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FM Multiplexing for STEREO

by LEONARD FELDMAN

All-about FM stereo, including the signal, converting, operation, servicing, and alignment of multiplex circuits . . . plus descriptions of the latest multiplex equipment.

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FM MULTIPLEXING FOR STEREO

by Leonard Feldman

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PREFACE

FM broadcasting has been with us now for almost three decades, since its invention by the late Major Edwin Armstrong. Never before has the promise of quality broadcasting been closer to mass acceptance. The final broadening of the FM receiver market will, in great part, be due to the newly authorized Stereophonic Broadcasting Standards approved by the FCC early in 1961.

Many have contributed to the development of these standards, and the system finally chosen is technically sophisticated but readily understandable by even the nontechnical reader.

With approximately 17 million FM receivers in the hands of the listening public, set owners will want to know what can be done to make their sets capable of receiving this type of programming. This book tells how to convert existing sets for stereo, and should be of interest to the thousands of service technicians who will be called upon to effectively make these conversions. Servicing and alignment procedures, as well as an analysis of test equipment and modern commercial multiplex circuits, are also fully discussed in this volume. The exact specifications of stereo broadcasting, as approved by the Federal Communications Commission, are reprinted in the Appendix.

The success of a book of this type depends, in great measure, on the cooperation the author receives from manufacturers and other experts in the field. In this connection the author wishes to thank the following manufacturers: Allied Radio Corp.; Altec Lansing Corp.; Bogen Company (Division of Siegler Industries); Crosby Electronics, Inc.; Crosby-Teletronics Corp.; EICO Electronic Instrument Co., Inc.; Fisher Radio Corp.; Harman-Kardon, Inc.; Heath Company (Division of Daystrom, Inc.); Boonton Radio Corp.; Pilot Radio Corp.; Zenith Radio Corp.; Sherwood Electronic Laboratories; General Electric Company; and others whose circuits were not actually used herein, but whose cooperation is nonetheless appreciated.

Finally, the author wishes to dedicate this volume to the man who has contributed much to the development of FM stereo broadcasting. To Murray G. Crosby, with whom the author has had the privilege of being associated these last several years, and who probably holds more patents in this field than any other living American, this book is fondly dedicated.

LEONARD FELDMAN

January, 1962

CHAPTER 1

INTRODUCTION TO FM STEREO

On April 19, 1961, the Federal Communications Commission held a press conference which initiated a new era in FM radio broadcasting. The long-awaited stereo standards were announced and approval was granted to stations to begin broadcasting as soon after June 1, 1961, as equipment installation would permit. This brief announcement actually represented the culmination of the efforts of literally hundreds of people who had been involved in the decision.

HISTORY

Long before the advent of the compatible stereophonic disc record (when tape was the only commercially available recording medium for stereophonic playback), engineers were already giving thought to "stereo radio" and its market potential. In 1953 Murray G. Crosby, a leading inventor in the field of FM and communications, applied for a patent which described a method of compatible stereophonic FM broadcasting using "multiplex" techniques originated by the late Major Edwin H. Armstrong, who is credited with the invention of FM broadcasting.

Around the same time, several enterprising radio stations began experimental stereo broadcasting utilizing an AM and an FM channel for the "left" and "right" signals. This technique was recognized by all as being strictly an interim measure, for it suffers from many limitations. For one thing, the AM "side" of the broadcast is inferior to the FM "side" in frequency response. Then, too, noisy reception of AM in many locations tends to detract from the little stereo effect present. Finally, the monophonic listener, equipped with *only* an AM or an FM receiver, hears what amounts to a one-sided program. Despite these inherent deficiencies, this "temporary" technique continued for many years and is still in limited use in areas which do not yet have all-FM stereo broadcasting.

In mid-1959 the National Stereophonic Radio Committee (NSRC), composed of industry representatives, was organized by the FCC and directed to look into several proposed systems for broadcasting both channels of a stereophonic program over one FM frequency. At one point, as many as nineteen systems of one form or another were under consideration. These were quickly narrowed down to six, which were field tested, under the supervision of the FCC in the summer of 1960. After several months of deliberations, the FCC rendered its decision in April of 1961. The system chosen was the one proposed by both the General Electric Company and Zenith Radio Corporation, although it contains most of the elements covered by the Crosby patent.

SPECIFICATIONS

An analysis of the system chosen shows how well the choice meets the criteria originally set forth by industry and the FCC. The Commission was concerned primarily with some sixteen to seventeen million FM receivers then in the hands of the public. The system to be chosen must in no way obsolete those receivers. Furthermore, it should not materially degrade monophonic reception *nor* alter its content. Ideally, the left and right channels should both blend together to form a monophonic equivalent. On the other hand, the stereo listener-to-be should have quality equal to or better than that afforded by other stereo media such as disc recordings and tapes. This meant the frequency response of both the left and right channels must be as good as the single-channel FM frequency response (flat from 50 cycles to 15,000 cycles). Moreover, full separation between channels must exist over the entire frequency range.

Another consideration, of greater significance to broadcasters than to the public, was provision for private subscriber services such as background music or "storecast." In 1955 the Subsidiary Communications Authorization (SCA) was established to permit "multiplex" transmission of such services. At that time, the FM stations were under considerable economic pressure, and at least one hundred and seventy stations welcomed the opportunity to supplement their income until stereo broadcasting became an important consideration. Therefore the ideal system would provide for stereo broadcasting and private subscriber services at the same time.

SUM-AND-DIFFERENCE TECHNIQUE

No introduction to FM stereo would be complete without a discussion of the "sum-and-difference" technique, which is the basis for the compatibility mentioned earlier.

If L represents the electrical signal corresponding to the left-channel program material, and R the electrical signal of the right channel, then it is possible to add and subtract these two electrical signals to form L+R and L−R. The latter signal (L−R) may be difficult to understand at first, but examination of the step-by-step diagrams in Fig. 1-1 should help clarify this technique.

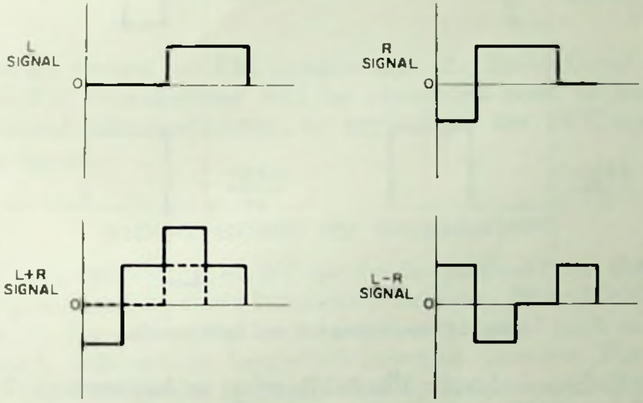


Fig. 1-1. Combining signals to obtain L + R and L − R.

Assume that L+R is transmitted by the FM station in the usual manner, but that the L−R is transmitted via a super-audible, modulated subcarrier (which will be explained in detail later). Then a monophonic listener equipped with an ordinary receiver will hear nothing but L+R. But L+R is, by definition, the sum of both the left and right signals—in other words, a monophonic equivalent of the stereo program. Thus, the mono listener is not even aware that stereo is being broadcast unless an announcer specifically mentions the fact.

Next, consider the stereo listener whose receiver not only recovers L+R signals, but is able to demodulate and receive the L−R signals as well, If, then, these two "sum" and "difference" signals are treated by further electrical adding and subtracting, the following results are obtained:

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

and,

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

In other words, separate L and R signals have again been recovered, as illustrated in Fig. 1-2. This concept, which is extremely important to the understanding of multiplex stereo circuitry, will be mentioned again and again in the discussions of specific circuits and servicing techniques throughout the book. The diagrams of Figs. 1-1 and 1-2 utilize square-wave

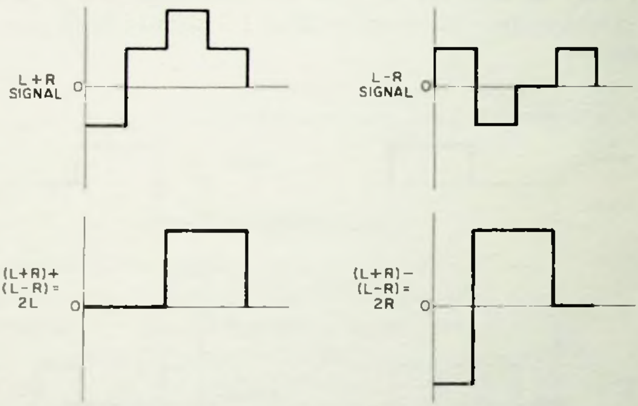


Fig. 1-2. Recovering left and right signals.

signals to more clearly illustrate what is happening. To convince yourself that the "sum and difference" principle works for any waveform, no matter how complex, try performing the adding and subtracting operations graphically to any L and R waveform you choose (such as two sinusoidal signals having different frequencies). If you plot carefully, elementary algebra will tell you that L and R will always be recovered intact and theoretically with no cross talk between the two.

CHAPTER 2

ANALYSIS OF THE APPROVED STEREO SIGNAL

A brief review of FM modulation in conventional monophonic FM broadcasting will be presented now, to help you understand FM modulation as applied to the FCC-approved stereo signal.

MONOPHONIC FM MODULATION

In FM broadcasting, an RF carrier is radiated from the transmitting antenna at some frequency between 88 and 108 megacycles. In the absence of any program material such as music or speech, this carrier frequency remains constant. For example, if the transmitter is modulated by a 1,000 cycle tone, the RF carrier frequency will be made to vary above and below its center point 1,000 times per second. The frequency shift above and below center is directly proportional to the amplitude of the modulating tone. According to the FCC standards for program material, this frequency shift must never exceed ±75 kc about the center carrier frequency. Thus, each station occupies a spectrum space of 150 kc. Additionally, a guard band of 25 kc is provided between stations. Thus, a total of 200 kc per channel is allotted. Fig. 2-1 illustrates the frequency-spectrum assignment for a station having its center frequency at 100 megacycles. It is not enough to specify the maximum variation of the carrier frequency under conditions of audio modulation; the range of audio modulating frequencies must be specified as well. Public program material is limited to the frequency range between 50 and 15,000 cycles. While these extremes are not the limits of human hearing, they are sufficiently broad to be classified as true high-fidelity transmission.

SCA Service

Background music, or SCA transmission, on a private, point-to-point basis has been in existence for several years. This secondary service, which is also provided for by monophonic transmission standards, was actually the first commercial use of "multiplex" techniques. It is completely analogous to stereo multiplex transmission, as you will see shortly.

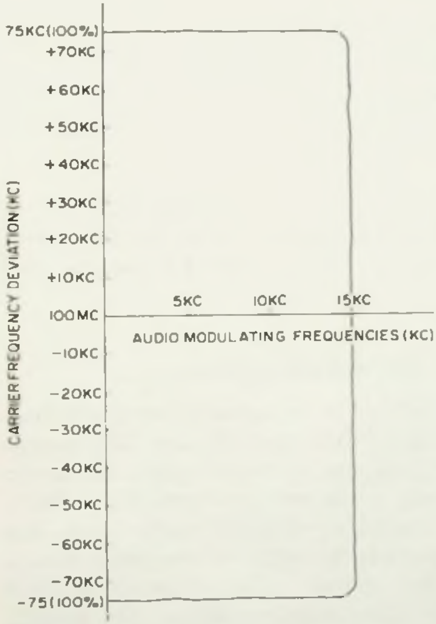


Fig. 2-1. Frequency distribution for FM station at 100 mc.

A superaudible tone between 25 and 75 kc (actually, 42 or 67 kc was most commonly used) modulates the main RF carrier in much the same way as the normal program material. This subcarrier, as the tone is called, is constant in amplitude and is always on the air while background music or other private services are being transmitted. In order to transmit additional intelligence, the subcarrier is altered in frequency by the secondary program material in much the same way the main RF carrier is modulated. Usually, the subcarrier is modulated ±7 or 8 kc about the subcarrier center frequency. Thus, if its frequency is 67 kc, the subcarrier will make maximum excursions up to 74 kc and down to 60 kc when maximum modulation is applied.

By this time you will surely ask, "How can main program material be allowed to modulate the main carrier ±75 kc and still have 'room' for applying a superaudible subcarrier in the same space?" Actually, when background music is transmitted, the modulation of the main program material must be "backed off" in order to accommodate the additional service. For example, suppose the subcarrier amplitude modulates the main RF carrier by 10% of its maximum permissible 75-kc modulation (or ±7.5 kc). The main program material modulation must

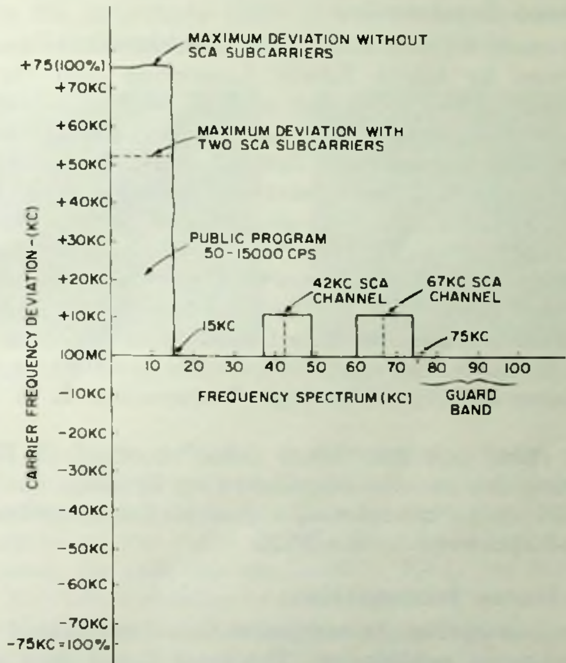


Fig. 2-2. Frequency distribution and deviation for monophonic plus SCA transmission.

then be restricted to a maximum of ±67.5 kc, so that the combined total of both forms of modulation never exceeds 75 kc. Fig. 2-2 shows spectrum distribution as well as percentages of modulation for the combination of public and subscriber programming for a station having a center frequency of 100 megacycles. The actual audio frequency of the main program material is not shown as a single tone, since it is presumed to include all frequencies from 50 to 15,000 cycles at various instances, depending on the nature of the program material itself.

FM STEREO MODULATION

From the above brief explanation of multiplex subcarrier use one could immediately suggest a simple means of transmitting stereo via FM. If the left-channel signal were used to modulate the main carrier, and if the right-channel signal were used to modulate a subcarrier, conditions for separate but simultaneous transmission of two channels of a stereo program would immediately be met.

Early Stereo Transmission

Such a crude form of stereophonic transmission was actually demonstrated by Major Edwin Armstrong over thirty years ago! "Crude" is not too harsh a term because virtually none of the criteria previously set forth for good stereo transmission were met: The monophonic listener could hear only the left side of the orchestra. The subcarrier (because of its lower energy contribution to the over-all radiated power) had far less range than the main channel. Finally, because the right channel occupies such a small portion of the total modulation spectrum and amplitude, its frequency response is greatly inferior to that of the main, or left, channel. (You may have noticed that the background music heard in restaurants and other public places usually lacks high frequencies. It is definitely lo-fi).

Having ruled out the "brute force" method of FM stereo broadcasting, let us now consider, step by step, the make-up of the total stereo modulating signal as finally refined by industry and approved by the FCC.

Adopted Stereo Transmission

The simplest part of the composite signal consists of the regular main-channel modulation. The total signal does not differ in appearance from conventional monophonic waveforms. You will recall, however, that the waveform itself is derived by adding L to R to form L+R. For the moment, let us simplify the discussion, by assuming that only a left signal—a single tone of, say, 400 cycles—is to be transmitted. Since there is no R signal at this instant, attempting to add L to R is the same as L+0, or just plain L. This signal is used for modulating the main RF carrier to 45% of the maximum allowable ± 75 kc.

Monophonic Considerations—The monophonic listener will still receive this 400-cycle tone on his FM receiver, but at an amplitude of only 45% of its previous maximum. Furthermore, the signal might just as well have been an R signal, with no L transmitted—the monophonic receiver would be unable to detect any difference between the two. As you can see in Fig. 2-3, this L only (or R only) main-channel modulation is nothing more than the familiar sine wave (approximately two complete cycles are shown).

Now suppose that a stereo program is being transmitted, but that for one instant, the left microphone picks up exactly the same signal as the right microphone, or L=R (both pickups being the 400-cycle tone shown in Fig. 2-3). Now the main-channel modulation waveform will look like Fig. 2-4, but will have twice the amplitude (90% of full 75 kc modulation) because the left and right microphones are picking up equal con-

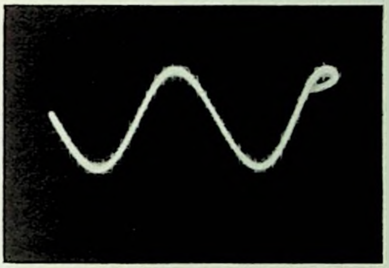


Fig. 2-3. Left-only 400-cps tone.

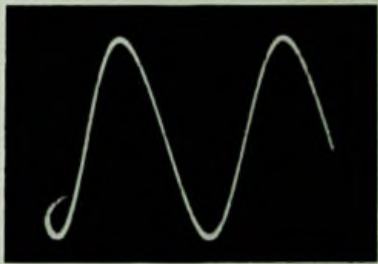


Fig. 2-4. Main-channel waveform when L=R.

tributions. Such an unusual instantaneous condition during stereo broadcasting is unlikely. In fact, it is a completely monophonic condition. In other words, any time the left signal is exactly equal to the right signal, the transmission is monophonic—since, by substitution, L+R = L+L (or R+R) = 2L (or 2R). For this condition of L=R, L−R will be zero. Hence, no stereo "difference" information is transmitted via a superaudible subcarrier. Not only is there no modulating signal for the subcarrier, but the subcarrier itself disappears automatically! Presently you will see why this happens when no "difference" information is to be transmitted in the approved stereo broadcasting system.

Let us return to the transmission of an "L only" signal, the main-channel modulation portion of which appeared in Fig. 2-3. Under these conditions, there is a difference signal (L−R), also equal to L, which must be transmitted. This difference signal is used to modulate a 38-kc subearrier, but the modulation itself is AM—not the FM that modulates the subcarrier when background music is transmitted. This technique is called "amplitude-modulated, double-sideband, suppressed-carrier transmission." To fully understand its importance, let us review some fundamentals of AM modulation.

38-kc Subcarrier

Fig. 2-5 shows the unmodulated 38 kc subcarrier used in stereo transmission, and Fig. 2-6 the familiar "envelope" waveform when the 38 kc subcarrier is amplitude-modulated by 400 cycles.

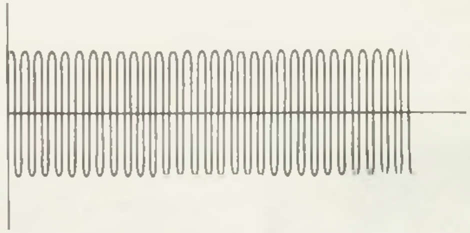


Fig. 2-5. 38-kc subcarrier.

Mathematically, it can be shown that this waveform actually contains three distinct frequencies—the subcarrier frequency (38,000 cycles), the sum of the subcarrier and the 400-cycle modulating tone (38,400 cycles), and the difference between the two (38,000-400, or 37,600, cycles). The sum and difference frequencies are called the upper and lower sidebands, respectively, of the amplitude-modulated waveform. It is these sidebands that actually carry the program information (the 400-

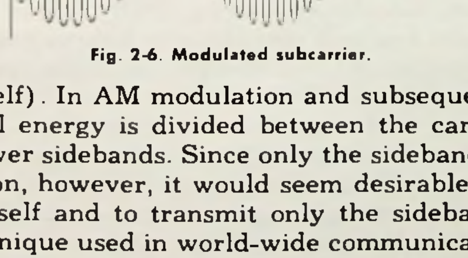


Fig. 2-6. Modulated subcarrier.

cycle tone itself). In AM modulation and subsequent transmission, the total energy is divided between the carrier and the upper and lower sidebands. Since only the sidebands carry useful information, however, it would seem desirable to eliminate the carrier itself and to transmit only the sidebands. This is, in fact, a technique used in world-wide communications, as any "ham" radio operator knows.

Carrier Suppression—Fig. 2-7 shows the upper- and lower-sideband waveform which results when the 38-kc subcarrier is removed, or *suppressed*. One point should be stressed—while this waveform was achieved through amplitude modulation, it will now be used, along with the main-channel 400-cycle tone in Fig. 2-3, to *FM modulate* the station RF carrier. Thus, actual stereo broadcasting is *all FM*. In fact, the main-channel wave-

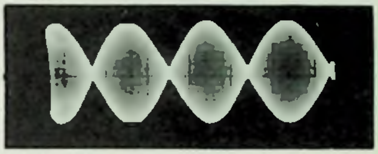


Fig. 2-7. Sidebands with carrier supressed.

form of Fig. 2-3 will now be added to the "double-sideband, suppressed-carrier" waveform of Fig. 2-7. This will produce the waveform in Fig. 2-8. Except for one minor missing element, it is all that will be used to transmit a left-channel-only, 400-cycle tone over the air. This will be done by modulating the RF carrier ±90% of 75 kc.

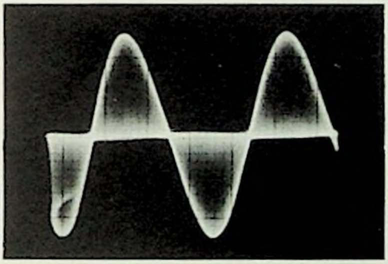


Fig. 2-8. L-only 400-cps waveform.

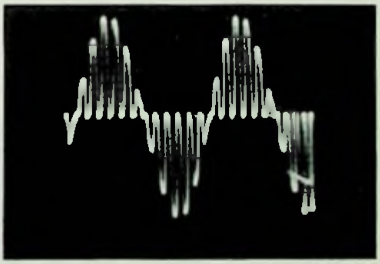


Fig. 2-9. L-only signal with higher-frequency tone.

Since 400 cycles was chosen to illustrate the procedure in the above figures, the presence of the sidebands themselves (37,600 and 38,400 cycles) is seen only as a white blur on the oscilloscope screen in the photograph. To clearly illustrate their presence, it was necessary to use a much higher tone and to take another photograph, as shown in Fig. 2-9. The modulating tone selected had to be a submultiple of the subcarrier frequency, so that the outline of the envelope as well as of the sideband frequencies could be synchronized on the scope and thus be more understandable to the reader. Note that there are twelve sine-wave excursions of the subcarrier sidebands for each tracing of the modulating tone itself, an indication that the modulating tone was approximately 3,166 cycles per second (38,000 divided by 12).

Pilot Carrier

There is one component missing from the composite signal. Up to this point, the signals discussed have never modulated the transmitter beyond the 90% point. The remaining 10% of spectral space has been reserved for a pilot carrier having a frequency of 19 kc (just half the frequency of the originally suppressed subcarrier) and an amplitude which will cause 10% modulation of the FM transmitter (in other words, an additional deviation of ±7.5 kc).

As has already been pointed out, the FM multiplex stereo receiver will have to recover both the L+R and L−R signals (in this example they are both equivalent to L). Otherwise, proper matrixing, as described toward the end of Chapter 1, cannot take place. The L+R audio is recovered at the discriminator output by the action of a conventional FM tuner or receiver. Also available at the output, along with this recovered audio, will be the pair of sidebands representing the L−R information. Being above the frequency limits of human hearing, however, both sidebands will of course be inaudible.

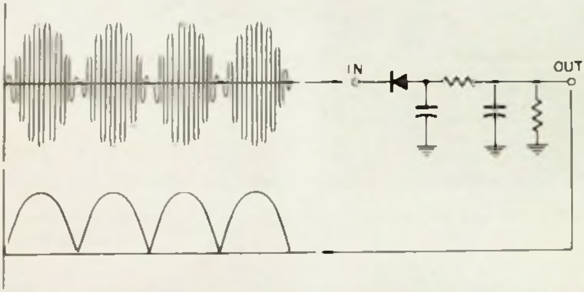


Fig. 2-10. Waveform obtained by rectifying sidebands without subcarrier.

What would happen if the sidebands were treated like an AM envelope. In other words, suppose they merely were amplified and then passed through a conventional AM diode detector to filter out the "residual RF"—just like the 455-kc IF envelope in an AM radio. The waveform recovered would look like Fig. 2-10—almost all second-harmonic distortion (with some higher order harmonics thrown in). None of the 400 cycles would be recovered at all.

Reinsertion of Subcarrier—There is one approach by which the envelope of the L−R information in Fig. 2-6 can be properly detected. This is to insert the "missing" 38-kc subcarrier at the receiving end, by having the receiver local oscillator produce a 38-kc output of sufficiently large amplitude to create an AM envelope waveform suitable for conventional AM detection. As will be more clearly seen later, however, the reinserted, locally created 38-kc output must have exactly the same phase relationship with the pair of sidebands in the recreated AM modulated envelope that it had back at the transmitter before the subcarrier was suppressed.

Oscillator Synchronization—For this reason, it is necessary to transmit a synchronizing pulse which can be used at the receiver to lock the local oscillator in both frequency and phase. This component of the total composite signal is the 19-kc pilot carrier. It is analogous to the horizontal locking pulses, transmitted at 15,750 cycles per second, to help lock the horizontal-sweep oscillator in your TV set.

Composite Signal

Adding this 10% worth of 19-kc pilot carrier to the signal thus far developed (for the left-channel only) results in the composite signal shown in Fig. 2-11. The 19-kc signal can be clearly seen riding on top of the waveform. Again, another photograph (Fig. 2-12), showing a higher modulating frequency

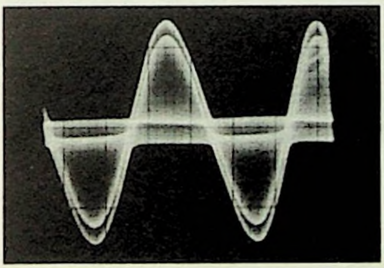


Fig. 2-11. Composite waveform with a left-only (or right-only) 400-cps tone.

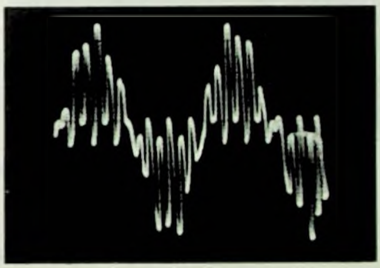


Fig. 2-12. Waveform using higher-frequency modulating tone.

for L, is presented so that details of the sideband frequencies as well as of the modulating tone itself may be clearly discerned.

It should be pointed out that a scope photograph of a right-only signal looks exactly like the left-only photos shown in this chapter, except that the envelope is reversed in phase. However, the synchronization of the oscilloscope will render the two waveforms identical in appearance.

Interleaving—Fig. 2-13 is perhaps the most important illustration in this entire discussion. It depicts one process known as *interleaving*—which, at first glance, might lead one to suppose that 2 + 2 = 2. Fig. 2-13 shows the composite modulating signal when L is 400 and R is 800 cycles. By scrutinizing the photograph, you can separately trace out the sine wave of greater duration (the 400-cycle tone) and the continuous sine wave of shorter duration (800-cycle tone). Note that the total amplitude of the composite waveform is no greater than when an L-only or R-only waveform was illustrated.

To understand clearly why this interleaving takes place and why the laws of addition are seemingly being defied, it will be necessary to consider the make-up of the two major portions of the signal—i.e., the L+R and the L−R sideband envelopes. If, at any moment, the instantaneous voltage of L is 0.45 volt and that of R is also 0.45 volt (though of a different frequency—remember, only one specific instantaneous value is being considered), then L+R will equal .9 volt. At that same instant, however, L−R will be 0.45−0.45, or zero. Suppose, now, that at some other instant L = 0.45 volt but R = −0.45 volt. Then, at this new time L+R will equal zero but L−R will have a

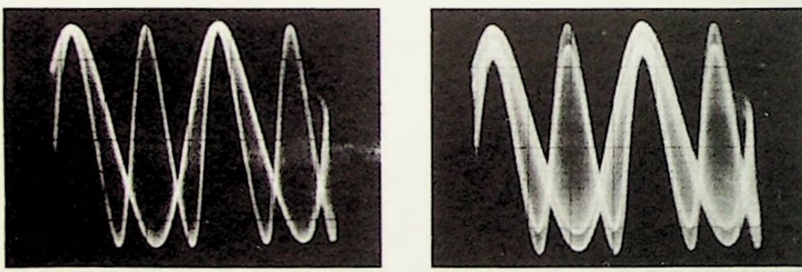


Fig. 2-13. Composite modulating signal with L = 400 cps and R = 800 cps.

sideband amplitude of 0.9 volt. An illustration using square waves (Fig. 2-14), will perhaps further clarify this concept. The important thing to remember is that under conditions of stereo transmission, the main-channel and subcarrier sidebands may each modulate the transmitter to 90% and the resulting instantaneous modulation of the transmitter will never exceed this figure! Had the subcarrier not been suppressed, the above efficiency of spectrum utilization would not be possible. The fixed 38-kc subcarrier would always be present and would use up some finite percentage of modulation, even in the absence of sideband (L−R) information.

Spectral Distribution—Bearing in mind that modulating frequencies from 50 cycles to 75 kc are available in each FM channel and that the amplitude of all frequencies added together must never cause the RF carrier to swing more than 75 kc, you are now ready to examine the spectral distribution of the radiated RF power under conditions of stereocasting. Fig. 2-15 lists the nature and spectral occupancy of the various components. As was shown in Fig. 2-1 (monophonic transmission only), the main-channel modulation consists of frequencies between 50 and 15,000 cycles per second. Their maximum amplitude at any instant must not be greater than the amplitude that will cause

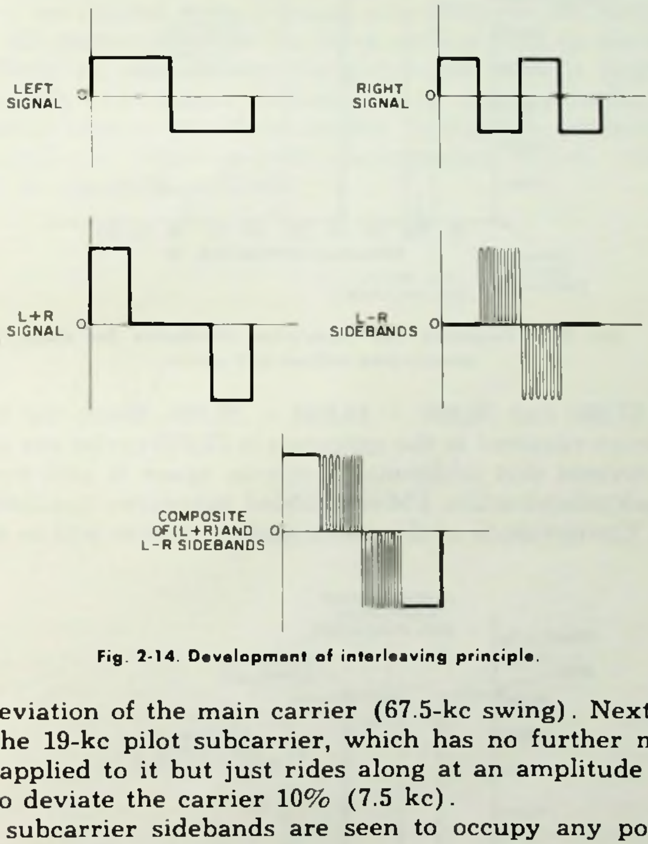


Fig. 2-14. Development of interleaving principle.

90% deviation of the main carrier (67.5-kc swing). Next consider the 19-kc pilot subcarrier, which has no further modulation applied to it but just rides along at an amplitude sufficient to deviate the carrier 10% (7.5 kc).

The subcarrier sidebands are seen to occupy any possible frequency from 23 to 53 kc, except from 37,951 to 38,049. Recall that the upper and lower sidebands are removed in frequency from 38 kc by an amount equal to the instantaneous L−R modulating frequency. Since L−R may at various times also contain musical frequencies ranging from 50 to 15,000 cycles, the limits of sideband frequencies present are automatically dictated. For the upper sideband, 38,000 + 50 = 38,050 and 38,000 + 15,000 = 53,000. For the lower sideband, 38,000 −

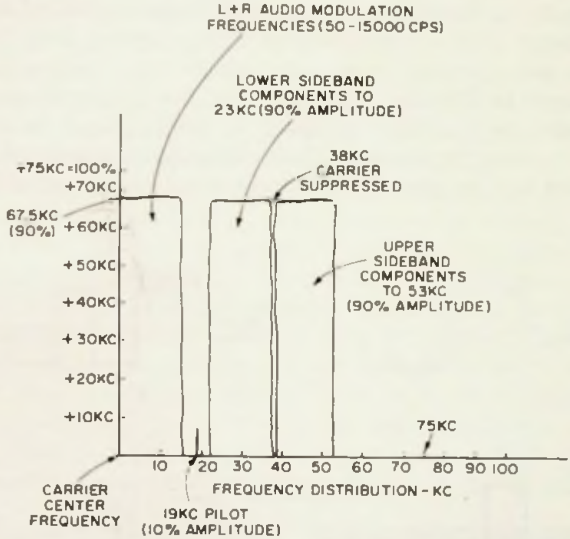


Fig. 2-15. Frequency and modulation distribution for stereo broadcasting without SCA service.

50 = 37,950 and 38,000 − 15,000 = 23,000. Since the highest frequency required in the spectrum is 53,000 cycles per second, it is obvious that additional spectrum space is still available for background-music FM-modulated subcarrier application at 67 kc. The deviation of this additional subcarrier will be limited

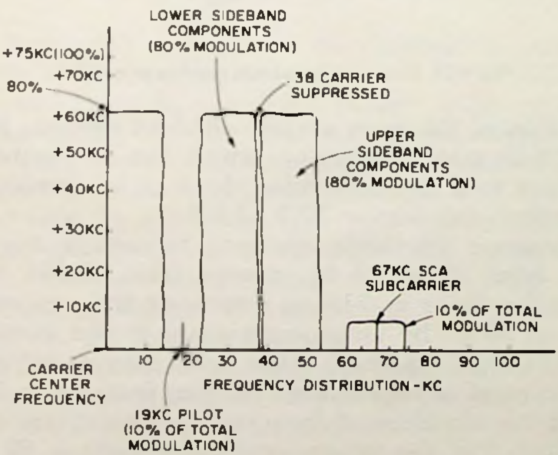


Fig. 2-16. Spectral distribution for stereo and SCA broadcasting.

to about ±7 kc, so that its lowest frequency excursion will be at 60 kc. This provides enough "guard band" between the background-music service and the upper extreme of the stereo frequencies (53 kc) for proper rejection filtering at both the transmitting and receiving ends of the system. Spectral distribution, showing the combined transmission of both stereo and private SCA (background music, etc.) services, is depicted in Fig. 2-16. Note that both the main-channel and subcarrier sideband amplitudes are limited so as to cause only 80% (60 kc) deviation of the RF carrier, because the SCA service uses up the extra 10%. Even so, monophonic signal-to-noise ratio is degraded only slightly more than 1 decibel, a figure which even the most discerning listener would not notice. It was this preservation of monophonic signal-to-noise performance which weighed heavily in the FCC decision.

CHAPTER 3

CONVERTING TO FM STEREO

With the advent of stereo FM, perhaps the most immediate problem confronting the audiophile is conversion of present equipment. The advisability of retaining one's present equipment and adding adapter or converter circuitry hinges upon a variety of factors which will be dealt with in this chapter. From the serviceman's point of view (and from the audio-equipment salesman's as well), offering advice to the potential stereo convert is a serious business. It's one of those dilemmas in which incorrect suggestions are likely to be offered solely because sufficient data about the capabilities of the customer's present equipment is lacking.

SENSITIVITY

In general, quality stereo reception makes more demands of every element in the system, from antenna to tuner output, than monophonic. FM receiver sensitivity, unlike that of AM, involves more than just a statement of "so many microvolts at the antenna input for so many volts of recovered audio output at the discriminator." There are, at present, two accepted ways of indicating receiver or tuner sensitivity.

I.R.E. Quieting Method

In recent years, FM set manufacturers have been indicating receiver sensitivity by giving the number of microvolts required at the antenna input terminals to provide a certain amount of *quieting* at the output.

This method of specifying receiver or tuner sensitivity indicates how relatively free the set is from noise during the absence of audio modulation at the transmitter. This form of noise is often called "background," or residual, noise.

The quieting figure is obtained by using the test setup recommended by the Institute of Radio Engineers and shown in

Fig. 3-1. An FM signal generator is connected to the antenna terminals of the receiver, and its output adjusted to about 1,000 microvolts. A fairly high-quality generator (such as the one manufactured by Boonton Corporation, Measurement Instruments, Marconi Instruments, etc.) must be used. Inexpensive generators of the "kit" variety, while perfectly suitable for general alignment work on FM receivers, ordinarily do not have accurately calibrated output attenuators. The generator is adjusted for a deviation of 30% of 75 kc (around 22.5 kc), and the modulating tone anywhere from 60 to about 800 cycles.

When the receiver under test is tuned to the generator frequency, the modulating tone will be heard. In the case of a tuner (less amplifier), a scope and an AC VTVM should be connected at the tuner output jack. In the case of complete re-

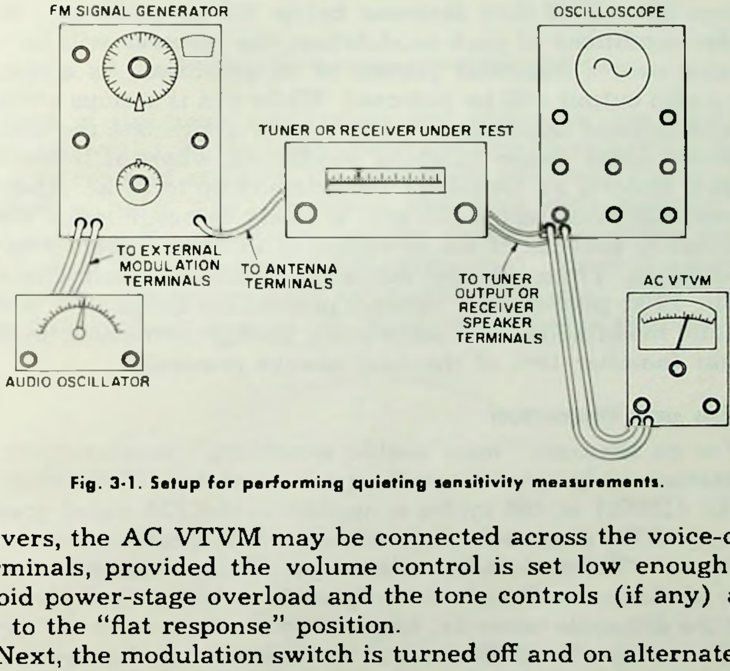


Fig. 3-1. Setup for performing quieting sensitivity measurements.

ceivers, the AC VTVM may be connected across the voice-coil terminals, provided the volume control is set low enough to avoid power-stage overload and the tone controls (if any) are set to the "flat response" position.

Next, the modulation switch is turned off and on alternately, and the level of background noise is plotted against the "signal on" level for various decreasing input values in microvolts. As the signal is made weaker, a point will be reached at which there is a 30-db voltage difference in receiver or tuner output between the time when the audio signal is modulating the generator and when it is not. As the signal is reduced, make certain the tuner or receiver is accurately tuned to the center of the channel, since tuning becomes more critical with weak signals. The number of microvolts required to produce this 30-db difference is the *quieting figure* for the particular unit. Obviously, the smaller this number, the more desirable the receiver, all other things being equal. This means that for 30 db of quieting, 5 microvolts is better than 10 microvolts.

IHFM Least Sensitivity Method

More recently, the Institute of High Fidelity Manufacturers (IHFM) has come up with what is perhaps a more meaningful sensitivity measurement called "least usable sensitivity." This particular measurement takes into account both background-noise limitations and distortion. In most receivers, not only does the background noise increase as the input signal is decreased but the IF and discriminator bandwidths tend to decrease as well. If they decrease below 150 kc (±75 kc), then under conditions of peak modulation, the receiver will be operating over a nonlinear portion of its passband. As a result, the audio output will be distorted. While this is serious enough in monophonic listening, you can readily appreciate the consequences under stereo listening conditions, where it is desired to pass *linearly* all frequency components up to 53 kc. Also, in stereo, full modulation (75 kc), is likely to occur much more frequently because of the presence of at least three forms of modulation. These are the subcarrier-sideband, main-channel audio, 19-kc pilot-carrier (always present for about 10% worth of total modulation) and sometimes, background-music modulation (another 10% of the total always present).

