of volts. We note the setting of the sensitivity switch and the scale reading, and, in turn, we know the percentage of harmonic distortion. For example, suppose that the setting of the sensitivity switch is 1%. Then, the full-scale meter indication is 1%. If the reading happens to be at half of full-scale, the percentage of harmonic distortion is 0.5%.

Percentage of harmonic distortion is usually measured with full rated power output from the receiver. This is the most demanding test because harmonic distortion tends to de-

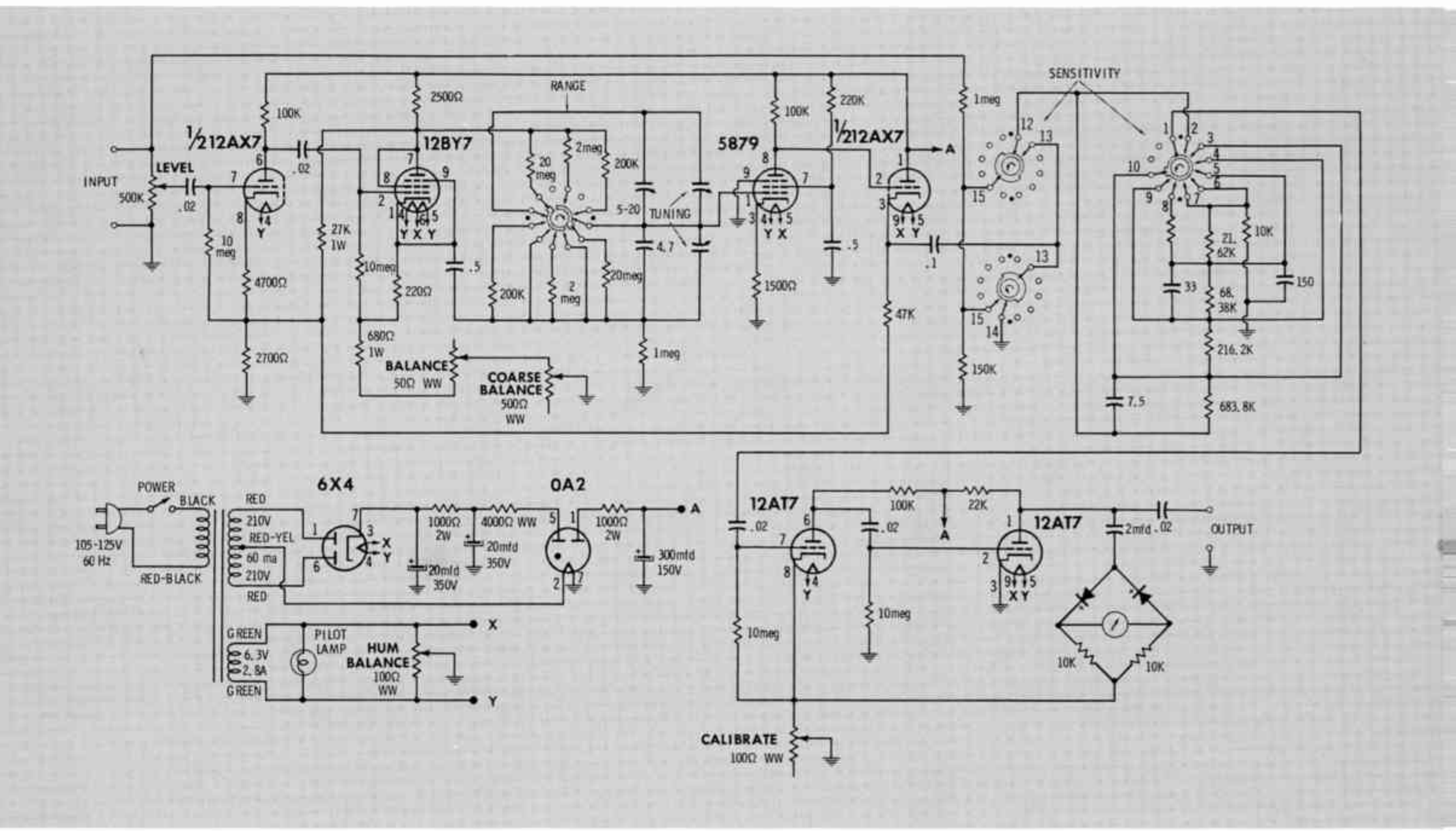


Fig. 2 Circuit diagram of a typical harmonic distortion analyzer.

crease at lower power levels. Let us observe typical test results for a good hi-fi system. Fig. 5 shows the results of a harmonic-distortion test at various audio frequencies for full rated power output, half-power output, and low-power output level. At full power output, the harmonic distortion is less than 0.1% at 400 Hz, and slightly greater than 0.1% at 1 KHz. At 10 KHz, the harmonic distortion is about 0.55%. At reduced power output the harmonic distortion is considerably less.

Localization of Trouble

To localize the source of trouble, make a harmonic distortion test at the output of the R audio amplifier, and at the output of the L audio amplifier (Fig. 1). If the percentage of harmonic distortion is low in the R-amplifier test but is high in the L-amplifier test, the trouble will be found in the L amplifier. However, if the percentage of distortion is virtually the same at the output of each amplifier, the trouble is probably in the multiplex section or the tuner section. To definitely clear an audio amplifier from suspicion, make a distortion test of the amplifier by itself, as shown in Fig. 6. The AC VTVM is used to adjust the drive for rated maximum power output, and the harmonic-distortion meter is operated as explained previously.

If the audio amplifiers are not at fault, turn your attention to the multiplex section. Percentage of distortion is checked with the test setup shown in Fig. 7. Note that composite audio output from the generator (not modulated RF output) is used. The generator is set for R-channel output when the harmonic distortion meter is connected to the R output terminals of the adapter, and vice versa. Normally, an adapter has very low harmonic distortion. However, defective tubes or transistors with leaky capacitors that cause incorrect biasing of tubes or transistors can also be responsible.

Poor alignment of the multiplex section can also cause objectionable harmonic distortion, particularly when a synchronized subcarrier oscillator is marginally locked. Therefore, let us consider step-by-step alignment procedures.

In the first analysis, there are two basic types of multiplex configurations. These are called matrixing and time-division. Operation of the

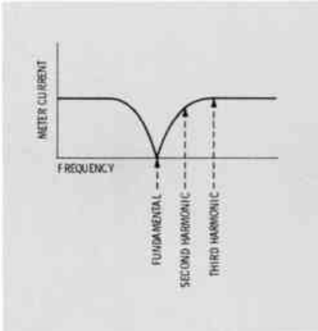


Fig. 3 Frequency-rejection curve for a typical harmonic distortion analyzer.

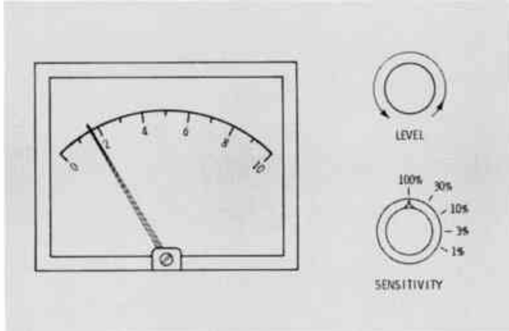


Fig. 4 Calibrated meter and sensitivity and level controls of a harmonic distortion analyzer.

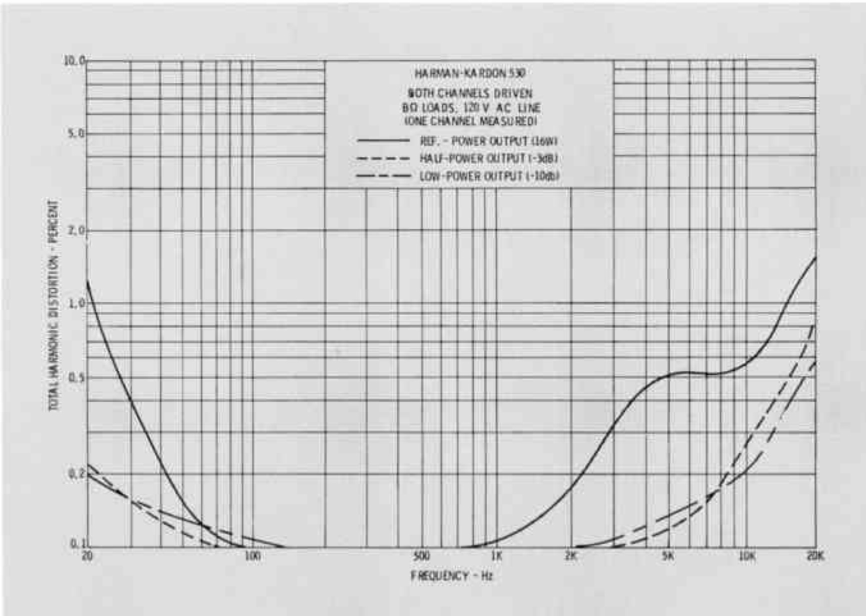


Fig. 5 Percentage of harmonic distortion at various audio frequencies in a good hi-fi system. Courtesy—Harman-Kardon.