Noise and Distortion

For an accurate "least usable sensitivity" measurement, a distortion analyzer of the null type is necessary. Modulation of 75 kc (100%) at 400 cycles is applied to the FM signal generator, and the distortion analyzer is used to measure *noise* and *distortion*. The modulation is left on at all times, since the function of the audio distortion analyzer is to measure everything *but* the 400-cycle tone—i.e. harmonic distortion, noise (if any), and even hum level. The output level of the signal generator is reduced until the difference between the 400 cycles recovered and the above "residuals" is again 30 db. The number of microvolts then applied to the receiver or tuner is said to be the "least usable sensitivity." The setup required for this measurement is illustrated in Fig. 3-2.

If a harmonic distortion analyzer is not available, it is possible to approximate the sensitivity by setting up a scope at the

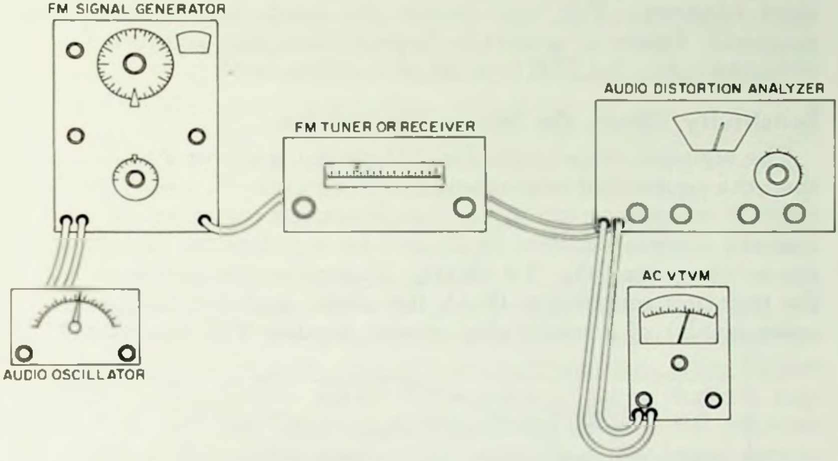


Fig. 3-2. Setup for measuring least usable sensitivity.

output of the tuner or receiver and observing the 400-cycle tone recovered. As the signal-generator level is reduced, a point will be reached at which the sine wave will show evidence of distortion (Fig. 3-3). As drawn, this sine wave contains approximately 3% distortion (or, the distortion products are 30 db below the fundamental 400-cycle tone). There is also evidence of some background noise, but in general, distortion is the factor which reaches the 30-db–below-signal point first in

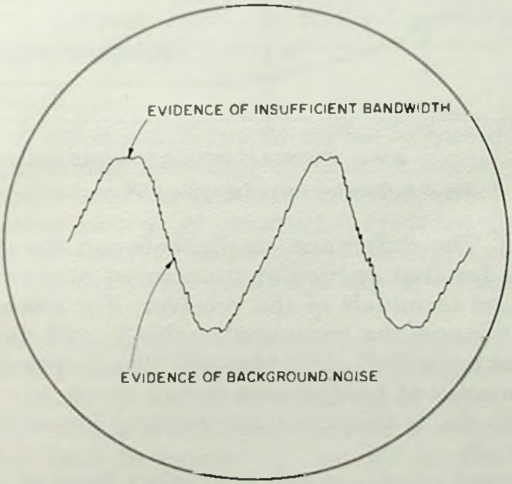


Fig. 3-3. Harmonic distribution in sine wave.

most receivers. For this reason the least usable sensitivity microvolt figure is generally *higher* than the quieting figure obtained using the IRE method of measurement.

Sensitivity Effects On Stereo Conversion

The signal-to-noise ratio for FM stereo is about 20 db *worse* than the equivalent monophonic performance. If, using the IRE method of measurement, a signal-to-noise ratio of 30 db is deemed acceptable, then 50 db will be required for satisfactory stereo reception. Fig. 3-4 clearly illustrates the problem. Here the quieting sensitivity (both the audio and the background-noise levels) of a reasonably priced popular FM receiver have

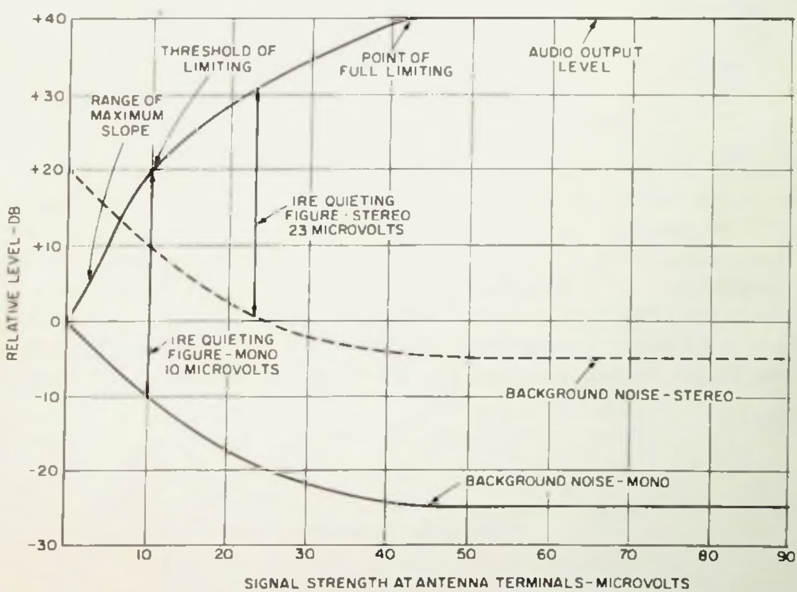


Fig. 3-4. Quieting curve for typical FM receiver.

been plotted. The difference, in db, between the two plots is the quieting for that particular number of microvolts applied to the antenna terminals of the receiver. For example, with a 10-microvolt input, the recovered audio is +20 db. When the modulation is turned off (IRE Method), the output (which then consists primarily of background noise) drops to −10 db. The difference, 30 db, is therefore the quieting figure for an input of 10 microvolts.

Superimposed on this graph is another level of background noise, but 20 db above the monophonic background noise line to correspond to stereo conditions of reception. From this stereo background-noise curve, you can see that approximately 23 microvolts are required at the receiver input in order to achieve the same 30 db of quieting under stereo listening conditions.

Incidentally, don't think that this factor of 2.3 times is a constant, and that it is all you need be concerned about when determining the advisability of converting a receiver to stereo. This particular receiver just happens to have a factor of 2.3. After making this measurement on a variety of tuners and receivers, I have found that sometimes stereo requires nearly a ten times stronger signal than mono. Conversely, some of the more sensitive and more carefully designed receivers will require virtually an unmeasurable increase in input voltage for equivalent performance. Fig. 3-4 helps to explain why. Notice that the audio-output curve is made up of three distinct segments. Before any limiting action by the IF and RF sections of the receivers, audio output will increase as the input microvolts do. Next, limiting begins to take place, and for a time the audio output increases linearly at the rate of 6 db for each 6 db that the input voltage increases. After full limiting is reached (in this example, at an input of about 40 microvolts), an increase in RF microvolts at the antenna will not raise the audio-output level. The nature in which any tuner or receiver develops these three segments of its quieting curve differs from unit to unit (and sometimes even between two similar models, because of production tolerances, etc.). It is therefore advisable to make the measurement described before advising conversion of a given receiver, unless the installation is in an area where the station in question comes in like a "powerhouse" under conditions of monophonic listening.

Signal Strength

Once the quieting sensitivity of the set in question has been determined, it will be necessary to measure the strength of the incoming signal from the station or stations which are or will be broadcasting stereo. A complex, expensive field-strength meter is not necessary. Station signal strength can easily be measured indirectly with your FM signal generator and a DC VTVM.

AGC Voltage Method of Measurement—Fig. 3-5 is a partial schematic of the last IF stage and ratio-detector stage of the new Bogen Model TP-60 FM-AM multiplex stereo tuner. B is the point from which AGC (automatic gain control) is developed and fed back to earlier IF and RF stages. As is well known, the AGC voltage developed at this point is directly

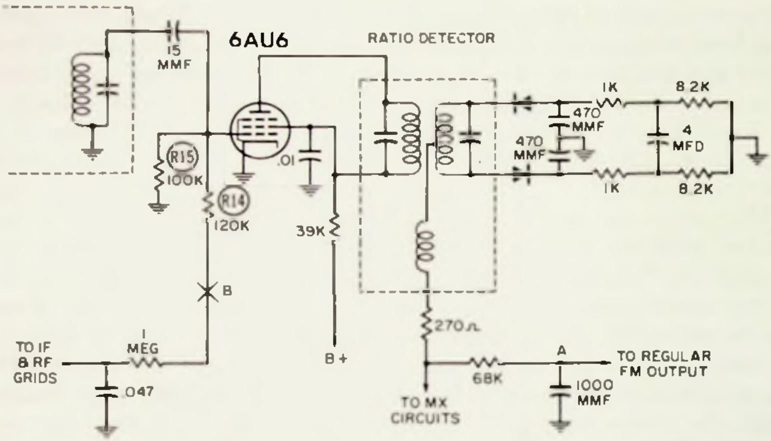


Fig. 3-5. Partial schematic of last IF stage and ratio detector of Bogen TP-60 tuner.

proportional to the input-signal strength. It is developed because of the grid-limiting action of the last IF stage. Since current is flowing through R15 from grid to ground, this voltage has to be negative at point *B*.

A plot of AGC voltage against increasing signal strength at the antenna would look like Fig. 3-6. This curve serves as a

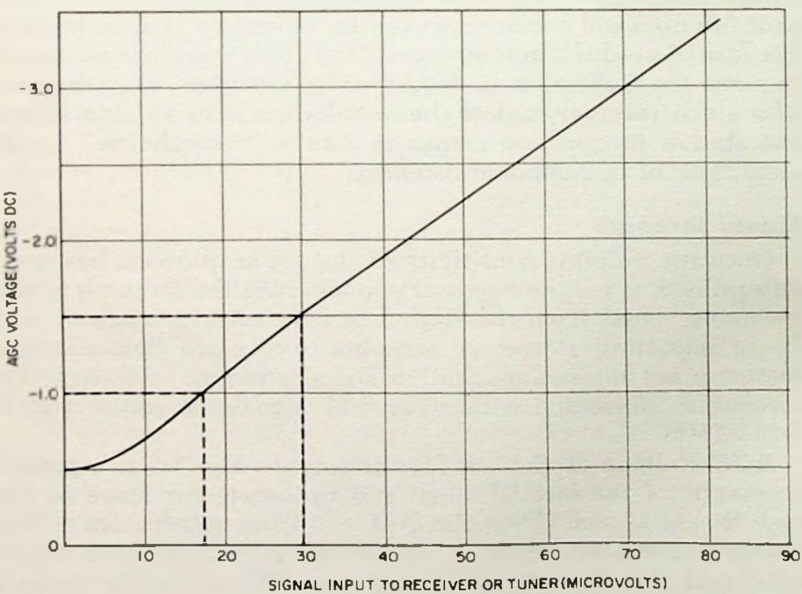


Fig. 3-6. AGC voltage developed at limiter grid.

convenient signal-strength reference. Having made the plot, let's connect the normal antenna to the receiver and carefully tune in the desired station or stations. With the station in tune, suppose -1.5 volts of AGC voltage is developed. By laying a straightedge horizontally across the graph, you will find that the AGC line is crossed at a point corresponding to about 30 microvolts, the signal strength at which the station in question is being received. If Fig. 3-4 represented the quieting capabilities of the given receiver (actually, it does not), you could easily determine that at 30 microvolts, even under stereo listening conditions, the receiver will provide quieting of about 36 db. It would therefore be safe to assume that the station in question will be received satisfactorily in stereo, at least as far as quieting is concerned. There are, of course, many other factors to be considered.

Suppose, however, that when some other FM stereo station is tuned in, the AGC voltage measured is only -1.0 volt. From the curve of Fig. 3-6 you can determine that this voltage represents an incoming signal strength of only 17 microvolts. Referring to Fig. 3-4 once more, you find that under stereo listening conditions, such a signal strength will result in a quieting of only about 25 db—despite the fact that in mono listening, the same station caused 45 db of quieting! At this point, the first thing to do is to assess the antenna installation.

Antennas — Frankly, FM receiver owners have become spoiled. It is not unusual to encounter an FM antenna consisting of nothing more than a short length of wire hidden under a rug in an otherwise elaborate high-fidelity installation. Thus, even in metropolitan areas, where theoretically the signal strength is more than adequate, setups are likely to be encountered which are no better than in fringe-area conditions. Of course, the obvious solution is a better antenna. The problem is more one of education than cost. The FM listener, who has been accustomed to adequate reception monophonically, finds it difficult to understand why his FM tuner or receiver should require a better antenna for stereo reception. "After all," he will protest, "the set hasn't changed." Of course, he fails to realize that the nature of the incoming signal *has* changed—considerably. One of the numerous commercially available outdoor antennas designed for FM reception is heartily recommended if the signal is particularly weak. If the situation is just under par, the use of a *two-set coupler* is recommended. This device enables the user to connect the FM tuner to his outdoor TV antenna with a minimum of interference between the two. While a TV antenna is not quite the right length for the FM band of frequencies, it is close enough to provide a great improvement over most indoor antennas. Orientation of the antenna is important, too. Previously, multipath problems in FM (similar to "ghost" images in TV) created virtually no ill effects. When stereo is broadcast, however, multipath can very often cause the local oscillator to shift phase (and hence vary the phase of the reinserted subcarrier) sufficiently to produce a "swooshing" sound and a shifting of the stereo separation characteristics of the program material. Certainly, the addition of better antenna installation can only improve reception, even where stereo reception is deemed adequate at first. After all, a weaker station in the vicinity may someday "go stereo"—in which case the problem may present itself anyway.

DE-EMPHASIS

Fig. 3-7 is a partial schematic of a typical discriminator circuit, including the normal de-emphasis network and audio-

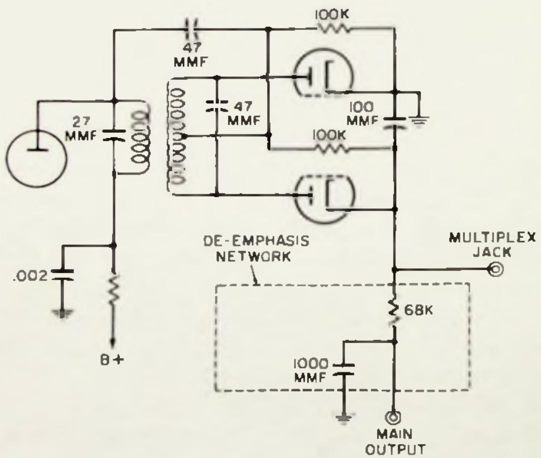


Fig. 3-7. Partial schematic of typical discriminator circuit.

output connection. You will notice that the multiplex jack is situated *ahead* of any de-emphasis components. A word about the meaning and purpose of de-emphasis is in order at this point.

Functions

Unlike in AM, the noise generated in the FM receiver increases directly with the frequency separation between the carrier and the interfering signal, as shown in Fig. 3-8. Further-

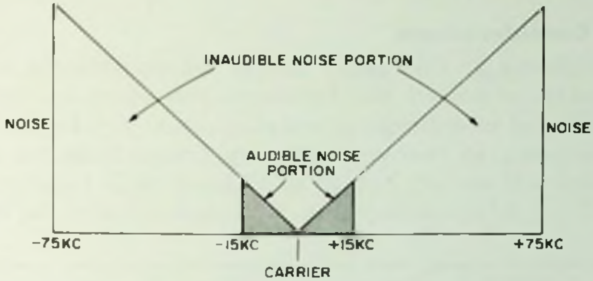


Fig. 3-8. Noise in FM reception.

more, most program or musical energy is contained in the lower or medium audible frequencies, and the most irritating form of random noise is that generated above 3,000 cycles or so. To reduce the effect of this noise, a pre-emphasis circuit is inserted into the audio section of the transmitter. Its function is to emphasize the higher frequencies (at an increasing rate as frequency goes up) in relation to the low and middle frequencies. At the receiver end, a de-emphasis circuit with the reverse properties attenuates the higher frequencies relative to the "lows" and "mids." The frequencies above 1,500 cycles are reduced to their original amplitudes. As the same time, the noise is reduced proportionately. The over-all effect is that the signal returns to its proper relative proportions, but with considerably less noise.

Noise Advantages—Another beneficial effect of the de-emphasis circuit is a reduction in the ever-present random noise. As you have seen, the greater the difference between the carrier frequency and interference source, the greater the FM noise produced (Fig. 3-8). Through the use of the de-emphasis network, the triangular noise response of Fig. 3-8 is modified to that of Fig. 3-9. The over-all noise is considerably reduced.

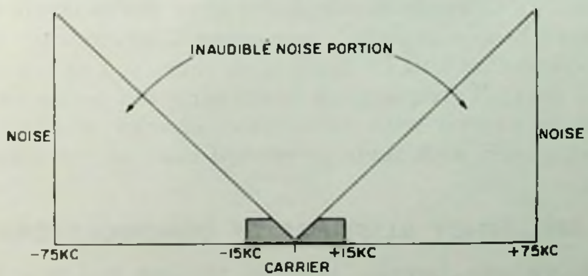


Fig. 3-9. De-emphasis pattern.

Stereo Considerations

De-emphasis at this point is fine where straight mono FM reception is concerned. But for stereo reception, the signal must not be allowed to undergo de-emphasis just yet. Remember that it is necessary to recover not only frequencies up to 15 kc (main-channel audio) but up to 53 kc as well (upper-sideband limit of L—R information). Fig. 3-10 shows that if the composite

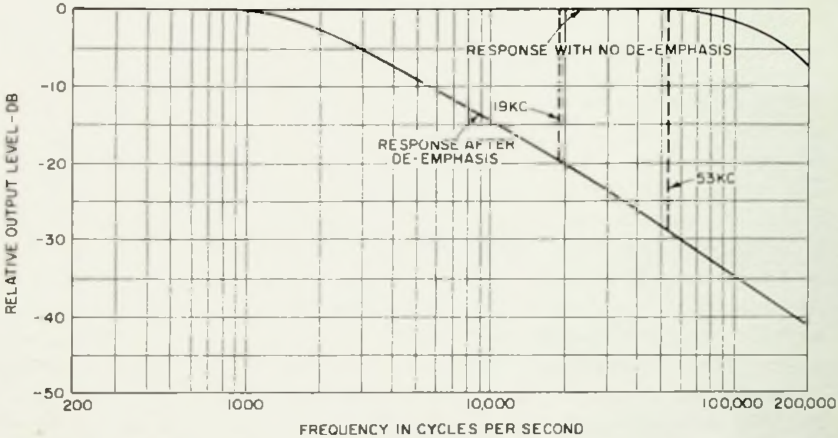


Fig. 3-10. Effect of de-emphasis on multiplex frequencies.

signal were recovered *after* de-emphasis instead of being extracted ahead of it, our 19-kc pilot carrier would be some 20 db lower in amplitude. Thus, if 0.1 volt of 19 kc was available from the composite signal just after the discriminator, only about 0.01 volt would be left after de-emphasis. Furthermore, the upper-sideband extreme frequency (53 kc) would be some 30 db lower than its unde-emphasized value—or, for all practical purposes, unusable. In stereo multiplex circuitry, it is therefore necessary to apply de-emphasis *after* the separate L and R signals have been completely recovered. The circuitry required to accomplish this will be dealt with shortly. For the moment, it is vital that *all* frequencies comprising the composite stereo signal be recovered with no relative change in amplitude or phase, compared with their generated state at the transmitter.

FREQUENCY RESPONSE OF DEMODULATORS

The above consideration leads to another test which should be made on existing FM equipment to determine its suitability for stereo conversion. This is a measurement of the discriminator or ratio-detector output frequency response. Unfortunately, many manufacturers assumed that when multiplex stereo finally arrived, it would consist of a system in which the subcarrier also would be FM modulated. Had such been the case, high-frequency response at the tuner output would have been of no great concern—for if the subcarrier were somewhat attenuated, it might easily be reamplified in the multiplex adapter before being applied to a conventional FM limiter and then some form of discriminator. Amplitude response with change of frequency would have been unimportant (just as it is for regular mono FM).

Effects on Separation and Subcarriers

It can readily be seen, however, that when the signal-carrying part of AM subcarrier sidebands consists of changes in *amplitude,* all amplitude relationships at all frequencies will have to be maintained precisely. Therefore, it is desirable to measure the total frequency response of the discriminator or ratio-detec-

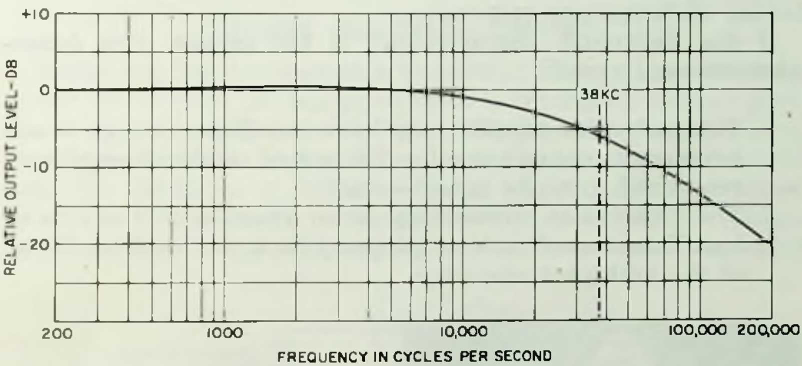


Fig. 3-11. Discriminator response with severe rolloff at subcarrier frequencies.

tor output of the set in question. This is done by applying, to an FM signal generator, external modulation of frequencies from 50 to at least 60,000 cycles per second. If the resultant curve is flat within 1 db or so, little or no trouble should be encountered in converting to stereo. Fig. 3-11 is a plot of tuner response having a fair amount of fall-off at the higher frequencies. In a left-only signal, for example, the amplitude unbalance between the subcarrier sidebands and the main-channel audio would be about 6 db, as shown in Fig. 3-12. Instead of the straight base line in Fig. 2-8 the resultant composite waveform recovered for a left-only signal will have a curved base line, as shown in Fig. 3-13. The theoretical separation then possible (if nothing were done to correct the situation in the multiplex circuitry itself) would be only 10 db. This is barely enough for satisfactory stereo performance. Fortunately, most adapter circuits make provision for such deficiencies. The subcarrier sideband

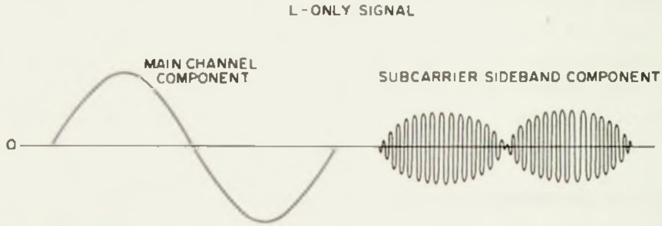


Fig. 3-12. Sidebands recovered in a receiver with discriminator response of Fig. 3-11.

contribution is overamplified, compared with the main carrier-modulation contribution. This is done to offset the reduction in available sideband contribution from the tuner due to its falling-off frequency response.

If the falling-off characteristic is too severe, two consequences may result:

1. The particular adapter may have insufficient range of adjustment to compensate for the lack of sideband amplitude recovered from the tuner section.
2. The 19-kc pilot carrier may be so attenuated that it is of insufficient amplitude to properly lock the local oscillator of the multiplex circuitry.

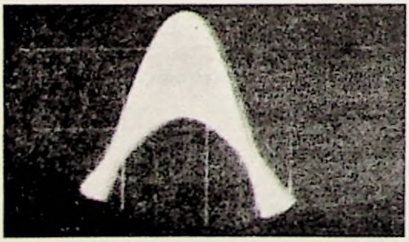


Fig. 3-13. Oscillograph showing unbalanced recovery of main and subchannel sidebands.

Response Measurement Without Generator

A word of caution before the above frequency response measurement is attempted. There are many FM signal generators which are not capable of linearly accepting modulating frequencies all the way up to 60 kc. In other words, the FM modulating circuits of the FM generator have a roll-off characteristic all their own. If there is any uncertainty about the quality of the FM generator in this respect, there is a more direct way of checking frequency response of the receiver output.

Connect a scope, by means of a low-capacity cable, to the output of the discriminator or ratio detector. Tune in a station that is broadcasting stereo at the time, and adjust the vertical gain of the scope so that peaks in the program material present a display over almost the entire height of the cathode-ray tube. Then wait for a moment of silence in the transmission and observe the amplitude of the 19-kc pilot carrier (which is all that will now be present). If the response of the tuner at this point is flat, the 19-kc amplitude will be exactly 1/10 the peak amplitude of the modulation previously observed. If it is 1/20th the peak amplitude, the 19-kc response is down about 6 db from the "main" audio and you may be fairly certain the sideband components will be down much further. Such results would clearly suggest that the given tuner is not properly adaptable to stereo without major rewiring and possible redesign as well.

TUNER MULTIPLEX OUTPUT

Assuming that conversion is the choice, first make certain the tuner or receiver is equipped with a multiplex output jack. Some do not, but can readily be modified to accept such a jack. A standard phono jack will do; it should be connected to the point on the tuner or receiver chassis where audio is obtained from the discriminator or ratio detector *ahead of the de-emphasis network*. Fig. 3-14 and 3-15 show the proper connection

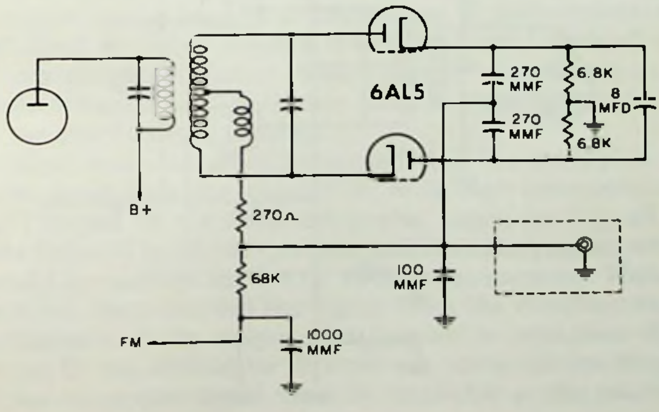


Fig. 3-14. Typical ratio-detector circuit.

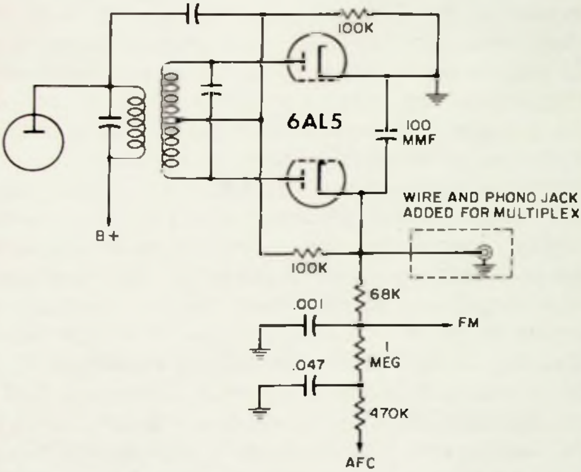


Fig. 3-15. Typical discriminator circuit.

points in the circuits of two common detectors. Some tuners have two audio outputs, for fixed- and controlled-level outputs. It may be convenient to disable the fixed output and make it the multiplex output.

Impedance

Lead lengths should be kept as short as possible; the longer the lead, the greater its stray capacitance, and therefore the

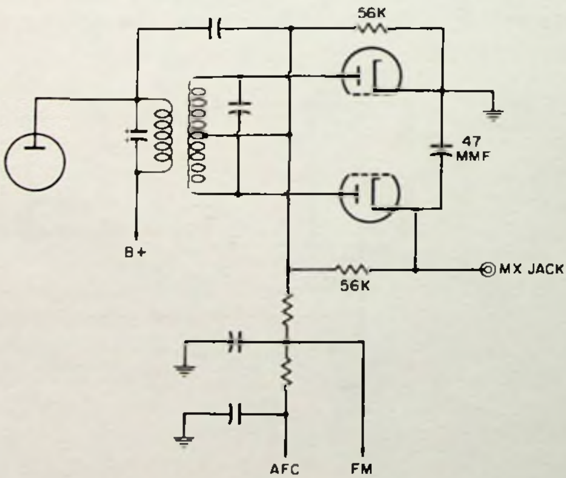


Fig. 3-16. Modified discriminator circuit.

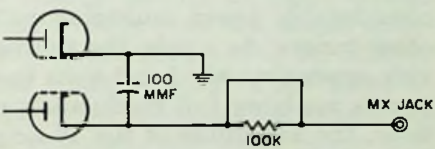
greater the attenuation of high frequencies. This is particularly true in discriminator circuits; their high output impedance makes them more susceptible to the capacitance of any cables attached. Because of this, you may find it beneficial, as I have, to modify the typical discriminator circuit in Fig. 3-15 to look like Fig. 3-16.

Reducing the two 100K resistors to 56K and the diode charging capacitor from 100 mmf to 47 mmf greatly reduces the output impedance of the discriminator and thereby renders it less susceptible to loading by capacitance of cables, for example. Do not attempt this change, however, unless the tuner initially had an output of at least 1 volt or more when full 75-kc modulation is present, because the modification will lower it slightly. Since most multiplex adapters have at least unity gain, however, the slight loss in audio level can in the majority of cases be tolerated easily.

Output-Jack Circuits

Some manufacturers, in preparing for the advent of FM stereo but not knowing which system would ultimately be approved, put series resistors between the discriminator and the multiplex output jack, as shown in Fig. 3-17. Their intent was

Fig. 3-17. Series resistor shorted out.



probably to isolate the discriminator from some form of adapter which might have a low input impedance. If such resistance is present, short it out or replace it with a short length of wire. It can only present a higher "looking back" impedance to any interconnecting-cable capacitance and thereby attenuate the "highs" as previously discussed.

Still other manufacturers presumed that the multiplex jack would be used only for extraction of a high-frequency subcarrier (instead of the total composite signal having all frequencies from 50 to 53,000 cycles), and that the regular output jack would be used for the lower, audible frequencies. With this idea in mind, they coupled the signal from the discriminator to the multiplex jack by means of a low-value capacitor of 100 mmf or so. It, too, should be shorted out, since *all* the frequencies of the composite signal must be available at the multiplex jack.

MATCHING ADAPTER TO TUNER

Much has been written in trade literature and advertising about the adaptability of certain multiplex adapters for use with certain tuners. While the owner of a good FM tuner or receiver must exercise care in the selection of an adapter, he generally need not confine himself to adapters built by the same manufacturer. For one thing, even tuners of one manufacturer vary in design from year to year and from model to model. In fact, no manufacturer we know of has suggested that his multiplex adapter is suitable for use only with certain models of *his own* tuners or receivers.

General Considerations

In general, the detector circuits of nearly all FM tuners or receivers are similar in design and manufacture, falling into the two categories of "ratio detector" or "discriminator" as previously mentioned. One notable exception is found in most of the FM tuners manufactured by H. H. Scott, Inc. Those made in the last several years employ an extremely wide-band IF and discriminator circuit. While ideal in terms of flatness of response and lack of phase shift, such a circuit results in a considerably lower multiplex output level than that of most other tuners. As a rule, the average tuner or receiver will provide approximately 1 to 2 volts from the detector when the station is applying full modulation of 75 kc. Under these conditions, the amplitude of the 19-kc pilot carrier will be between 0.1 and 0.2 volt (since it is transmitted as 10% of full modulation).

In H. H. Scott FM products, the recovered audio has a measured amplitude of between 0.2 and 0.3 volt under conditions of peak deviation of 75 kc. Thus, the amplitude of pilot available will range from 0.02 to 0.03 volt. These tuners usually incorporate a stage of audio amplification in their output circuitry so that the output from the main FM output jack will be equivalent to that of more conventional tuners and receivers. With multiplex stereo, however, the signal must be extracted *ahead* of this stage of amplification—i.e., prior to de-emphasis, or at the output of the detector. In most instances, the level (both of signal and pilot carrier) is therefore too low for adapters designed to be used with higher-output tuners or receivers. H. H. Scott does, however, manufacture extremely well-designed adapters for use exclusively with its own tuners and complete receivers.

About most tuners and receivers this much must be said, in all fairness. *No* tuner designed, produced, and aligned some time ago is ideally ready for *any* commercially available "universal" stereo adapter. On the other hand, almost every such adapter will provide reasonably good stereo when correctly used with almost any tuner or receiver. Performance is dependent largely on the quality of the tuner or receiver (as well as its present state of alignment, which is well worth checking prior to any attempts at conversion to stereo). All sets we have tested provide at least *some* stereo after conversion. But in many of them the quality of stereo reception, noise level, and stereo separation after conversion left much to be desired.

Output-to-Input Voltage

Let's consider specific criteria relating to both the receiver and adapter. The first pair of major concern are the output voltage available from the multiplex jack (usually stated for conditions of full 75-kc modulation) versus the input voltage required by the adapter being considered. The two numbers should be the same, of course, or at least the ranges of voltages required should overlap. Thus, a tuner with a stated multiplex output voltage of 1.2 volts for 75-kc modulation might be expected to work well with an adapter having an input voltage specified between 0.5 and 2.0 volts. Aside from this simple specification, very little of any value in the selection of an adapter can be determined from stated characteristics of the tuner or receiver. Of course, some adapters, by virtue of their excellence of design and completeness of circuitry, will work better with a given receiver than other adapters. Moreover, these superior adapters will generally work better with *other* receivers to which they are attached. As is true for so many products, the manufacturer's cost and the ingenuity of his engineers determine the performance. Probably before very long, trade periodicals and other recognized product-testing groups will evaluate the myriad of FM products now on the market.

Component Concept

In view of the complexities already outlined, the prospective buyer who has neither FM nor stereo is definitely better off buying a tuner or receiver with the multiplex circuitry already built in. Granted, this is contrary to the rest of the "component hi-fi" philosophy—which generally encourages a separate tuner, amplifier (sometimes even separate preamplifier and power amplifier, speakers, record player, etc.—and to which I heartily subscribe).

For example, suppose that after building the best FM circuit he can within a given price range, a manufacturer finds that his discriminator still has a rolling-off response (usually accompanied by a phase-shift characteristic) like that in Fig. 3-11. Incorporating a corrective inverse response in the first amplifying stage of the multiplex decoder circuitry would then be a relatively simple matter. The resultant response would be flat, with virtually no phase shift involved. Furthermore, the correction would hold from set to set of a given design. Thus, better stereo separation is more likely to be obtained if kept under the manufacturer's total control.

INSTALLATION

Installation of adapters with existing tuners and receivers is relatively simple, particularly for one who has interconnected components into his mono or stereo hi-fi system. Two common cases relating to hi-fi component type of installations, will be illustrated first. Finally, the complete FM receiver (containing a single speaker and previously operated monophonically) will be discussed, since most exsisting FM facilities probably are in this category.

Separate Component System

Fig. 3-18 illustrates the normal interconnections in a stereo system which includes a stereo amplifier, two speaker systems,

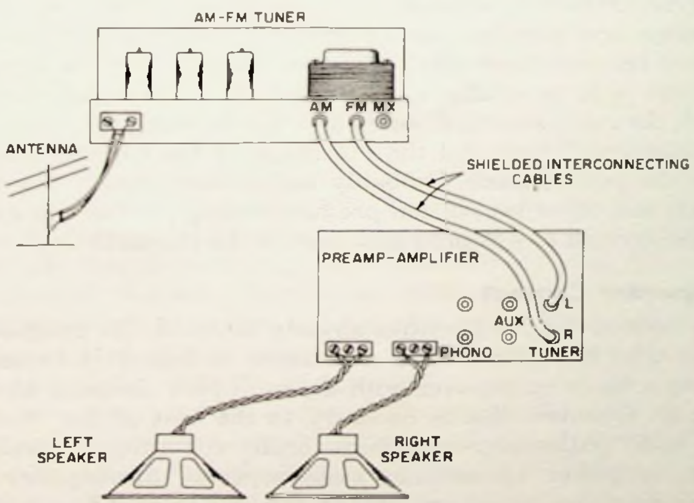


Fig. 3-18. Typical stereo component setup.

and an FM-AM tuner (of the popular simulcast, or simultaneous FM and AM, type). It has been connected to receive interim AM-FM stereo broadcasts. Note that the AM and FM output jacks of the tuner are interconnected to the appropriate input jacks on the preamp-amplifier by the usual pin-to-pin shielded cables. Also note that two speakers are already connected to the stereo amplifier, for it is utilized for stereo records, too. The MX jack on the tuner is not yet in use.

Fig. 3-19 shows the "after" view in the "before-and-after" sequence. A cable has been connected to the multiplex adapter

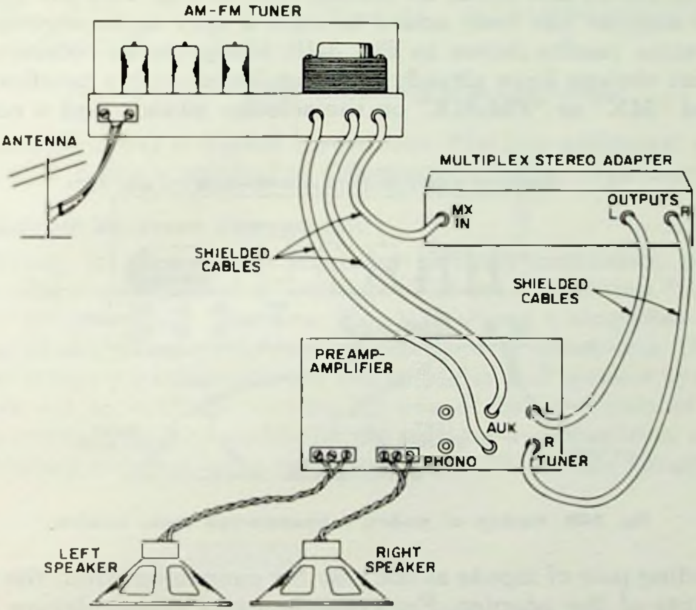


Fig. 3-19. System after multiplex adapter has been added.

from the MX jack on the tuner. Left and right output cables are connected from the adapter outputs to the two tuner inputs of the amplifier. (Actually, any pair of high-level amplifier inputs will do; these are usually labeled "auxiliary," "TV," or "spare.") In addition the AM and FM outputs of the tuner are still connected to another pair of inputs to the amplifier. These cables (not needed for FM or FM stereo listening) are maintained because the listener may be in an area where AM-FM stereo is still broadcast, or he may wish to hear AM programs (monophonically) which are not available on FM. The third reason is that some weak-signal FM stations will be noisier if listened to *through* the adapter rather than directly, as previously explained. In some adapters, Mono-Stereo switches eliminate this problem by deactivating all or part of the subchannel circuitry. Even so, the user may prefer not to have to manipulate a control on the adapter, since it may be hidden out of sight behind the tuner, for example.

Component-Receiver System

Fig. 3-20 illustrates the use of a complete receiver (still of the component hi-fi type with separate speakers) before the advent of FM stereo and needs no elaboration. In Fig. 3-21 the multiplex adapter has been added in such a way as to accomplish the same results shown in Fig. 3-19. Many stereo receivers of recent vintage have already been equipped with a position labeled "MX" or "FM-MX" on the selector switch, and a corre-

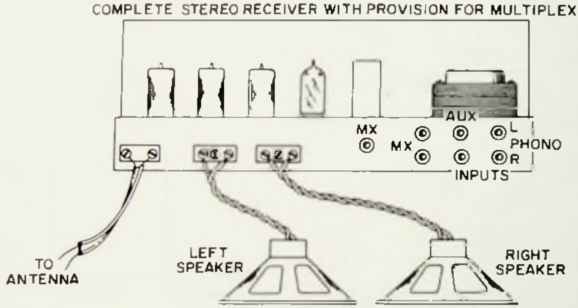


Fig. 3-20. Hookup of modern component-type stereo receiver.

sponding pair of inputs at the rear for connection from the two outputs of the adapter. For consistency of nomenclature and simplicity of operation, these inputs should, of course, be used. If no such inputs are delineated, any pair of high-level stereo inputs may be used, as before, in making connection from the output of the adapter. The shielded cable for the adapter input must be kept as short as possible. Its inherent parallel capacity is 20 to 100 mmf per foot, and the loading effect of a substantial length of cable might be enough to seriously reduce the amplitude (and alter the phase) of the higher-order frequencies of the composite stereo signal. For this reason, some manufacturers are supplying precut lengths of high-quality, low-capacity cable for this connection. Do not use a substitute if such a cable is supplied. If it is left to your discretion, keep the lengths no greater than one or two feet. In fact, low-capacity cable (under

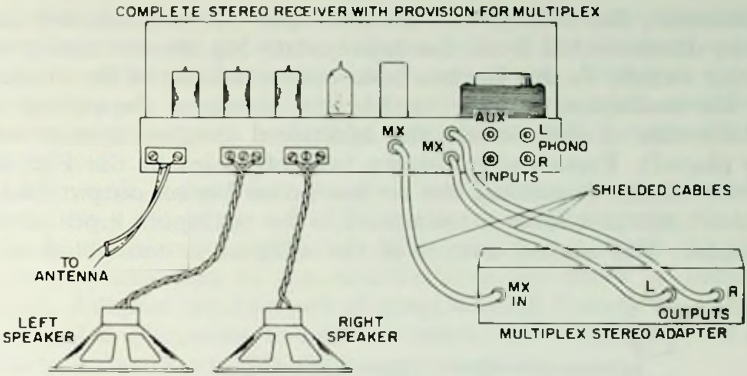


Fig. 3-21. Interconnection after addition of multiplex adapter.

50 mmf per foot) is a good investment. The few additional pennies may make a noticeable difference in stereo reproduction.

Packaged Receiver Conversion

Finally, let us consider the more difficult installation problem—the conversion of a "package" FM set (integrated "furniture" console unit) to stereo. Fig. 3-22 shows a simplified diagram of the usual switching arrangement of these sets. Obviously it would be desirable for the amplifier and speaker system in the set to continue serving as one of the channels of the final stereo setup. Accordingly, the selector-switch wiring must be altered somewhat, as shown in Fig. 3-23. In this simplified

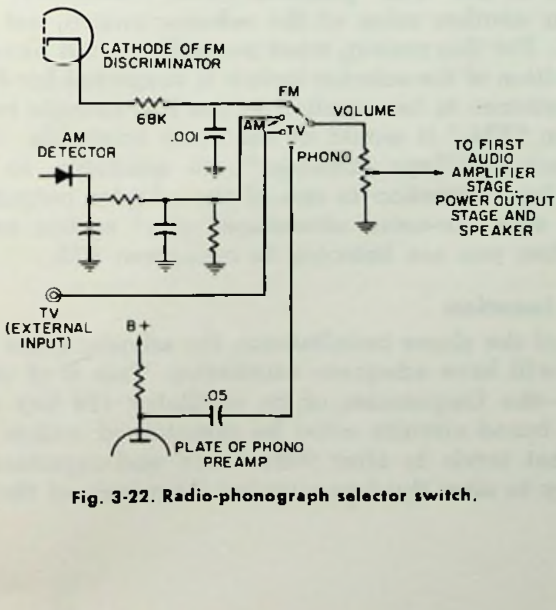


Fig. 3-22. Radio-phonograph selector switch.

schematic, the normal FM tuner output (de-emphasized) has been disconnected from the appropriate lug on the rotary selector switch. To this lug has been connected *one* of the outputs of the multiplex adapter (either left or right, depending on which side of the console the additional speaker system will be placed). From the multiplex take-off point in the FM detector circuit (assuming the set has no multiplex output jack), a short shielded cable is connected to the multiplex input of the adapter. The second output of the adapter is connected to a

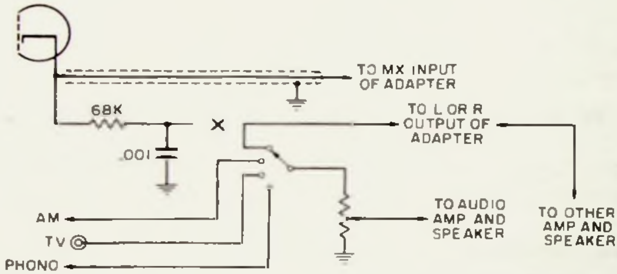


Fig. 3-23. Switch rewired to accommodate multiplex adapter.

separate amplifier which feeds the second-channel speaker system. The latter is appropriately positioned in the listening area for best stereo effect.

To prevent a drain on the power supply, in many sets of this type no B+ voltage is supplied to the FM circuits unless the selector is in the "FM" position. (This changeover is accomplished by another rotor of the selector switch, not shown in Fig. 3-23). For this reason, what normally would have been the "FM" position of the selector switch is suggested for FM-stereo. If B+ continues to be supplied to the FM circuits in positions other than "FM," it would be far more advisable, if external inputs such as "Tape Recorder" are available, to use such an input for connection to one of the adapter outputs. In this way, the signal-to-noise advantages cited earlier can be obtained when you are listening to nonstereo FM.

Adapter Location

In any of the above installations, the adapter must be placed where it will have adequate ventilation. This is of utmost importance—the frequencies of its oscillator (19 kc) and other critically tuned circuits must be maintained within stringent limits. Heat tends to alter inductance and capcitance values sufficiently to alter the free-running frequency of the 19-kc oscillator. While extremely stable, the temperature-compensated components used in most of these critical circuits may not be able to offset extreme rises in temperature. Adapters should therefore not be placed next to power-output tubes, rectifiers, power transformers, and other components which normally become hot.

Above all, if your first stereo listening experience proves disappointing in terms of signal-to-noise performance, do not blame the newly purchased adapter just yet. Look to your antenna, and then to the sensitivity of the tuner or receiver itself. Adapter troubles are of great enough variety to warrant a detailed discussion in later chapters. However, they are not likely to be responsible for noisy reception.

CHAPTER 4

MULTIPLEX DECODER CIRCUITS

From the foregoing chapters it is apparent that whether you are concerned with so-called multiplex adapters or complete FM-stereo tuners (or complete receivers, for that matter), the actual circuitry will be the same. In analyzing circuits or in troubleshooting, we shall confine ourselves to that portion of the FM receiving system beyond the discriminator or ratio detector, but ahead of the stereo audio amplifier. To attempt a complete discussion of FM would be a monumental task indeed and inconsistent with the purposes of this book. Those who feel that their background knowledge of regular FM circuitry and techniques needs refreshing might well consult one of the many excellent books on the subject.

This chapter will be concerned with a general explanation and detailed analysis of the various circuits which comprise the multiplex stereo decoder. Circuit and block diagrams of commercially available sets will be relied upon heavily. Obviously, many excellent manufacturers produce this sort of equipment. The choice of diagrammatic material, therefore, is more for the purpose of illustrating a particular circuit feature or innovation than for specific recommendation of equipment.

A block diagram of an early adapter is shown in Fig. 4-1. This diagram will be used in tracing the steps in the decoding action of nearly all forms of multiplex circuitry employing the *matrixing* principle. Later, adapters using the time division switching approach will be examined.

OPERATION OF MATRIXING CIRCUITS

In the basic FM receiver, the complete composite signal is recovered from the FM carrier and presented at the multiplex output jack. From here it is connected, through a shielded cable, to the multiplex adapter. Remember that this signal con-

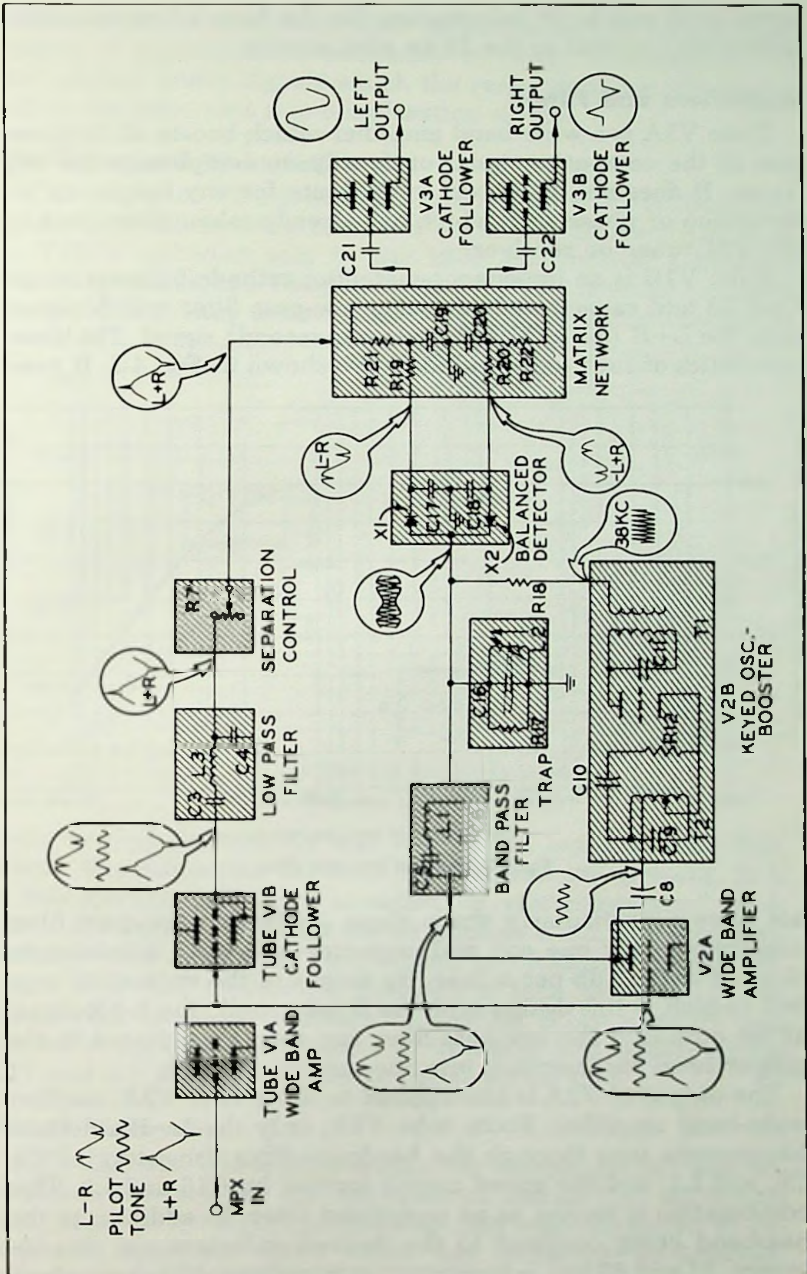


Fig. 4-1. Block diagram of Heath AC-11 adapter.

tains L+R and L−R information (in the form of superaudible sidebands) as well as the 19-kc pilot carrier.

Amplifiers and Filters

Tube V1A is a wide-band amplifier which boosts *all* frequencies of the composite signal uniformly, to compensate for any losses. It does not, however, compensate for any frequency attenuation or phase shift which has already taken place back at the FM tuner or receiver.

Tube V1B is an impedance-matching cathode-follower stage. Coil L3 and capacitor C4 act as a low-pass filter which passes only the L+R (50- to 15,000-cycle-per-second) signal. The characteristics of such a general filter are shown in Fig. 4-2. It need

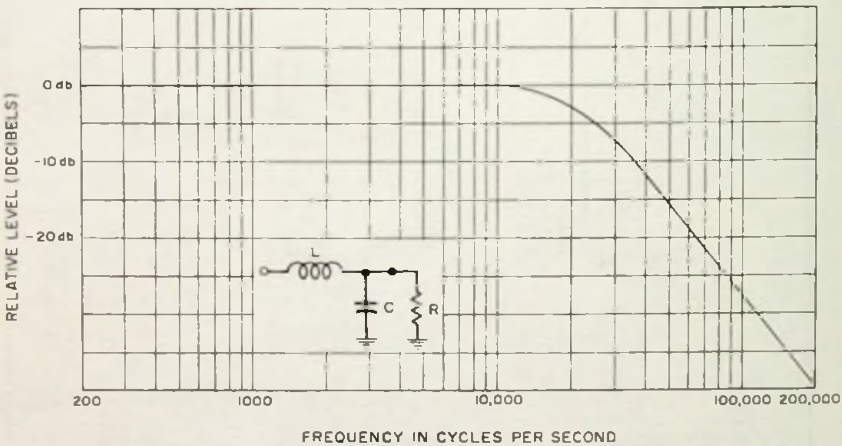


Fig. 4-2. Curve of low-pass filter

not have a particularly sharp slope. In fact, a low-pass filter comprising only one coil and capacitor will have a maximum slope of only 12 db per octave. By means of the separation control (which in this design is on the front panel), the L+R signal at the output of the low-pass filter can then be adjusted to the proper level for insertion into the matrix network.

The output of V1A is also applied to the grid of V2A, another wide-band amplifier. From tube V2A, only the L−R sideband components pass through the bandpass filter consisting of C5, C6, and L1, and the tuned circuit formed by C16 and L2. This combination is known as an *m*-derived filter. In addition to the passband being confined to the desired extremes (in this instance, 23 and 53 kc), a frequency exists (here, 67 kc) at which a high degree of attenuation or rejection takes place. The response of this five-element filter is shown in Fig. 4-3. A high degree of attenuation is necessary to reject any storecast or background music signals which the radio station might transmit at the same time it is broadcasting stereo.

Oscillator and Re-insertion

The output of V2A is also fed through C8 to a synchronized oscillator, V2B. Because of its self-resonant frequency and high Q, V2B is locked-in only by the 19-kc pilot-carrier portion of the composite signal. The plate circuit of V2B is tuned to 38,000 cycles for doubling action. The 38-kc output of the doubler is

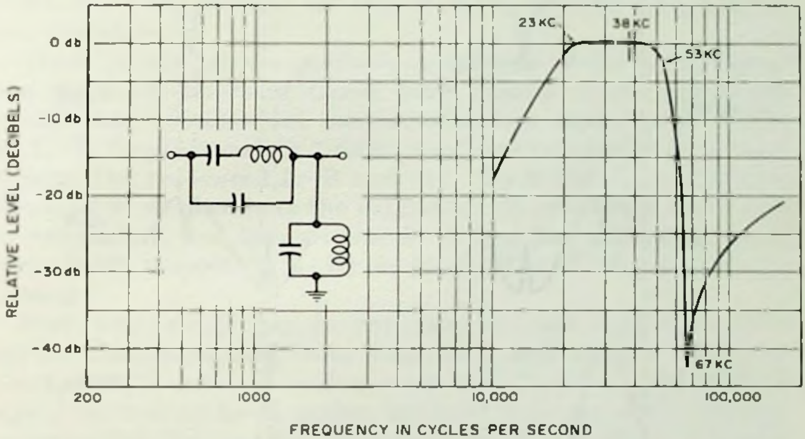


Fig. 4-3. Response of bandpass filter.

then coupled through T1 and R18 to the common load R17, where the subcarrier sideband components are present. It is at this junction point that so-called "carrier reinsertion" takes place. A scope at this point would disclose a most conventional looking "AM modulated" waveform, as illustrated in the diagram. This AM modulated envelope is then detected, or demodulated, by diode detectors X1 and X2. Since they are connected in opposite polarities, a L−R signal will appear across C17 and a −(L−R) or —L+R signal across C18.

Matrix Networks

R19, R21, and C19 form a matrix (or adding circuit) and de-emphasis network for the left channel. Recall that up to this point, none of the signal components had been subjected to de-emphasis, since the entire signal was extracted, as detected, prior to any de-emphasis network in the tuner or receiver. R20, R22, and C20 are the counterparts for the right channel.

The matrix, or resistive adding, circuit recombines the L−R and −L+R signals with the L+R signal as follows:

If L+R is added to L−R as shown in Fig. 4-4, the instantaneous sum will be 2L. The spikes, or R portion of the waves, cancel each other. But the rest of the waves, or L portion, reinforce each other to reproduce the original left-channel sine

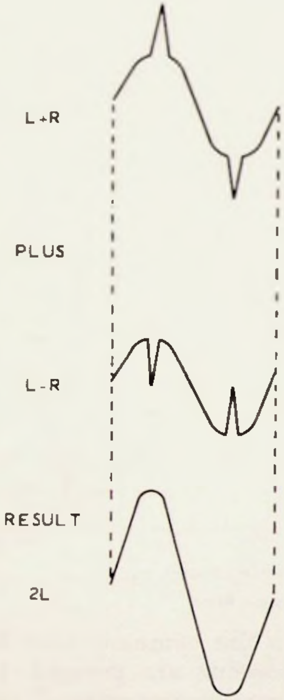


Fig. 4-4. Adding L + R to L — R.

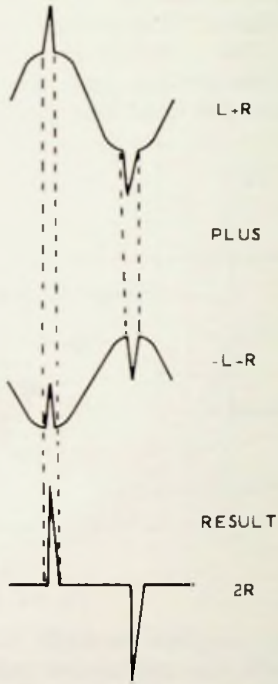


Fig. 4-5. Subtracting L — R from L + R.

wave. This signal is fed from the output of the matrix, through C21, to the grid of tube V3B.

If L+R is added to −L+R (just another way of saying that L−R is *subtracted* from L+R) as shown in Fig. 4-5, the result will be 2R. The sine waves, or L portion of the signal, will now cancel each other. But the instantaneous sum of the spikes (R) will reinforce each other to produce 2R. The output of this side of the matrix network is fed, through C22, to tube V3A.

Output

V3 is a dual cathode-follower stage that provides a low output impedance. Thus it prevents excessive signal loss or induced hum when the adapter is connected to the stereo amplifier (or pair of mono amplifiers) through long shielded cables. R25 and R28, in the cathodes of V3A and V3B, are right- and left-channel output-level controls.

Separation Control

The channel-separation control was mentioned briefly in discussing the L+R section of the block diagram. It will now be analyzed in detail, since doing so will help in an over-all understanding of how the adapter operates. Moreover, the channel-separation control will be encountered again and again as representative commercial circuits of the matrixing type system are examined.

Theoretically, if the available amplitude of the L+R signal and the L−R sideband signal were exactly equal, and if the rectification efficiency of the demodulating diodes were 100%, an L+R "separation" or "dimension" control would be unnecessary. The recovered L−R and the −L+R signals would each be equal in amplitude to the original L−R created at the transmitter matrix, and this original L−R had just sufficient amplitude (with respect to L+R) to cause 2L and 2R upon rematrixing.

Such ideal conditions do not occur, as you have seen from earlier discussion and from Figs. 3-12 and 3-13. In the first place, the FM detector seldom recovers as much L−R sideband signal as it does L+R audio, because of losses at the higher superaudible frequencies. Nor does the simple diode demodulation process described in the block diagram (Fig. 4-1) have 100% rectification efficiency, either. That is, if the modulation on the input envelope waveform has an amplitude of 1 volt peak-to-peak, the audio recovered at the diode output (after the remaining 38 kc has been filtered) will be somewhat less than 1 volt peak-to-peak. (The exact rectification efficiency depends on the amplitude of the applied signal, and on the ratio between the load impedance of the diode and its internal resistance. The efficiency tends to be consistent for large signal voltages, but falls off when they are only a few volts.)

For these reasons, the amplitude of the L+R signal must be separately controlled so that its amplitude ratio to the L−R sidebands is the same as it was in the original transmission. Only in this way can matrixing yield "pure" L and R signals.

But what happens if the L+R and L−R amplitudes are not exactly equal just prior to matrixing? Suppose, for example, that 0.5 (L−R) is added to and subtracted from L+R in the usual matrixing network (that is, not enough L−R is matrixed in for pure L and R recovery in the output). The algebraic results of matrixing will be:

$$(L+R) + 0.5(L-R) = 1.5L + 0.5R$$
$$(L+R) - 0.5(L-R) = 1.5R + 0.5L$$

In other words, the supposedly L-only signal will contain about one-third residual R signal, and the so-called R-only signal the same amount of residual L signal. This is like saying that the maximum separation possible is only 10 db.

Blend Control—Many stereo amplifiers include a "Blend" control, which deliberately enables the user to reduce stereo separation by mixing in a little of the left back into the right and vice versa. Such a control has some merit—particularly when one is listening to exaggerated stereo recordings, which are sometimes displeasing because of the extreme stereo effects. Others find the Blend control useful where the speakers must be placed far apart in the listening area (usually because of decor considerations). In any event, setting the separation, or dimension, control on a multiplex adapter so that *more* L+R than L−R is provided to the matrixing network will have the same effect as a Blend control set between full stereo and mono operation.

Suppose, however, that the L+R contribution to the matrix is *less* than that of L−R and −L+R. Let us examine the algebraic results of this situation:

$$0.5(L+R) + (L-R) = 1.5L - 0.5R$$
$$0.5(L+R) - (L-R) = 1.5R - 0.5R$$

Being a bit more unusual, these recovered signals must be analyzed carefully. The so-called L signal will again contain about one-third R, but this R content will be out of phase with the R coming from the opposite speaker. Similarly, the R signal will contain one-third L but this residual amount will also be out of phase with the remainder from the L speaker. The audible effect (provided it is not carried to extremes) is to actually enhance or intensify the stereo effect, or to make the separation seem greater than it was on the original recording.

If such intensification seems impossible, consider what took place at the recording studio in the first place. Two microphones were set up to record the two channels. The left microphone, while picking up substantially "left side of orchestra" sounds did, after all, pick up *some* from the right side as well. It is this total musical pickup that has so far been called the "left-only" signal—meaning only that it is the total signal recorded from the left microphone; nevertheless, it may contain some sounds, however slight, from all locations. Now suppose on reproducing this sound, that some "negative right" (in moderate amounts) is added to the left-only signal (which already contains some positive residual right signal from the original pickup). It follows that whatever quantity of right signal was originally picked up by the "left only" microphone will be further cancelled. The result in theory should be (and is in actual audible demonstration) a further purification or more complete isolation of the left-only signal. The same argument holds for the opposite channel also, of course.

Now, all of this is brought up for only one reason. Some manufacturers claim that their separation controls can adjust from full monophonic (when only L+R is added in to the matrix) to an enhanced stereo effect (when more L−R than L+R is added into the matrix). This claim, if properly interpreted, is actually true. Of course, if no L+R is added in, you would be listening to L−R from one speaker and −(L−R) from the other. This would be equivalent to some sort of mono listening—and with speakers out of phase at that. So, do not get the idea that by setting the separation control to the opposite extreme you will be able to achieve infinite separation, because it just doesn't happen.

ACTUAL MATRIXING CIRCUITS

Now let's examine, stage by stage, the multiplex circuitry used in various matrix adapters. The first stage is generally an amplifier, which provides gain to the entire composite signal and also isolates later stages from the discriminator or ratio-detector output to prevent severe loading.

Input Stages

Fig. 4-6 is a simplified schematic of the input stage of the Heath AC-11 (Fig. 4-1). The signal is coupled to the grid of this stage by a 0.1-mfd capacitor. The grid resistor, being 470K, is large enough so that no phase shift or amplitude attenuation occurs at low frequencies (which must be passed at this point) down to 50 cycles. By leaving the 1,500 ohm cathode bias resistor unbypassed, degeneration takes place. This has the effect of further increasing the effective input grid impedance, insofar as its possible loading effect on the discriminator is concerned. Of course, this unbypassed cathode resistor also limits the maximum gain that may be achieved from the triode. But this is of no consequence, since very little gain is required at this point.

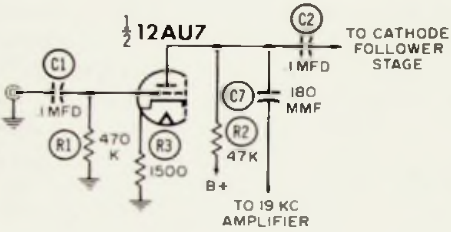


Fig. 4-6. First stage of Heath AC-11 adapter.

The plate-circuit signal of this stage, is passed on to two points. A large coupling capacitor passes the entire signal to the second amplifying stage, where it will be separated into L+R and L−R sidebands by suitable filtering. Simultaneously, a 180-mmf capacitor couples only the higher frequencies to another stage, where the 19-kc pilot carrier will be used to lock the local oscillator.

Fig. 4-7 is a partial schematic of the first stage of the Bogen Model PX-60 adapter. Except for a somewhat different grid

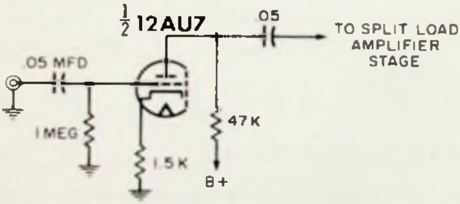


Fig. 4-7. First stage of Bogen PX-60 adapter.

load resistor value, its design approach is the same as the circuit previously described.

The first stage of the Crosby Model 101 circuit is shown in Fig. 4-8. Here, the input grid circuit is the same as that shown in earlier circuits, except that the distribution of outputs is to three distinct circuits, right from the first stage instead of two. A 1.0-mfd capacitor couples the signal to the low-pass filter (L+R only) which will follow. The desired 19-kc pilot carrier

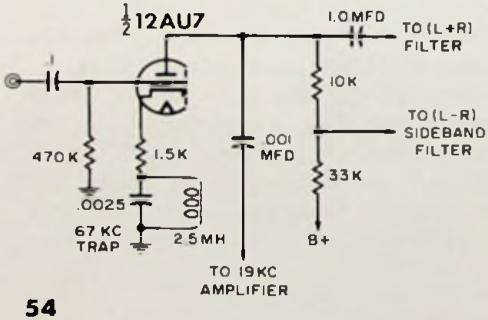


Fig. 4-8. First stage of Crosby 101 adapter.

is taken from the entire plate load (10K and 33K in series) for maximum amplitude and is passed through a low-value capacitor for subsequent amplification and locking of the local oscillator. The (L−R) sidebands are taken from the junction point of the 10K and 33K plate load resistors for application to the bandpass filter (23 to 53 kc) to follow. This method of coupling provides not only the desired amplification of the various signals, but also the relatively low input impedance needed for proper operation of the bandpass filter to follow.

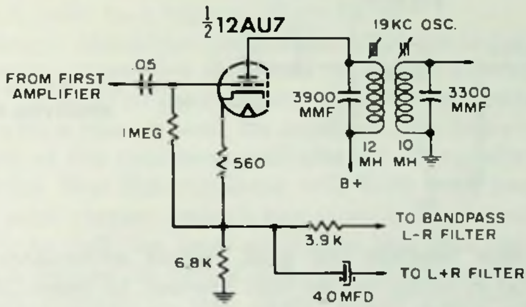


Fig. 4-9. Second (split-load) stage of Bogen PX-60.

Second Stage

Usually, a cathode-follower stage follows the first amplifying stage of the multiplex circuitry. This second stage isolates the various elements of the composite signal, as shown in Fig. 4-9. Its plate circuit is tuned to 19 kc by means of a parallel L-C tank circuit. From the cathode load of this Bogen circuit, the L+R audio and subcarrier L−R sidebands are extracted. Each signal will be fed to its respective filter for proper isolation from the other.

As shown in Fig. 4-10, the 19-kc pilot carrier and the L−R sidebands had been coupled to another amplifying stage prior to the second signal stage. In this case, the cathode follower

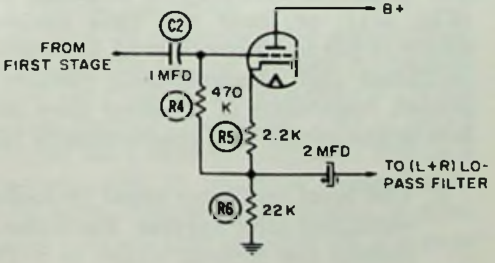


Fig. 4-10. Second stage of Heath AC-11.

merely serves as a take-off point for the L+R part of the stereo signal. The low-impedance output characteristic of the cathode follower is ideal for feeding the low-pass L+R filter which follows.

In the second stage (Fig. 4-11) of the Crosby MX-101 circuit, only the 19-kc pilot carrier is amplified (other elements of the signal already having been directed to their respective filters in the plate circuit of the first stage). The .001-mmf cou-

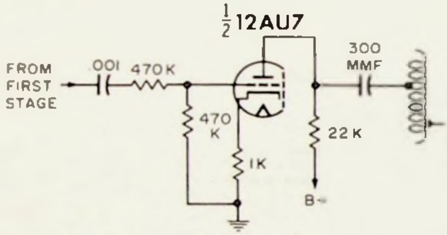


Fig. 4-11. Crosby 101 19-kc amplifying stage.

pling capacitor feeding the grid circuit attenuates the low-frequency L+R components still present to some extent. The series 470K resistor, aided by the input capacitance of the grid circuit of the 12AU7, helps to roll off the higher frequencies of the sideband signal so that the 19 kc is already favored for amplification at this point.

Oscillators and Doublers

Because the 19-kc oscillator to follow is a high-*Q* tuned circuit, one might suppose that only the 19-kc pilot will trigger or lock the oscillator. Nevertheless, residual quantities of L+R and sideband modulation information, if amplified through the oscillator stage and subsequent doubler, might modulate the 38-kc carrier slightly, in such a manner that distortion is introduced and separation adversely affected.

Local Oscillator Requirements—The design and synchronization of a local oscillator are much the same, whether the problem of stereo recovery is attacked from the matrix approach (Fig. 4-1), or from the time division (switching) approach which is yet to be considered in detail. For this reason, as many oscillator and synchronizing circuits as possible will be presented, regardless of whether they are extracted from one system or the other. The requirements for each form are the same:

1. The local oscillator must be locked to exactly 19 kc by the incoming pilot carrier. Facilities must be provided for adjusting the lock-in phase to within 3° of the phase of the

incoming 19-kc pilot carrier if separation in excess of 30 db is to be achieved.

2. The local oscillator must operate with sufficient amplitude so that, after subsequent doubling, there is enough 38-kc voltage for proper distortionless carrier reinsertion (or, in the case of switching circuits, sufficient voltage to properly and positively polarize the diode detectors for conduction and nonconduction in the presence of a sideband component). At the point where the 38-kc voltage is applied to the demodulating diodes, amplitudes vary from a low of about 4 volts to a high of 20 or 25.
3. Frequency and phase stability is perhaps the most important requirement for the local oscillator. Obviously, if the L or C element of the oscillator tank circuit varies appreciably with a rise in heat, for example, the self-resonant frequency of the oscillator will change. Very often this does not mean that the oscillator will drift with respect to the 19-kc pilot carrier (which has usually been amplified sufficiently to pull the local oscillator in), but only that substantial errors in phase may occur. This results in decreased separation and often, more distortion. Care in layout of components for minimum heat conduction, and choice of proper temperature-compensating capacitors and stable inductors, are as important here as they are in the design of the local RF oscillator in superheterodyne FM or AM front-ends.

Consider, for a moment, the tolerances involved. With a fairly strong locking signal available, a free-running drift of about 10 cycles is enough, under locked conditions, to alter the phase relationship by about 7°. In turn the stereo separation would be reduced to a theoretical maximum of 24 db (provided the amplitude relationships were perfect at the time).

A change of 10 cycles in 19,000 represents a stability of .00053, or approximately .05%. Translated to the more familiar oscillator stability requirements encountered in FM, this would mean a drift of about 50 kc for a 100-mc oscillator.

Ideally, a separation of at least 30 db should be maintained. For this reason, the phase shift (due to any change in component characteristics with heat, etc.) should be less than 3°. Such stability would represent a free-running oscillator drift of about 4 cycles (empirically measured on a typical adapter); this is a stability factor of approximately .000215, or .02%. Again translated to FM RF oscillators, such stability represents a total drift, from a cold start, of only ±20 kc. This is about all one ever encounters in FM front-ends of the most careful and expensive design.

Typical Oscillators—A typical Hartley oscillator, used in the Bogen Model TP-60 AM-FM-multiplex tuner, is shown in the partial schematic of Fig. 4-12. Half of a 6AN8 tube is used. The plate circuit includes a tank circuit tuned to 38 kc, for doubling. The method of injecting the synchronizing signal (19-kc pilot carrier) is somewhat unusual in this circuit—it is applied through transformer action. That is, the primary of T2 is the plate load of the previous pilot-carrier amplifier, whereas the secondary of T2 is the inductance of the L-C tank circuit of the 19-kc local oscillator. Vacuum-tube oscillators tend to synchronize with an injected voltage of approximately their resonant frequency. The amount by which the oscillator frequency will deviate—in order to synchronize with an injected locking signal—increases as the amplitude of that signal does and as the

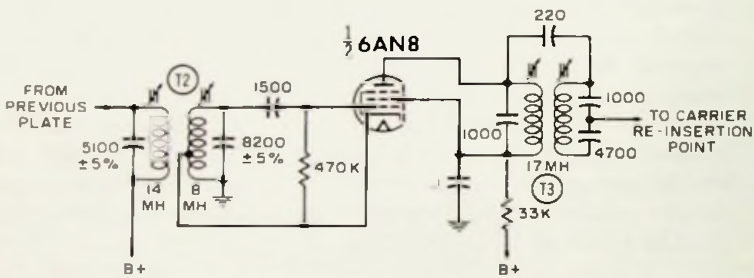


Fig. 4-12. Bogen TP-60 oscillator circuit.

frequency stability or amplitude of the local oscillator decreases. Thus, in this and all other oscillators of the same type, an ideal relationship must be established between amplified 19-kc pilot carrier and the local-oscillation amplitudes. For 10 volts of local oscillation, about 2 to 3 volts of synchronizing signal is ideal.

The waveform shown in Fig. 4-13 represents the doubled, or 38-kc, output of the stage. Note that every other sine wave is somewhat larger than its adjacent neighboring sine wave. The reason is that pure doubling has not taken place; some fundamental 19 kc is still present in the output waveform. This need not cause any concern at this time; it will not affect proper 38-kc carrier reinsertion, but will only appear as a small amount of 19 kc in the rectified audio output of the adapter. The residual 19 kc may be suppressed at the output, as will be shown later, or additional tuned circuits may be employed at this point to purify the 38 kc. Bogen elects to further "clean up" the 38 kc here, by tuning the secondary of T3 prior to using the 38 kc for carrier reinsertion.

Pilot Carrier Amplifiers—Before considering other local oscillators, it might be well to point out what you may have already concluded; that a local oscillator is not even necessary! After all, the 19-kc pilot carrier *is* available in the composite signal. Moreover, if amplified sufficiently and doubled, it can provide the source of 38 kc required, either in the matrix carrier reinsertion or the switching method of detection.

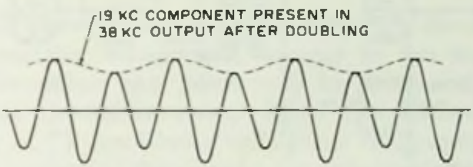


Fig. 4-13. Typical 38-kc output after single-stage doubling.

Such a circuit was, in fact, proposed and demonstrated by the General Electric Company as part of its documentation of the proposed stereo system before the FCC. The system involved the successive amplification of the 19-kc pilot-carrier signal. This was continued until its amplitude was sufficient to be fed to the grid of a doubler tube operating over a nonlinear portion of its characteristic curve, so that doubling would take place in the plate circuit. Provided the amplification is sufficient, there is no reason why this system should fail. If amplification is marginal, however, and the incoming 19 kc is lower than the optimum amplitude, the amount of 38 kc developed in the doubler plate will fall short of the amplitude required for proper carrier reinsertion. As a result, the recovered L-R will be distorted.

Altec Lansing Corporation has approached the circuit design problem from the "no local oscillator" point of view. Moreover, it has made certain that the amount of 19-kc pilot will always be sufficient to do the job, by incorporating sufficient stages of amplification as well as a diode limiter, or clipper, circuit to insure proper development of 38-kc signal. Actually, four full stages of amplification (including the 38-kc doubler) are used. As can be seen from the schematic of Fig. 4-14 these high-gain stages employ three 12AT7 triodes followed by a 6AU6 pentode.

The tank circuit (labeled "pilot phase") in the grid of the first 12AT7 amplifier is tuned to 19 kc. By tuning slightly above or below resonance, however, the phase of the 19-kc pilot carrier may be altered substantially without materially decreasing the amplitude in the grid circuit. In alignment, this adjustment is made to provide 38 kc, at the output of the circuit, having the correct phase relationship with the subcarrier sidebands which may have undergone some phase shift, either in the tuner section or in the multiplex circuit itself. In actual practice, proper

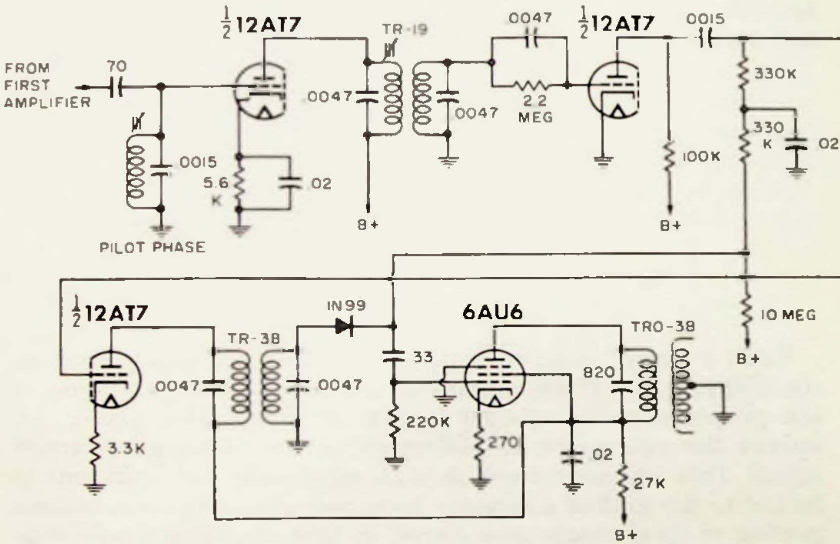


Fig. 4-14. Unusual circuit for developing 38-kc voltage.

phase relationships are established when best separation is achieved. The degree of separation is used as an indicator of proper phase relationship, rather than a direct measurement of phase in degrees.

Further amplification of the 19-kc signal takes place in the three succeeding 12AT7 stages. Tuned-grid or tuned-plate circuits are employed in two of these stages. The grid circuit of the 6AU6 includes a clipping diode which produces a high proportion of second-harmonic component (38 kc). Because of subsequent amplification by the 6AU6 and the presence of a tuned circuit in its plate, the 38 kc is purified and almost all other components are rejected. Thus, the 38-kc carrier has as much amplitude as if a local oscillator had been employed.

Other Typical Oscillators—Fig. 4-15 illustrates the oscillator circuit used by Sherwood Electronic Laboratories in its Model

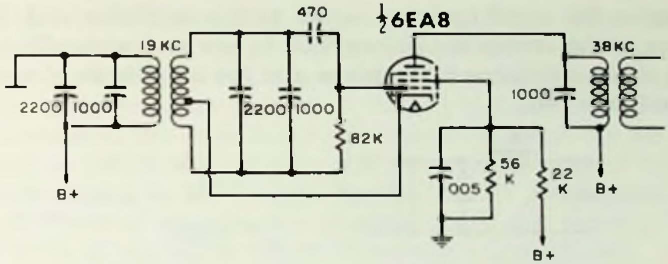


Fig. 4-15. Sherwood S-8000 oscillator circuit.

S-8000 FM-MX stereo receiver. At first glance, the circuit looks identical (except for component values) to the Bogen circuit discussed earlier. One new point may be learned from careful scrutiny of this circuit, however. Note that both the primary and secondary of T1 (oscillator coil) have *two* capacitors (.0022 and .001 mfd) instead of the usual only one—which, in this case, might have been presumed to be .0033 (nearest standard RETMA value to the parallel combination used). In all probability, the manufacturer did not use four capacitors—when two would have done the job—just because he happened to have a heavy inventory of .001-mfd and .0022-mfd capacitors! Rather, he was attempting to properly stabilize tuned circuits with respect to changes in temperature. Small-valued capacitors which *decrease* in capacity with increases in temperature are available. Conversely, most inductances exhibit an increase in inductance with a rise in temperature. Thus, by careful choice of a capacitor so that its reduction in capacitance will raise the resonant frequency to an amount exactly equal to the lower frequency caused by the rising inductance, nearly perfect temperature compensation may be effected.