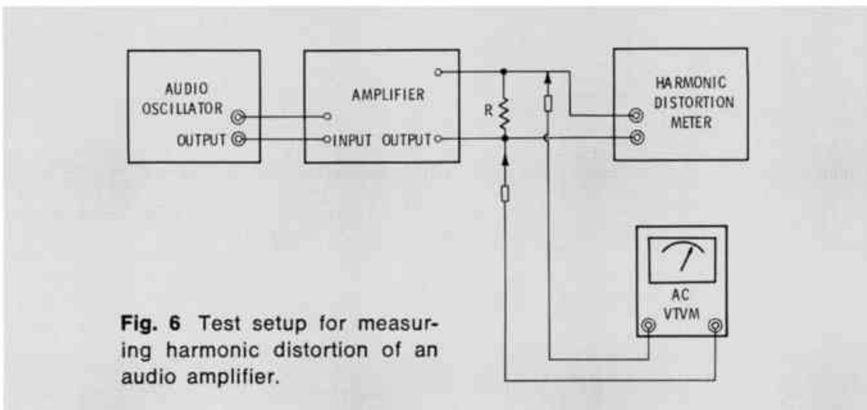


Fig. 6 Test setup for measuring harmonic distortion of an audio amplifier.

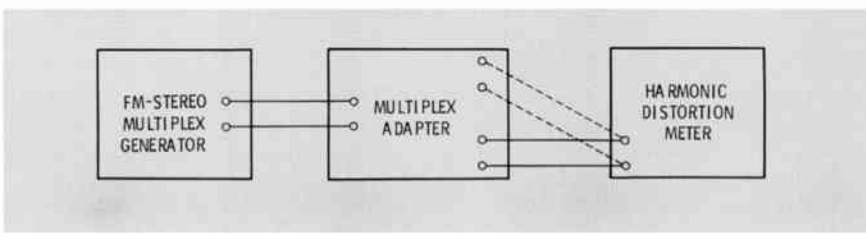


Fig. 7 Harmonic-distortion test setup for a multiplex adapter.

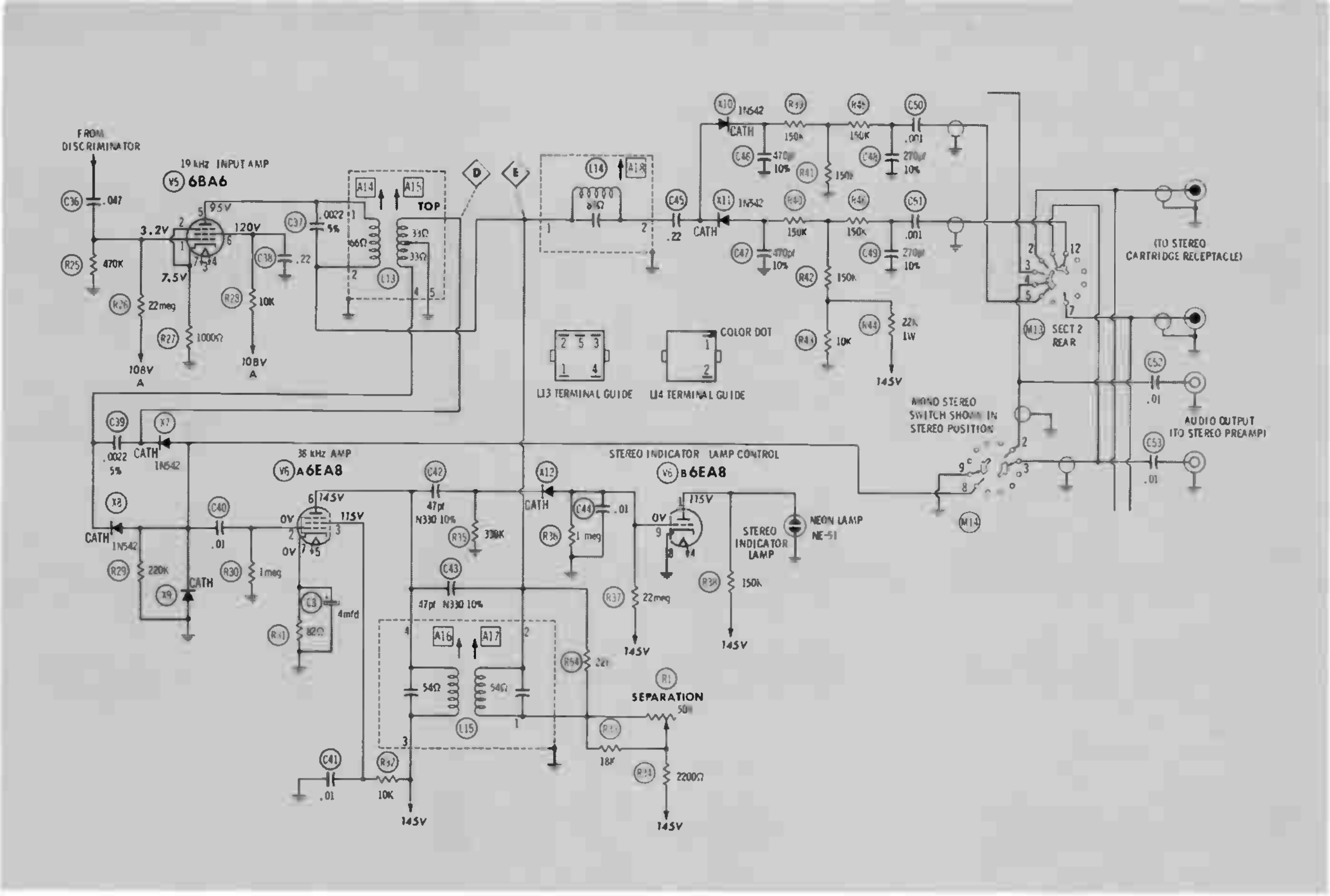


Fig. 8 Multiplex system using an envelope detector circuit.

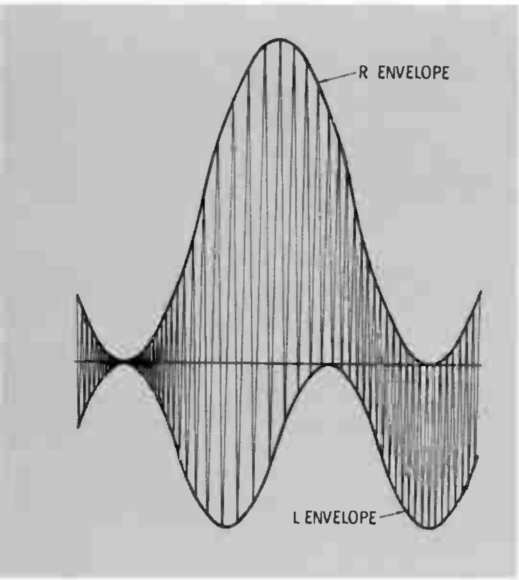


Fig. 9 Undemodulated multiplex waveform with L and R envelopes indicated.

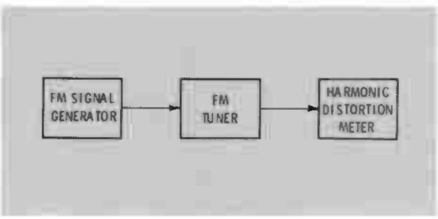


Fig. 10 Test setup for measuring the harmonic distortion of an FM tuner.

TABLE 1

FM stereo multiplex alignment using FM stereo signal generator (±.0001% accuracy)

High side to point C, low side to ground.

	Generator Frequency	Indicator	Adjust	Remarks
1	19 KHz	Vert. Amp. of scope thru 47K to point D, low side to ground.	A14, A15	Adjust for maximum. Set scope to lock in 2 cycles of 19 KHz waveform.
2	19 KHz	Vert. Amp. thru 47K to point E. low side to ground.	A16, A17	Adjust for maximum 4-cycle waveform.
3	67 KHz	Vert. Amp. thru 47K to point E, low side to ground.	A18	Use Audio Oscillator if necessary. Adjust for MINIMUM

To align multiplex section using an air signal, first make sure FM section is properly aligned. Tune in a strong FM stereo signal. Follow steps 1 thru 3 above except in step 3 adjust to eliminate whistle or interference.

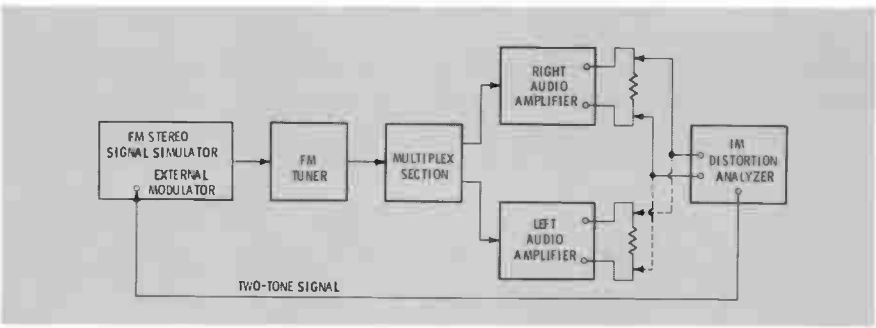


Fig. 11 Test setup for measuring the intermodulation distortion in an FM multiplex stereo system.

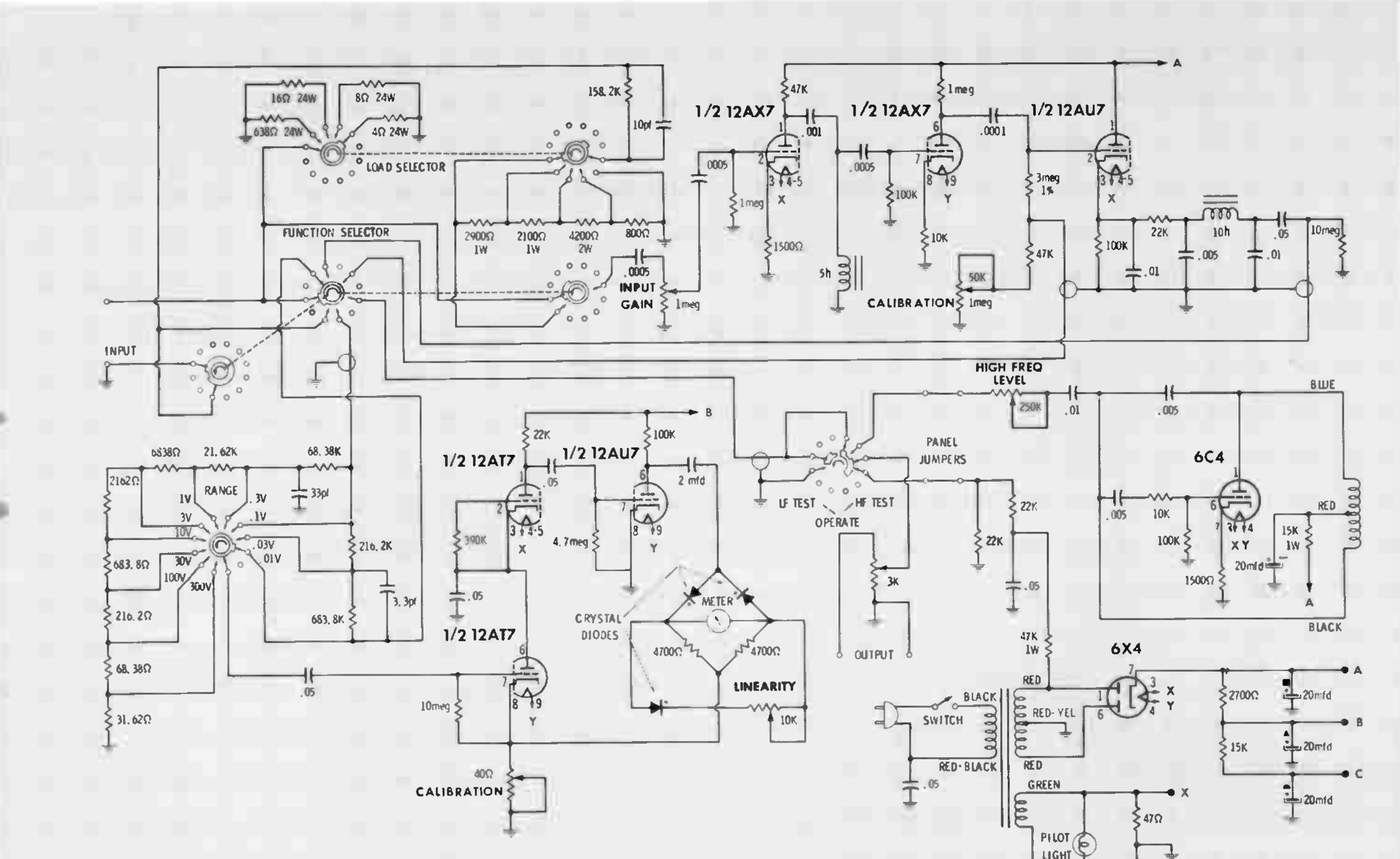


Fig. 12 Circuit diagram of a typical intermodulation distortion analyzer.

matrix type of decoder was explained previously. Therefore, let us consider the alignment procedure for the time-division type of decoder and, in particular, the envelope detection configuration. Fig. 8 depicts a typical envelope-detector circuit. A normal undetected (undemodulated or undecoded) waveform is shown in Fig. 9. Note that after the 38-KHz subcarrier is reinserted into the discriminator output signal, R-signal and L-signal envelopes become available.

With reference to Fig. 8, the output from V5 is fed to L13 and to L14. A 38-KHz subcarrier is fed from L15 into L14. Thus, the R and L envelope signals are fed to diodes X10 and X11. Note that X10 passes the positive envelope signal, and rejects the negative envelope signal. Conversely, X11 passes the negative envelope signal and rejects the positive envelope signal. It is evident that if misalignment causes the subcarrier to have an incorrect phase, both envelope signals will be distorted and the decoder will develop an objectionable amount of harmonic distortion.

If there is no component defect, and distortion is caused simply by misalignment, the procedure tabulated in Table 1 will restore the operation to normal. L13 is peaked to the pilot-subcarrier frequency, and L15 is peaked to the subcarrier frequency. L14 is a 67-KHz trap, and is aligned for minimum output. Thus, the step-by-step alignment procedure is comparatively simple.

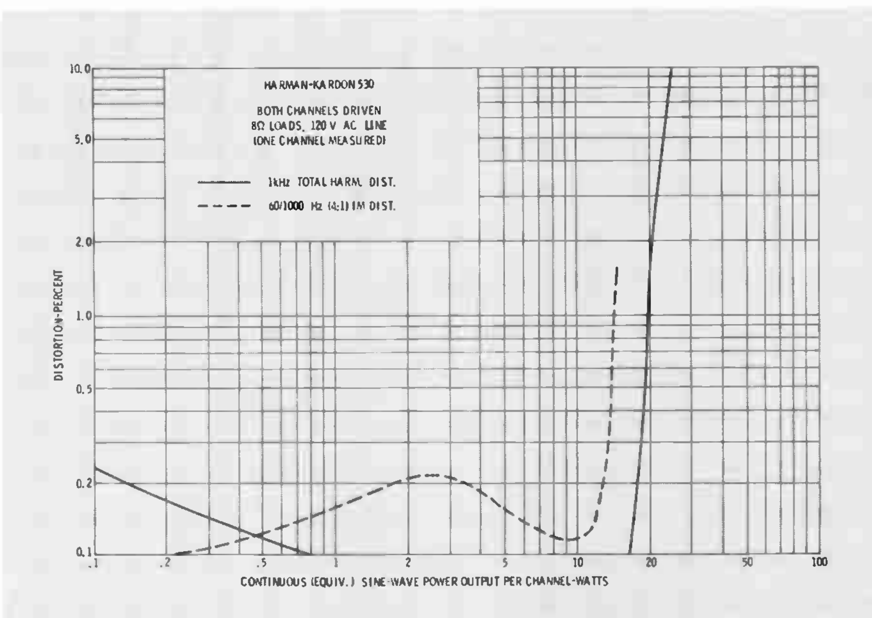


Fig. 13 Curve comparing intermodulation and harmonic distortion at various power levels. Courtesy—Harman-Kardon.

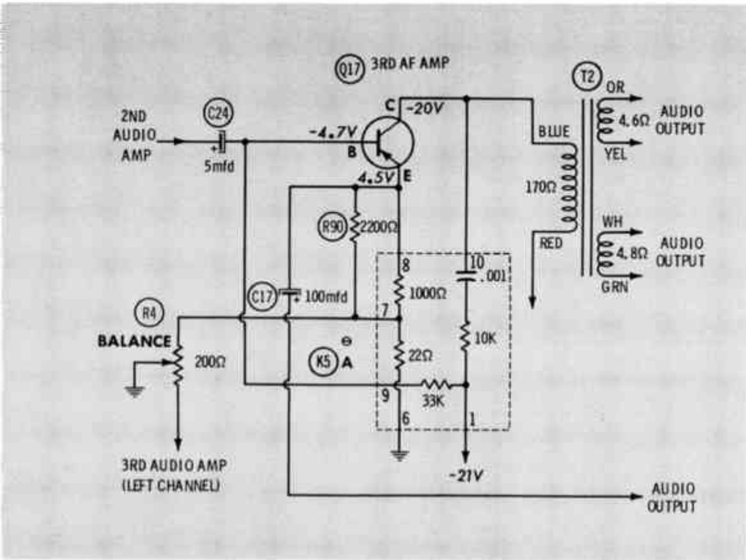


Fig. 14 Leaky coupling capacitor, C24, caused distortion in right channel.