A more common way of locking the local oscillator is shown in Fig. 4-16. Except for actual component values, it is the approach used in the Heath AC-11 and Crosby MX-101 adapters and in the Crosby RMX-30 FM receiver. A 300-mmf capacitor

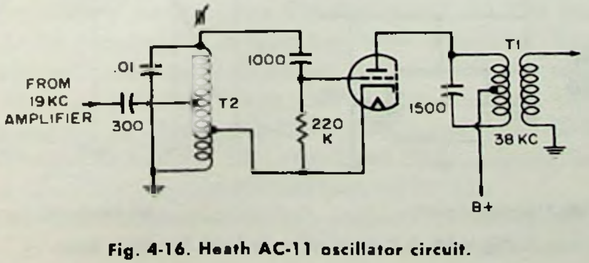


Fig. 4-16. Heath AC-11 oscillator circuit.

couples the synchronizing signal to the oscillator coil. By coupling to the center tap rather than to one end, virtually no loading of the oscillator takes place and the amplitude of oscillation is not reduced.

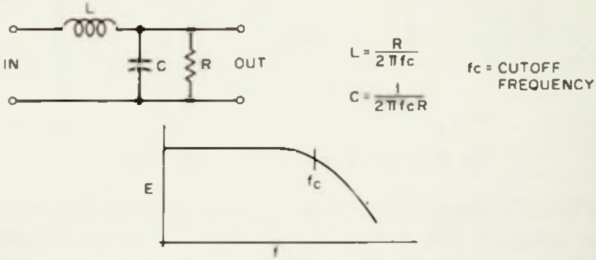


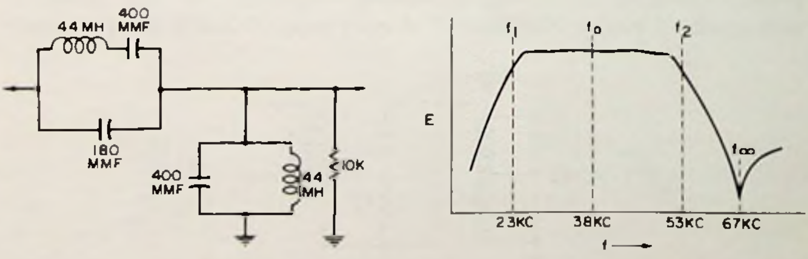
Fig. 4-17. Constant-K low-pass filter.

Filters

All the matrix adapters described so far require certain bandpass, low-pass, and single-frequency filters. But adapters and multiplex circuits of receivers which employ time division switching require frequency filters of a somewhat different configuration. Before analyzing the demodulating process in both types of circuits, let's examine the filter requirements of multiplex stereo circuitry in a general sense.

The simplest filter required in a matrix circuit is the low-pass type for the L+R signal path. Fig. 4-17 shows the usual circuit of this filter, the design formulas, and a plot of the frequency response. The terminating and source impedances should be as nearly equal as possible to insure flat, unpeaked response in the passband. In general, 5% tolerance components should be used in production.

Fig. 4-18 illustrates a practical circuit, component values, and desired response of an *m*-derived bandpass filter used in con-



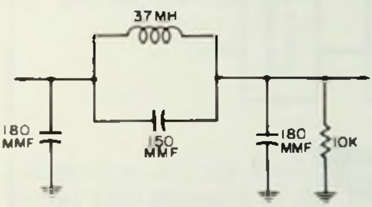
(A) Practical circuit.

(B) Response curve.

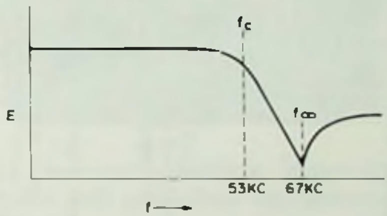
Fig. 4-18. A 5-element, *m*-derived, bandpass filter.

nection with matrix circuitry. An *m*-derived filter differs from the simple constant-K type (Fig. 4-17) in that the circuit performs *two* functions. First, a passband is established. That is, all frequencies between 23 and 53 kc are passed through the circuit unattenuated. In addition, a single frequency (67 kc, the frequency at which the background music subcarrier is transmitted) is caused to have theoretically infinite attenuation. In actual practice, of course, the attenuation is not infinite, but can be 40 to 50 db below the midfrequency amplitudes.

Fig. 4-19 shows another low-pass filter, this time of an *m*-derived configuration and with a greater frequency range of bandpass. As will be seen shortly, in the "time division switching" approach to demodulation, the entire composite signal



(A) Practical circuit.



(B) Response curve.

Fig. 4-19. Practical low-pass filter for switching demodulators.

(from 50 cycles to 53 kc) must be fed to the demodulating circuit. It is still imperative that frequencies around 67 kc (background-music subchannel frequency) be rejected. For this reason, the lowpass filter of Fig. 4-17 would not be adequate.

OPERATION OF "TIME DIVISION" CIRCUITS

Before analyzing the various demodulators, or detectors, used in stereo circuitry, let's examine another fundamental decoding circuit. Sometimes called switching or time division multiplex, this technique has been referred to in a general way throughout the chapter. The block diagram (Fig. 4-20) of the multiplex circuitry used in the Crosby RMX-30 FM stereo receiver, will be helpful in explaining how it works. First, let's deal quickly with the early, familiar parts of the circuit. As before, the first amplifying stage raises the level of the entire composite signal. A 19-kc amplifier, oscillator, and 38-kc doubler are all used as before. The resultant 38-kc signal, however, will *not* be used for carrier reinsertion as before.

The amplified composite signal, instead of being broken up into an L+R component and L−R subcarrier sideband compo-

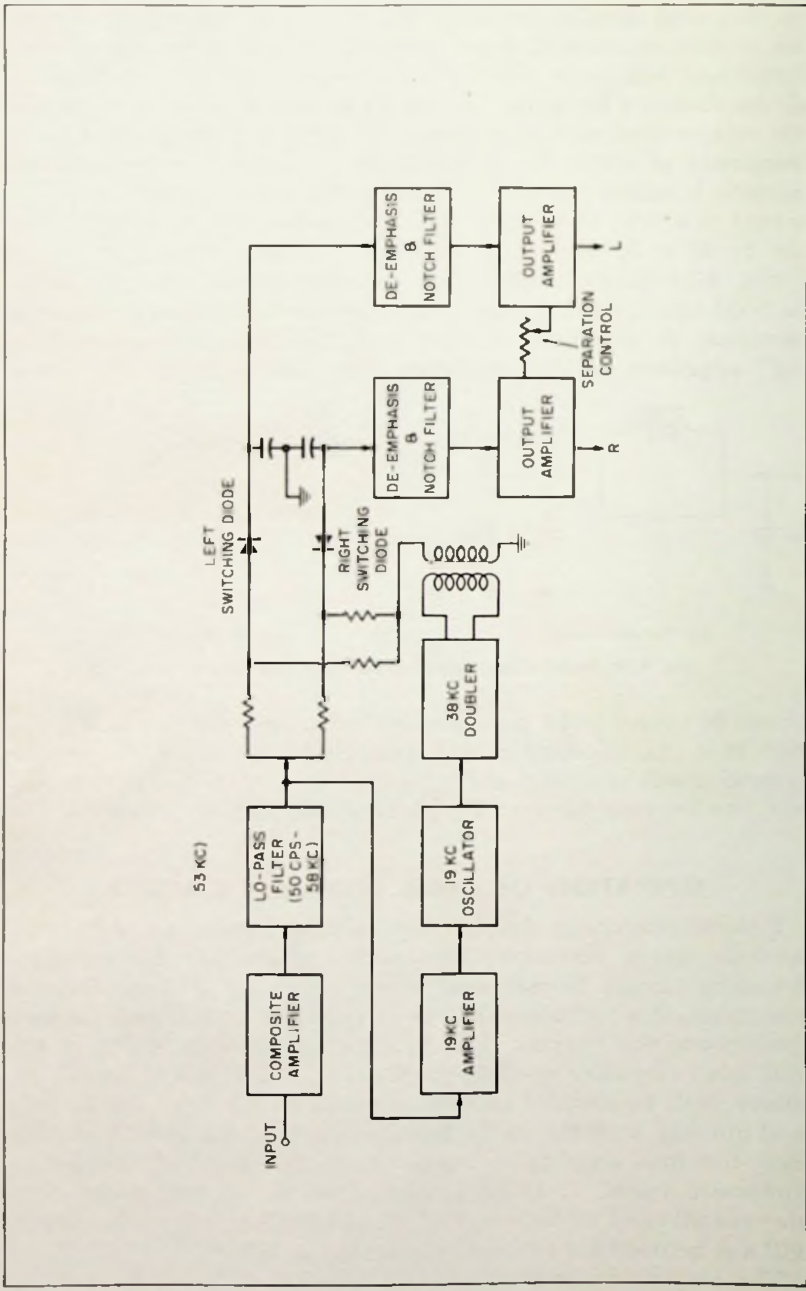


Fig. 4-20. Block diagram of time-division decoder.

nents, is passed intact through a low-pass filter having a cutoff frequency of 53 kc. The waveform seen at the filter output is therefore the total composite signal as previously presented in the scope photograph of Figs. 2-11 and 2-12—or if the 19-kc pilot component is ignored, as shown in Figs. 2-8 and 2-9. For the purpose of this discussion it will be helpful to display a composite signal (less 19-kc pilot) wherein the modulating frequency of a left-only tone is even higher than that presented in

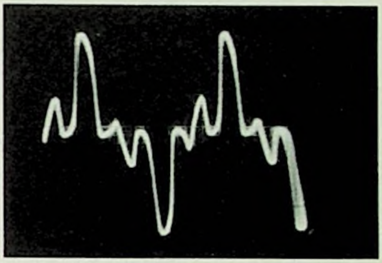


Fig. 4-21. A left-only signal.

either Fig. 2-9 or 2-12. Fig. 4-21 represents such a left-only signal (the absence of the 19-kc pilot will not affect the discussion for the present).

Detection

The composite signal, therefore, is applied to each of the detecting diodes (which are polarized oppositely from each other). If no other voltages were applied to these diodes, the output from the diode labeled L would be positive half-wave rectified pulses, as shown in Fig. 4-22, and not the sine-wave

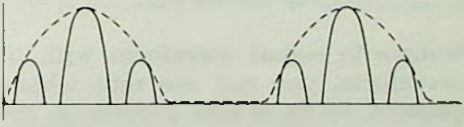


Fig. 4-22. Results of detecting diode L signal "as is."

outline which we know to be the original left-only signal. As for the R diode, its output would also be half-rectified pulses, but in a negative direction—when in fact the R diode should have produced *no* output, since we have been dealing with a left-only signal.

Referring to Fig. 4-20, you can see that in addition to the composite signal, the two diodes are also fed a polarizing or switching voltage at a 38-kc rate. The phasing of this 38 kc is precisely arranged to coincide with correct periodic on-off switching of each diode so that it conducts only when a left signal (in the case of the left diode) or right signal (in the case of the right diode) is present.

The left-only signal of Fig. 4-21 has been carefully redrawn (Fig. 4-23) and directly below it, the 38-kc switching voltage is plotted. The diodes are so polarized that when a positive voltage is applied at the junction point of the two diodes, only the L diode will conduct, whereas when a net negative voltage is applied, only the R diode will conduct. By comparing the

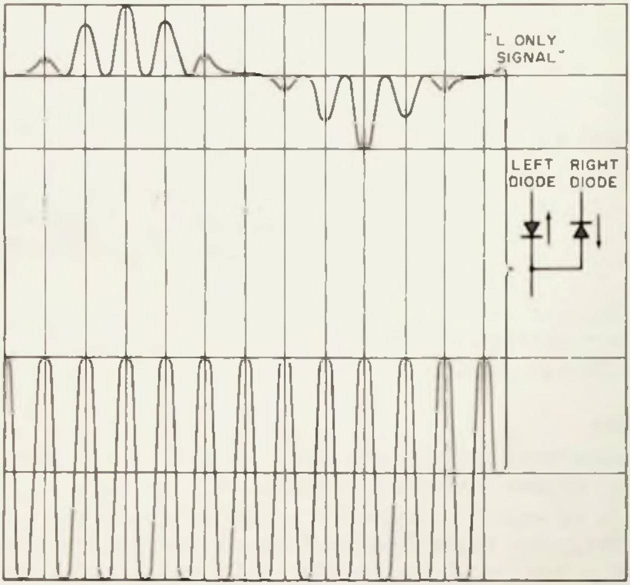


Fig. 4-23. Left-only signal (above) and switching voltage (below).

composite signal waveform with the 38 kc switching-voltage waveform, you can see that whenever the L signal pulse is present, 38 kc is also present in the polarity that causes conduction of the L diode. This is true even when the L composite signal peaks go negative (over the second half of the audio sine-wave cycle), for at this time the positive 38-kc switching voltage is still positive by a greater amplitude than the signal itself is negative. The net polarization, therefore, is still positive as far as the L diode is concerned and so it conducts. Prior to any further filtering, the recovered signal at the output of the L diode would appear as in Fig. 4-23. After filtering (just as in any RF detection), the 38-kc components would be smoothed out and a single sine wave traced out. This sine wave is, of course, the original left-only signal.

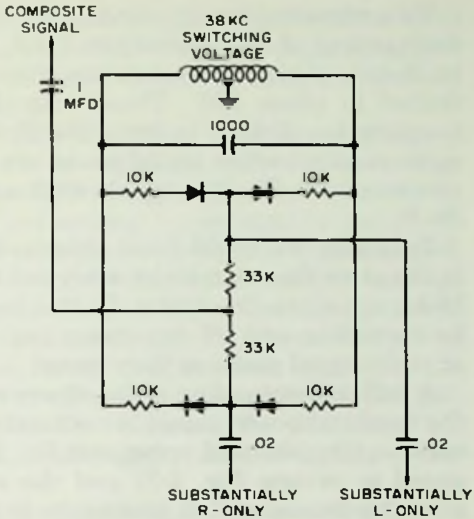


Fig. 4-24. Sherwood S-8000 demodulator circuit.

To complete the picture, consider what the R diode has been doing all this time. Recall that the R diode is in a position to conduct when the junction point of the two diodes is *negative*. Further examination of Fig. 4-23 will disclose that every time the R diode *is* negative, no composite signal modulation is present. Therefore, the only net conduction will be of the 38-kc switching voltage; and this voltage will be filtered out after detection, just as it was for the L diode. Thus, while some average DC level will be established at the output of the R diode, no AC or audio voltage will be developed. This is, of course, as it should be, since with an L-only signal there obviously is no R signal to be recovered.

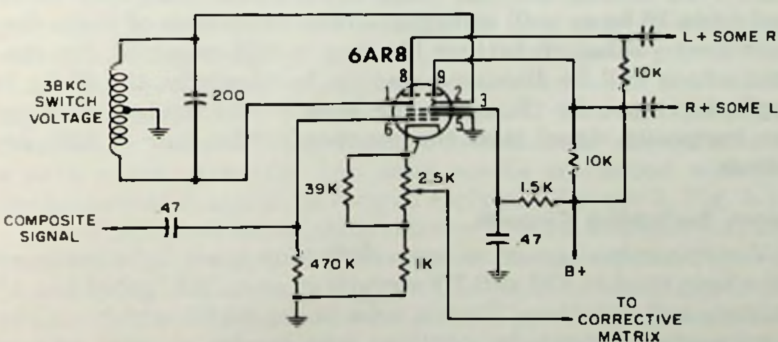


Fig. 4-25. Altec Lansing 359A demodulator circuit.

Two more important points are necessary for a complete understanding of this technique. First, had we been dealing with an R-only signal, the composite signal peaks would have been shifted in phase 180°. Thus, with the 38-kc phase remaining constant (as it does, in fact), the R diode would have been set up to conduct when signal peaks are available and would have recovered an R-only signal, with no AC output from the L diode.

Secondly, we could have polarized the two detecting diodes in the *same* direction and merely fed two equal but out-of-phase 38-kc signals to the diodes. In this way, alternate diodes would be turned on and off in correct sequence to detect proper left or right signal peaks as they occur.

A full understanding of the above method of demodulation of the total composite signal is necessary for a real analysis of the various circuits used commercially. Towards this end, you are urged to review Fig. 4-23 and the accompanying explanation until the demodulation procedure is completely clear.

DEMODULATOR CIRCUITS

Fig. 4-24 is a partial schematic of the demodulator circuit in the Sherwood Model S-8000 FM-MX stereo receiver. At first glance, this circuit might appear to represent still another technique for demodulation. Actually, as will be seen, it is really a refined form of switching circuit very much like the one just analyzed. The difference is that four diodes are used in a familiar bridge configuration. The composite signal is fed to the junction of the two 33K resistors, whereas the 38-kc switching voltage is fed to the ends of the 10K resistors. This configuration, known as a balanced demodulator, has the added advantage of cancelling residual components of RF (actually 38 kc and some 19 kc as well as higher-order harmonics of these frequencies)—although further filtering is still required, for reasons which will be discussed shortly. Incidentally, the 38 kc is applied in opposite phases to the ends of the bridge, whereas the composite signal is at the junction of the pair of 33K resistors.

Beam Switching Circuits

Vacuum tubes known as *beam deflection* types have for some time been used in FM and TV circuits as so-called "gated beam" limiters and detectors. Such a tube is the 6AR8 which, unlike a conventional pentode, contains *two* anodes as well as two deflection plates. The voltage applied to these deflection plates determines which anode will conduct partial or total cathode current. The circuit of Fig. 4-25 will be used in explaining how this type of tube can serve as a stereo demodulator and thereby eliminate the need for diodes.

As can be seen in Fig. 4-25, the 38-kc switching voltage is applied to plates 1 and 2 of the 6AR8 in such a manner that when plate 1 is most positive (allowing conduction to anode 8), plate 2 is most negative (preventing conduction to anode 9). The total composite signal is applied to common signal grid 6; it is this signal which determines the total emitted cathode current, as in any amplifying vacuum tube. The destination of the emitted current is determined by the polarization of plates 1 and 2. Thus, the previous function of the two oppositely polarized diodes of Fig. 4-20 is replaced by the action of this particular type of vacuum tube. The tube has the additional advantage of providing gain, whereas the diodes usually introduce some small loss.

It is interesting to note that Zenith Radio Corporation used a form of this circuit in its 1960 field tests. Relatively few manufacturers have so far chosen this method, probably because the 6AR8 tube (as well as others in its family) is more expensive than two (or even four) diodes. Nevertheless, excellent left and right recovery are possible with this circuit, provided the two halves of the tube are balanced (both internally and in terms of external components used). Altec Lansing uses 3% tolerance resistors (10K) in the anode circuit of the 6AR8 tube.

Demodulation and Separation

Rather than being labeled L and R, note that the outputs in Figs. 4-24 and 4-25 are referred to as "substantially R" and "substantially L" and "L + some R" and "R + some L." In other words this circuit, by itself, will not recover pure L or pure R. There are several reasons for this, one of which involves a rather lengthy mathematical analysis. It proves that in an *average* detector (the configuration in most of the switching diode demodulators), the equivalent L−R available is less than the L+R, which this form of demodulator always sees as a peak value. Actually, you need not be concerned with the mathematical analysis. In fact, as early as Chapter 3, Fig. 3-13, a case of sideband attenuation relative to main channel was discussed solely in terms of FM receiver characteristics. Fig. 4-26 is essentially the same as Fig. 3-13, except the condition is not so severe. The difference in amplitude between main-channel (L+R) and subchannel sidebands (L−R) is only 3 db (with the high-frequency sidebands being the lower). But this discrepancy is sufficient to limit ultimate separation to a maximum of only 15 db between left and right channels.

Fig. 4-27, a detailed, hand drawn view of Fig. 4-26, shows that for a left-only signal the R diode, when able, will conduct some of the L sine wave and feed it ultimately to the R output. (Normally the R diode will have no signal peaks to conduct if the

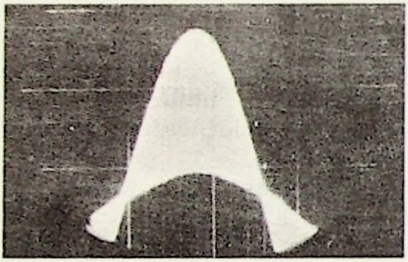


Fig. 4-26. Left-only signal attenuated 4 db.

composite signal is as it should be—i.e., as in Fig. 4-23). Conversely the L diode, when polarized to conduct in the presence of an R-only signal, should have had no subcarrier sideband peaks to conduct at that instant. Because of the unbalance, however, the L diode will in fact conduct some of the R sine wave and feed it on to the L output.

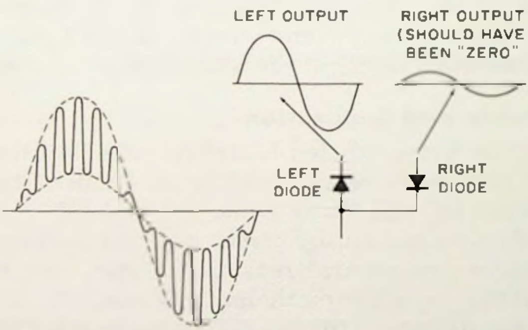


Fig. 4-27. Detail of signal in Fig. 4-26.

The residual L coming out of the R diode will be in phase with the desired L from the L diode. Likewise the residual R from the L diode will be in phase with the desired R from the R diode. This is also true for the beam-switching circuit of Fig. 4-25.

The signal at the output of the L diode is L + 0.2R, and from the R diode, R + 0.2L.

If you carefully examined Fig. 4-25, you have already surmised that this situation is not hopeless. The arm of the potentiometer in the cathode of the 6AR8 goes to a corrective matrix. Here is what happens:

The cathode is unaware of any switching taking place in anodes 1 and 2 as 38 kc is applied to them. Thus the L−R sidebands are, in effect, canceled in the cathode; and only a fixed 38-kc voltage, together with the L+R component, is present across the cathode resistor. Now the L and R signals in the two anode circuits of the 6AR8 tube are 180° out of phase with the signal in the cathode because of the usual phase reversal in any amplifier tube.

With reference to the output signals, the cathode can be said to contain −(L+R) component, or −L−R. Suppose the arm of the potentiometer in Fig. 4-28 picks off a signal equal to

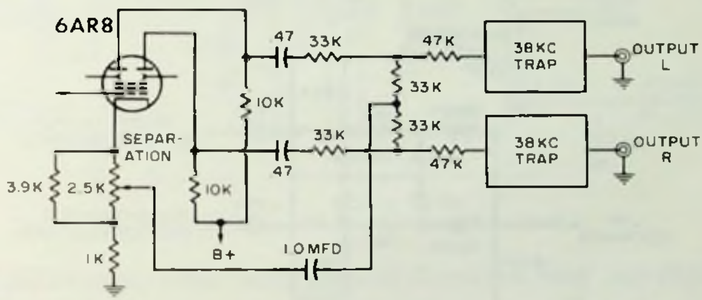


Fig. 4-28. Portion of Altec Lansing 359A demodulator circuit.

0.2(−L−R) and adds it algebraically (through resistive matrixing) to the two improperly separated signals at the two outputs of the 6AR8. In the case of the predominantly L signal, the result will be:

$$L + 0.2R - .2L - 0.2R = .8L$$

In the case of the predominantly R signal:

$$R + 0.2L - 0.2L - 0.2R = .8R$$

In other words, addition of the proper amount of out-of-phase (L + R) signal to an improperly separated stereo signal pair improves the separation (theoretically it would be perfect). For this reason, the potentiometer is called a separation control and is analogous to the L+R separation control seen in earlier block diagrams and circuits of the matrix type of multiplex circuitry. Incidentally, the 0.8 factor is only a relative term. As long as there is enough signal available at each output to drive the respective amplifier, do not be concerned over the slight change in output level when adjusting the control for optimum separation between left and right signals.

Detection and Direct Matrix Demodulators

Having examined basic switching-circuit demodulators, let's look at some circuits which use the more conventional carrier reinsertion, detection, and direct matrix approach. (This system has already been shown in block-diagram form in Fig. 4-1.)

A partial schematic of an actual detection and matrix circuit appears in Fig. 4-29. After the bandpass filter (shown earlier) has separated the sideband components from the rest of the composite signal, they are mixed in proper phase with the internally generated 38-kc carrier to produce the familiar AM-

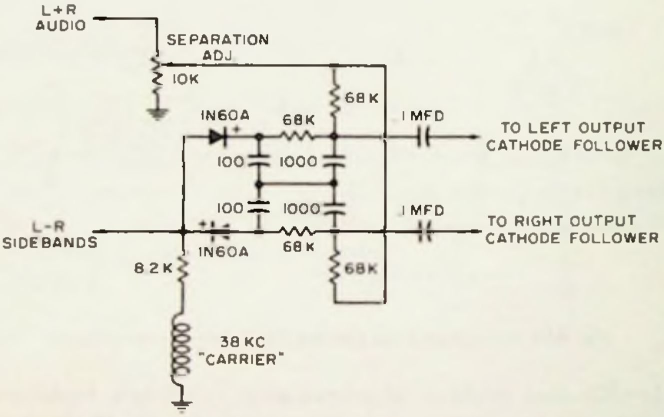


Fig. 4-29. Heath AC-11 detection and adding circuit.

modulated envelope at the junction of the 8.2K resistor and two 1N60A diodes. To prevent an overmodulated waveform envelope, at least three times as much 38-kc carrier than sideband component is required at this point. The matrixing process to follow will require both L−R and −(L−R). These two out-of-phase (L−R) audio signals are provided by two diodes connected in opposite polarity, so that one diode will detect the upper half of the modulation envelope and the other the lower half. The 100-mmf capacitors at each diode output serve to filter out some of the 38-kc carrier. A scope connected to the output will show a clearly defined recovered (L−R) signal (or −L+R, as the case may be), although a great amount of 38-kc component will still be observed. To these recovered signals is added the L+R signal, so that L and R exist independently at the junctions of the pairs of 68K resistors. The proper de-emphasis response, which was still missing up to this point, is provided by shunting the output of each 68K resistor with a 1,000-mmf capacitor. The presence of these capacitors aids immeasurably in further reducing the 38-kc RF to an almost negligible value. The L and R signals are passed on to a pair of cathode followers and then to their respective stereo amplifiers.

Fig. 4-30 shows still another matrix type of detector circuit, that of the Bogen Model PX60 multiplex adapter. Its principle of operation is the same as that of the circuit shown in Fig. 4-29.

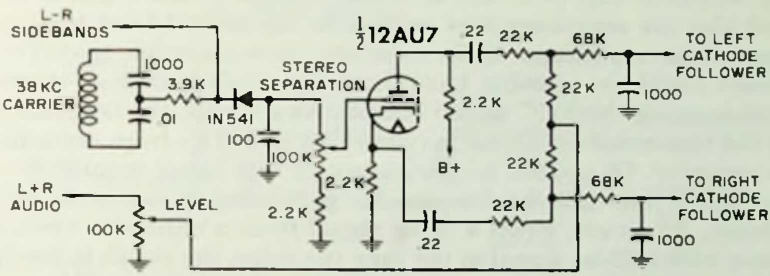


Fig. 4-30. Bogen PX-60 additive matrix circuit.

Unlike in many other such circuits, however, only one diode is used.

The L−R recovered in the usual fashion is fed to a phase-splitter triode—the cathode circuit of which contains L−R while the plate circuit, through normal amplifier phase inversion, contains −L+R. Both outputs are mixed with L+R of sufficient amplitude (as determined by the separation and the stereo balance controls) to provide proper L and R signals. These signals then undergo de-emphasis in the network consisting of an 68K resistor and 1000-mmf capacitor in each channel.

Residual Frequencies

Since the 38-kc signal generated internally is much larger in amplitude than the recovered audio, the demodulating circuits illustrated thus far cannot remove all of this residual 38-kc RF, even after filtering and normal FM de-emphasis.

At first thought, the presence of some 19-kc and 38-kc frequencies in the output of the multiplex circuitry(and even 57 kc and 76 kc due to distortion products generated in nonlinear portions of the circuits) does not seem to be objectionable, because these frequencies cannot be heard. Actually, the presence of any one of them can be quite serious. For one thing, nearly all high-fidelity amplifiers are capable of supplying power to a speaker at 19 kc. While most people cannot hear this frequency, the tweeters may be damaged in speaker systems which were never designed to handle such large, continuous amounts of high-frequency power.

A more serious objection to the presence of these residual frequencies is the distortion they introduce when stereo is being recorded on tape (certainly one of the most promising aspects of the entire FM stereo picture). Tape recorders contain an oscillator that provides an erase voltage to the erase head, and also the necessary bias voltage to the record head for distortionless recording. Unfortunately, tape-recorder manufacturers could not possibly have foreseen which system of stereo broadcasting the FCC would choose. As a result, the frequency of the bias-erase oscillator may vary 23 to 100 kc from recorder to recorder. Of course, no frequency in this range is audible—any more than are the frequencies generated in the multiplex circuits. However, when a 19-kc signal from a multiplex circuit beats with a 23-kc signal in the tape recorder, the result is likely to be a steady whistle at 4 kc which is *definitely* audible—in fact, it may be loud enough to ruin a carefully recorded selection.

For these reasons, it is desirable to get rid of such residual signals at the output of the multiplex circuitry. For proper operation with many home tape recorders, any 19 kc in the output must be at least 30 to 35 db below the audio signal, and 38 kc should be at least 40 db below.

Cancelling in Switching Circuits—Additional filters can be installed at the outputs of the left and right signal channels in the adapter or receiver, ahead of the recorder take-off point. Various forms of these filters will be dealt with a little later. For the moment, before leaving detection and demodulation circuits, let's examine one form of switching detector which not only recovers L and R, but in doing so, also rejects any 19 and 38 kc still present. The circuit shown in Fig. 4-31 is the demodulation system used by Pilot Radio Corporation in its Models 100 and 200 FM multiplexers.

In addition to the usual pair of switching diodes (oppositely polarized and therefore fed from a common 38-kc switching source), two more diodes are connected at the output points of the first two diodes. Each of the latter has the opposite polarization from that of its associated switching diode and is in series with a network consisting of a 220K resistor and 500-mmf capacitor in parallel. The audio voltages in the composite signal, being of a low fundamental frequency, are dropped across the 220K resistor. The 38-kc carrier (or switching voltage), however, is conducted through the 500-mmf capacitor and rectified by the series diode in a polarity opposite from that of any re-

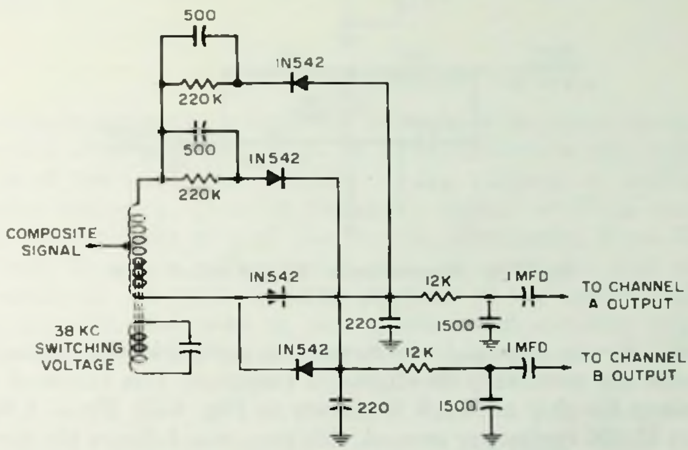


Fig. 4-31. Pilot Radio 100 and 200 demodulator circuit.

sidual 38 kc that the switching diodes themselves might have passed. At the junction of each pair of diodes, residual RF is therefore cancelled out; and no further filtering is required, other than the usual de-emphasis network needed to establish a flat tonal response. The de-emphasis network shown has a much lower value than the usual 75 microseconds, because the 500-mmf capacitors associated with the RF canceling diodes aid in roll-off of the high-frequency audio. The latter is due to the "bucking" action of the higher frequencies, which "see" a lower impedance than the middle and lower audio frequencies.

Filters and Networks—Manufacturers not wishing to employ four diodes have resorted to other means for removing 38 kc and other spurious components from the output signal. Typical of these is Harman-Kardon who, in its model MX-500, employs the well-known twin-T notch filter shown in Fig. 4-32. Those of you who are familiar with the normal values of a twin-T RC-type rejection filter will recognize that the circuit values are a bit unusual and unsymmetrical. The reason is that the twin-T performs two functions at once. Primarily it greatly attenuates any 38 kc present. Since the attenuation at other frequencies (particularly just below the frequency of maximum desired attenuation) can be readily controlled by choice of component

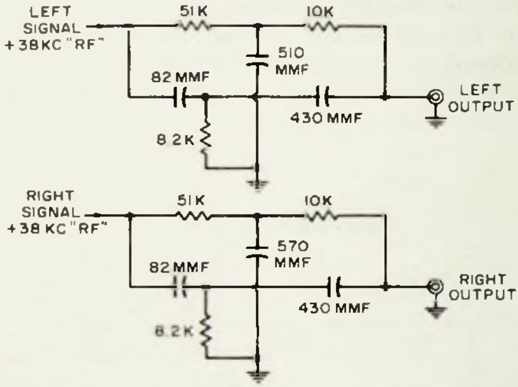


Fig. 4-32. Harmon-Kardon MX-500 twin-T notch filter (38 kc).

values, it was desirable to have this network simultaneously provide the necessary de-emphasis response. The curve of total response for this network is shown in Fig. 4-33. From 1,500 to about 15,000 cycles per second, this response follows the desired de-emphasis characteristic shown earlier in Fig. 3-12. Beyond 15 kc, the response falls off more rapidly until maximum attenuation of about 45 db is reached at 38 kc.

This form of network, while ideal for maximum rejection of a single frequency, will not attenuate the 19, 57, 76 kc, etc., frequencies. Although usually present in much smaller amounts than the 38 kc, they must be dealt with at other points in the

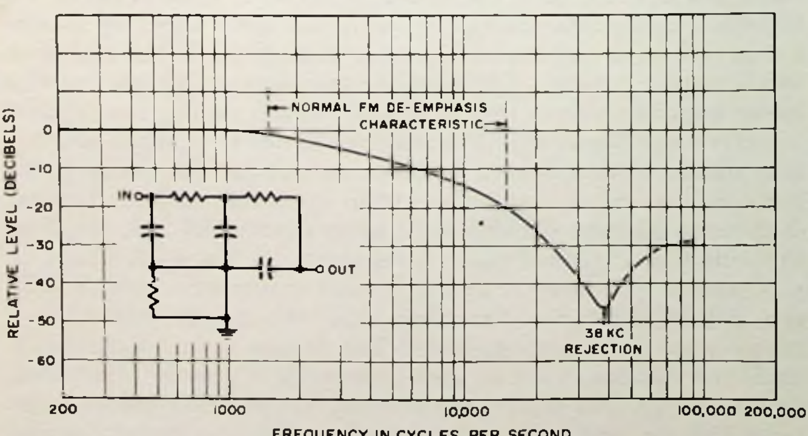


Fig. 4-33. Response of the filter shown in Fig. 4-32.

circuit. Nearly all manufacturers surveyed except Pilot use some form of twin-T filter arrangement at the output. Moreover, the component values chosen are all so similar that more circuits of this type need not be illustrated at this point. Readers wishing to examine them in more detail are referred to Chapter 7.

SCA INTERFERENCE

A simple corrective circuit for SCA interference is shown in Fig. 4-34. A trap, tuned to 67 kc and installed in the cathode circuit of the very first amplifier in the adapter or multiplex receiver circuitry, impedes further progress of 67-kc components through the rest of the circuit. Obviously, if no 67-kc (actually 60 kc to 74 kc) signal ever gets past the first stage, the nonlinear elements in later portions of the circuit will be of no concern. This trap, in combination with existing traps in the bandpass or low-pass filter previously shown, is usually

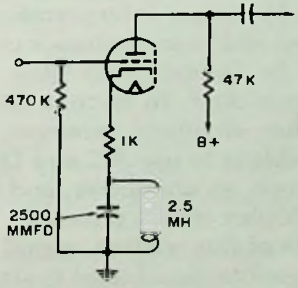


Fig. 4-34. Trap (67 kc) for reducing SCA interference.

enough to attenuate any SCA signals by more than 60 db below the desired stereo signal. As a result, the interference will be completely inaudible. Of course, all tuned circuits designed to attenuate the 67-kc (or so) signals must be carefully tuned. If too low in frequency, they may attenuate frequencies in the desired passband (53 kc and lower). The result will be no high-frequency stereo separation. If tuned too high in frequency, they may fail to properly attenuate the SCA cross talk.

The sound made by this type of interference is not audio *per se*. Rather it is a "swooshing" or "variable whistling" sound which is particularly annoying during quiet passages of the desired program material. Happily, the circuits illustrated in this book have all overcome this early problem by having sufficient SCA rejection capability.

CHAPTER 5

SERVICING MULTIPLEX CIRCUITS

A book of this type must resort to a generalized approach in attempting to describe servicing procedures for multiplex circuitry. Moreover, it must also assume that readers attempting to service or troubleshoot such equipment are familiar with the use of certain standard electronic test equipment. Many troubles can be analyzed, of course, by listening to the output and by proper interpretation of the effects of the various front-panel and rear-of-chassis controls. In this chapter, troubles that can be located with little or no test equipment will be considered first. In circuits as sophisticated as those described in earlier chapters, however, often the only way to locate the trouble is to use AC and DC vacuum-tube voltmeters, an oscilloscope, an ohmmeter, and sometimes even a specially designed multiplex signal generator.

As of this writing, signal generators for simulating the actual composite signal used in stereo broadcasting retail between five-hundred and a thousand dollars. For readers who service multiplex equipment professionally, the purchase of such a generator is strongly recommended, but would hardly be justifiable for "do-it-yourselfers" and others having only a casual interest in the subject.

Some currently available multiplex signal generators will be discussed in the next chapter. For the present, only such troubles that can be located by the use of common sense or, at worst, a minimum of inexpensive test equipment likely to be already owned by the reader will be considered.

Before discussing the troubles likely to be encountered, let's start with the premise that the manufacturer of any multiplex equipment, either adapter or complete receiver, was convinced that the design of his product was sound and that he took pains to check out each unit shipped. These troubles can be broadly classified into four general areas:

1. Misalignment.
2. Incorrect selection of adapter, where present equipment is converted.
3. Component failures.
4. Instability.

The first category will be dealt with in more detail in Chapter 6, which gives the actual step-by-step alignment procedure recommended for the various multiplex circuits. The criteria for selecting adapter equipment for a particular receiver were discussed in Chapter 3. Nevertheless, certain troubles in matching may crop up which at first might not be recognizable as such.

GENERAL SEPARATION

The largest single customer complaint is lack of adequate separation. Before tearing into the adapter (or its associated tuner or receiver), let's define "lack of separation." Happily, many musical recordings made in the last few years have shied away from the spectacular stereo demonstration or "ping-pong" effects of earlier years. Directionality of sound is one of the most outstanding characteristics of stereo—but by no means the only one. A symphony orchestra, for example, is not two halves of an orchestra, each playing in a separate recording studio; rather it is a fusion of instruments and sounds across an expanse of space.

A far better way to judge stereo separation, during an actual stereo broadcast, is to wait for those all-too-few statements by the announcer that he is "coming to you from the left (or right) speaker." Helpful programming includes such periods of left-only or right-only transmission. This is the proper time to quickly adjust your separation control, to see whether the single-speaker source being described can be achieved. If not, your quest for troubles begins.

Obviously, the first step is to check the connection from the tuner or receiver to the adapter. The adapter may be connected to the main (de-emphasized) rather than the multiplex output. So, the adapter circuits would not be actuated because no sub-carrier sidebands and very little 19-kc pilot carrier would be extracted from the tuner or receiver.

Next, make sure all stereo amplifiers are in the Stereo mode if they also have a Mono setting. The mere presence of sound in both speakers does not mean the amplifier is set to the stereo function. In the mono position, the amplifier mixes L and R all over again. As a result, the same L+R (i.e., mono) signal comes out of each speaker—even though the adapter had originally unscrambled and separated the two.