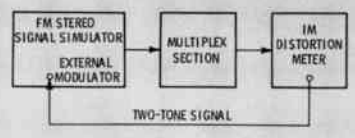


Fig. 15 Test setup for checking intermodulation distortion in a multiplex section.

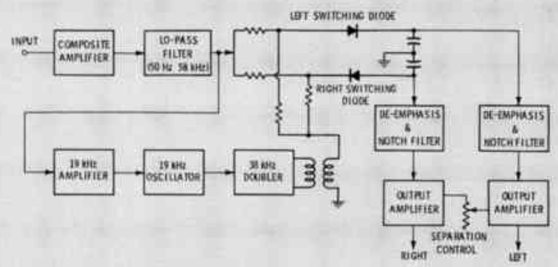


Fig. 16 Block diagram of a time-division demodulator employing a 19-KHz locked oscillator.

Next, if objectionable harmonic distortion persists, check for defective components as follows:

1. Test the tubes with a tube tester, or check by substitution.
2. Check capacitors for leakage. An open capacitor can also be the cause of distortion. For example, C45 (Fig. 8) might be leaky, or C38 might be open.
3. Next, we check diodes X17 through X11 for proper front-to-back ratio. Note that if X12 is defective, only the operation of the stereo indicator is affected.
4. Infrequent, though possible, causes of distortion include off-value resistors, noisy controls, and shorted turns in a coil or transformer.

Now, let us return to the troubleshooting procedures used to localize harmonic distortion to a particular receiver section. If the multiplex section checks normal, the trouble probably will be found in the tuner section. If this is the case, the test procedure shown in Fig. 10 will indicate an objectionable amount of harmonic distortion. Correct the trouble as explained in the first article; that is, make preliminary tube tests, and then follow a step-by-step alignment procedure, if necessary. If there are no component defects and the tuner is properly aligned, the test depicted in Fig. 10 will indicate a very low value of harmonic distortion.

After the cause of distortion has been localized and corrected, the final system check shown in Fig. 1 is repeated. Note that the total distortion of the overall system will always be greater than the distortion of any particular section because harmonic distortions in the various receiver sections add up (often in a somewhat unpredictable manner). In case the final system check does not indicate an acceptably low percentage of distortion, it is necessary to repeat the sectional tests. Also, make certain that each section is in optimum alignment.

System Intermodulation Distortion

To make an intermodulation distortion (IM) test of the complete stereo system it is necessary to use an FM generator that can be externally modulated. That is, an IM distortion analyzer that supplies a pair of test signals (two-tone signal) that are used to modulate an FM RF carrier. The modulated RF signal is applied to the input terminals of the front end as depicted in Fig. 11. For example, an FM stereo signal simulator is one type of FM generator that provides an external-modulation terminal for hi-fi tests.

Intermodulation analyzers were briefly discussed in a previous article in this series. The circuit diagram for a typical instrument is shown in Fig. 12. Note that power resistors having values of 4, 8, 16 and 600 ohms are built into the analyzer. Therefore, it is seldom necessary to use the accessory load resistors depicted in Fig. 11. Merely switch the input selector of the instrument to a suitable load value.

The operating principle of an intermodulation analyzer can be compared with that of a broadcast radio receiver. The two-tone signal (a mixture of low- and high-frequency sine waves) is fed into the analyzer which amplifies the high frequencies but rejects all low frequencies except those actually modulating the higher frequency. Thus, only the high-frequency signal and its sideband are passed.

This modulated high-frequency signal is set to a predetermined level and is then demodulated. The remaining signal appears as a low-frequency component (envelope frequency) and is passed through a low-pass filter to remove any residual high frequencies. In turn, this remaining signal is fed to the meter, which is then calibrated in terms of percentage of intermodulation distortion.

A 12AX7 operates in the high-pass amplifier section. The high-frequency signal is amplified and fed through an LC high-pass filter to the grid of the second half of the 12AX7. After further amplification, the high-frequency signal is fed to the grid of the detector tube (one-half of a 12AU7). The reference level of the signal is set in the detector grid circuit.

The detector is an infinite-impedance (cathode-follower) type, and the output signal is taken from its cathode. The low-pass filter in the cathode circuit removes all demodulation products except the envelope frequency. This envelope signal is then passed into the VTVM section for energizing the meter. Although the operation of an intermodulation analyzer is not as simple as that of a harmonic distortion meter, a little practice with the instrument is all that is needed. If the step-by-step instructions given in the instrument manual are observed, the operation is quite simple.

There is no simple relation between percentages of IM distortion and harmonics distortion. For example, Fig. 13 shows comparative test results for a good FM stereo-multiplex system. IM distortion was measured with a 60-Hz and 7-KHz two-tone signal, and harmonic distortion was measured with a 1-KHz signal. Observe, in this example, that IM distortion is less than the harmonic distortion at low power-output levels, but IM distortion is greater than harmonic distortion at medium power level. At a high power level, IM distortion increases faster than harmonic distortion.

Since an IM system test may show a widely different percentage of distortion compared with a harmonic-distortion test, it is good practice to make system measurements of both types of distortion. If you find that IM distortion is excessive, follow a step-by-step localization procedure similar to that employed for tracking down excessive harmonic distortion. That is, progressively eliminate the audio amplifiers, multiplex section and FM tuner from suspicion. After the trouble has been localized to a particular section, turn your attention

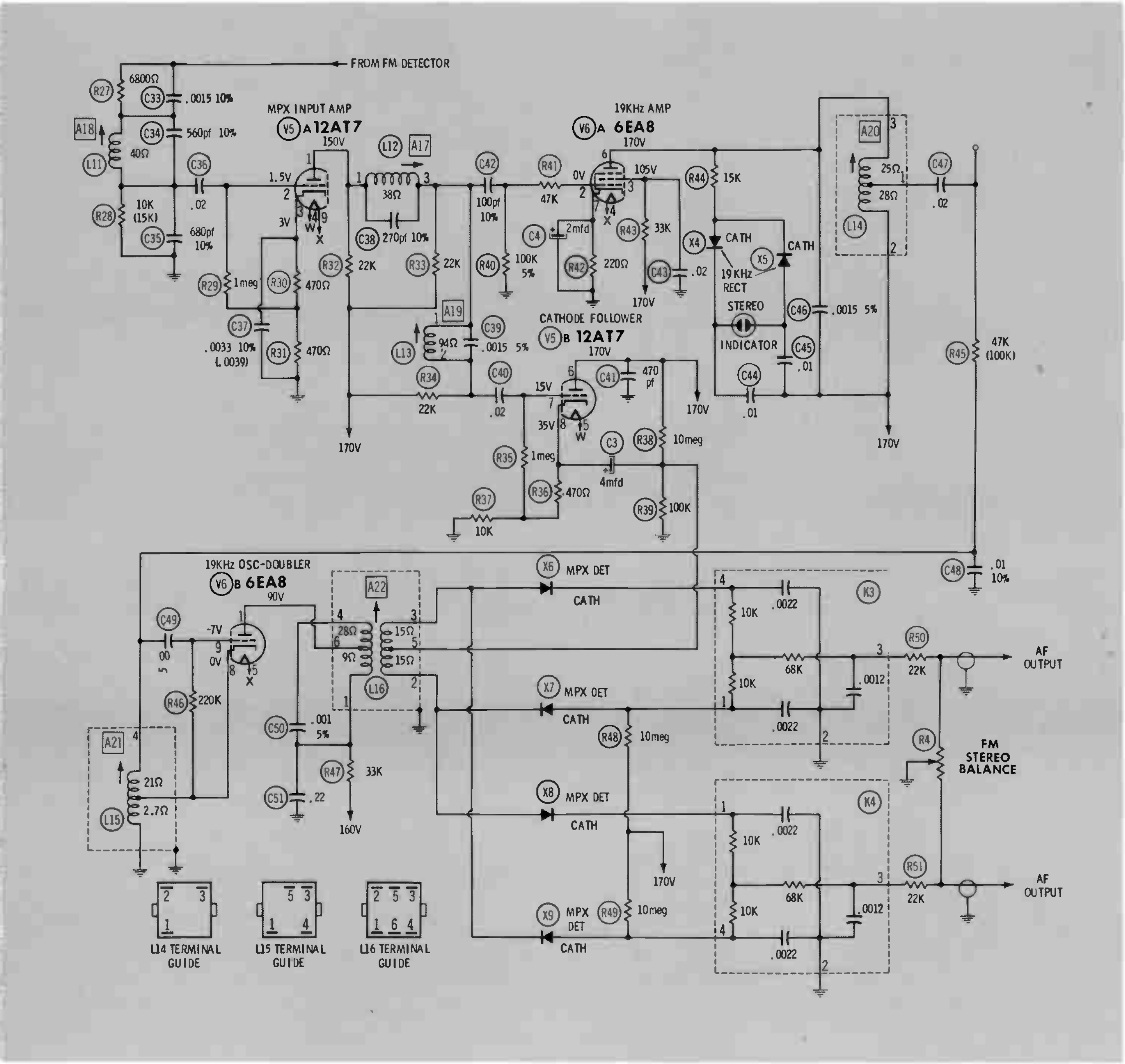


Fig. 17 Typical multiplex circuit employing a 19-KHz pilot-subcarrier oscillator.

to possible defects that can cause nonlinear operation. In general, the same component defects that cause harmonic distortion will also cause IM distortion.

Preliminary checks might show, for example, that IM distortion is satisfactory in L-channel operation, but objectionably high in R-channel operation. If this is the case, turn your attention at once to the right audio channel.

In a typical case history, C24 in Fig. 14 was leaky. This type of defect can be pinpointed either by DC voltage measurements at the associated transistor terminals, or by a careful waveform analysis at the base of the transistor. Defective audio-output transformers are less common, but this possibility should not be overlooked. If an output transistor is short-circuited, the associated resistors should be checked when the transistor is replaced.

Next, consider a situation in which the audio amplifiers check normal and attention logically is focused on the performance of the multiplex section. The IM test setup is shown in Fig. 15. If misalignment causes the subcarrier to be inserted with an incorrect phase, an objectionably high percentage of IM distortion will be indicated. This condition is aggravated when a synchronized 38-KHz oscillator "pulls." Similarly, a "pulling" 19-KHz oscillator also can cause objectionably high IM distortion. Fig. 16 shows a block diagram for a multiplex section with a 19-KHz locked oscillator. Tight locking requires precise alignment of the oscillator circuit and a normal signal-output level from the 19-KHz amplifier.

Alignment and Final System Tests

Fig. 17 shows a typical multiplex circuit with a 19-KHz pilot-subcarrier oscillator. To align the circuits, adjust L12 for minimum response at 67 KHz, L11 for minimum response at 71 KHz, and L13 for minimum response at 19 KHz. L14 and L15 are adjusted for maximum response at 19 KHz; L16 is adjusted for maximum response at 38 KHz. Finally, slight touch-up adjustments are made (if required) to obtain optimum separation of L and R signals. Note that L15 determines the free-running oscillator frequency. Tight locking depends upon precise adjustment of this frequency and injection of a normal 19-KHz signal level from L14.

After alignment and separation have been verified (or corrected), make another IM distortion measurement of the multiplex section. If there are no component defects, the percentage of distortion should be very low. However, if objectionable distortion is measured, look for component defects that can cause nonlinear operation. Assuming that the tubes and diodes have been checked, the most likely culprits are leaky capacitors which upset normal bias voltages. Off-value resistors are less-likely suspects, but the resistors should be checked if capacitors have been cleared.

Finally, let us consider the situation in which the multiplex section checks normal in the initial IM test. This leaves the FM tuner as the source of IM distortion. Tuner alignment procedures were explained previously. After the step-by-step system alignment has been completed, verify the end result by repeating the IM test depicted in Fig. 11. The percentage of IM distortion should be acceptable now, inasmuch as each section in the system has acceptable performance. However, in rare cases, the IM distortion might be a bit higher than anticipated. If such is the case, it is probably due to cumulative sectional distortions; consequently, individual sectional checks must be repeated.

It is advisable, at this point, to repeat a precaution mentioned earlier: Generators used in the foregoing step-by-step test procedures must have better characteristics than the receiver under test, otherwise, it is pointless to make distortion measurements of a hi-fi system. Even the best test equipment will occasionally require attention. It is good practice to feed the output from an audio oscillator directly into a harmonic-distortion meter to verify that the harmonic-distortion meter has acceptably low distortion. Crosschecks of all hi-fi test equipment should be made regularly for verification of performance.

The final installment of this series will discuss cartridge and turntable tests. ▲

Part 5

Cartridge and turntable measurement tests

by Robert G. Middleton

Cartridge and turntable measurement tests are required to obtain and verify performance within accepted high-fidelity limits. These tests are listed below.

Cartridge Tests

1. Measurement of needle pressure
2. Checking of cartridge output voltage
3. Hum Tests
4. Test of frequency response
5. Measurement of tracking distortion
6. Verification of stylus angle
7. Observation of stylus tip
8. Verification of tracking angle

Turntable Tests

1. Measurement of turntable speed
2. Check for wow
3. Check for flutter
4. Level test
5. Tests for wobble and/or eccentricity
6. Approximate measurement of rumble.

Measurement of Needle Pressure

Fig. 1 shows a typical needle-pressure gauge. Other types may be used, but all indicate the pressure in grams.

Normal needle pressures are different for various types of recordings. A typical hi-fi, super lightweight, pickup employs 1, 2.5 or 3 mil radius sapphire tips. (One mil equals 0.001 inch.) A 25-ohm pickup of this type uses a nominal needle pressure of 9 grams, ±1 gram. On the other hand, a 400-ohm pickup operates with a nominal needle pressure of 5 grams, ±1 gram. For precise evaluations, stylus pressure must be checked within specified tolerance limits against tracking distortion. This measurement is discussed later in this article.

The needle-pressure gauge should be placed so that the indication is obtained at the normal playing level of the pickup. This level varies in most changers; therefore, measurement is made at the average playing level. Although the pressure usually varies as the pickup is raised, and decreases as the pickup is lowered, this variation should not exceed 25%. If a greater variation is measured, look for a defect in the arm assembly. Obscure defects must be corrected by replacement of the arm assembly.

Checking of Cartridge Output Voltage

Cartridge output voltage is checked with an AC VTVM or with a calibrated oscilloscope. The advantage of a scope is that hum voltage can be distinguished from the normal cartridge output. Only the VTVM method will be discussed in this section.

The test setup for checking cartridge output voltage is shown in Fig. 2. It is desirable to employ a test record that provides a constant level at a standard test frequency, such as 1 KHz or 400 Hz.

A typical hi-fi super lightweight pickup that works into a 25-ohm load (R in Fig. 2) normally provides about 10 mv rms. If the output is weak, it is advisable to check the stylus tip, as explained later. If the tip is in good condition, but output is weak, the cartridge should be replaced.

The 25-ohm pickup described previously is often used with a 1-to-5 ratio transformer. Normal loading of the transformer results in a secondary output voltage of approximately 300 mv rms. On the other hand, a 400-ohm pickup normally is used with a 1-to-6 ratio transformer. In this case, about 40 mv rms output will be measured across the secondary. Note that the secondary must be connected to the correct value of load impedance to obtain a meaningful voltage measurement.

With reference to Fig. 3, Np denotes the number of primary turns, and Ns denotes the number of secondary turns. The winding ratio is equal to Np/Ns.

The voltage ratio of the transformer is the same as the turns ratio (except for losses due to leakage flux under normal load). The current ratio of the transformer is the inverse of the turns ratio (except for losses). The impedance ratio is equal to the square of the turns ratio. This means that if the cartridge has an impedance of 400 ohms, and the transformer winding ratio is 1-to-6, the impedance ratio is 1-to-36, and the secondary impedance is 14,400 ohms. Thus, the nominal secondary load resistor has a value of 14,400 ohms. Note, however, that a somewhat different value of secondary load impedance might be chosen to compensate for incidental transformer characteristics such as distributed capacitance. This topic is explained in greater detail in this article under "Test of Frequency Response."

The normal output voltages of common types of cartridges may be summarized as follows: A variable-reluctance cartridge will provide from 10 to 30 mv rms; a ceramic cartridge, 500 to 1200 mv rms; and crystal cartridges, 0.5 to 5 volts rms. These values assume that the cartridge works into its specified load.

The load is an impedance which consists of both the load resistance, cable capacitance, and any leakage reactance of the transformer primary. The variable-reluctance pickup works into a load impedance of

5,000 to 50,000 ohms, with a stylus pressure of 16 grams or slightly more. The ceramic pickup works into a load impedance of approximately 1 megohm, with a stylus force of about 22 grams. The crystal pickup works into a load impedance as high as 5 megohms, with a stylus force of less than 70 grams.

Hum Tests

Pickups that employ iron and coil construction are more likely to be affected by stray hum fields than crystal pickups. Of course, a poorly shielded lead from a high-impedance pickup is susceptible to stray electrostatic hum fields. Pickups with iron and coil construction are generally used with solid-steel turntables, because magnetic shielding is thereby provided against stray hum fields from the motor and power supply.