The same will occur if the adapter itself has a Mono setting and is not in the Stereo mode. There is a good reason for having a mono position on the adapter itself: The signal-to-noise ratio in stereo is much poorer than in mono. Thus, when a station is not broadcasting stereo, the music or program material will sound the same in either mode (same mono program from each speaker), but there will be less background noise or hiss in the mono mode.

On some stereo adapters and receivers, an indicator light glows while a stereo program is in progress. Usually, the indicator is actuated by amplifying and then rectifying the incoming 19-kc pilot carrier so that it will trip a relay or light a neon glow tube. While a handy device at the outset, it has some drawbacks. Nothing in the FCC stereo rules prevents the station from transmitting the 19-kc pilot carrier even with no stereo broadcasts in progress, and at least one station is known to engage in this practice. Furthermore, many stations eventually will broadcast stereo almost exclusively. When this happens, it will nullify what usefulness the indicator light might now have. Usually, the subcarrier recovery circuit in adapters is switched in and out at the same time (and by the same means) that the indicator is actuated. Thus the listener is spared the trouble of switching the adapter or amplifier from Mono to Stereo.

Let's assume that although stereo is definitely being broadcast and the above precautions have been observed, there is still no audible stereo effect. One more point must be cleared up before looking inside the circuits, and that is to make certain no motorboating (a low-pitched "putt-putt" sound) is coming from the speakers during a stereo program. Such a symptom clearly indicates that everything is working normally except the local oscillator—it is not being locked in by the incoming 19-kc pilot carrier. The simple alignment procedure described in Chapter 6 will in all probability rectify the situation.

Lack of separation would be impossible to analyze in terms of every commercially available adapter. For this reason, only two circuits will be singled out in this chapter. Each represents one of the two basic configurations—the matrix circuit of the Heath Model AC-11, and the switching circuit of the Knight (Allied Radio Corporation) Model KN-MX adapter (Figs. 5-1 and 5-3).

Matrix Circuit

From Fig. 5-1, the reasons for lack of stereo may be:

1. No or insufficient L+R signal.
2. No or insufficient L−R signal.
3. Improper matrixing (open component, lead, etc.)

No or Insufficient L+R Signal—This is the easiest trouble to check. With the equipment turned on, short to ground the pair of 100-mmf capacitors (C19 and C20) at the output of each diode. This will completely deactivate the subcarrier-sideband circuitry and leave only the L+R channel operative. As separation control R7 is rotated clockwise, the volume of L+R to both channels should increase, and should decrease to zero when R7 is fully counterclockwise. Presence of sound at this point with R-1 midrange indicates that the entire L+R channel is functioning normally. If no sound is heard, short across L3; it may be open. (Filter coils used in multiplex circuits are generally in the millihenry range and contain fairly heavy windings of fine wire which are easily broken.) Although V1 might be suspected, analysis will disclose that since some sound had been heard before C19 and C20 were shorted to ground, all audio (L−R and L+R) must have originated through V1. Therefore, it could not be the source of trouble.

Turn off the power to the adapter and, with an ohmmeter, check C4 for a possible short. Substitute capacitors for C3, C2, and C1. This is about all the components there are between the adapter input and matrix resistors R21 and R22. By this time, the source of trouble should have been eliminated and the L+R signal recovered.

Should the lack of stereo not be caused by absence of L+R component signal, it is time to investigate the second cause listed above.

No or insufficient L−R Signal—Remove the shorting jumpers across C19 and C20, and rotate the Separation control fully counterclockwise. If all were well within the circuit, you would now hear equal L−R audio signals from each speaker. But if you hear no sound at all, or the sound you do hear is below the L+R level or is distorted, the need for alignment is indicated. This procedure will be dealt with in the next chapter. The best test point for examination of the subchannel circuitry is the junction of the two 1N60A diodes and the 8.2K, 38-kc feed resistor R18. A scope is literally a necessity in this circuit, and even in the absence of a stereo signal generator, much can be learned by osberving the waveforms.

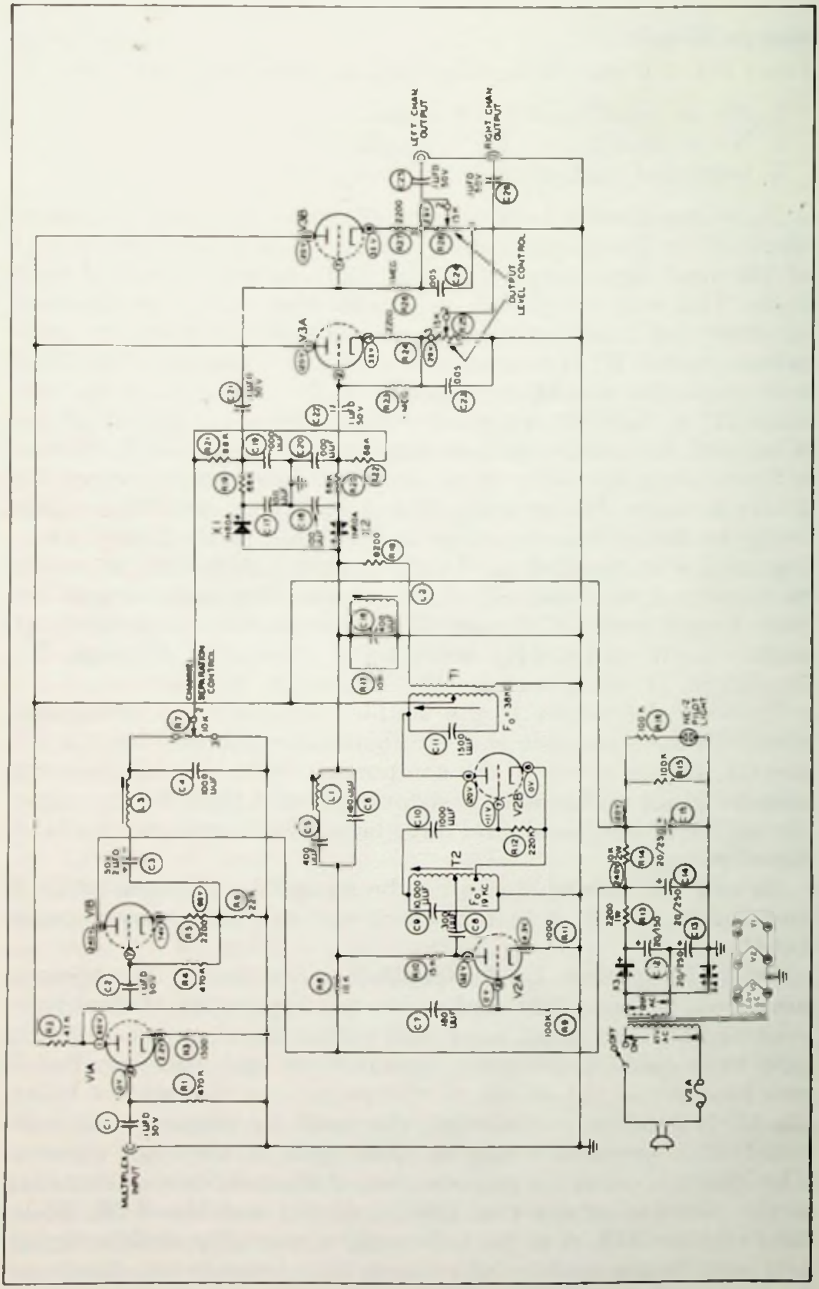


Fig. 5-1. Heath AC-11 adapter circuit.

For the most usable display, set the horizontal sweep of the oscilloscope somewhere between 400 and 1,000 cycles per second. If all circuits were operating normally, the scope display would look somewhat like Fig. 5-2, a large-amplitude 38-kc signal (self-generated in the adapter) upon which the amplitude (i.e., the L−R) modulation is superimposed. If such a waveform is observed during a program and there is still no L−R audio, the trouble must lie beyond this point. It could be in the 1N60A diodes (which might be open). Other possibilities are a shorted C17 or C18 and an open R19 or R20.

If the recovered L−R signal became severely distorted when the separation control was turned counterclockwise, the scope display is likely to show modulation-sideband products, but insufficient or no 38-kc carrier will be available for reinsertion. It is easy to determine the level of 38 kc with respect to the modulation subcarrier-sideband products. This is done by examining one while the other is deactivated. First, deactivate the local 19-kc oscillator (and thereby the 38-kc doubler) by shorting from the cathode of oscillator V2B to ground. The stage will stop oscillating, and only the subcarrier sidebands will be seen, varying in amplitude in accordance with the stereo program material. On the face of the scope, mark the peak amplitude level with a soft crayon.

Next, short out from pin 7 of V1B to ground (to prevent the composite signal from reaching the bandpass filter consisting of L1, C5, and C6), and remove the short at the oscillator cathode. The waveform would now consist entirely of 38-kc (unmodulated), with perhaps some small but constant 19 kc.

Let's assume, first, that when the oscillator was shorted out, the sideband modulation was 1 inch high on the scope tube. If distortion is to be held to a minimum, the 38-kc waveform with the sidebands shorted out should be at least 3 inches in amplitude. A lower amplitude would indicate either that the 19- or 38-kc tuned circuits require adjustment, or that one or more operating voltages (shown on the schematic) have decreased due to a faulty power supply or other component.

If *no* 38 kc was visible in this last operation proceed with signal tracing. First, move the scope lead to the top of the T1 secondary winding. (More 38 kc should normally be present here than at the junction point.) Then move to the grid (pin 7) of V2B, and observe the 19-kc sine waves there. If the oscillator itself is dead, check all operating voltages. Then, with the power turned off, measure the continuity of oscillator coil T2. Another way to check the oscillator is to measure the DC voltage at the grid (pin 7) of V2B. An oscillator of this type generally draws a considerable amount of grid current, since basically it operates as a class-C amplifier. This grid current charges capacitor C10 to about −11 volts DC. If it is necessary to change any of the parts associated with the oscillator or doubler stage, the frequency and phase of these stages are so critical that they will require realignment even though the same-valued parts have been substituted.

Now suppose 38 kc was present at the original junction test point, but no sideband modulation. Signal tracing becomes relatively simple, involving only the elements of the bandpass filter and the pair of plate load resistors of V2A. C5, C6, or L1 may have opened. The mere fact that V2B is oscillating does *not* mean it is being locked by the incoming 19-kc pilot carrier signal. Therefore V2A, which conducts both the pilot signal and subcarrier sidebands, may be malfunctioning. As a matter of fact, a scope applied at the junction of R8 and R10 would still show a total, composite waveform with L+R (which will be removed later by the bandpass filter). This signal may be traced all the way back to the grid circuit of V1B, in the usual manner, until the source of trouble is located.

Let's say the correct waveform of Fig. 5-2 has been established at the junction point of diodes X1 and X2. Also, let's assume L+R is available at the arm of the separation control. Now there is little else, componentwise, to account for lack of

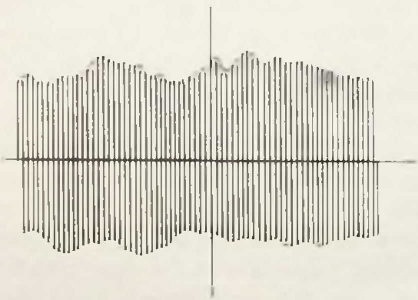


Fig. 5-2. Random modulation at junction of diodes X1 and X2 of Fig. 5-1.

separation, provided the set is properly aligned. The final cause, improper matrixing, is the least likely to occur.

Improper Matrixing—A short between the two grids of the output cathode followers (V3A and V3B) will cause a blending of the L and R signals again. So will a short between the normally unconnected ends of R21 and R22 (because of the proximity of leads, for example), or between any similar pair of components in the cathode circuits of V3. Such mixing will recreate an L+R monophonic signal and thereby destroy the stereophonic effect.

Switching Circuit

Let us now examine lack of separation in terms of a switching circuit (Fig. 5-3). Here improper separation, unless due to poor alignment, is not so simply found, for the entire composite signal is applied to the switching tube at once. More than likely, improper separation will also be accompanied by distortion.

The primary causes are:

1. Improper or no 38-kc switching voltage.
2. Defective demodulator circuitry.

Signal Tracing—If any audio at all is heard in the output, you can safely assume there are no open coils or other interruptions of the signal path. Of course, always check tubes first, to make sure the composite signal waveform itself is not being clipped or otherwise distorted.

But if no sound is heard, signal tracing must proceed from the input to the switching tube output. Convenient check points for looking at the total composite signal are:

1. Pin 2 of the 6AU8 tube (to check the signal waveform from the tuner section or receiver).
2. Top of L2, which feeds pin 6 of the 6AR8 beam switching tube (to check the continuity of all filter elements and to compare the signal level at this point with the input-signal amplitude).
3. The 38-kc deflection plates of the 6AR8 (pins 1 and 2).

One of these check points is sure to disclose a circuit discontinuity when there is no sound from the multiplex adapter or section.

Switching Voltage—Correctness of amplitude of the 38-kc switching voltage may be analyzed and traced in much the same way as for matrix adapters, since both circuits have the same 19-kc oscillator and doubler as well as the same method of feeding this voltage. As before, the final amplitude of the 38-kc switching voltage must be many times larger than that of the composite signal. Otherwise, distortion is likely during the recovery or demodulation process.

Demodulation Circuitry—A short between pins 1 and 2 of the 6AR8 tube would result in the entire L and R signal being transmitted through both channels, since individual plate conduction would be taking place continuously. As a result, monophonic program would emanate from both channels.

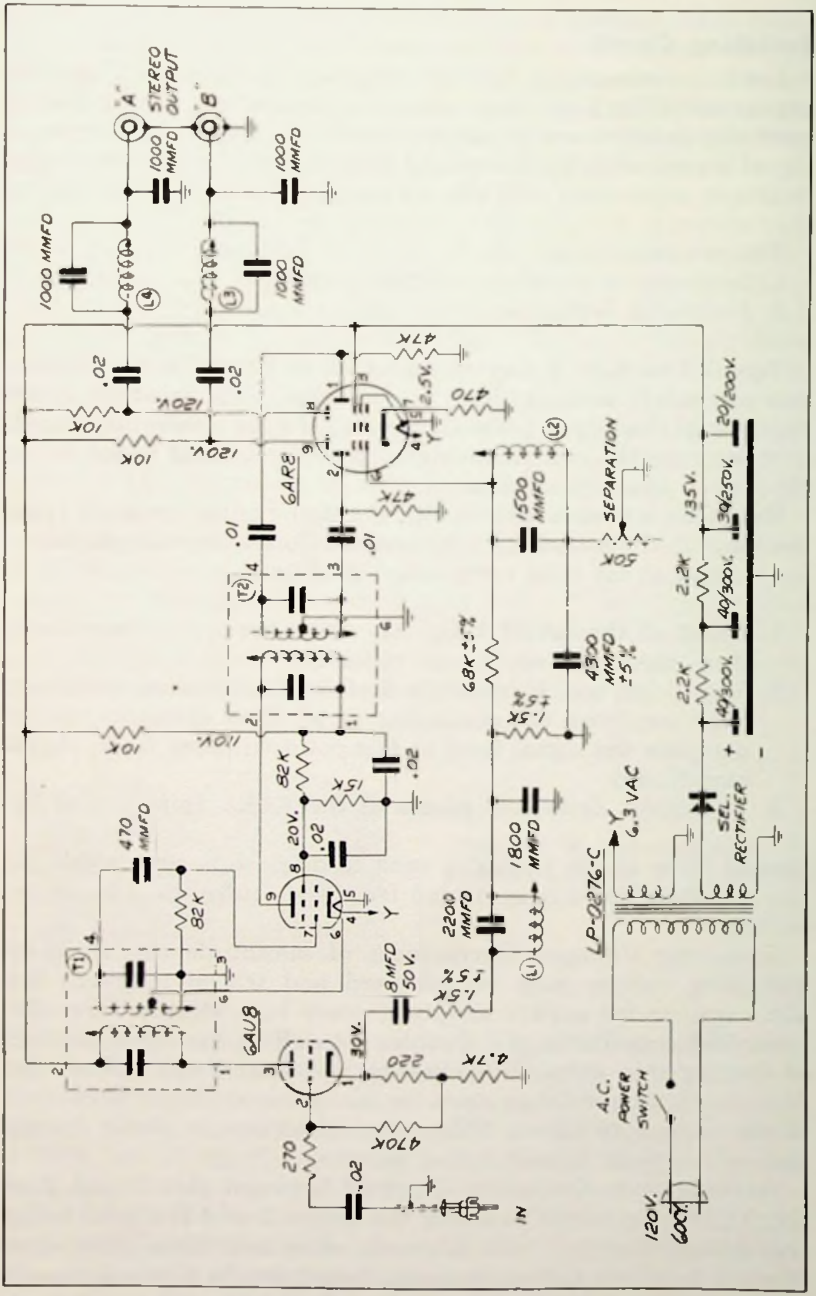


Fig. 5-3. Knight KN-MX adapter circuit.

CHANNEL SEPARATION

Another likely trouble in multiplex circuitry is lack of separation over a particular band of frequencies, rather than over the entire band. As pointed out earlier, the FCC specifications provide for excellent separation from 50 cps to 15 kc. Furthermore, each station is required to maintain at least 30 db of separation in each channel over this entire frequency range. Because of the complexity of filters, tolerance of components, and other circuit considerations, it is extremely difficult for reasonably priced multiplex receivers or adapters to meet these specifications (although many of those in this book actually surpass the minimum FCC specifications).

Given equally stringent tolerances for components and alignment, the switching circuit generally is capable of better overall separation than the matrix type. This is not to imply that a matrix adapter *cannot* have excellent separation. In fact, the General Electric adapter field-tested by the FCC in 1960 at Uniontown, Pennsylvania, maintained a separation of over 20 db, even at 15 kc—yet consisted of only a *single* tube (dual triode) plus a couple of diodes and the necesary filters. Moreover, this separation was achieved with a standard tuner, and the one-tube adapter and amplifiers were located over 40 miles from the transmitter!

A generalized plot of typical channel separation is shown in Fig. 5-4. Notice that best separation is usually achieved at mid-band frequencies, but tends to fall off fairly rapidly as the high frequencies are approached and slows down gradually at the low frequencies. While not ideal, such a separation characteristic is normal with a great many adapters. This discussion will instead be concerned with virtually no separation over one band of frequencies. As shown by Fig. 5-5, separation is lost above 4,000 cycles or so. Such lack of separation would definitely be noticed by the listener.

Matrix Circuit

In a matrix adapter (Fig. 5-1) there are two causes for sudden diminution of separation:

1. The amplitude of either L+R or L−R changes as the audio frequency is increased.
2. A phase shift takes place in the subcarrier-sideband channel, with no corresponding phase shift in the L+R channel.

L+R or L−R Amplitude—The first cause is relatively easy to check out; no specialized test equipment is needed. Connect

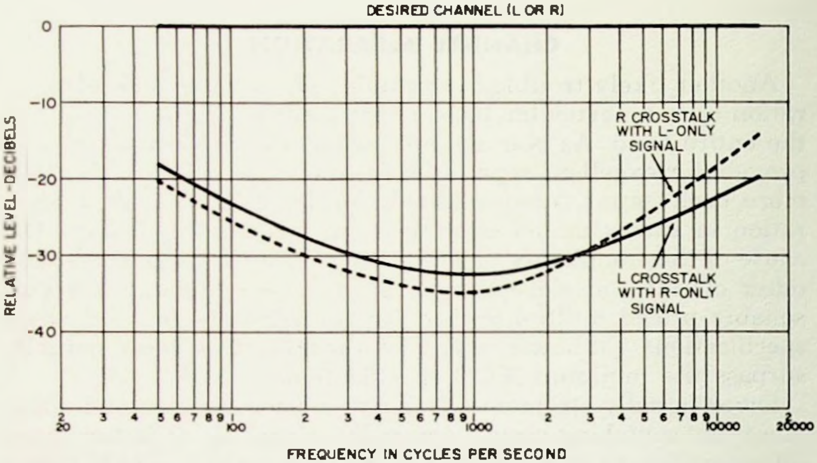


Fig. 5-4. Channel-separation curves.

an audio oscillator to the adapter input. Disable the local oscillator by shorting from the cathode (pin 8) of V2B to ground (chassis). Connect a VTVM (AC type) at the junction of the 1N60A diodes, and set the input voltage from the audio oscillator between 0.25 to 0.5 volt. Then measure the response of the subchannel circuits over a range of frequencies from 23 to 53 kc. If the response is not flat over this range (except possibly for a drop of several db near the 53-kc limit), one of the filter elements (C5, C6, L1, L2, or C16) may be defective or the setting of L1 or L2 may be off.

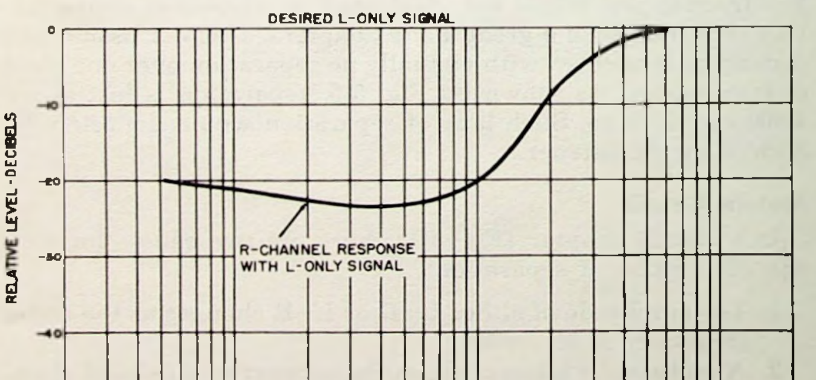


Fig. 5-5. Separation at high audio frequencies.

Important: Always adjust L1 for maximum null (i.e., minimum output) at 67 kc.

If the response of the bandpass filter circuitry is flat or nearly so, check the main channel, or L+R, circuit next. With the audio oscillator still connected to the input, transfer the VTVM probe to the arm of separation control R7. This time apply frequencies ranging from 50 to 15,000 cycles. If the response is not flat or nearly so, suspect C4 or the alignment of L3. Adjust L3 so that the frequency response of the L+R section is down about 3 db at 20 kc, outside the filter passband.

Phase—The phase response of the filters is unlikely to change, unless amplitude-response measurements indicate major realignment is necessary.

Switching Circuit

Referring to Fig. 5-3 once more, it is clear that once the total composite signal is fed to the switching tube, there is nothing in the circuit which could cause one range of frequencies to exhibit better separation than another (assuming the design was basically sound to begin with). Thus, partial loss of separation is not likely to be encountered in a switching circuit.

OSCILLATOR SYNCHRONIZATION

Still other common problems may occur in both types of adapters. So let's continue to examine Fig. 5-3 and discuss loss of oscillator synchronization. Such a trouble will be heard as a low-pitched motorboating, or if complete loss of pilot-carrier signal has occurred, a thoroughly distorted wavering sound having no separation whatsoever. In the circuit of Fig. 5-3 (or circuits of its general type), the incoming pilot carrier may be traced from input to oscillator locking point fairly simply with just the aid of an oscilloscope. Check first at the input of the adapter (maybe the broadcast station has temporarily lost its 19-kc pilot carrier). Remember that the composite signal contains the 19-kc pilot carrier, and that it is the only part of the signal remaining on, at constant amplitude, regardless of the program material. Thus, during a quiet moment in the music or narration, this 19-kc signal should be clearly visible on the scope. It should also be visible at the plate circuit of the first half of the 6AU8, but at a much greater amplitude because of the gain provided by the triode amplifier. Next, disable the oscillator by shorting from the cathode (pin 6) of the 6AU8 to ground, and check the signal at the plate (pin 9) of the 6AU8. If the oscillator is not shorted during this last check, the scope will display the 19 kc of the oscillator itself (or nearly 19 kc, if the oscillator is free-running) and thereby mask the incoming pilot-carrier signal. If the pilot carrier is available at this point but there is still trouble, the capacitor across the primary of T1 may be open or the oscillator coil (secondary of T1) may be so far out of tune that the oscillator cannot be synchronized by the locking signal. Realignment of the oscillator coil (see Chapter 6) is called for in the latter case.

FREQUENCY AND PHASE

The importance of uniform frequency and phase response from the tuner or receiver being converted has been emphasized many times. Up to now, little has been said about corrective measures which might be taken in either the tuner (receiver) or adapter, because such measures are not truly servicing.

Compensating Networks

A sensitive adapter (one which requires less signal than is available from the tuner or receiver) may be corrected for deficiencies in tuner response or phase characteristic. This is done by the introduction of a compensating network into the input-stage circuitry. Such compensation is already built into the Pilot Radio Models 100 and 200 FM multiplexer units, as shown in Fig. 5-6. The network ahead of the input cathode-follower grid is a frequency-sensitive voltage-divider and phase-shifting type. Low frequencies are not affected by the presence of the 39-mmf capacitor in the network. The voltage is divided by the 1 and .68-meg resistors. The grid resistor can be ignored, although it partially shunts the 1-meg resistor because the stage is a cathode follower and therefore the input impedance is ac-

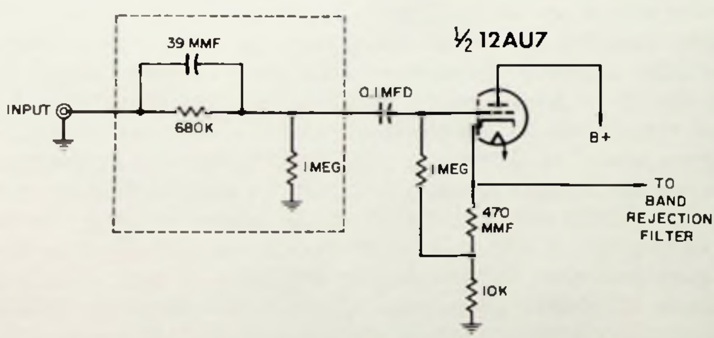


Fig. 5-6. Partial circuit of first stage of Pilot 100 and 200.

tually much higher than 1 megohm. Approximately 60% of the total low-frequency voltage is passed on to the grid itself. This represents a loss of about 4.5 db, which the adapter must be able to withstand without loss of oscillator lock or deterioration of the audio signal-to-noise ratio.

Assuming the adapter sensitivity is sufficient to withstand this initial loss, let's see what happens at higher frequencies. At 53 kc the impedance of the 39-mmf capacitor is only 75K, so that for all purposes the 680K resistor is shorted out as far as the AC signal is concerned. Thus, at this high critical frequency the compensation is nearly 4.5 db. That is, the gain at 53 kc is 4.5 db higher than at low and middle frequencies. This means that a tuner or receiver with a normal falling-off discriminator characteristic of 4.5 db at 53 kc would, by the addition of this network between the discriminator and adapter input, be almost perfectly compensated for proper separation all the way out to 15 kc.

Now you can take your cue from Pilot Radio Corporation (and also Altec Lansing, which employs much the same technique at the input to its first amplifier, as shown in Fig. 5-7). After proper measurement of the response of your own tuner

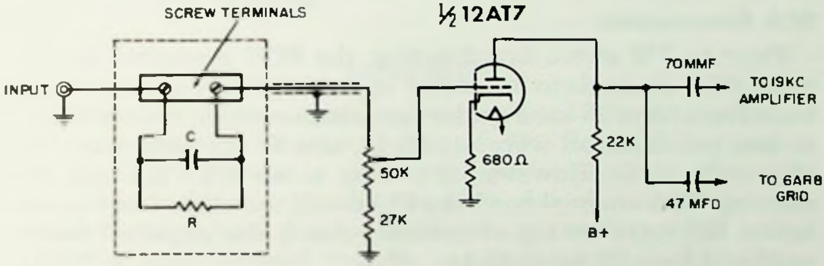


Fig. 5-7. Altec Lansing 359A input circuit.

or receiver discriminator characteristic, you can arrive at a network which will at least partly if not completely correct for gradual loss of separation caused at the high frequencies by tuner-circuit deficiencies. Table 5-1 shows network compensating values for two input grid-impedance conditions and for several degrees of attenuation of 53 kc as received from the tuner-circuit discriminator or ratio detector.

Do not attempt to compensate beyond a 6-db roll-off at 53 kc. The network deemed proper, in accordance with Table 5-1, should be inserted at the adapter input rather than the discriminator output. Frequency-response measurements should also be made there. In this way, any attenuation introduced by the

TABLE 5-1.
Compensation network values.

Attenuation at 53 kc	Grid: R = 470K, C = 81 mmf	Grid: R = 1M, C = 39 mmf
1 db	R = 53K	R = 110K
2 db	R = 120K	R = 250K
3 db	R = 200K	R = 430K
4 db	R = 270K	R = 560K
5 db	R = 370K	R = 820K
6 db	R = 470K	R = 1 meg

interconnecting cable between tuner (or receiver) and adapter will be included in the measurements and calculations.

SCA INTERFERENCE

One of the problems encountered by users of multiplex equipment stems from the transition of radio stations from mono to stereo. A review of Fig. 2-15 will disclose the presence of a subscriber, or SCA, channel. This channel occupies the space from 60 to 74 kc, with the center at 67 kc.

SCA Frequencies

Prior to FM stereo broadcasting, the FCC permitted one or more SCA channels to be located anywhere in the channel spectrum from 25 to 75 kc. Popular frequencies which became more or less standardized were 41, 42, 57, and 67 kc—and occasionally, even 37 kc. However, it is easy to see why the only remaining SCA channel has been 67 kc—all the other frequencies would fall right on top of (or very near) the required stereo passband from 23 kc to 53 kc.

Effects on Multiplex Reception

Unfortunately, stations not yet engaged in stereo FM broadcasting are not required to alter their SCA service frequency or frequencies until they do convert to stereo. In listening to such stations monophonically, there is no evidence of the SCA channel. When multiplex circuitry (which specifically responds to frequencies in the range from 23 to 53 kc) is permanently added to the circuit, a loud whistling or swishing sound will be heard if a station is also broadcasting an SCA channel at, say 42 kc. Sometimes the noise will almost mask the program completely. As pointed out previously, certain multiplex equipment is provided with a Mono-Stereo switch which, when set to the Mono position, effectively cuts out the circuitry responsive to the subcarrier frequencies. Other circuits perform this function automatically. There are some, however, which have neither feature. Unless thought is given to this problem during installation, SCA interference may plague the listener on non-stereo stations (which, despite the rapid increase in stereo FM stations, are likely to be around for many years).

Corrective Switching—The proper approach when installing corrective switching is to provide for a straight FM position on the amplifier (if a separate amplifier is used). On most complete receivers (those with built-in stereo FM) an "FM-Mono" position on the selector switch accomplishes the same purpose. If a complete receiver is being equipped with a stereo adapter, the adapter outputs should definitely be fed to one pair of auxiliary inputs on the receiver. In this way the user will still be able to use his radio position for normal FM mono reception and then switch to "Aux" (some older receivers labeled the position "MX" in anticipation) when some other station is broadcasting stereo.

Hookup diagrams in Chapter 3 have already illustrated the proper procedure for making these interconnections, but now an additional and important reason for these specific instructions has been clarified.

DETECTORS

Detector problems in multiplex circuitry are rare, particularly where semiconductors are used. The "match" between a pair of diodes used in a given multiplex circuit is difficult to check under dynamic conditions. However, the diodes may be disconnected (one end only) if this portion of the circuit seems defective, and forward and back resistance of each in turn measured with a standard ohmmeter. The respective readings should be about the same for both diodes. In a properly operating diode, the forward resistance will generally be a few hundred ohms or less, whereas the backward (non-conducting) resistance will usually measure several hundred thousand ohms. A diode that measures low resistance in *both* directions is shorted (or partially shorted) and should be replaced. The symptom of such a condition is monophonic reception (a mixture of L+R) from one channel, while the other channel continues to perform properly (L only or R only). An open diode (reading very high resistance in *both* directions) will present the same symptoms in a matrix circuit; but in a switching circuit there will be almost no output from the defective channel. Whatever small output one may hear or measure is due to some arbitrary setting of the corrective matrix or separation control, or to the small amount of leakage current still getting through the defective diode.

Vacuum-tube detectors (such as the 6AR8 beam-switching tube previously discussed) are not so simply analyzed. The best approach, in the absence of a stereo signal generator, is to check operating voltages at the pins of the tube, in accordance with the manufacturers' diagrams. If the voltages seem correct but the performance symptoms still point to the detector circuit, simply replace the tube before going on to individual circuit components such as resistors or capacitors.

REGENERATION

Another seemingly insoluble trouble in multiplex circuits has begun to show up in certain installations—including converted tuners or receivers. The user, having thoroughly checked his equipment in accordance with Chapter 3, connects a stereo multiplex adapter. To his chagrin he not only gets very poor stereo separation, but suddenly encounters more distortion than he had ever heard monophonically—particularly if the station is weak, but not weak enough to have been ruled out.

Nine times out of ten the trouble is with the tuner or the receiver itself. The tuner was improperly aligned in the first place, and often the loading of the multiplex jack by some capacity (such as by the interconnecting shielded cable) is enough to "trigger" the tuner into IF regeneration or oscillation. Such regeneration is more prevalent on weak signals because the tuner or receiver AGC voltage is so small that each stage is operating at or near maximum gain. Regeneration to any degree in a tuner or receiver will narrow or compress the bandwidth of the IF circuits because the entire IF strip becomes an oscillator at 10.7 megacycles and this increases the effective Q of the total circuit. Of course, recovery of high-frequency subcarrier-sideband components now becomes a hit-and-miss proposition. Moreover, what little subcarrier component is recovered is usually severely distorted, both in amplitude and in phase.

Alignment

Fig. 5-8 illustrates a test-equipment setup suitable for sweep-aligning an FM tuner or receiver while the multiplex adapter is connected. A filament transformer supplies a low AC voltage used to externally modulate an FM signal generator ±75 kc. This voltage is also applied to the horizontal plates of the oscil-

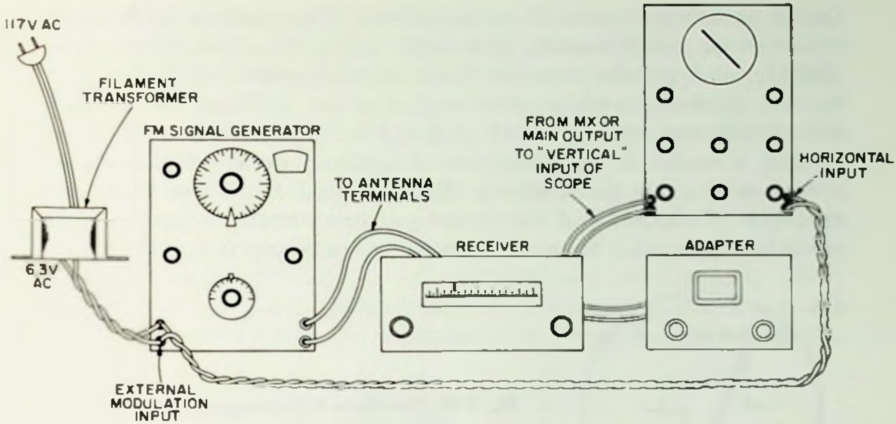
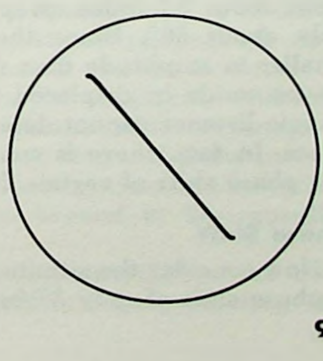


Fig. 5-8. Sweep-alignment setup.

loscope, which is set to the External Horizontal Deflection function to provide a 60-cycle sweep. The tuner (or receiver) and generator are tuned to the same arbitrary frequency, and the resultant scope trace examined for a 1,000-microvolt RF signal at the antenna terminals. Usually the presence of such a strong signal will erase any traces of regeneration, and only the straight line in Fig. 5-9 will be seen on the scope.

Next, the signal applied to the antenna terminals is gradually reduced in amplitude until some fundamental change is observed in the scope trace. The RF frequency of the generator (or receiver) may require slight retuning as this is done. If the receiver is indeed marginally regenerative, the scope trace will suddenly break at some lower signal strength and look like Fig. 5-10. This is an indication that the bandwidth is now but a fraction of the minimum ±75 kc required for decent stereo reception. The IF transformers should now be touched up slightly until the straight-line trace of Fig. 5-9 is restored

Fig. 5-9. Scope trace for ±75-kc bandwidth.



(or is as nearly restored as possible). Experiment with each transformer individually; tune both top and bottom slugs only slightly, and return them to their original positions if you notice no improvement. In this way, the set will not be thrown completely out of alignment. A good indication that no regeneration remains is the presence of random noise on the scope screen when the generator is disconnected from the tuner or receiver. If blocking of the scope pattern occurs, however, the set is fairly certain to oscillate with a weak input signal.

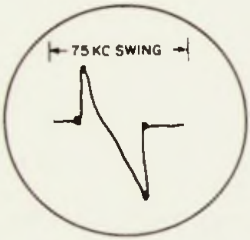


Fig. 5-10. Waveform indicating regeneration.

RF SIGNALS AND REFLECTION

The need for a strong input signal at the receiver, for proper reception of stereo, was pointed out earlier. But signal strength alone is not enough. In some questionable installations you may run into a disturbing effect called "multipath" interference—a term equivalent in FM reception to the familiar "ghosts" associated with TV reception. FM RF frequencies are in the same range as TV frequencies—in fact, the FM band (80 to 108 megacycles) is between TV channels 6 and 7. Moreover, FM signals have the same reflective characteristics as TV signals. In conventional monophonic reception, such reflections are delayed only a microsecond or so and are of no consequence except in extreme conditions. Fig. 5-11 shows that even a 15-kc tone, if delayed 0.5 microsecond (which represents a reflection from about 2.7 miles away), would be displaced in phase by only about 30°. Since the reflected wave is usually much smaller in amplitude than the primary wave, the two received waves would be displaced only 10° or so in phase. The monophonic listener cannot detect such small phase shifts of single tones. In fact, there is some question whether he can detect the phase shift of certain frequencies at all.

Phase Shift

Now consider the situation in stereo. As already pointed out, a phase shift of only 3° between suppressed 38-kc subcarrier at the transmitter and restored 38-kc carrier, or switching voltage, at the receiver will limit stereo separation to 30 db. The received 19-kc pilot, of course, determines the phase relationship of the internal 38 kc and the subcarrier sideband components. Fig. 5-12 shows the relative phase displacement of the 19-kc pilot carrier for a 2.7-mile reflection (similar to that in Fig. 5-11). The net phase displacement, after the "true" and reflected signals are added together is about 12° to 15°. Fig. 5-13 shows that the maximum separation would be about 18 db. Such multipath reflections are not unusual, particularly when the receiver is many miles from the transmitter and there are large steel buildings or mountainous terrain between them,

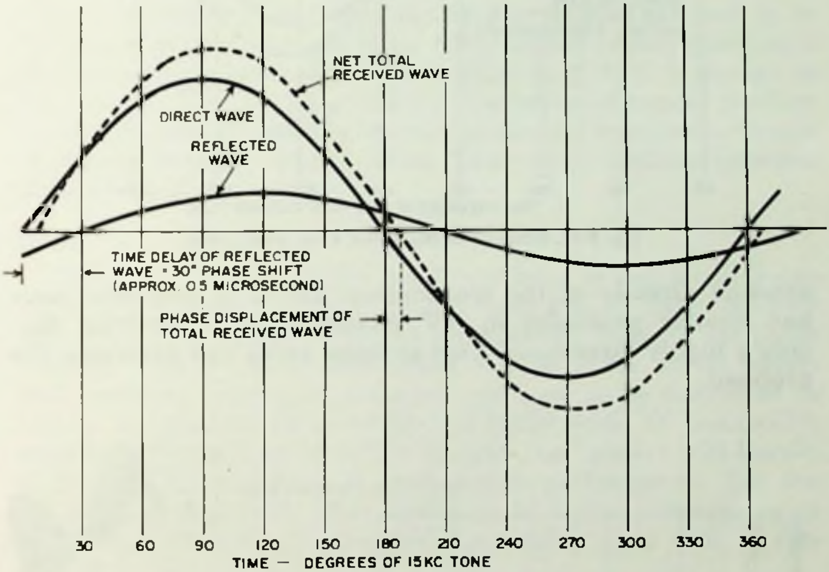


Fig. 5-11. Effect of multipath on 15-kc signal.

to the left or right of the line of sight. A maximum of 18 db of separation is not too serious—many stereo phonograph cartridges do not have much more. What is serious, however, is that these reflections generally waver in amplitude with momentary changing atmospheric conditions. This wavering is definitely audible; in fact, it is so severe that orchestral instruments sound as though they are wavering from their correct location to "center stage," and even beyond to the opposite channel.