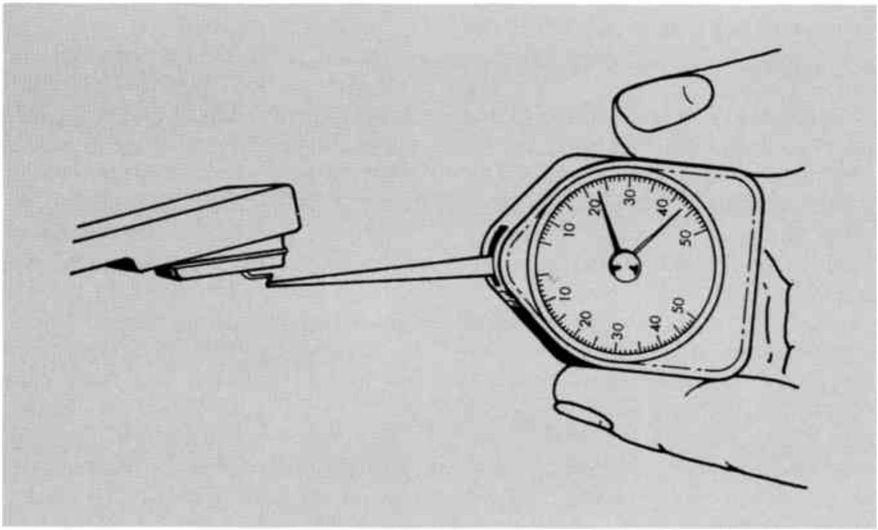


Fig. 1 A typical needle-pressure gauge used for measuring the pressure exerted at the tip of the needle.

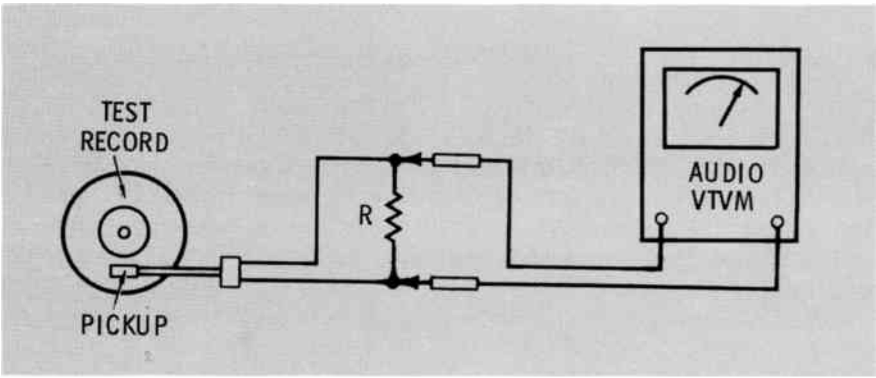
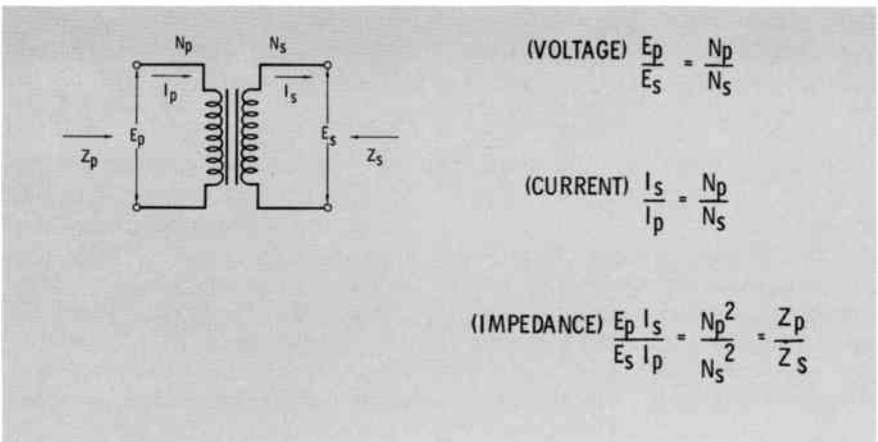


Fig. 2 Test setup for checking cartridge output voltage.



To check for hum, use the test setup shown in Fig. 2, with the pickup held slightly above the record surface. Any reading on the meter is probably the product of hum pickup.

A better test for hum is made by using a scope in place of the VTVM in Fig. 2. Hum voltage is then seen as a 60-Hz (or 120-Hz) waveform, and can be definitely identified. Note that 120-Hz hum is most likely to stem from stray fields of a filter inductor. When playing a test record that has a 1-KHz tone, hum shows up with the tone signal superimposed on the low-frequency waveform, as shown in Fig. 4. The hum level can be minimized in any case by correction of shielding defects to prevent stray fields from gaining access to the pickup.

Test of Frequency Response

Frequency response is checked with a test record and an audio VTVM, using the setup shown in Fig. 2. Various types of test records are available. A **gliding-tone** test record can be compared to an audio sweep generator. One standard test record starts at 14 KHz and ends at 10 Hz. Another starts at 10 KHz and ends at 50 Hz. A **banded-tone** test record provides only a 1-KHz tone in steps with increasing output levels. This type of record is suitable for harmonic-distortion tests, but not for frequency-response checks. Some test records provide both banded-tone and gliding-tone outputs. For example, a banded-tone sequence may be followed by alternate gliding and constant audio frequencies from 3 KHz to 30 Hz. **Two-tone** test records are used only for IM distortion measurements.

The frequency response of a good ceramic or crystal cartridge should be reasonably uniform from 50 Hz to approximately 15 KHz. Variable-relutance cartridges should have reasonably flat frequency response to approximately 20 KHz. Some types of high-output cartridges have an upper frequency limit of 8 KHz. Although one or more peaks in frequency response will often be found, the peak amplitude(s) should be less than 2 dB. An audio VTVM with a dB scale is particularly useful for these tests.

Load-impedance values can have considerable effect on the frequency response of a ceramic or crystal cartridge. Poor frequency response can sometimes be improved by a change in load resistance and/or capacitance. Equalizer circuits for standard recordings are shown in Fig. 5. If the high-frequency response of a pickup drops off gradually, this characteristic will often be acceptable, inasmuch as the preemphasis employed in most recordings must be equalized in the hi-fi system.

Measurement of Tracking Distortion

An ideal tone arm and pickup would track a record (see Fig. 6A) so that the stylus would always move perpendicularly to the tangent at the record groove. In practice, this can occur only at one groove in the record because the pickup moves on a curve from the outer to the inner grooves, as shown in Fig. 6A. A reasonably good test of the tracking angle is to place the stylus at the outer grooves, then at an intermediate groove, and finally at an inner groove, and to observe whether the edge of the cartridge is **reasonably** tangent to the grooves.

A tracking error of 3° or 4° is normally expected at the outer and inner grooves. Large tracking errors are plainly visible, but small errors must be detected by careful measurement. Excessive errors in the tracking angle produce second-harmonic distortion, as shown in Fig. 6B. Harmonic distortion is measured with the test setup shown in Fig. 7. In such tests, it is assumed the preamplifier has been checked previously for distortion, using a quality audio oscillator.

Note that the percentage of harmonic distortion measured in Fig. 6B is not entirely accounted for by tracking error and possible preamplifier distortion. That is, the pickup might have inherent harmonic distortion. At this point in the procedure, we are concerned with minimizing the harmonic distortion due to tracking error. Since this is a function of stylus pressure, it is advisable to try the effect of increasing the stylus pressure within reasonable limits. Some pickups are rated for minimum stylus force to provide optimum tracking. If a recording or turntable is warped, or slightly eccentric, a 50% increase in stylus force may be required to obtain

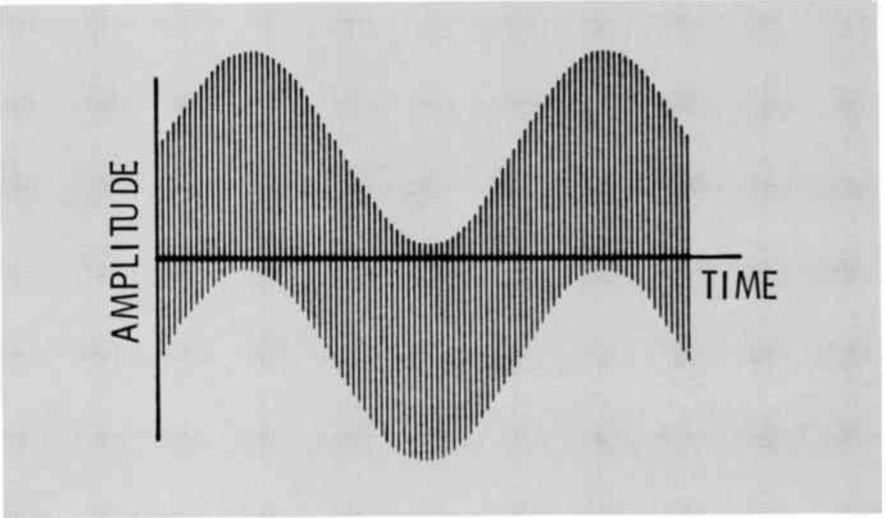


Fig. 4 Hum voltage superimposed on low-frequency waveform.

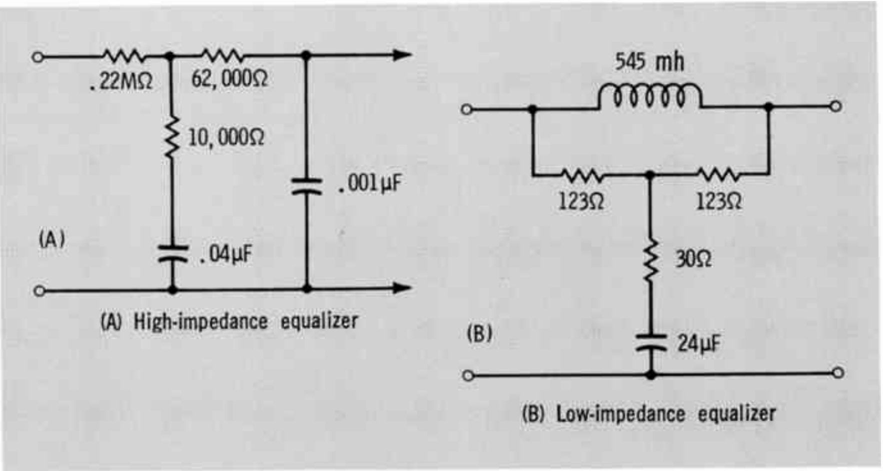


Fig. 5 Equalizer circuits can be used to improve frequency response.

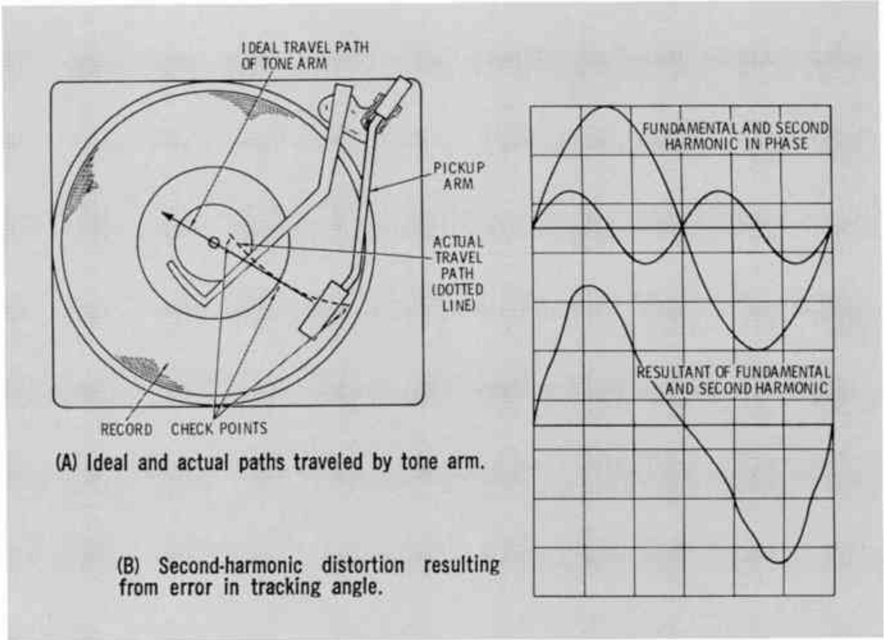


Fig 6 Illustrations showing tracking characteristics and harmonic distortion resulting from improper tracking.

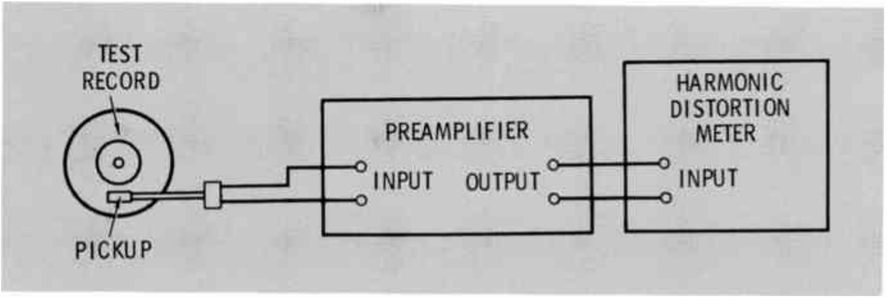


Fig. 7 Test setup for measuring harmonic distortion.

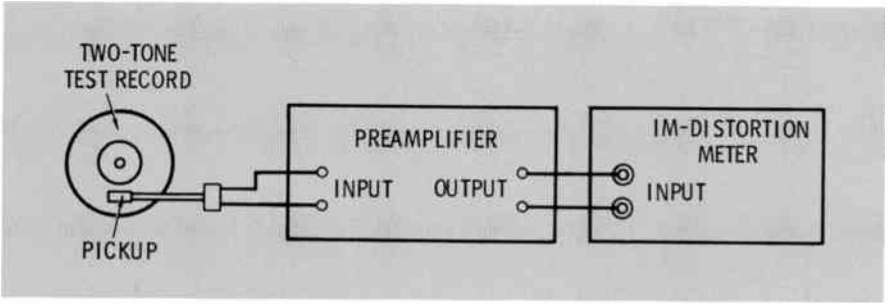


Fig. 8 Test setup for measuring intermodulation distortion using two-tone test record.

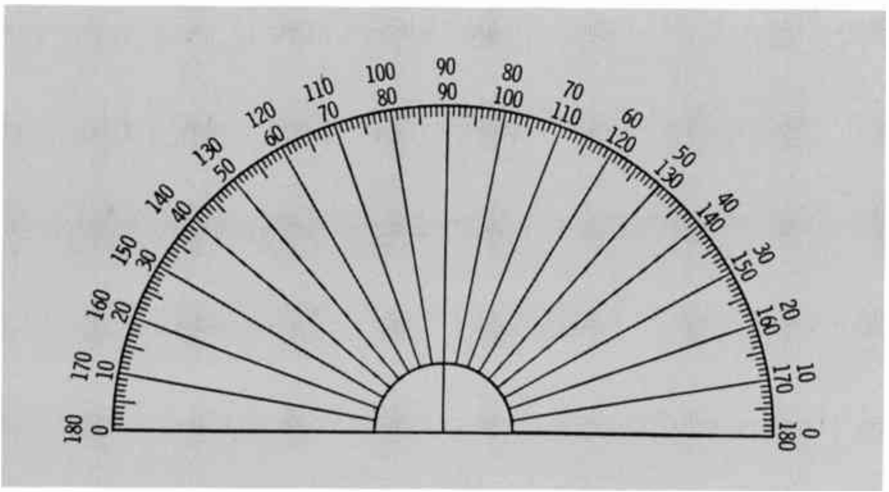


Fig. 9 Protractor can be used for measuring stylus angle.

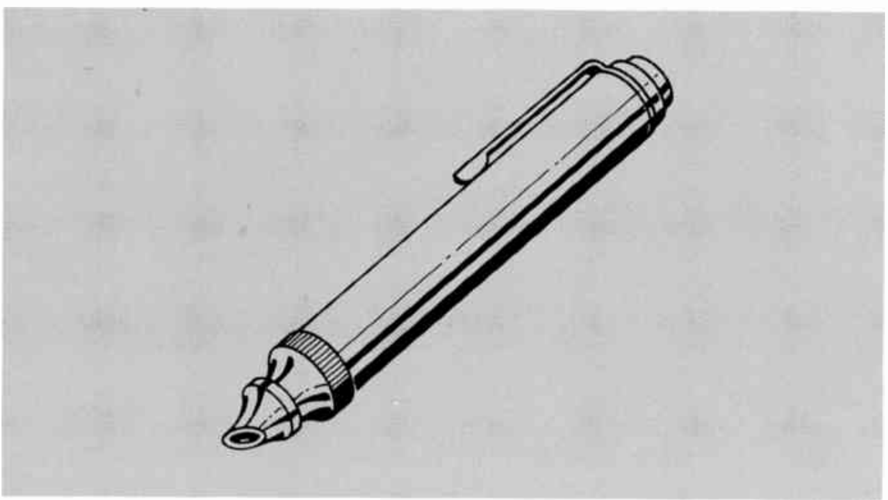


Fig. 10 Fifty-power microscope for visually inspecting condition of stylus.

good tracking.

After checking for harmonic distortion, it is good practice to make an intermodulation distortion test, as depicted in Fig. 8. The two-tone test record must provide suitable frequencies. Assuming that the preamplifier has been checked previously and found to have acceptably low IM distortion, an objectionably high meter reading is due to nonlinearity in the pickup. A banded-tone 1-KHz test record with progressive output levels may show that IM distortion is negligible except at high output level. In any case, a defective pickup must be replaced.

Verification of Stylus Angle

Even if the needle pressure is optimum, distortion will occur if the stylus angle is incorrect. Therefore, it is good practice to check the stylus angle against the manufacturer's specifications, if available. Angles can be conveniently measured with a protractor, such as depicted in Fig. 9. If the stylus angle is out of tolerance because of mechanical distortion, it is generally impractical to correct the trouble—replacement should be made at the outset.

Observation of Stylus Tip

Unless a new stylus is used in making performance measurements, it is a good practice to inspect the stylus for wear with a 50-power stylus inspection microscope, as shown in Fig. 10. If the stylus shows signs of wear, it should be replaced before playing a test record. By the same token, a test record should be checked periodically for harmonic and IM distortion, using a good stylus. When the test record begins to show appreciable distortion, it should be discarded.