The solution, of course, is optimum orientation of the outdoor antenna. This often means more than merely pointing the

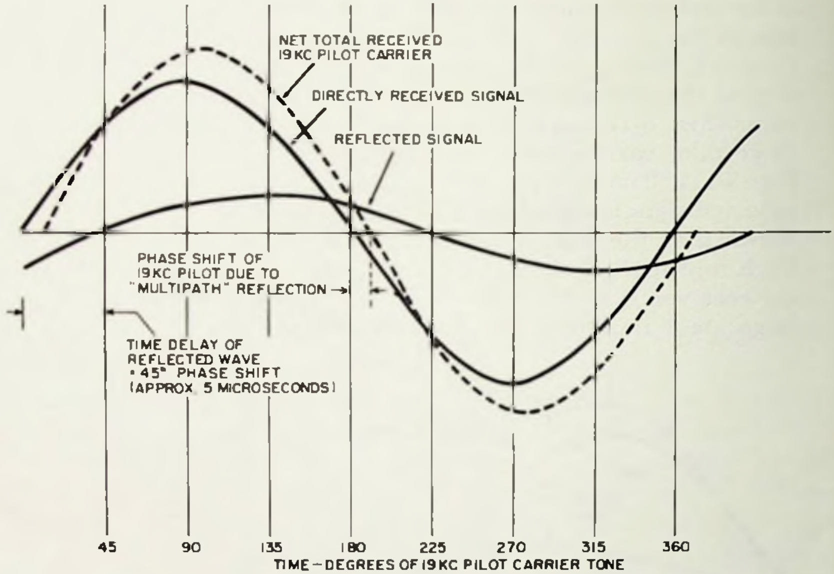


Fig. 5-12. Effect of multipath on 19-kc pilot carrier.

antenna directly at the transmitter. Those of you who have had similar problems in TV reception will recognize that only a highly directional Yagi antenna array can overcome the problem.

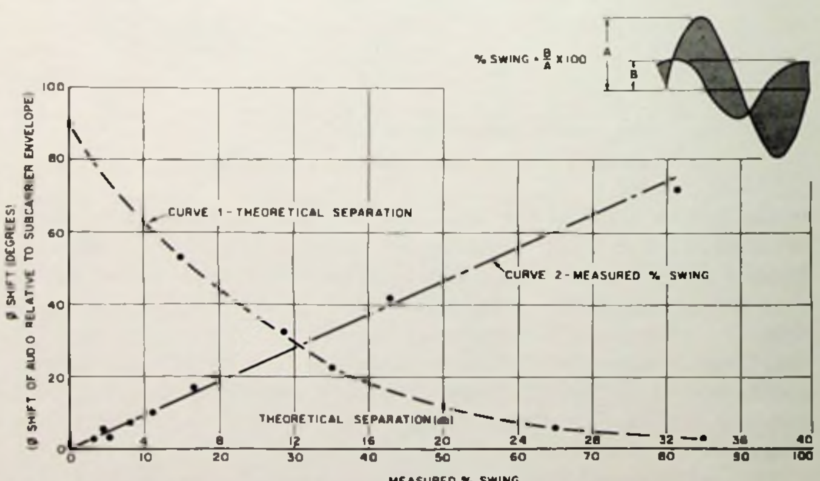


Fig. 5-13. Effects of phase shift on separation.

The moral of the preceding discussion is this: Identify the problem through audible effects. Don't pounce on the receiver or multiplex adapter; both are blameless.

TUNING

It is more important to tune in a stereo than a monophonic station properly. In many tuners and receivers, off-center tuning can deteriorate the separation as much as (and sometimes more than) a completely misadjusted separation control or local 19-kc oscillator. In receivers equipped with AFC (automatic frequency control), the AFC should be defeated (if a Defeat position is available) before the station is tuned in to the center of the channel. If no AFC Defeat switch position is available on the receiver or tuner (that is, if AFC is always in the circuit), find the approximate "center-of-channel" position experimentally by turning the tuning control from one extreme of station "lock-in" to the other. Then set it midway between these two points.

Drift

Drift in the phase of the local oscillator is not all that can degrade stereo performance. So can basic FM RF drift. All new sets described in this book can be assumed to have oscillator stability. But no such assumption can be made about tuners and receivers previously acquired and now to be converted to stereo. A tuner or receiver having fairly wide IF bandwidth may drift as much as 50 to 100 kc, and the listener will hardly notice any deterioration of monophonic performance. Yet the same amount of drift, after warm-up in stereo listening, may well result in little or no separation and a good deal of distortion.

It is therefore good practice, when listening to a stereo broadcast on an older tuner or receiver with adapter, to retune within fifteen minutes to a half hour. Usually, heat stabilization will have taken place by then, and no further retuning will be necessary.

CHAPTER 6

FM STEREO TEST EQUIPMENT AND MULTIPLEX CIRCUIT ALIGNMENT

Manufacturers of FM stereo receiving equipment were delayed pending the development of laboratory and production test equipment (just as radio stations were, by the lack of immediately available transmitter modification equipment, in converting to stereo broadcasting after the FCC decision of April 19, 1961). By late July, 1961, however, at least two manufacturers began shipping specially built FM stereo generators.

STEREO GENERATORS

The Crosby-Teletronics Model SG-292 stereo signal generator and the H. H. Scott Model 830 stereo multiplex generator came out almost simultaneously. Although these two designs differ—in much the same way that matrix and time-division-switching receiver circuits do—both develop identical composite signals. In other words, the difference is in design approach rather than final results.

Since these two pieces of equipment appeared on the market, at least two other manufacturers have announced similar test equipment—the Boonton Model 219A and Calbest Model MX 625 SG. The Crosby and Scott units retail at about $1,000 each. The Boonton Radio generator (Fig. 6-1) retails at about $750; and the price of the Calbest generator, designed more as a service than production instrument, has been announced at $495. The Calbest instrument also provides a complete RF signal (From 88 to 90 megacycles, nontunable) modulated by the composite stereo signal. The other three instruments provide an extremely accurate composite audio signal for modulating a standard FM signal generator. The generator to be modulated must have a flat response and phase characteristic from 50 cycles to 75 kc and the necessary sensitivity to be driven to full 75-kc deviation by the outputs of the stereo generators.

To further your understanding of the generation and reception of the composite stereo signal, let's examine the circuitry and operation of both the Crosby SG-292 and the H. H. Scott Model 830 in detail.

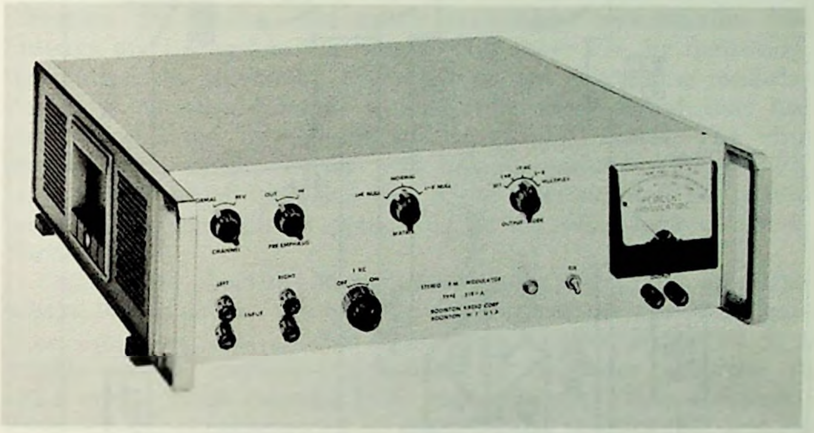


Fig. 6-1. Boonton Radio 219-A stereo FM modulator.

Crosby SG-292

A complete block diagram of the Crosby Model SG-292 is shown in Fig. 6-2, and the actual schematic diagram (less power supply) in Fig. 6-3. In both, the two audio signals (any frequency from 50 cps to 15 kc) comprising the left and right portions of a stereo signal are applied to input terminals A and B, respectively, on the front panel. These terminals are designated as A and B rather than "left" and "right" for purposes of generalization. Of course, in actual equipment use this arbitrary designation may be reversed.

Audio Input—Both the left and right input terminals are ungrounded and may be fed externally from either grounded or ungrounded sources having low impedances (up to approximately 2,000 ohms). With the A Phase switch in the "Normal" position and the Input Selector switch in the "A&B" position, the two signals are added in phase by one secondary winding of transformer T2 and one of T3. In this way the main, or L+R, channel is produced.

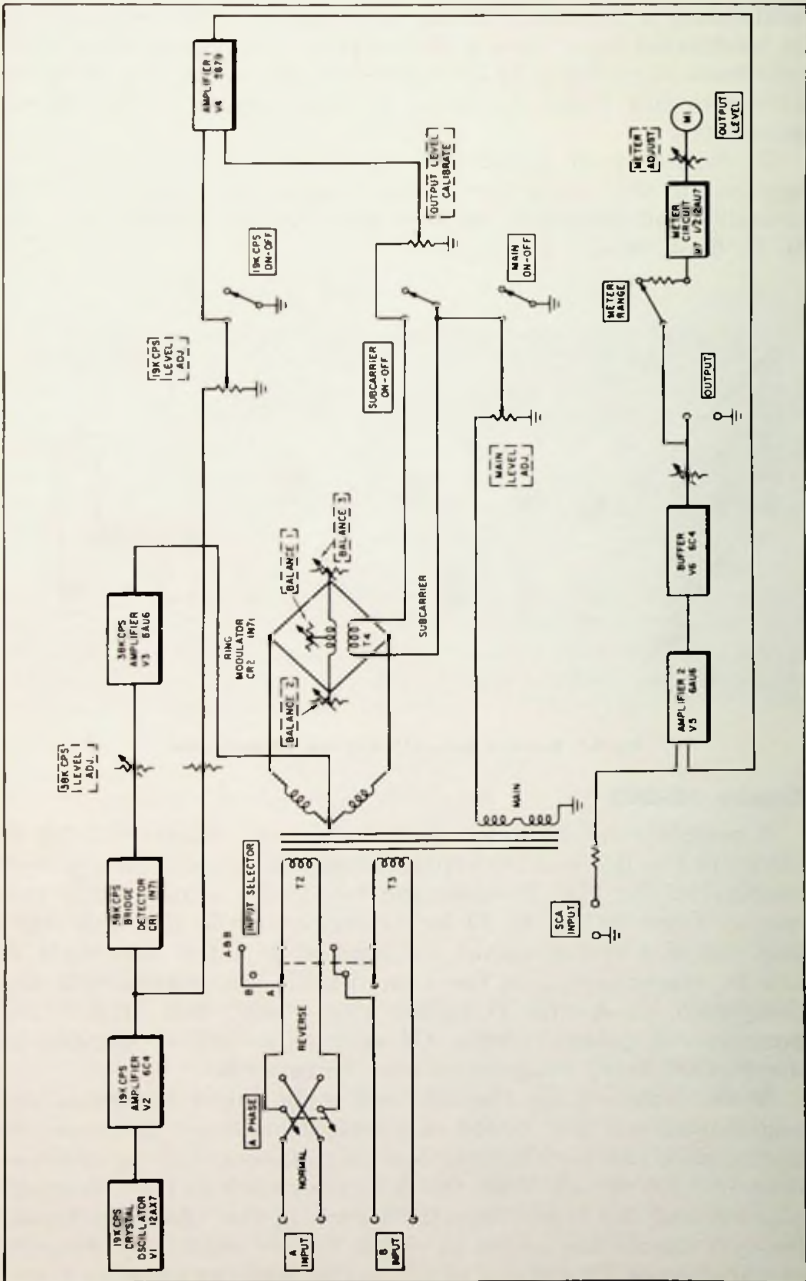


Fig. 6-2. Block diagram of Crosby-Teletronics SG292.

Ring Modulator—The difference (L−R) subcarrier channel is produced from the remaining two secondary windings of transformers T2 and T3. The windings are so connected that the R input signal is reversed in phase. Thus an audio signal, L−R, is generated across them. This signal and the internally generated 38-kc carrier signal are applied to Ring-modulator circuit CR2. Whenever carrier frequencies are applied across one pair of ring-modulator terminals and modulating frequencies are applied across the other, the carrier is nulled, or canceled. As a result, only the sum and difference frequencies between the audio modulating frequency and carrier frequency are present in the output. But these are, by definition, the subcarrier sidebands. Thus, the output of the ring modulator is a double-sideband, suppressed-carrier signal that has been modulated by the difference (L−R) signal. The ring modulator is supplemented with balance controls (P3, P4, P5, and C34), which prevent the 38-kc carrier and the audio modulating signal from appearing at the secondary of transformer T4.

Generation circuits—Before continuing along the amplifier chain, let's examine the circuits which generate the 38-kc carrier signal as well as the 19-kc pilot carrier. Stage VI is a highly stable, cathode-coupled, 19-kc oscillator accurate to ±2.0 cycles (thus meeting FCC requirements). It is decoupled from V2, the succeeding 19-kc amplifier, by resistors R7 and R8 to avoid any loading effects on the oscillator itself. The 19-kc pilot carrier is taken off V2 and developed across resistor R9. It is then coupled to the 19-kc level-adjust potentiometer through resistor R10 and capacitors C9 and C30. These two capacitors are selected, during alignment of the SG-292, so that the 38-kc subcarrier and 19-kc pilot carrier cross the zero axis simultaneously, as seen in the output of the instrument. This is the all-important phase relationship discussed earlier. The secondary of transformer T1, in the plate circuit of stage V2, applies the amplified 19-kc signal to bridge rectifier CR1. A parallel tuned circuit (L2, C10, and C11) selects the desired second-harmonic (38-kc) signal. This signal is amplified in V3 and applied to the ring modulator, as described.

Amplifier—Stage V4 employs a low-noise amplifier (type 5879). The 19-kc pilot carrier is added, at V4, to the main channel and the subcarrier-sidebands channel, which were added before being applied to the grid of V4. A 76-kc rejection filter (L3, C15, and C15a) and a 38-kc rejection filter (L4, C16, and C16a) reduce leakage of these channel frequencies to V4. No time delay has to be inserted when the main and subcarrier

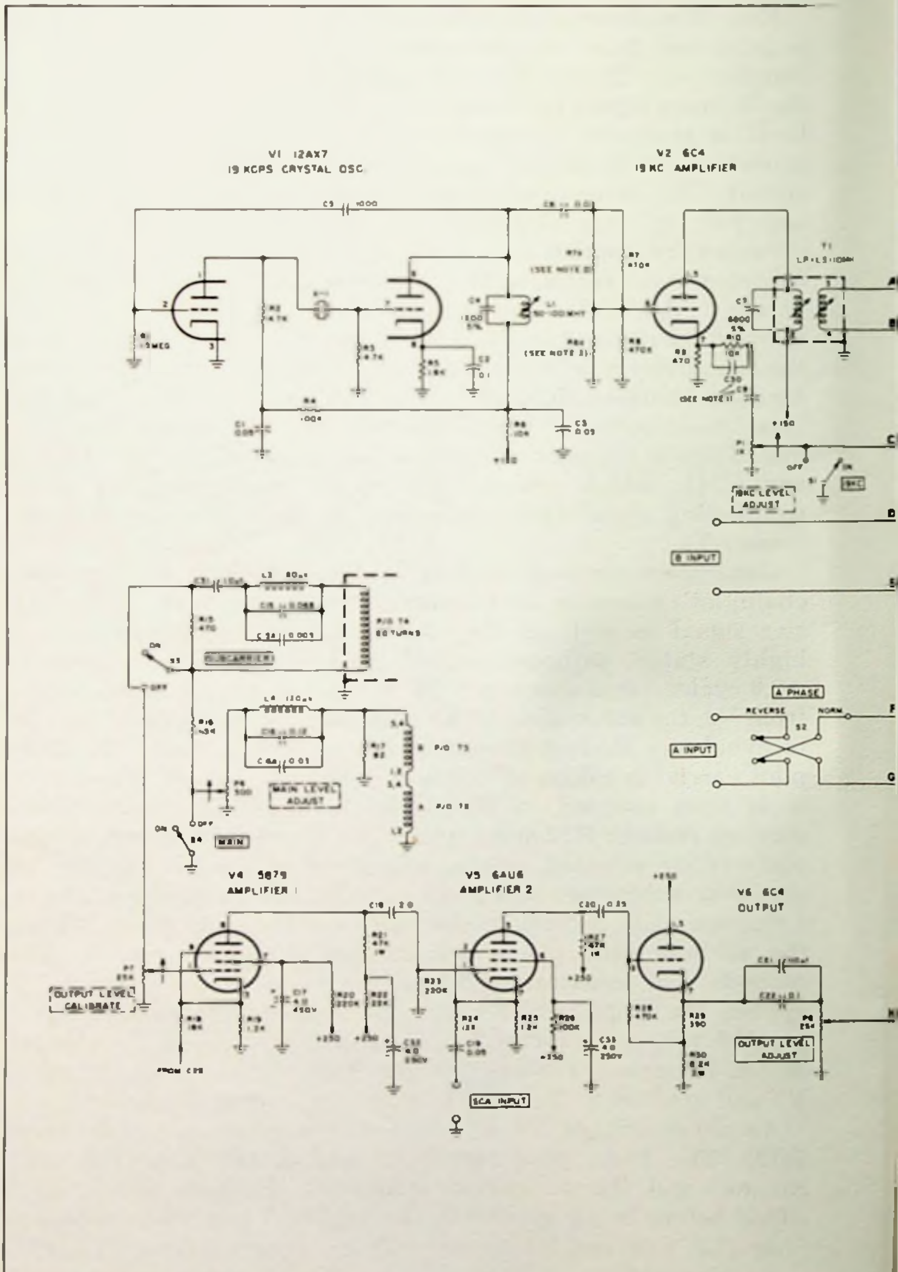
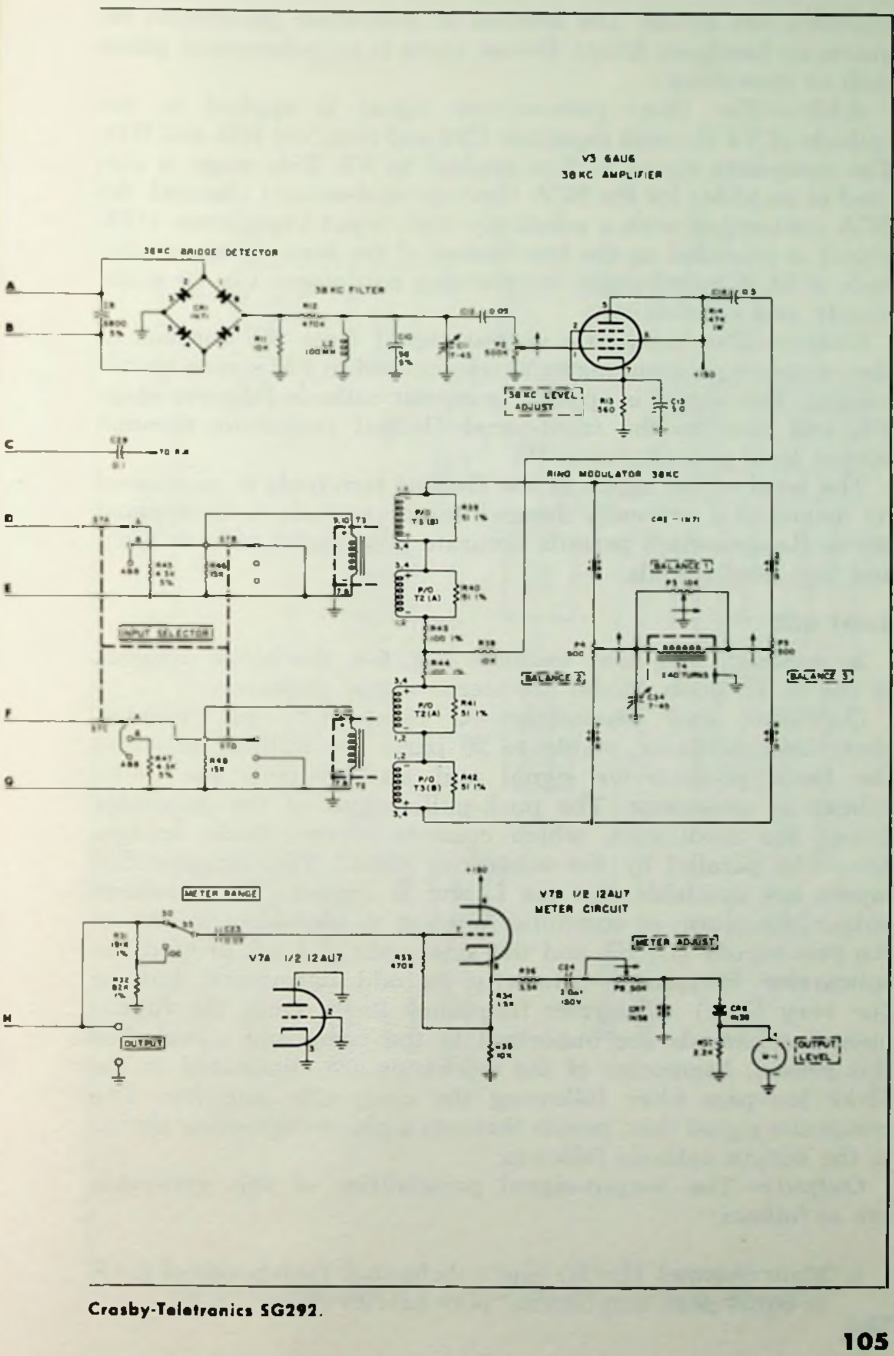


Fig. 6-3. Circuit of



Crosby-Teletronics SG292.

channels, are added. The method of subcarrier generation requires no bandpass filters. Hence, there is no subsequent phase shift or time delay.

Adder—The 19-kc pilot-carrier signal is applied to the cathode of V4 through capacitor C29 and resistors R18 and R19. The composite signal is then applied to V5. This stage is also used as an adder for the SCA (background-music) channel. An SCA connection with a relatively high input impedance (12K ohms) is provided on the front panel of the instrument, so that tests of SCA interference in receiving equipment can be made simply and conclusively.

Output—The composite output signal from V5 consists of the complete program material transmitted in FM stereo broadcasting. This signal is applied to output cathode-follower stage V6, and then to the front-panel Output connector through output level potentiometer P8.

The level of the signal at the Output terminals is monitored by means of a critically damped meter circuit. A front-panel Meter Range switch permits accurate adjustment of both high- and low-level signals.

Scott 830

In contrast, let's next examine Fig. 6-4, the block diagram of the H. H. Scott Model 830 stereo signal generator.

Oscillator and Modulator—A 19-kc (±2-cps) crystal-controlled oscillator, stable to 20 parts per million, provides the basic pilot-carrier signal and also controls the 38-kc subcarrier generator. The push-pull output of the generator drives the modulator, which consists of two diode bridges driven in parallel by the subcarrier signal. Two single-ended inputs are available, for the L and R signals. The resultant output waveform at summing junction A contains the sum of the two inputs (L+R) and the sidebands of L−R around the subcarrier frequency—primarily its odd harmonics, but no (or very little) subcarrier frequency itself. Only the fundamental sidebands are important in the composite signal. For this reason, harmonics of the sidebands are eliminated in the 53-kc low-pass filter following the composite amplifier. The composite signal then passes through a phase-correcting circuit to the output cathode follower.

Outputs—The output-signal possibilities of this generator are as follows:

1. Main channel (L+R) and subchannel (sidebands of L−R in equal peak amplitudes; pilot carrier off).

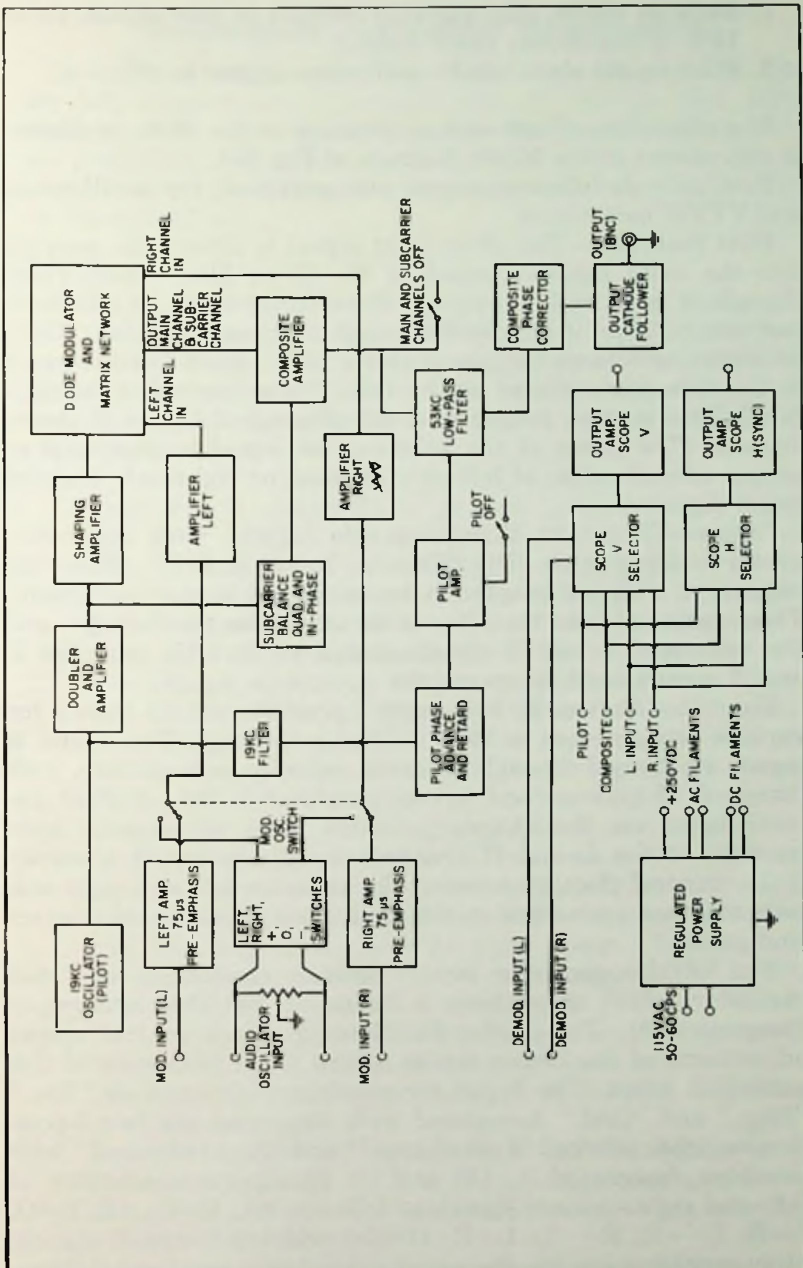


Fig. 6-4. Block diagram of H. H. Scott 830 stereo multiplex generator.

2. Same as above, plus variable amount of pilot signal, up to 10% of maximum rated output.
3. Pilot signal alone (sub- and main channels off).

The pilot-channel network originating at the 19-kc oscillator, is also shown in the block diagram of Fig. 6-4.

Two cathode-follower stages are provided, for oscilloscope and VTVM monitoring.

Pilot Insertion—The 19-kc pilot signal is ultimately inserted into the main signal channel at the 53-kc filter input. First, though, it is passed through a 19-kc tuned filter to eliminate harmonics. Then it is passed through a phase corrector, which advances or retards the phase of the pilot signal (with respect to the subcarrier phase) as the Pilot Phase control is manipulated. This insures proper over-all phasing of L and R stereo signals. (The phase of the pilot-carrier signal is important to proper identification of left-only channel or right-only channel input signals.)

Inputs—There are three separate inputs, with input-connector binding posts. The "Demod. Input, L & R" allows the outputs of a stereo adapter to be connected to the instrument. These terminals can then be connected to the oscilloscope, and the recovered L and R signals compared directly with the L and R inputs used to create the composite signal.

The "Modulation, L & R input" provides stereo inputs for various sources such as tape or disc recordings. The L and R inputs are passed through separate, equal-gain amplifiers, with standard 75-microsecond pre-emphasis included. A dual potentiometer on the chassis provides level adjustment; gain equality of the L and R channels is adjusted with a screwdriver control (factory preset). The separate left and right outputs are then connected to the modulator input signal connection points.

The "Audio oscillator input" permits connection of a test oscillator which must have a balanced 500/600-ohm output (ungrounded). The Audio Oscillator Balance control allows adjustment of the center tap to insure exact balancing of this push-pull input. The input terminals are denoted as "Pos.," "Neg.," and "Gnd." Associated with this input are two 3-position switches marked "Left channel" and "Right channel," with positions designated +, Off and −. This permits selection of left- and right-channel signals as follows: $\pm L, R=O$; $\pm R, L=O$; $L=R$; $L=-R$; $R=-L$; $L+R=O$. The selected L and R signals, after amplification by the equal-gain stages mentioned above, appear as inputs to the modulator circuit.

PRELIMINARY ALIGNMENT ADJUSTMENTS

Several manufacturers of multiplex products maintain that the only correct way to properly align or realign their equipment is with a multiplex signal generator. Of course, there are preliminary adjustments that can be made without the aid of such expensive alignment equipment; these procedures will be discussed first.

Filters

Filter alignment, whether the circuit is of the "matrix" or "switching" variety, is quite simple and requires no more complex equipment than an audio oscillator, an AC VTVM, and possibly an oscilloscope. Refer once more to Fig. 5-1 and follow the step-by-step alignment instructions prescribed by the manufacturer.

1. Short lug D of coil T2 to ground (chassis) with a short clip lead. (This point is the top of the oscillator tank circuit. Shorting it out defeats the oscillator.)
2. Connect an audio generator to the "Multiplex Input" jack. Set the generator to 67 kc. Adjust its output level to 0.3 volts AC. (The accuracy of calibration of the audio generator used here is very important. An error of only 3 or 4 kc would completely upset the next alignment step.)
3. Connect a scope and AC VTVM to lug 4 on terminal strip BB. (Schematically, this is the junction point of detector diodes X1 and X2.)
4. Plug in the multiplex adapter and turn on the power.
5. After the unit has warmed up (at least a ten- to fifteen-minute warm-up is recommended), adjust coil L1 for minimum output. The voltmeter should read .02 volt or less. This adjustment will be quite sharp. (The "infinite" attenuation point of the *m*-derived bandpass filter is being adjusted here. The more carefully you perform this step, the less trouble you are likely to run into when a station broadcasts SCA and stereo simultaneously.)
6. Disconnect the shorting clip lead of Step 1. (Oscillator now functioning.)
7. Leave the generator connected to the "Multiplex Input" and set it to 19 kc ±100 cps (actually, even greater accuracy is preferred, if possible). Adjust the generator output level to 0.1 volt AC..
8. Leave the scope and voltmeter connected as before and adjust oscillator coil T2 for maximum output. Observe the

scope to make sure the 38-kc output of the doubler is locking on the 19-kc input signal. NOTE: This adjustment will be very sharp and should be made carefully. (Unfortunately, the mere "locking in" of the 19-kc and self-generated 38-kc signals is insufficient proof that they have the proper phase relationship. The final adjustment must be performed during a stereo broadcast, or with a stereo signal generator if available. See below.)

9. Next adjust coil T1 for maximum output. (T1 is the 38-kc tuned-plate tank circuit. It will also be quite sharp and should be peaked carefully.)
10. Finally, adjust coil L2 (the bandpass-determining element of the *m*-derived bandpass filter) for maximum output. This adjustment will be very broad, so you may have to rock the slug back and forth to find the peak.
11. The voltmeter should read about 8 volts AC. (This voltage reading represents all internally generated 38-kc carrier, since no composite signal is as yet being fed to the adapter.)
12. Reset the audio generator to 1,000 cps and adjust its output level to 3 volts. (This is equivalent to an L+R, no subcarrier-sideband components, signal.)
13. Disconnect the scope and voltmeter, and reconnect them to the "Left Channel Output" jack. Set the voltmeter on 10-volt range.
14. Rotate both Output Level controls fully clockwise.
15. With the Channel Separation control turned maximum counterclockwise and the voltmeter on the 10-volt range, there should be no output. (Actually, *plenty* of output will be seen on the scope if its gain is turned up. This output will be residual 38-kc superaudible subcarrier, however; it will not be the 1,000-cycle audio tone being fed in at the input. The tone is not now contributing to the matrixed output because the L+R or separation control is completely turned down.)
16. With the Channel Separation control set fully clockwise, the voltmeter should read between 6 and 10 volts and the scope show a clean sine wave. (This step merely checks the L+R channel, including stages V1A and V1B as well as the V3A and V3B cathode-follower output stages. The L−R recovery has not yet been checked.)
17. Reconnect scope and voltmeter connections from "Left Channel" to "Right Channel" output jack.
18. Repeat Steps 16 and 17 to check the L−R channel and right-channel output.

Adjustment With Broadcast Signal

With respect to matrix adapters such as this one, the importance of the following instructions cannot be overemphasized. Fully 90% of all complaints can be solved by adhering to these simple instructions.

Tune in a stereo broadcast. Do not take it for granted that stereo is being broadcast unless you hear the announcer actually say so. Published schedules cannot anticipate technical difficulties.

Oscillator and Separation—Once you have established beyond a doubt that stereo is being broadcast, turn the Channel Separation control (L+R) completely counterclockwise to ground out the L+R signal contribution. You should now hear equal sound from both speakers, unless your 19-kc oscillator is completely out of lock with the incoming 19-kc pilot carrier. The sound, however, will most probably be low in amplitude and quite distorted. To correct, *slowly* rotate the 19-kc oscillator slug to the left or right slightly until the sound is maximum in volume and minimum in distortion. Rotation too far to the left or right will cause the oscillator to go "out of lock" and a motorboating sound will be heard which rises in pitch as the slug is rotated further off-tune. Somewhere between these two end points of "lock in" there is only one correct setting of the slug, corresponding to maximum output and minimum distortion. If there seems to be more than one peak, choose the one farthest from either the left-rotation or right-rotation "out-of-lock" point. This will usually be the point at which the sound is loudest and clearest. On most adapters, less than one full rotation of this adjustment is enough to cause the oscillator to go out of lock on either the high or low side of 19 kc.

Once its correct setting has been found, the 19-kc local oscillator should never have to be readjusted. This adjustment must be made, when adapters are connected to tuners or receivers, because of the slight phase shift which nearly every tuner or receiver introduces to the higher subcarrier and pilot frequencies as compared with the main-channel audio. Most manufacturers try to "average out" this condition in their factory alignment. However, this touch-up adjustment is essential during installation if optimum results are to be obtained with *any* adapter.

Once the 19-kc oscillator adjustment has been completed, adjust the Channel Separation control (L+R) for best stereo separation as heard from the speaker systems. Best separation can be recognized by one of the following conditions:

1. Maximum difference between the program material heard from the left and right speaker systems.
2. When the Channel Separation control is set for minimum sound from the opposite speaker system during a single channel broadcast.

Once properly set, the Channel Separation control should not have to be readjusted. However, variations in the program material being broadcast and between signals from different stations may make occasional readjustment necessary. For this reason, some manufacturers place this control on the front panel.

Alignment of the 19-kc adjustment and separation controls in a switching-type adapter or multiplex circuit under actual listening conditions is a bit more tricky because it is impossible to deactivate the L+R signal while adjusting sub-carrier L−R recovery independently. Recall that in these circuits the entire composite signal is handled as an entity. Nonetheless, misadjustment of any coils relating to frequency and phase of the 38-kc switching voltage (that is, the 19-kc coil itself, the 38-kc doubler coil, or any other 19- and 38-kc tuned circuits in amplifier stages, etc.) reduces the amount of recovered (L−R) component. Since the main audio (L+R) is unaffected by the switching voltage, however, it is always fully recovered. Thus, adjustment can be made by carefully tuning the 19-kc oscillator until maximum total output is heard with the Separation control set to its midpoint. Moreover, severe misadjustment—either clockwise or counterclockwise—will cause the oscillator to "lose sync" and produce a motorboating sound. Between the end points, however, is a point of maximum output *and* best separation. Maximum output occurs because the L−R (although not separately heard as such) adds to and subtracts from the fixed L+R while being adjusted to maximum. As the L−R reaches its correct value (or its highest value for the particular setting of the Separation control), the separation approaches its best value. A graphic illustration of what takes place in this adjustment procedure is shown in Fig. 6-5.

Finally, the separation adjustment (a signal cross-coupling affair for varying L+R or L−R in some common-cathode configuration of an output stage) is set for best separation.

PROFESSIONAL ALIGNMENT

In their customer manuals, almost all manufacturers recommend the use of a professional FM stereo generator for accurate adjustment and alignment of any multiplex equipment. Although no one can dispute this recommendation, not every reader has such equipment, costing several hundred dollars, available for doing the job in the most precise manner. For this reason empirical, "off-the-air" alignment techniques have been stressed thus far.

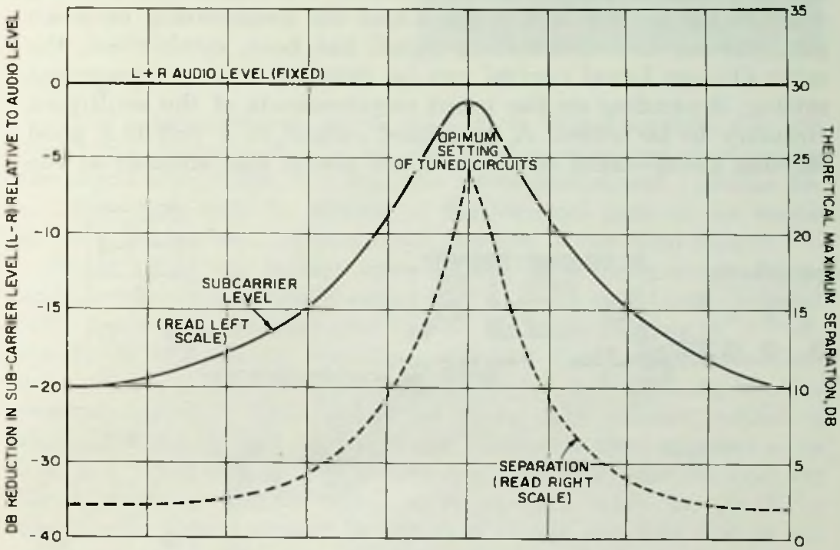


Fig. 6-5. Separation curve.

The difference between 20 and 30 db of separation in stereo program material may be discernible to only the most experienced listener. Nevertheless, to "get the maximum" from multiplex equipment requires the use of such professional alignment equipment.

Setup

Fig. 6-6 illustrates the test setup for checking and aligning multiplex equipment using the Crosby SG-292 signal generator. In this instance, only the multiplex circuitry is to be checked (as if the composite signal had already been recovered from the tuner section of the receiver).

The handiest way to use this instrument is to connect a single audio oscillator to the A and B inputs. The switching facilities on the instrument itself will enable you to check A only, B only, A=B (monophonic), or even A = −B (all subcarrier-sideband information).

To set up the instrument, turn off the Main Channel and Subchannel toggle switches. The output will then consist of only 19-kc pilot carrier. Turn the Output Level control fully clockwise to maximum. The front-panel meter will read approximately 10 (on the "30" scale). Whatever the reading, apply nine times as much audio input with the Input Selector switch set at "A&B" (monophonic), so that the combined reading reaches 100 on the "100" scale. Once the relationship between pilot carrier and modulating signal has been established, the main Output Level control can be reduced to any convenient setting, depending on the input requirements of the multiplex circuitry to be tested. A combined output of 1 volt is a good starting point—most tuners recover about this amount at the

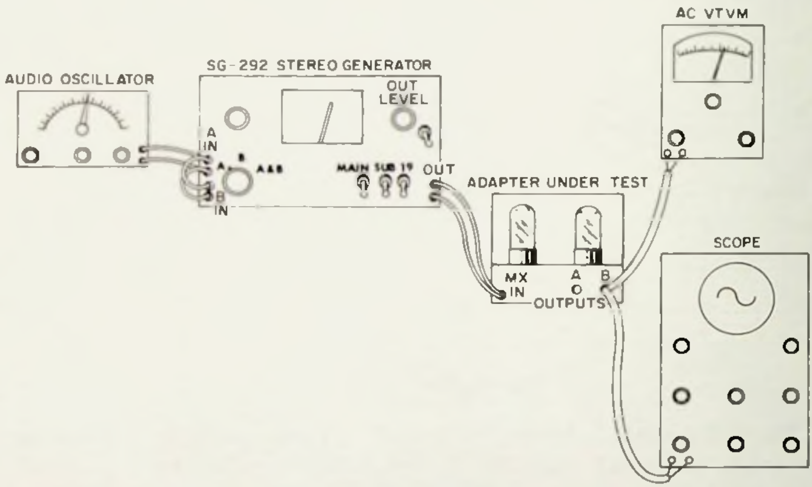


Fig. 6-6. Setup for aligning and testing multiplex equipment.

discriminator output under full modulation. Since the pilot carrier has already been set up as being one-tenth of the total composite signal, a setting of 1 volt of combined output means that approximately 0.1 volt of pilot carrier is available at the output. With the selector switch set in the "A&B" position for the moment, the output will consist of only a sine wave of the audio input with the 10% amplitude pilot carrier superimposed on it, as shown in Fig. 6-7. To familiarize yourself with this or any other such test instrument, it is advisable to hook up a scope and observe the waveforms as they are illustrated.

Connect the output of the generator to the input of the multiplex circuitry. Also connect a VTVM and scope to either the left or right output of the multiplex circuitry.

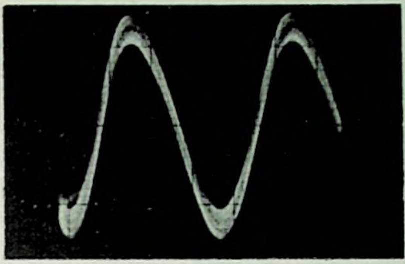


Fig. 6-7. L = R with pilot (19 kc) also present.

Operation

Since a monophonic signal (A=B) is being applied, the left and right outputs should be identical sine waves duplicating the input waveform, but less the 19-kc component. (Some 19- or 38-kc fuzz may be visible, if the vertical gain of the scope is too high, but should be a small portion of the total output.)

Phase—For the fastest over-all check of adapter performance, reverse the phase of either the A (left) or B (right) input with respect to its alternate signal. By thus making A = −B, the L+R component vanishes and the L−R component becomes 2A (or 2B). The total modulation now consists of subcarrier sidebands only (plus the 19-kc pilot carrier, which is always present). The output of the generator then appears as in Fig. 6-8. The left or right outputs, however, should still be sinusoidal and exactly equal to what they were for A=B in amplitude. (They should be equal in phase, too, but this is not easily checked by mere scope observation. If equal amplitude is observed under the two conditions stated, the phase characteristic probably is also being maintained.) Usually the sine wave observed at the outputs for A=−B has a smaller amplitude than for A=B. If so, the unit needs adjustment or alignment. Adjust or trim the tuned circuits of the 19-kc amplifiers, oscillators, etc., and of the 38-kc doubler, always tuning for maximum recovered sine wave at either the left or right outputs of the multiplex circuitry.

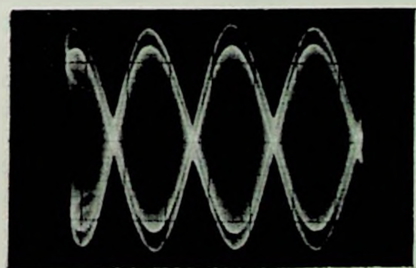


Fig. 6-8. L = (− R) signal.

Separation—If the highest signal falls short of being equal in amplitude to the recovered sine wave under L=R conditions, the remainder may be obtained in switching-type circuits by varying the separation control. In matrix types, after best recovered sine wave is obtained under L=−R conditions, the Separation control may then be used to vary the L=R recovered signal upward or downward until it equals the subcarrier-sideband recovered signal (L=−R). It is quite simple to observe both conditions as these adjustments are being made. Since the SG292 contains a toggle switch which reverses the A signal with respect to the B input, you can alternately create an A=B or an A=−B signal with a flip of the switch. When the two amplitudes of recovered audio have been made as equal as they can be, it is time to switch the selector to A-only or B-only. The "active" channel output will display a recovered sine wave of approximately half the previous amplitude observed, whereas the null channel should have virtually no output. Final trimming of the Separation control is now accomplished by rotating it for the best null obtainable from the inactive channel.

This procedure can be repeated at any frequency of audio input. It is advisable, however, to get the best null at some mid-frequency such as 1,000 cycles and then to measure the separation (from L to R or R to L) at other frequencies.

Step-by-Step Alignment Procedure

To arrive empirically at a compensating network such as those described in the previous chapter, it is advisable to test separation at some fairly high frequency. Altec Lansing, in its instruction manual covering the operation and alignment of the Model 359A *Stereoplex* unit, recommends making separation measurements at 11 kc. These instructions are reproduced here, since the professional alignment procedure they represent is an ideal way to guarantee performance. A full schematic diagram of this unit is presented at the beginning of Chapter 7, and it is suggested that you refer to this diagram while following the step-by-step alignment procedure.

1. Connect a shielded cable from the multiplex output jack of the tuner or receiver to the "MPX In" jack on the adapter.
2. Turn on the receiver, adapter, signal generator, etc., and allow at least 15 minutes warm-up time.
3. Tune the receiver in on the RF signal from the generator. (The manufacturer is suggesting that the stereo generator

be used to externally modulate a regular FM signal generator so that the entire receiver is actually being checked, rather than just the multiplex circuitry. For this test setup, see Fig. 6-9). Note: This tuning must be extremely accurate. If AFC is available, tune with the AFC OFF, and then switch to AFC "ON." (Altec Lansing confirms our earlier warnings in this regard!)

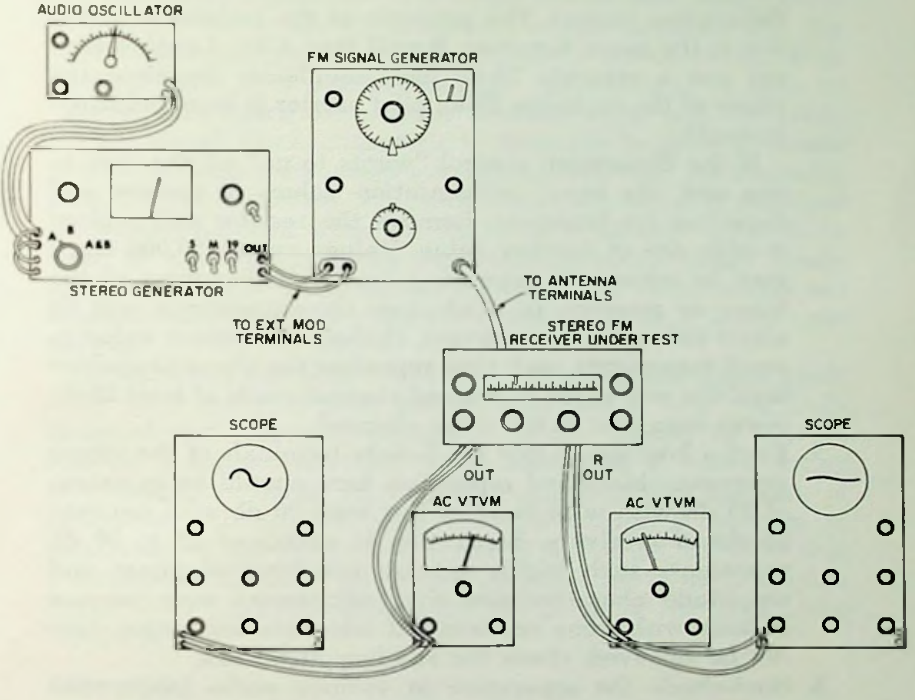


Fig. 6-9. Setup for checking performance of FM stereo receiver system.

4. Feed an audio signal (approximately 11 kc) into the "L In" jack of a stereo signal generator. Adjust the input-signal level for 50% modulation (of the FM generator).
5. Connect an AC VTVM to either the L or R output of the adapter. Note: Two meters will facilitate adjustment, since both channels may be monitored simultaneously. One of the outputs should have a higher reading, indicating some measure of stereo separation (even prior to final alignment).
6. Monitor the lower-output channel by turning the Pilot Phase adjustment slightly. When a dip is reached in the

meter reading, turn the separation control slightly in either direction, and retune the pilot-phase coil for another dip. If it is lower than the previous dip, the Separation control is being moved in the right direction. By adjusting it and then the Pilot Phase control, the point of greatest null, or dip, in meter reading should be obtained. (On most other adapters, you would be alternately adjusting the 19-kc oscillator coil, 38-kc coil, and the Separation control. The principle of the technique to follow is the same, however. Recall that Altec Lansing does not use a separate 19-kc local oscillator; therefore the phase of the incoming 19-kc pilot carrier is being adjusted instead).

If the Separation control "wants to go" all the way to one end, the input compensation values of resistor and capacitor are incorrect. Remove the resistor and replace it with one of another value. Values up to 250,000 ohms may be necessary, depending on the IF bandpass of the tuner or receiver. In most cases the capacitance will be about 150 mmf. In any event, change the resistor value in small increments, each time repeating the above procedure until the null of the unwanted channel reads at least 35 db lower than that of the other channel.

7. Feed a 2-kc signal into the L-only terminals of the stereo generator. Measured separation here should be in excess of 25 db with wide-band and at least 20 db with narrow-passband receivers. Separation in excess of 25 to 30 db represents such highly precise matching of phase and amplitude characteristics that adjustment may become tedious with some receivers. If adequate separation cannot be achieved, check the receiver alignment.
8. Spot-check the separation at various audio frequencies with left and then right input. Slightly adjust the Separation control to optimize the separation on both channels (or average it out between channels) for all frequencies.

Finally, Altec Lansing ends its instructions with the admonition that "An accurate FM stereo signal generator must be employed for adapter alignment."

19-kc Synchronization

On most commercial generators, the 19-kc pilot carrier can be turned on and off without altering the rest of the composite signal. This feature is useful in determining the lock-in ability of the local oscillator. In fact it is a good idea, when first adjusting or aligning multiplex equipment, to turn the 19-kc pilot off altogether. In this way, the free-running frequency of the local oscillator may be adjusted very close to 19 kc by listening for (or observing) the absence of motor-boating in the speaker outputs. Motorboating indicates that the internal oscillator is oscillating at other than 19 kc and that the resultant doubling is not at 38 kc—but at some other frequency which "beats" with the subcarrier sidebands to produce the characteristic "putt-putt-putt." The lower the frequency of the motorboating, the closer the free-running local oscillator is set to 19 kc. With extreme care, it is possible to align the oscillator so that no "beats" take place for several seconds, even with no pilot-carrier signal. Such alignment is desirable because it means proper and almost effortless synchronization with the incoming pilot. To put it another way, much less pilot signal is required for synchronization when the oscillator is already tuned at or near 19 kc. This becomes important in considering "weak-signal" reception of stereo, where often, the receiver has not reached full FM limiting and the available 19 kc is lower than normal.

An excellent test is to lower the total composite signal gradually until the recovered stereo "goes out of lock". A well-designed and -adjusted multiplex circuit, should be able to tolerate a signal reduction of at least 5 to 1 before losing synchonization. Also, in quality equipment the amount of separation will remain fairly constant despite a decrease in amplitude of the composite signal. This would indicate that a decrease in the 19-kc pilot-carrier voltage fed to the multiplex circuit does not appreciably alter the phase of the local 19-kc oscillator.

Distortion

Distortion measurements may be made on multiplex equipment by connecting conventional harmonic-distortion analyzers to the channel outputs—provided certain inherent limitations of this method are recognized. As was pointed out, the residual RF (consisting of some 19 kc and 38 kc, and possibly higher harmonics of these frequencies) in a well-designed multiplex circuit should be at least 30 db below full output. A harmonic-distortion analyzer, however, cannot discriminate between extraneous signals which are harmonics of the desired audio signal, and other random additional signals. A level of −30db represents approximately 1% of signal. Thus, for distortion measurements on quality equipment to be meaningful, a sharp filter (low-pass) having a cutoff frequency of about 12 to 15 kc should be inserted between the channel outputs and the distortion analyzer. The filter will eliminate the residual RF so that only the harmonics of medium to medium-high audio frequencies (the true distortion products) will be read.

Alignment Without Stereo Generator

In the absence of a stereo generator, certain filter alignment adjustments can be made with an accurately calibrated audio generator, an AC VTVM, and a scope. These adjustments were covered in Chapter 5. However, a tabulation of desired responses is summarized in Table 6-1.

The detailed descriptions of adjustment and alignment using stereo signal generators may seem a bit complex to those of you who have no such equipment. On the other hand, familiarity with multiplex circuitry comes with having worked on many units, and with a complete understanding of the individual circuit stages which translate what at first looks like a rather complicated signal into clean, recognizable left and right stereo channels. As an aid in familiarizing you with various forms of multiplex circuitry, Chapter 7 is devoted to an analysis of numerous commercially available circuits.

TABLE 6-1.

Adjusted-filter response characteristics.

Filter Type	Circuit Type	Description of Response
15-kc low-pass.	Matrix.	Output flat from 50 cycles to 15 kc. Down 3 db at approx. 20 to 23 kc (depending on specifications).
19-kc tuned circuits.	Matrix and switching.	All circuits peaked at precisely 19 kc (local oscillator temporarily disabled). Flat response from approx. 25 to 50 kc; down 3 db at 23 and 53 kc. Max. atten. at 67 kc.
Bandpass.	Matrix and switching.	Flat response from approx. 25 to 50 kc; down 3 db at 23 and 53 kc. Max. atten. at 67 kc.
Low-pass.	Switching.	Flat response from 50 cycles to 53 kc; max. atten. at 67 kc.
Notch filter (adjustable).	Matrix and switching.	Mat. atten. at 38 kc (in some types de-emphasis should be —11 db at 8.2 kc).
SCA band elimination.	Switching.	Mat. atten. from 60 to 75 kc.

CHAPTER 7

LATEST MULTIPLEX CIRCUITS

In the field of electronics, and more particularly in the fast-moving field of stereo high fidelity, it is difficult if not impossible to publish circuit data which is as up-to-date as today's newspaper. In the brief span of time since the first multiplex adapters and receivers became commercial available, many manufacturers have already come up with "Model 2" and some, even "Model 3."

Model changes do not always mean complete redesign. Just as in the automotive field, next year's models of multiplex equipment may look different outwardly and will undoubtedly contain internal changes as well. They will, however, be based on last year's circuits with improvements in performance or production efficiency. For this reason, a presentation of as many current circuits as possible will be useful for present understanding, and as a basic background for the multiplex circuits which are sure to follow in the years ahead.

ALTEC LANSING 359A

The circuit of the Altec Lansing Model 359A "Stereoplex" adapter is shown schematically in Fig. 7-1. A photograph of the unit appears in Fig. 7-2. The input is fed from the demodulator of the tuner or receiver in the usual fashion. Any response deficiencies in the preceding tuner portion are compensated for by the RC network shown in series with the 50K level control at the grid of V1 (12AT7). The first triode of V1 is operated as a conventional RC-coupled amplifier with a moderate amount of cathode degeneration by virtue of the unbypassed 680-ohm cathode resistor. The entire stereo composite signal appears at pin 1 of V1. From there, the signal is fed along two paths: The total signal, coupled through a 0.47-mfd capacitor, is fed to the grid of V4 (6AR8), the dual beam-switching demodulator tube. Simultaneously, frequencies above 15 kc are coupled, through

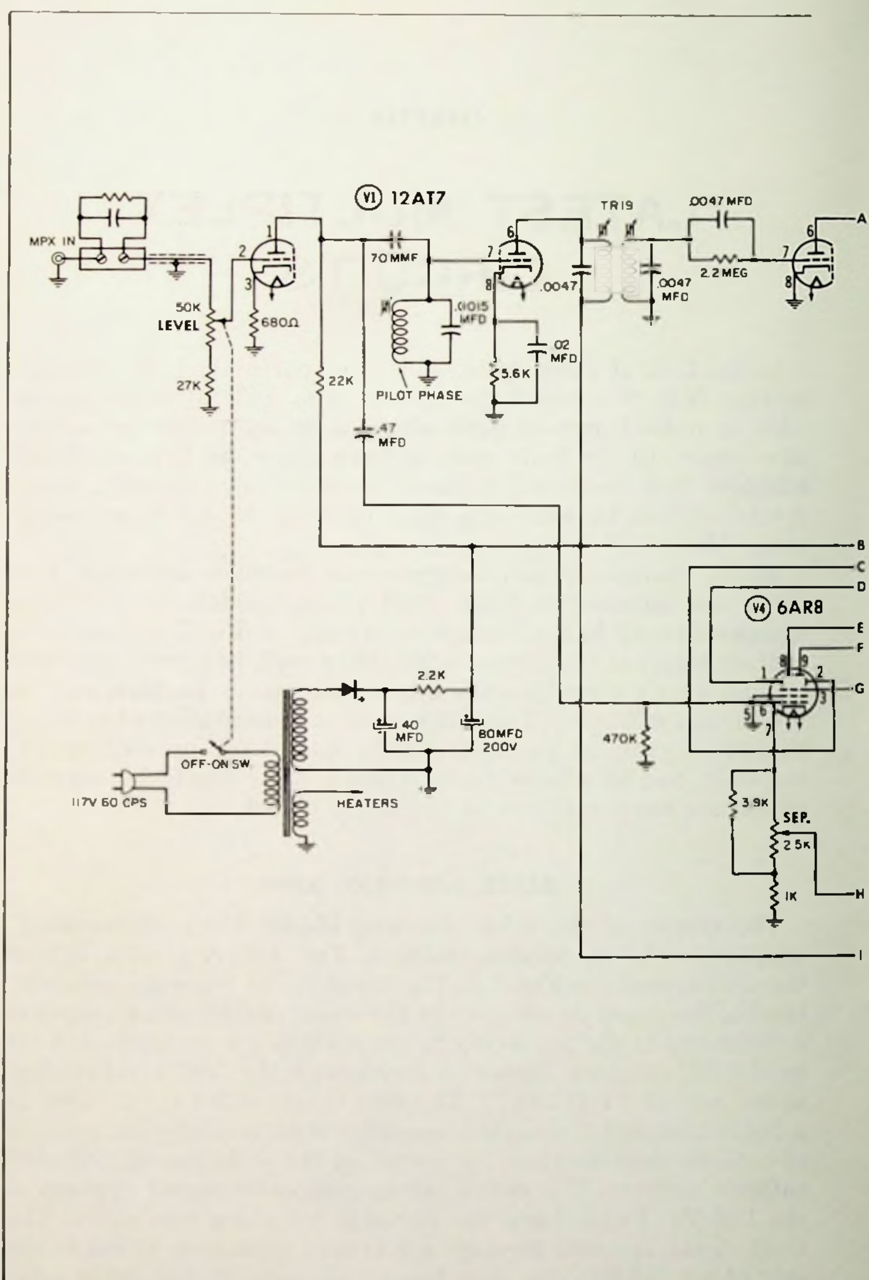
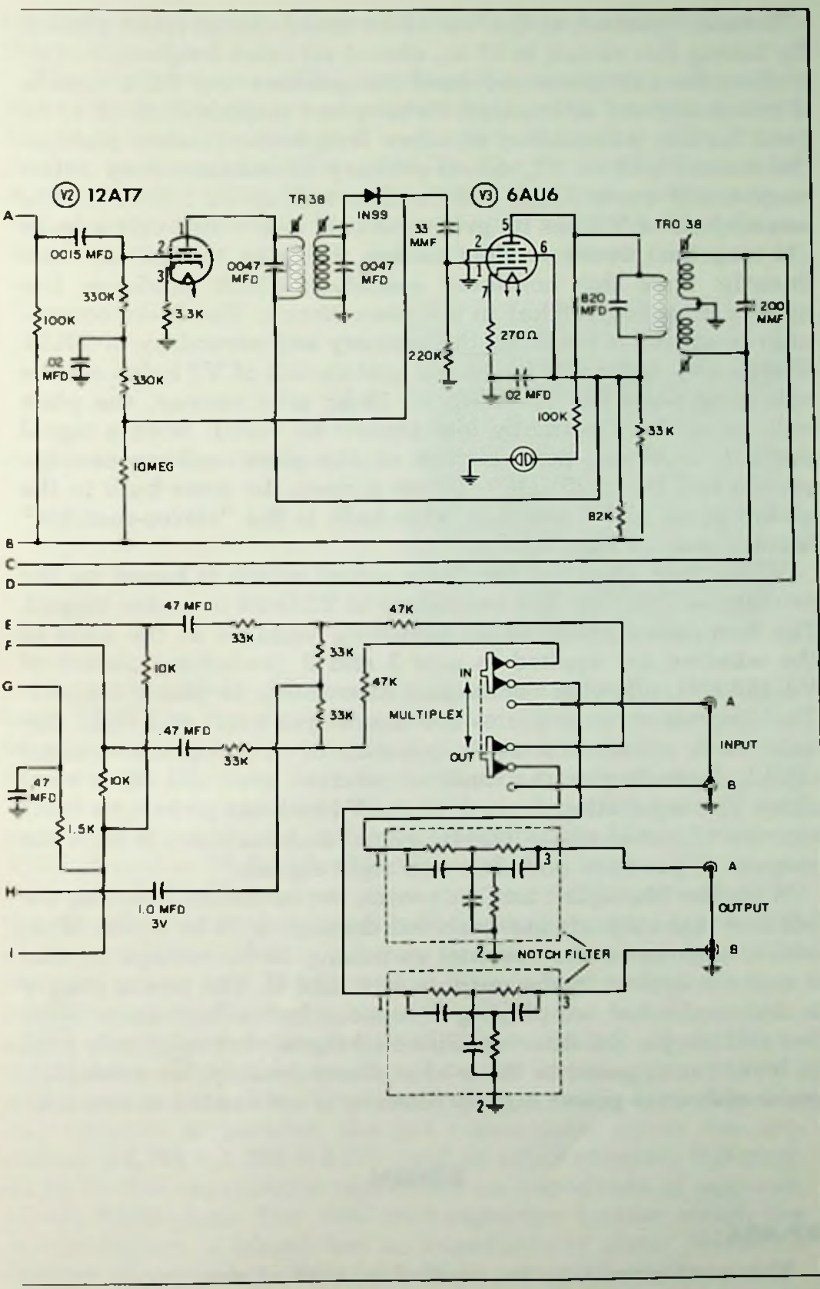


Fig. 7-1. Altec Lansing



359A adapter circuit.

a 70-mmf capacitor, to the first 19-kc tuned circuit (pilot phase). By tuning this circuit to 19 kc, almost all other frequencies (including the subcarrier sideband components and SCA signals, if broadcast) are attenuated. Subsequent amplification of 19 kc (and further attenuation of other frequencies) takes place in the second half of V1, tuned-primary–tuned-secondary interstage transformer TR19, and the first half of V2 (12AT7). The second half of V2 has its grid returned to B+ through a large (10-megohm) resistor. This causes the tube to operate nonlinearly. It is this nonlinear operation which produces frequency doubling (38 kc) in the plate circuit. The 38-kc component produced is tuned by the primary and secondary of TR38. The diode limiter (1N99) in the grid circuit of V3 helps set the operating point for V2. With no 19-kc pilot carrier, the plate voltage of V3 is normally low (below 65 volts). With a signal present, however, peak swings of the plate voltage can approach full B+ (125VDC). When it does, the neon bulb in the plate circuit of V3 will fire. This bulb is the "stereo-monitor" feature seen in Fig. 7-2.

V3 further amplifies the 38-kc signal which is tuned by the primary of TRO-38. The secondary of TRO-38 is center tapped. The two out-of-phase 38-kc switching voltages at the ends of the winding are applied to pins 1 and 2 (switching plates) of V4 (6AR8), allowing conduction alternately to plates 8 and 9. The outputs at these plates are almost pure left and right signals; each contains a small quantity of the opposite-channel signal, depending on the tuner or receiver used and other variables. The separation control, in the V4 cathode picks up a finite amount of out-of-phase (L+R) signal and matrixes it with the outputs to produce pure left and right signals.

With the Multiplex In-Out switch set to the In position, the left and right signals are each fed through a 38-kc notch filter, which attenuates the residual switching 38-kc voltage so that it will not appear in the outputs at A and B. The power supply is uncomplicated, employing a semiconductor half-wave rectifier and simple RC filtering. Since all signals are relatively high in level (as opposed to those of a phono preamp, for example), more elaborate power-supply filtering is not needed in this unit.

BOGEN

RP-40A

The next circuit to be studied is that of the Bogen Model RP-40A AM-FM-stereo receiver. A schematic of the multiplex circuitry in this set is shown in Fig. 7-3, and a photograph of the unit in Fig. 7-4.

The tuner section (FM) employs a ratio-detector demodulator circuit. The output of this detector is fed through a short length of shielded cable and is coupled, through a .05-mfd capacitor, to the grid of V24 (12AU7). The first half of this tube is a conventional amplifier that raises the level of the entire composite stereo signal. The second half is a split load amplifier. Its plate circuit is tuned to 19 kc by the oscillator transformer

Fig. 7-2. Altec Lansing 359A multiplex adapter.

primary. The 19-kc pilot-carrier signal synchronizes the oscillator (V25) by transformer action, since the secondary of the transformer shown is actually the inductance of the oscillator tank circuit. The two halves of V25 are connected in parallel to reduce the output plate impedance and permit greater current conduction in the oscillator circuit. The plate circuit of V25 is tuned to 38 kc (double the oscillator frequency) by the primary of the 38-kc doubler transformer. The secondary of this transformer is also tuned to 38 kc. Approximately one-fifth of the available voltage is fed on as *reinsertion carrier* to the junction of the two 1N542 diodes. This is accomplished by capacitive voltage-divider action rather than the usual resistors, which might load down the Q of the secondary of the 38-kc transformer. The total capacitance across the secondary is .0047 mfd in series with .001. Since capacitors in series are computed like resistors in parallel, the net capacitance across the secondary is 4,700 × 1,000 ÷ 5,700 mmf, or approximately 825 mmf. At 38 kc this capacitance represents an impedance of approximately 5,000 ohms. The .0047-mfd capacitor (across which the carrier voltage is taken) has an impedance of about 900 ohms at 38 kc. Thus, about 18% of the 38-kc voltage actually appearing at the secondary of the doubler transformer is passed on to the junction of the two detecting diodes. As stated earlier, this junction (labeled TP, denoting test point) is the most useful point for scope and meter observations in a matrix circuit.

Returning to the second half of V24, the cathode signal feeds two distinct paths. The upper circuit path, isolated by a 3.9K resistor, is through the bandpass filter which eliminates all frequencies below 23 kc and also rejects any 67-kc SCA signals.

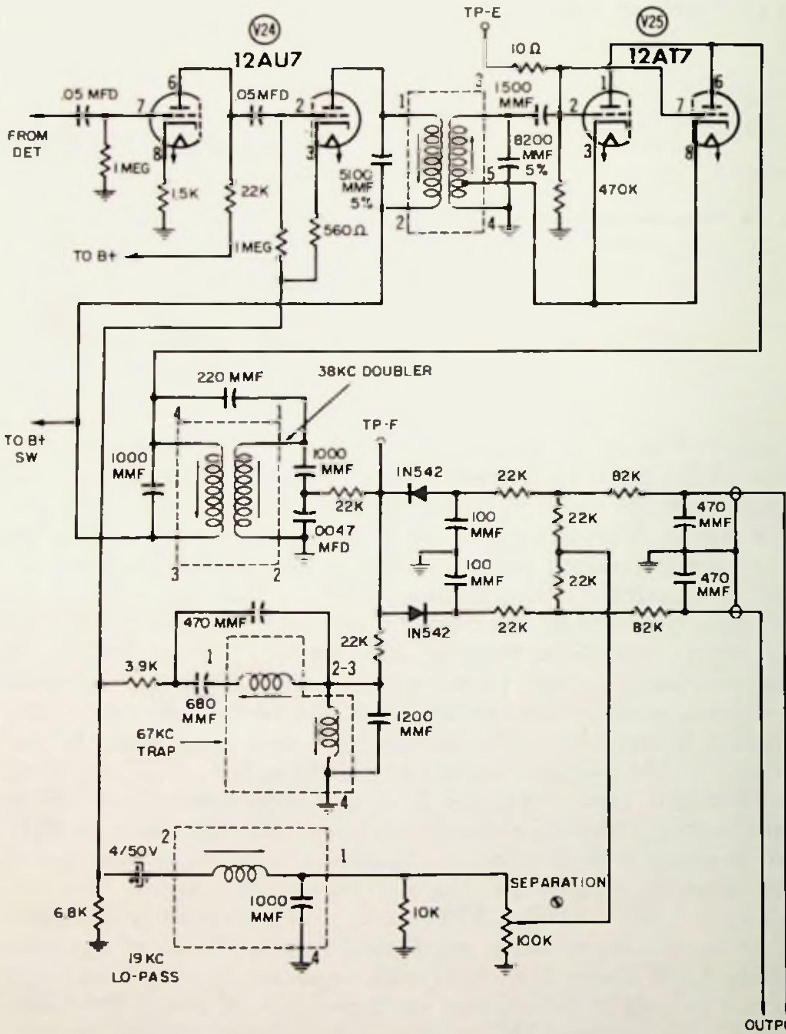


Fig. 7-3. Bogen RP-40A multiplex section.

Both inductance elements of this complex filter need to be tuned accurately, the series element for a 67 kc null and the shunt element for correct bandpass requirements (uniform response from 23 kc to 53 kc). Only the subcarrier L−R sideband components will appear at the output of this filter. They are then added to the internally generated 38-kc carrier at the junction of the detector diodes. The upper and lower diodes therefore detect (L−R) and −(L−R) signals.

Fig. 7-4. Bogen RP40A stereo receiver.

The lower signal path includes a simple 19-kc low-pass filter which rejects all but L+R audio signals. Equal quantities of L+R (as determined by the setting of the separation control) are added to (L−R) and −(L−R) through the four 22K matrixing resistors. This is done to recover L and R, which at this point have still not undergone de-emphasis. De-emphasis is accomplished by the 82K resistor and 470-mmf capacitor in each channel "leg." The resulting L and R outputs are fed out to the main-receiver selector switch shown in another part of the diagram.

The separation control is a screwdriver adjustment. Here is a manufacturer who has the advantage of mating the receiver and multiplex circuitry himself. In such instances, a front-panel dimension control for optimizing separation is not absolutely necessary, provided all stations transmit in strict accordance with FCC standards. Under these circumstances, separation may well be considered a factory setting, needing only an occasional re-adjustment—as when tubes age or are replaced.

One other interesting fact is that 1N542 diodes were chosen for detectors. These are nothing more than 1N541 diodes which have been specially matched in performance.

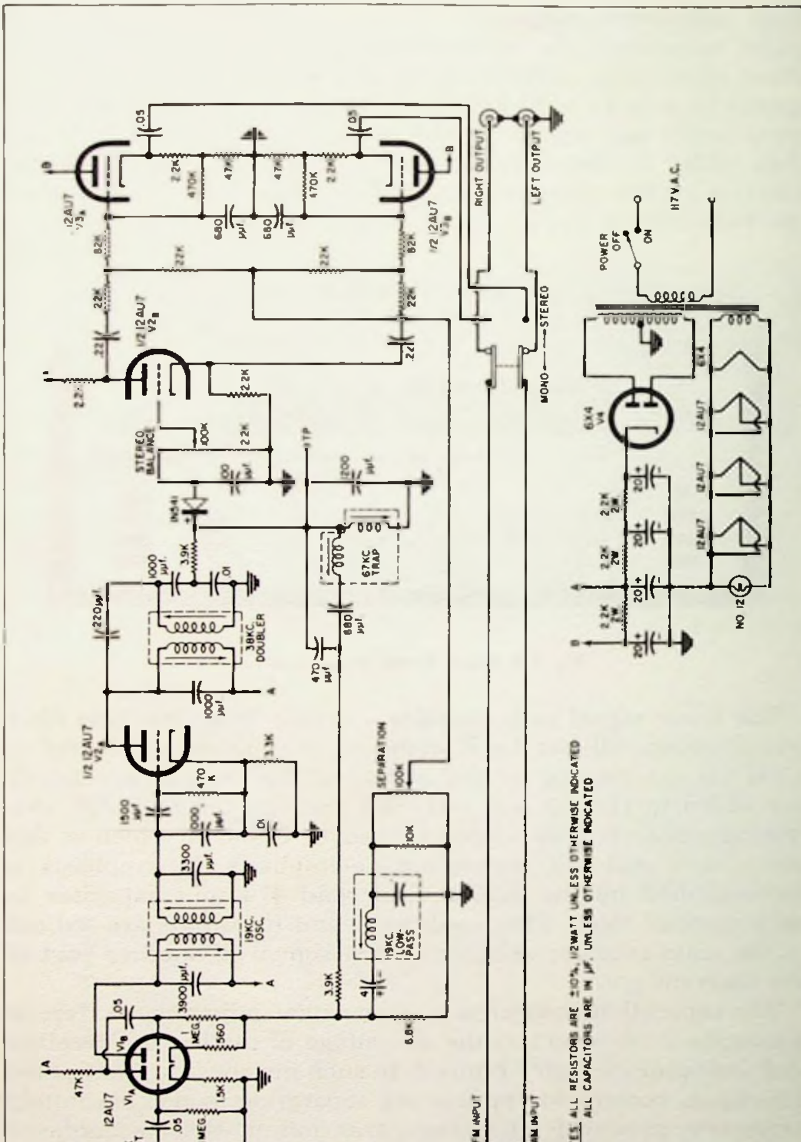


Fig. 7-5. Bogen PX-60 adapter circuit.

PX-60

Another Bogen circuit, shown in Fig. 7-5, is the one used in their Model PX-60 self-powered adapter. A photograph of the unit appears in Fig. 7-6. Being self-powered and designed to operate at some distance from the stereo amplifiers with which it is used, this unit requires four tubes (including rectifier) and a diode detector—as opposed to the previous Bogen circuit, which required only two tubes and two diodes. One of the additional tubes, V4, is the full-wave 6X4 rectifier. The other extra tube is a dual triode 12AU7 used in a pair of low-impedance cathode-follower output stages which can feed long lengths of connecting cables with no increase in hum level or capacitive loss of high-frequency response.

In many respects the adapter is similar to the other Bogen circuit. Detected output from the FM tuner or receiver is amplified by V1A and coupled to V1B. Two cathode outputs are taken from V1B. The first reduces the signal to L+R component only, by means of the now familiar low-pass filter. The second output is filtered down to L−R sideband components by the five-element bandpass filter, which also rejects any 67 kc SCA transmission present. Plate output from V1B selects the 19-kc pilot signal by means of a tuned circuit which, by transformer coupling, feeds doubler stage V2A. The output of V2A is doubled by the plate tuned circuit and the resultant 38 kc is fed, in part, to the junction of the 1N541 diode for L−R detection.

Fig. 7-6. Bogen PX-60 multiplex adapter.

Here the circuit differs from its preceding companion. Only one diode is used and only the L−R signal is detected. This signal is then fed to a split load (also known as a paraphase) amplier in which (L−R) appears across the cathode load resistor, whereas $-(L-R)$ appears across the plate load resistor. These two polarities of L−R are then matrixed with the L+R signal, the amplitude of which is determined by the separation control. This time the separation control is a front-panel owner adjustment and may be seen in Fig. 7-6 as the Stereo Dimension control. Properly matrixed across the four 22K resistors, the separate L and R signals are then de-emphasized as before and applied to individual sections of V3, the cathode followers. Outputs from the cathode followers are then ready for connection to the stereo amplifier. For monophonic use, the adapter circuits are completely bypassed by a slide switch. This switch is effective only if the regular FM output of the tuner is connected to the FM input of the adapter. Another input jack, labeled "AM input," is also made available on the adapter, in case AM-FM stereo simulcasts are still broadcasted in the area. For mono use, the power supply of the adapter need not even be turned on, since none of its powered circuits are being used. As will be seen in subsequent circuits, monophonic operation might also be obtained (with the same noise-reducing benefits) by merely deactivating the subchannel portions of the circuitry while allowing the L+R amplifying circuitry and cathode followers to continue functioning normally.

CROSBY CIRCUITS

MX-101

The Crosby Model MX-101 schematic is shown in Fig. 7-7. The circuit consists of four tubes (including the EZ-80 rectifier) and a pair of matched 1N541 diodes. FM composite signals taken from the discriminator or ratio-detector output of a tuner or receiver are applied to V1A. Three separate circuits are taken from this conventional isolating amplifier. The upper output is applied through a low-pass filter to the pair of 22K (plus part of the 100K potentiometers) matrixing resistors. Information in this circuit consists of L+R modulation only. The center output of V1A is applied, through a bandpass network, to the detecting diodes. Signals present after the bandpass filter are the L−R sidebands in the 23- to 53-kc stereo subchannel range. SCA (background music) is also rejected in the bandpass filter. The lower output of V1A is applied, through a voltage divider, to V1B. This stage amplifies the 19-kc pilot-signal portion of the composite signal. The amplified 19-kc signal at the output of V1B is then used to lock, or synchronize, the local 19-kc oscillator V3. Synchronization is accomplished by feed-

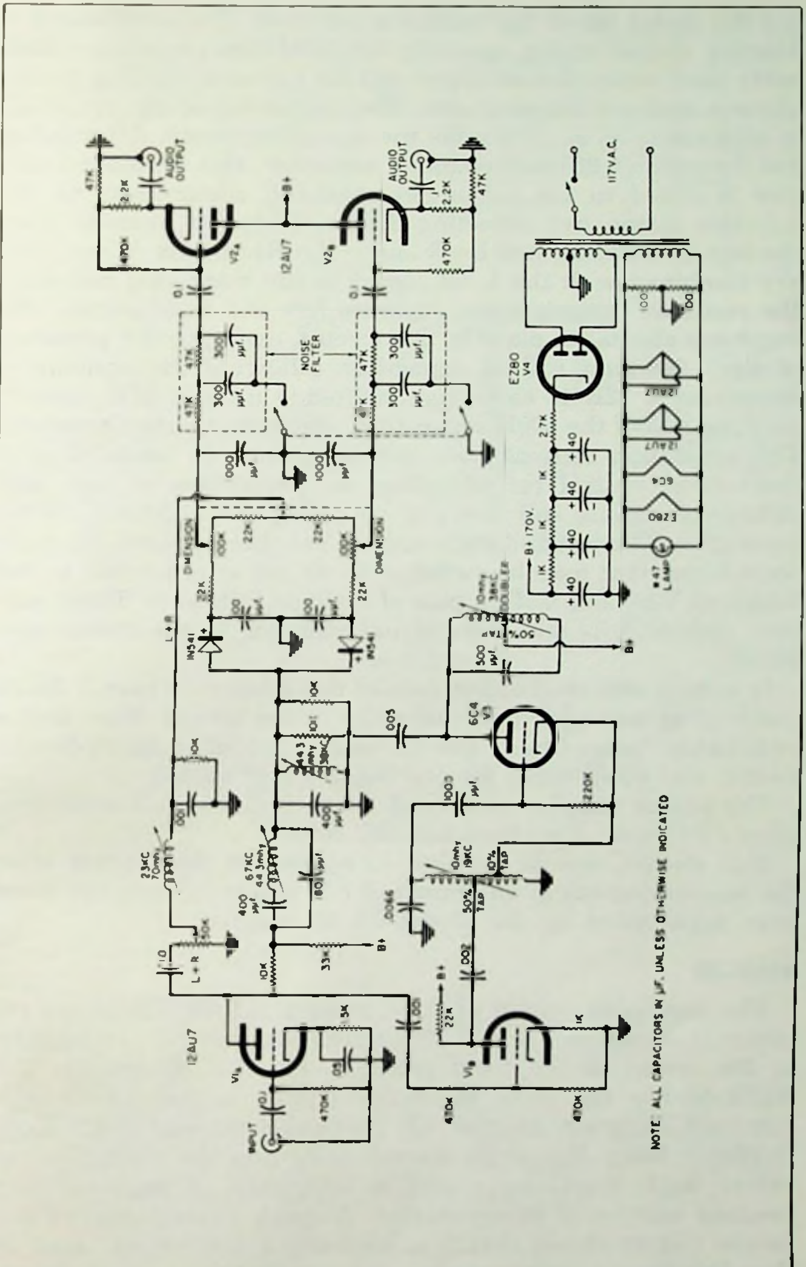


Fig. 7-7. Crosby MX-101 adapter circuit.

ing the center tap of the oscillator coil itself. The oscillator is a Hartley circuit with a specially designed film capacitor (.0066 mfd) used across the oscillator coil for extreme stability under various ambient temperatures. The output tuned circuit of V3 is adjusted to 38 kc, or double the input frequency. After being fed through a 0.005-mfd coupling capacitor, this developed carrier is added to the subcarrier sideband components at the junction of the two detecting diodes. Detected outputs from the two diodes consist of L−R and −(L−R) signals. When they are combined with the L+R signal in the matrixing resistors, the resultant outputs are a separate left and right signal. De-emphasis also takes place in this circuit, owing to the presence of the 1,000-mmf roll-off capacitors following the equivalent resistance of 72K in each channel (made up of a 22K resistor plus one half of the 100K separation adjustment potentiometer). The separation control (two potentiometers in tandem) is a front-panel control for adjusting the proportion of sum and difference signals and thereby changing the apparent stereo separation. The left and right signals are then applied (through noise filters that may be switched in or out as required) to the inputs of V2A and V2B, a pair of cathode followers. Their outputs deliver left- and right-signal voltages to the stereo amplifier.