Measurement of Turntable Speed

A turntable must rotate at constant speed, even during lightly and heavily recorded passages. Turntable speed is conveniently measured with a stroboscopic disc illustrated in Fig. 11. The strobe disc is viewed under a neon or fluorescent light powered from a 60-Hz source. One of the barred circles will appear to stand still, or to rotate slowly clockwise or counterclockwise. Motionless bars indicate the calibrated

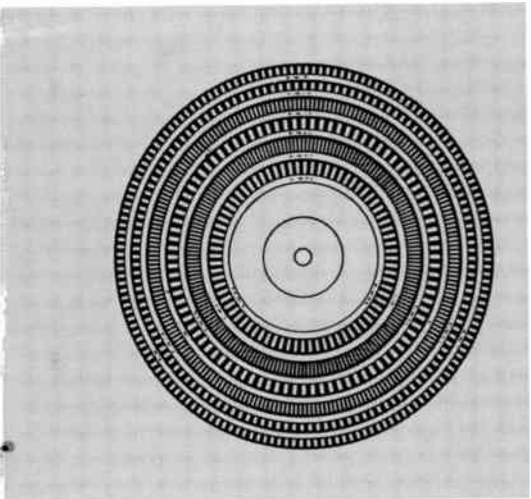


Fig. 11 Stroboscopic disc for measuring turntable speed.

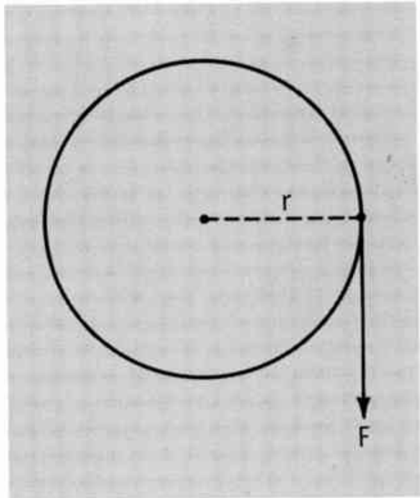


Fig. 12 Illustration of twisting force called torque.

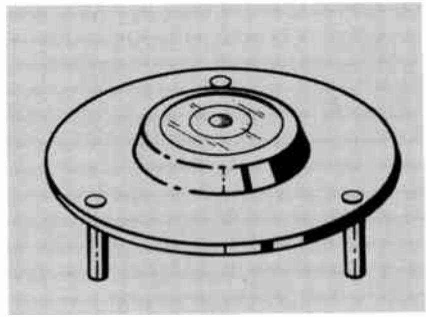


Fig. 13 Device for leveling a turntable.

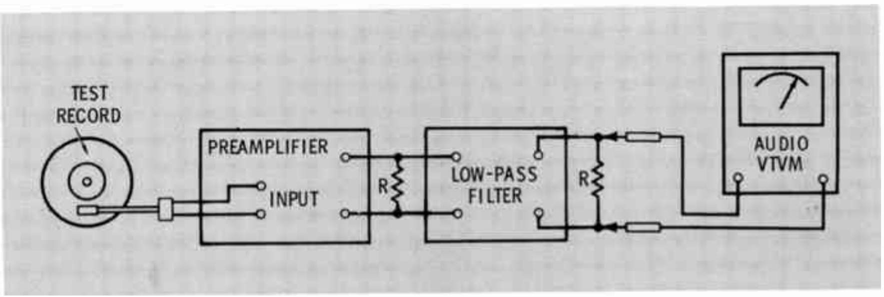


Fig. 14 Test setup for measuring rumble.

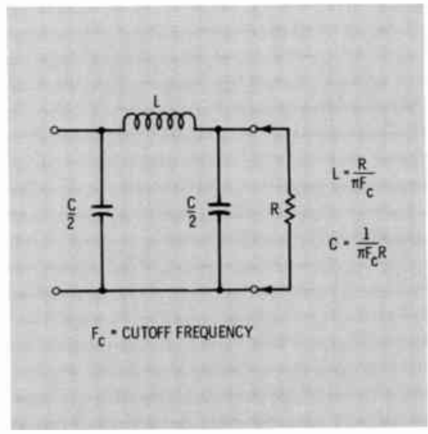


Fig. 15 A single-section, low-pass Pi filter used in rumble test.

speed of the particular barred circle. Clockwise rotation indicates that the actual speed is slower.

In some cases, the barred circle will appear to stand still momentarily, rotate for a while, and then to stand still again. This trouble is sometimes caused by oil or other foreign substances on the drive wheel. These parts can be cleaned with alcohol.

The same trouble symptom can be caused by defective turntable or motor bearings. Also, there may be unsuspected mechanical interference, such as the turntable rubbing against some surface or object.

Wow

If a turntable does not rotate at constant speed, the condition is technically termed **wow.** It is basically a mechanical form of frequency modulation. Economy-type turntables may be unsatisfactory for hi-fi reproduction because of marginal motor power and lack of sufficient flywheel effect. A strobe disc test under playing conditions will disclose wow which, in this case, cannot be eliminated.

Motor and turntable units are often specified by manufacturers for motor type, power consumption, and the torque required to brake the turntable from a given speed to a lower speed (such as from 78 rpm to 77 rpm). Torque denotes a twisting force, as depicted in Fig. 12. The magnitude of torque is expressed as the product of force and distance (F and r in Fig. 12.) Braking torque for a turntable is stated in ounce-inch units.

Of course, wow will become evident if the motor is operated at subnormal line voltage. Noticeable wow will be observed on heavily recorded passages. Another obvious cause of wow is a warped record; the tolerable limit of warp is about 1/16 inch. Special instruments are available to measure wow, and indicate the percentage in rms deviation of a tone frequency with respect to its average frequency. However, a strobe disc serves the same purpose for routine shop tests. The only requirement in strobe-disc application is that the changes in apparent speed of bar rotation be observed carefully. When wow is slight, it might be overlooked in an off-hand test.

Flutter

Flutter is related to wow, but has a comparatively higher deviation rate. The human ear is more sensitive to flutter than to wow. Flutter is usually caused by small defects in the motor or the mechanical drive system. It tends to occur in units that have been in extended service, with little or no attention to lubrication. The resulting wear causes poor mechanical fit, which must be corrected by replacement of the faulty parts. Although flutter meters are available, a strobe disc serves adequately for routine shop tests. Both wow and flutter meters are comparatively expensive instruments, and their cost can be justified only by the larger shops or labs.

Understandably, a turntable must be level to provide optimum performance. A typical turntable level is illustrated in Fig. 13. The turntable must also be free from wobble, and should maintain its level against moderate vertical pressure near the edges. If a wow and/or flutter meter is available, note that a meter reading of less than 1% is considered acceptable. A good hi-fi turntable may show as little as 0.1% wow or flutter, even on heavily recorded passages.

Rumble

Both wow and flutter are distinguished from **rumble.** Rumble consists of a series of random pulses (not a cyclic low-frequency interference). It is caused by vibrations within the player, and is often decribed as noise "like furniture moving around upstairs." A scope test quickly shows whether interference has a rumble waveform—the pattern is random and does not have a steady cyclic form in the case of rumble. Although rumble can be reduced or eliminated by means of an amplifier rumble filter which cuts off sharply below 50 Hz, it is obviously desirable to correct the cause of the rumble if it is practical to do so.

When rumble is a problem, check the motor for vibration. Any mechanical defect that causes vibrations to be transmitted to the turntable can cause rumble. Note that rumble can be measured, using the test setup shown in Fig. 14. The low-pass filter has a cutoff frequency of approximately 300 Hz. The test procedure is as follows:

First, the low-pass filter is removed from the circuit, and the output is measured using a 1-KHz test record. The preamplifier controls are adjusted for maximum-rated output and flat frequency response. Then, the low-pass filter is connected into the test circuit, and the VTVM sensitivity is advanced to obtain a convenient reading. Finally, lift the pickup from the test record and note any change in VTVM reading. The amount of rumble is expressed as the number of dB below maximum rated output from the preamplifier.

Let us briefly consider the component values used in a low-pass filter. With reference to Fig. 15, choose a cutoff frequency, fc, and a terminating resistance, R. Then find the required values of L and C as indicated. For example, if fc= 300 Hz, and R=75 ohms, then L= 80 mh and C=0.071 mfd. One-half of C, or 0.035 mfd is connected at each end of L to form the filter. A sharper cutoff characteristic can be obtained by connecting several filter sections in series. (Note that the inductors of the multi-section filter must not couple into one another.)

Conclusion

Although other types of tests are made in audio laboratories, these are generally out of the question for the service shop because of the costly test equipment that is required. For example, measurement of the static compliance of a stylus requires a calibrated shadowgraph and a sensitive balance. However, the simpler tests that have been explained in this series of articles permit almost any service problem to be solved with a minimum of time and effort. Apprentice technicians are often awed by ordinary harmonic and intermodulation distortion meters, and occasionally by a scope. However, practice makes perfect, and the modern shop cannot hope to remain competitive unless modern test equipment is utilized. ▲