In subsequent production runs of this adapter, a twin-T 38-kc notch filter was added permanently in the circuit. Now that a switchable "noise filter" was no longer needed, a Mono-Stereo switch was substituted for the Noise Filter switch.

The power supply consists of a conventional full-wave rectifier (V4) with a multisection RC filter.

This circuit, one of the first to appear on the market after the announcement of an approved FM stereo system, has since been superseded by the Model SA-40 adapter.

RMX-30

The multiplex circuit of the Crosby RMX-30 FM stereo receiver is shown in Fig. 7-8. The multiplex controls can be seen at the center of the front panel in Fig. 7-9. Essentially the RMX-30 has the same multiplex circuit as the SA-40 self-powered, separate adapter. Of particular interest is Crosby's thinking. Since this is its second entry into the multiplex receiver field, this later model is indicative of improvements deemed worthy of incorporation. A quick examination of the newer circuit shows that it is basically a "switching" type of decoding circuit, rather than a matrix type like its predecessor. The first stage, fed from the discriminator output, is still a

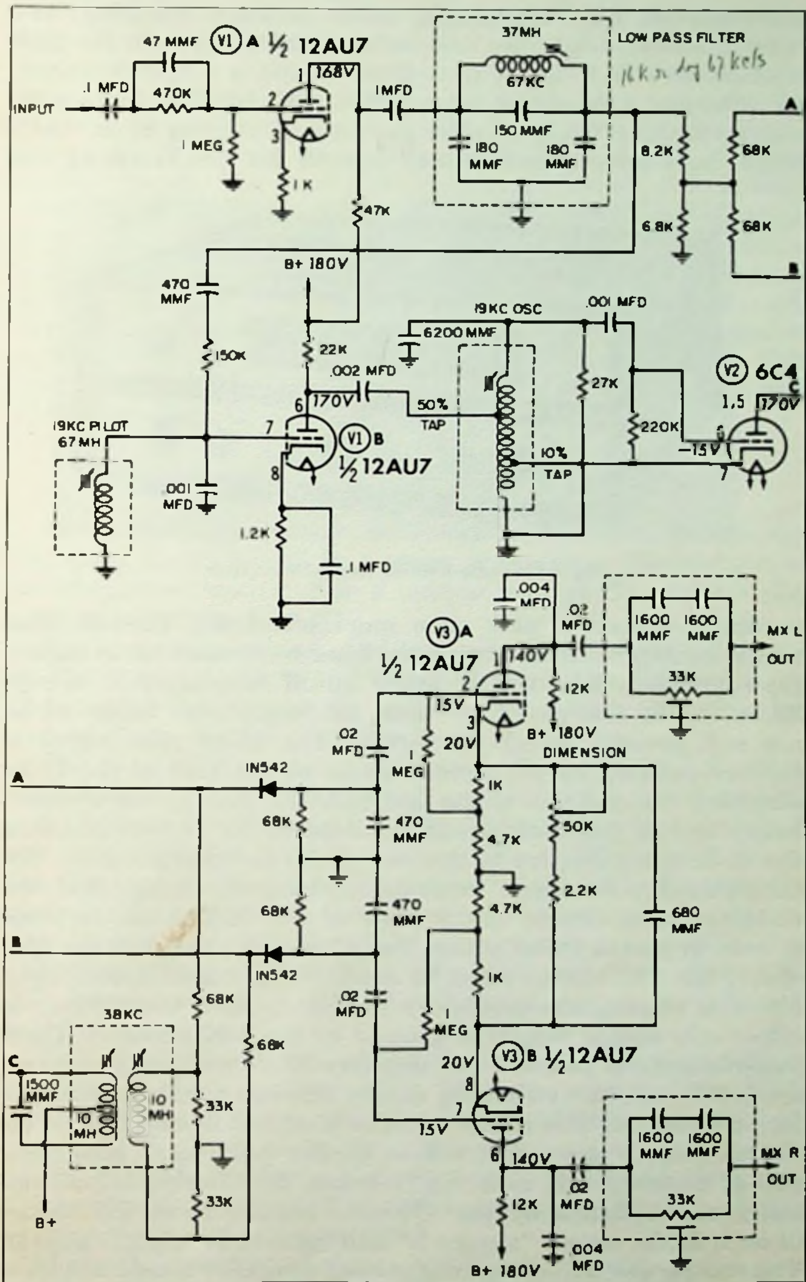


Fig. 7-8. Crosby RMX-30 multiplex section.

conventional, high-impedance, triode isolation amplifier. This time, however, the entire composite signal is fed from the plate voltage divider to a low-pass filter having a cutoff frequency of 53 kc and a maximum attenuation at 67 kc. The 19-kc pilot-carrier signal is taken off after the filter. Thus, any 67 kc which might have already been mixed in with the synchronizing and

Fig. 7-9. Crosby RMX-30 stereo FM receiver.

oscillator signals is now even more effectively filtered. This would be impossible if a bandpass filter were used (as in matrix type circuits), because its lower cutoff frequency is always 23 kc. In the low-pass filter used, all frequencies below 53 kc are still present at full amplitude. The 19-kc pilot signal is further isolated by the tuned circuit in the grid of the 19-kc amplifier (second half of the first 12AU7). Tuning the 67-millihenry coil of this tank circuit establishes the correct phase of the 19-kc pilot relative to the rest of the composite signal. The oscillator-doubler stage remains unchanged, except that the doubler tuned circuit in the plate of the 6C4 oscillator stage is now, in part, a transformer. The 38-kc switching voltage necessary for this circuit must be available in opposite polarities. For this reason, the secondary of the doubler transformer is effectively center tapped to ground by the 33K resistors. Thus, instantaneously positive and negative 38-kc switching voltages are fed to the two switching diodes through a pair of 68K isolating resistors. The entire composite signal is also fed to the diodes (which are connected in similar polarities) through a pair of matched 68K isolating resistors. RF filtering is substantially accomplished by the 470-mmf capacitors at the output of each diode, where "almost L" and "almost R" signals appear. The signals are fed to a pair of output-amplifier triode 12AU7's. Cross-coupled corrective matrixing takes place in the cathode circuits of the output amplifiers. As a result, the amount of inverse L−R signal present is varied until each channel contains its respective L−only or R−only signal. The two outputs are fed to a combined de-emphasis–38-kc notch filter network, and finally to the master selector switch of the receiver. No Mono-Stereo switch is required; to listen to monophonic FM, the user merely changes the setting of the selector switch to FM, after which all multiplex circuits are bypassed. While this arrangement requires one extra de-emphasis resistor and capacitor back at the tuner section output, less complex switching is needed. This circuit is powered by 150 to 180 volts B+ and requires no more tubes than the earlier MX-101. The number of tuned circuits remains the same also, since the L+R (low pass to 15 kc) filter is no longer required, even though a new 19-kc tuned circuit has been added.

EICO MX99

A novel multiplex circuit is the EICO (Electronic Instrument Co., Inc.) Model MX99. In a sense, it combines the features of switching and matrixing in a unique manner. The circuit diagram is shown in Fig. 7-10.

The composite stereo signal is received from a ratio-detector or discriminator output and is amplified by V1A. The input impedance of this first stage is made very high in order not to load down the discriminator or ratio detector.

The 19-kc pilot carrier is isolated and amplified by V4. Frequency doubling is accomplished by a full-wave rectifier (CR1 and CR2) at the plate of V4. The 38-kc pulses thus obtained are used to lock-in a 38-kc local oscillator (V5), which drives the ring modulator (CR3-CR6 and R12-R17). In addition to the strong 38-kc component, the full-wave rectifier delivers a negative DC voltage when the 19-kc pilot carrier is received. This DC voltage cuts off switching tube V1B and thus ignites the MX-Stereo neon pilot lamp I1, indicating that a stereo broadcast is in progress.

V1A also delivers the amplified composite stereo signal to the grid of phase inverter V2A. The two outputs from V2A, now 180° out of phase, are alternately sampled by the 38-kc output of the ring modulator. The two sampled outputs are added by V2B. The output in effect is an amplified version of the input signal multiplied by a 38-kc switching function of zero average value and odd symmetry.

The audible portion of this signal (L−R) appears at the output of V2B and is mixed with the two outputs of V2A (L+R,

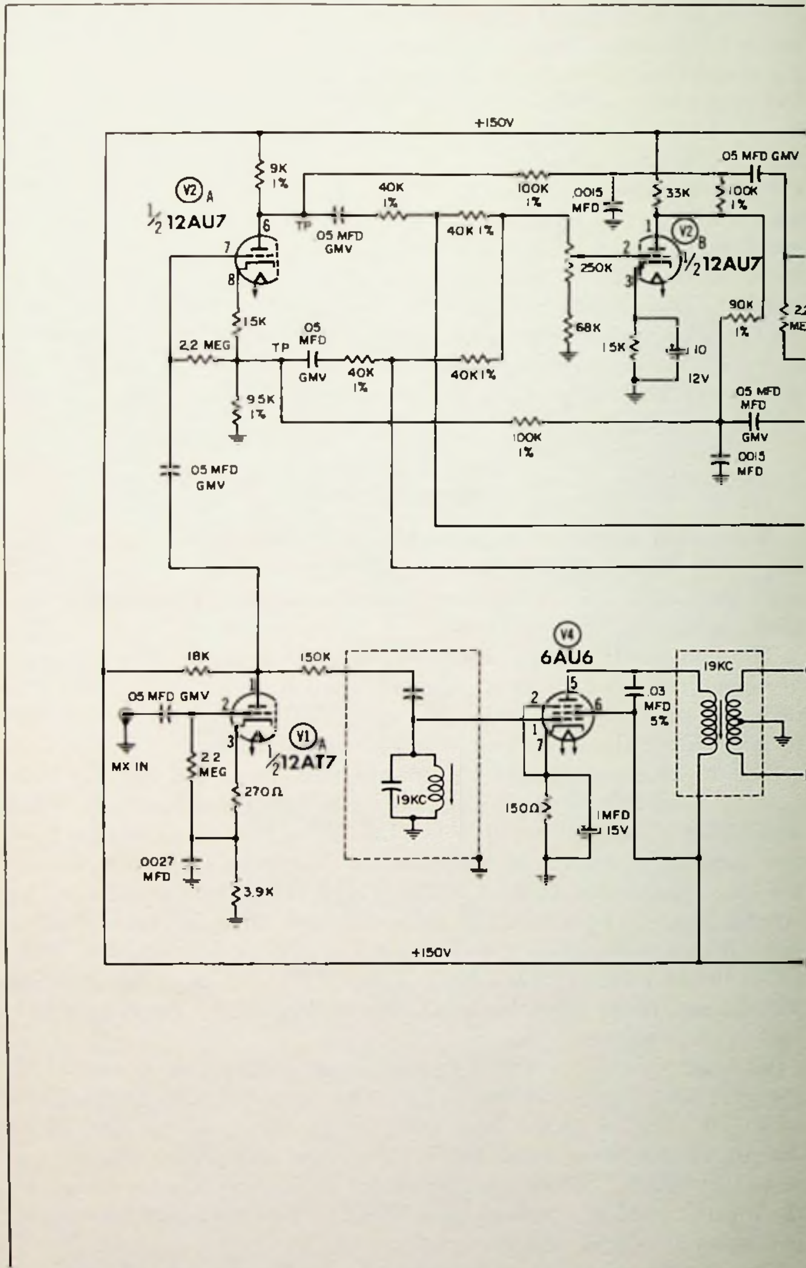
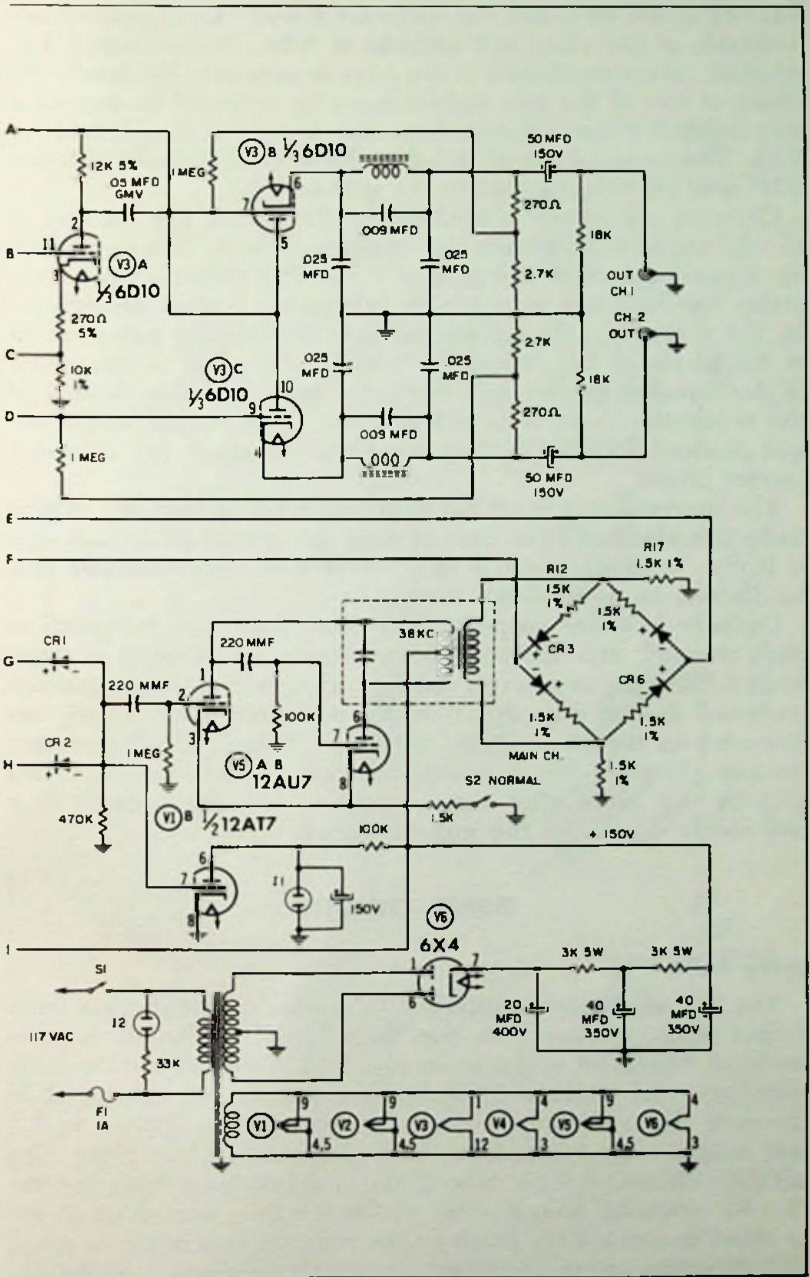


Fig. 7-10. Eico MX-99



adapter circuit.

−L−R) to derive L and −R. (The L+R and −L−R outputs are available at the plate and cathode of V2A, respectively.) De-emphasis is accomplished in the mixing network. Evidently the phase of one of the two signals must be reversed to duplicate the original phase relationship, and this is accomplished by V3A. The inverted signal is fed to the grid of cathode follower V3B, and the other signal to the grid of V3C.

Channel separation is controlled by varying the amount of (L−R) signal injected into the mixing network. This is effected by a gain control in the grid of V2B. The output-signal level under the condition of optimum separation is thus determined by the direct (L+R) component and is virtually independent of the phase of the reinjected carrier. An error in the phase of the injected carrier will require simply a higher setting of the separation control to achieve the same output amplitude and channel separation that would be obtained for optimum carrier phase.

The demodulator is of the balanced type so that the potentially troublesome 38-kc carrier does not appear in the outputs. If it did, it might produce beat notes with tape-recorder bias oscillators, as pointed out earlier.

Cathode-follower outputs (V3B and V3C) are provided in each channel, and the output waveforms are filtered in these stages. Filtering attenuates the high-frequency components introduced during demodulation. Even though substantially attenuated by the de-emphasis networks, these components may become objectionable in a tape recorder. In addition, the filters remove the 19-kc pilot signal which is usually present to a noticeable degree in the output signals.

FISHER CIRCUITS

Early MPX-100

The first multiplex equipment to appear on the market from Fisher Radio Corporation, was their Model MPX-100. A schematic of its circuit is shown in Fig. 7-11. Five tubes, two diode rectifiers, and a diode bridge power supply are employed in the unit. Signals from the tuner or receiver are applied to V1, and outputs are taken from both the cathode and plate. The cathode circuit of V1A now contains a bandpass filter for the (L−R) sideband frequencies, which are then passed on to the product detector V4A. Such a tube requires two separate grids, and its grid-voltage to plate-current relationships must be linear. The regenerated 38-kc carrier is fed to one grid, and the

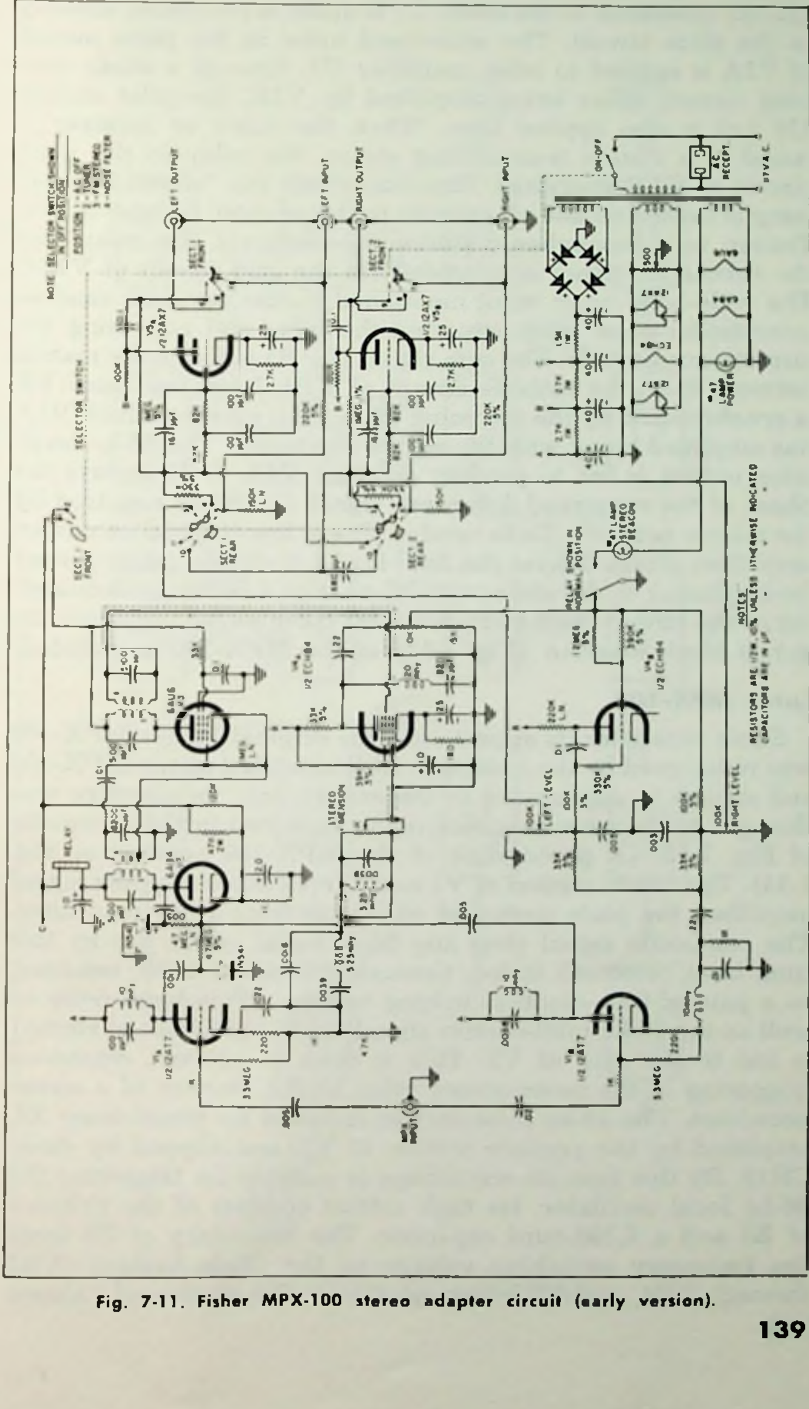


Fig. 7-11. Fisher MPX-100 stereo adapter circuit (early version).

(L—R) sidebands to the other. L—R audio is recovered directly in the plate circuit. The wide-band noise in the plate output of V1A is applied to relay amplifier V2, through a diode clipping circuit. After being amplified by V1B, the pilot carrier (19 kc) is also applied here. When the tuner or receiver is tuned to a station transmitting stereo, the relay in the plate circuit of V2 is energized. Simultaneously the "stereo beacon" lamp is turned on and all circuits in the adapter become active. Tuning to a monophonic station de-energizes the relay and the subcarrier signal is grounded (in the grid circuit of V4B). The wide-band noise input mentioned earlier prevents random interstation noise from actuating the relay and activating the circuits erroneously. The sum signal (L+R) is fed to the matrix network from the cathode circuit of V1B. Local oscillator V3, is synchronized by the incoming 19-kc pilot carrier, which V1A has amplified along with the rest of the signal. The 38-kc oscillator output is fed to product detector V4A. V4B inverts the phase of the recovered difference signal (L—R) as required by the matrix network. Twin triodes V5 are low-impedance output amplifiers which deliver the final L and R signals (after proper de-emphasis) to the stereo amplifiers. For a better understanding of the circuit itself (which is a bit complex, as these circuits go), a block diagram (Fig. 7-12) of the MPX-100 is included.

Later MPX-100

Some time after its appearance, the original Fisher MPX-100 was redesigned by the company. Still identified as the MPX-100 and similar in appearance to the earlier unit, its circuitry was changed to the switching technique, as shown in the schematic of Fig. 7-13. (A photograph of the MPX-100 appears in Fig. 7-14). The triode section of V1 now serves as a composite signal amplifier, the plate circuit of which contains a low-pass filter. The composite signal (less any SCA signal which has by this time been rejected) is fed, through individual 100K resistors, to a pair of four-diode switching bridges. Wide-band noise as well as the 19-kc pilot-carrier signal (in the presence of stereo) is fed to the grid of V2. This is done to prevent erroneous triggering of the mono-stereo relay in the absence of a stereo broadcast. The 19-kc pilot carrier is tuned by transformer Z2, amplified by the pentode section of V1, and clipped by diode CR10. By this time its waveshape is suitable for triggering the 38-kc local oscillator. Its tank circuit consists of the primary of Z3 and a 5,100-mmf capacitor. The secondary of Z3 feeds the necessary switching voltage to the diode bridges (CR1 through CR4, and CR5 through CR8). The recovered "almost

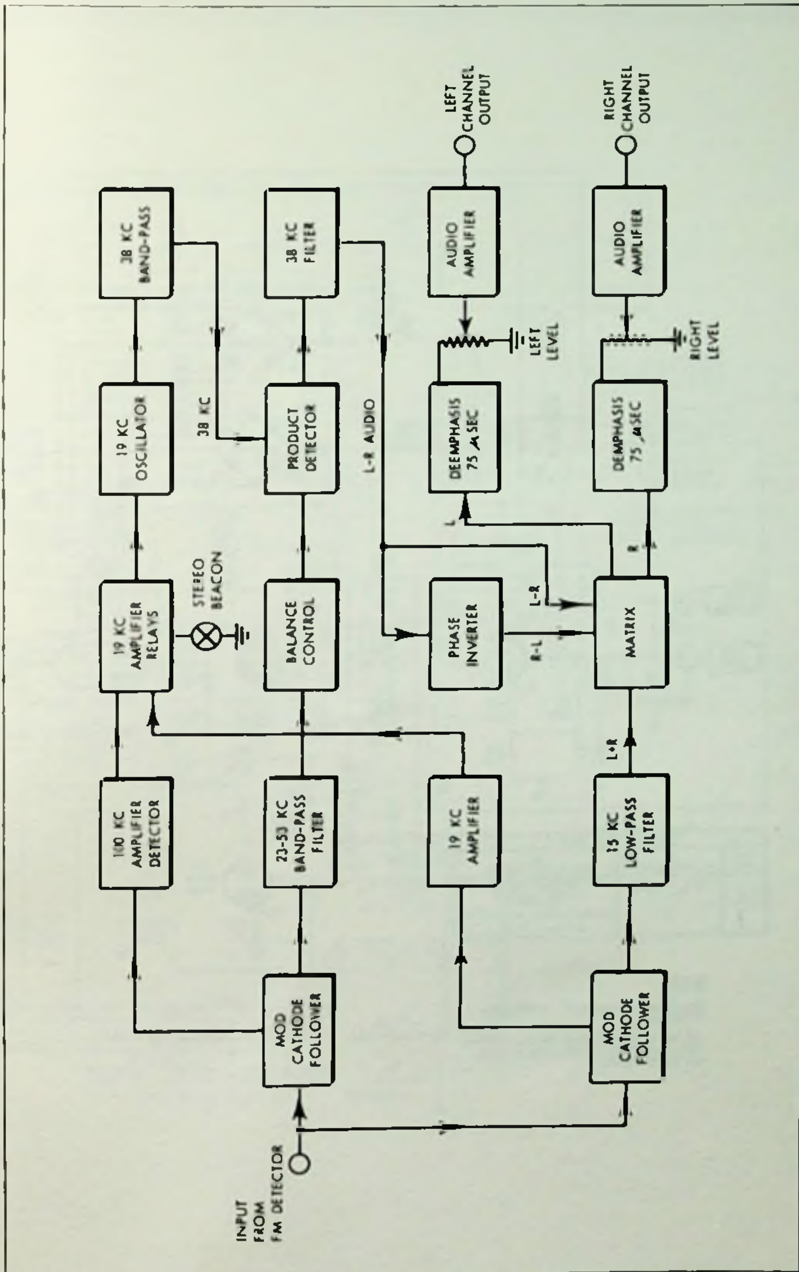
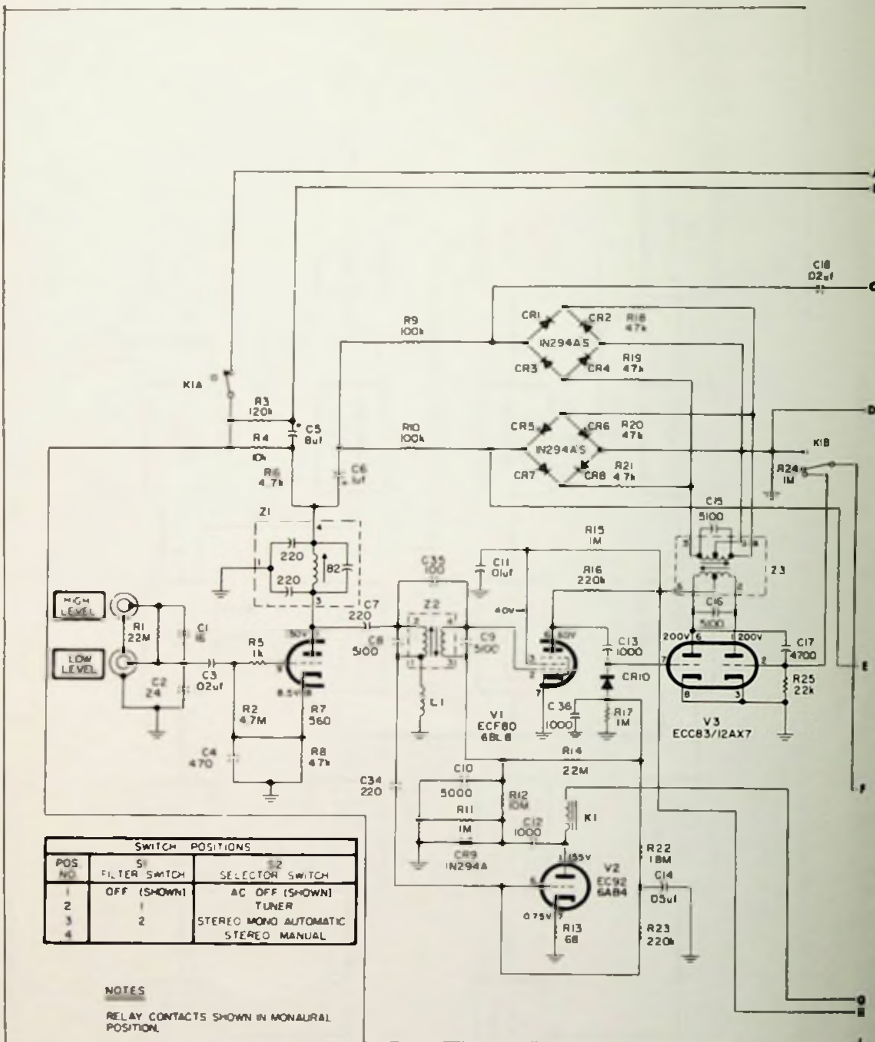


Fig. 7-12. Block diagram of Fisher MPX-100.



POS NO	S1 FILTER SWITCH	S2 SELECTOR SWITCH
1	OFF (SHOWN)	AC OFF (SHOWN)
2	1	TUNER
3	2	STEREO MONO AUTOMATIC
4		STEREO MANUAL

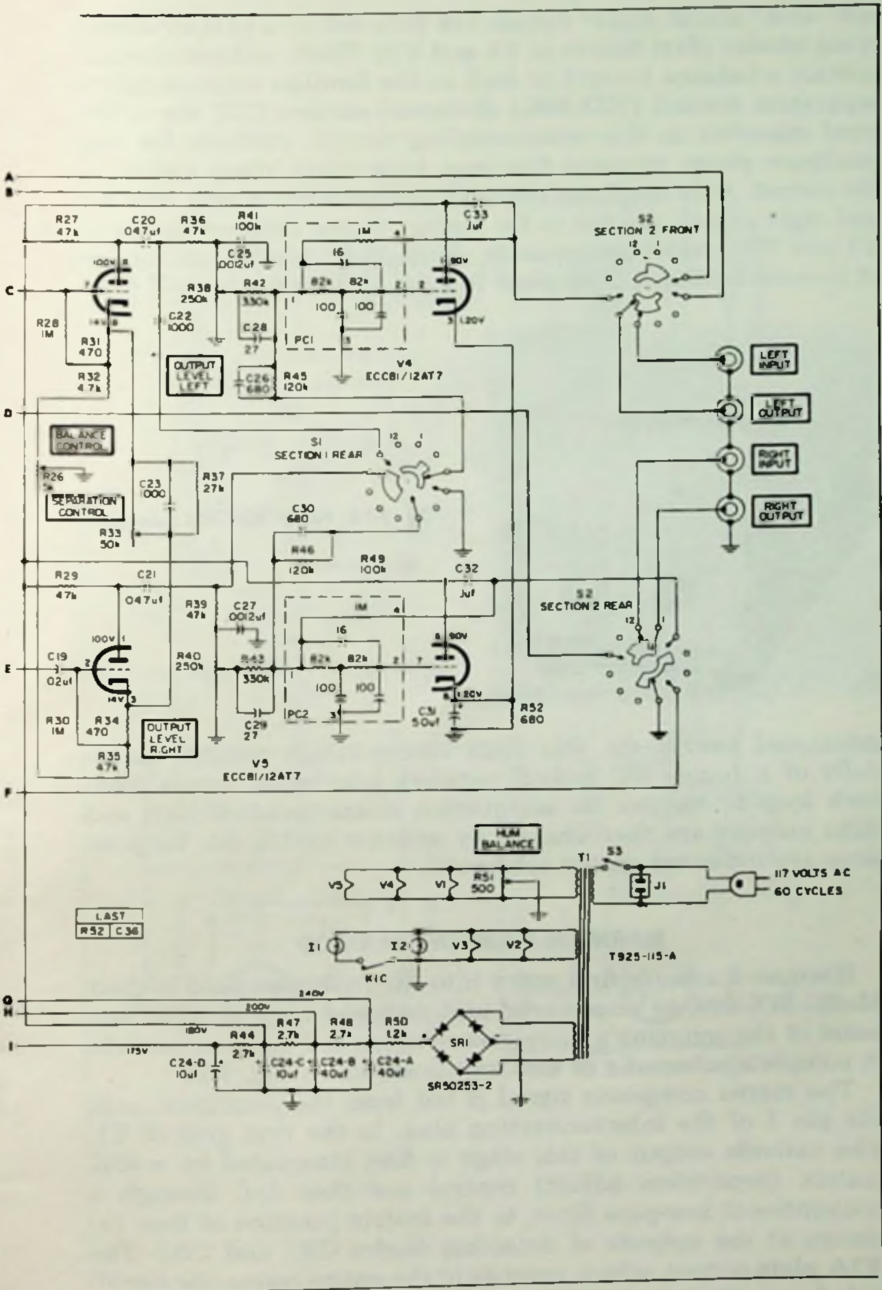
SWITCH POSITIONS

NOTES

RELAY CONTACTS SHOWN IN MONAURAL POSITION.

DC VOLTAGES ARE MEASURED WITH VTVM AND REFERENCED TO CHASSIS GROUND NO INPUT SIGNAL.

Fig. 7-13. Fisher MPX-100



adapter circuit (revised).

left" and "almost right" signals are then fed to a pair of amplifying triodes (first halves of V4 and V5). Their cathode circuits contain a balance control as well as the familiar cross-coupling separation control (R33-50K) discussed earlier. C23, the 1,000-mmf capacitor in this cross-coupling circuit, corrects for any nonlinear phase response that may have taken place earlier in the circuit. A de-emphasis network follows each triode. The left and right signals are fed to the output stages (second halves of V4 and V5), which incorporate sharp 38-kc filters making use of inverse feedback from plate to grid. Unlike the twin-T filter

Fig. 7-14. Fisher MPX-100 adapter.

mentioned previously, this filter configuration consists essentially of a double RC roll-off network interjected into a feedback loop to steepen its attenuation characteristics. Left and right outputs are then chosen by selector switch S2. Its positions are indicated on the schematic.

HARMAN-KARDON MX-500

Harman-Kardon's first entry into the multiplex field is their Model MX-500, an unpowered unit designed to be plugged into some of the company's more recent tuner and receiver models. A complete schematic of this unit is shown in Fig. 7-15.

The stereo composite signal is fed from the associated unit, via pin 1 of the interconnecting plug, to the first grid of V1. The cathode output of this stage is first attenuated by a 50K matrix (separation adjust) control and then fed, through a conventional low-pass filter, to the matrix junction of four resistors at the outputs of detecting diodes CR1 and CR2. The V1A plate output, which consists of the entire composite signal, is fed on to V1B for further amplification and isolation. Most of

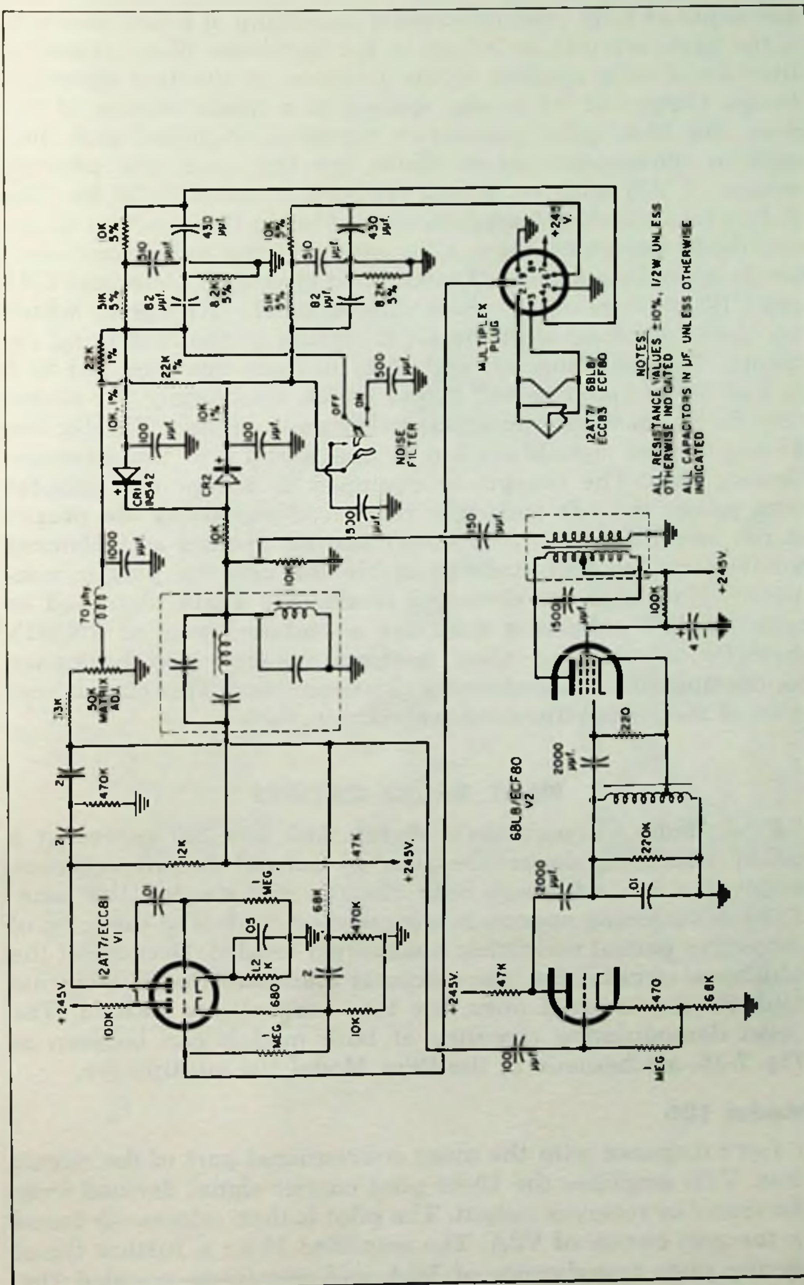


Fig. 7-15. Harmon-Kardon MX-500 adapter circuit.

the output of V1B (voltage divider consisting of a 12K and 47K in the plate circuit) is fed on to the bandpass–67-kc rejection filter and finally applied to the junction of the two detecting diodes. Output of V1 is also applied to a triode section of V2. Here the 19-kc pilot subcarrier signal is amplified and then used to synchronize an oscillator (in this case, the pentode section of V2) with an output frequency tuned to 38 kc. The 38-kc carrier is then transformer-coupled to the junction of the two diode detectors. Here it is added to the subcarrier sidebands to produce a full AM modulated envelope. Detectors CR1 and CR2 produce out-of-phase difference (L−R) signals which are then combined with the L+R signals in the matrixing resistors. The resulting left and right outputs are then fed to a pair of 38-kc notch twin-T filters which also supply the necessary de-emphasis characteristic to the audio signals. Finally, the de-emphasized signals are fed to pins 4 and 6 of the interconnecting plug. The receptacle equipped to accept this adapter then passes the left and right recovered signals to the proper stereo amplifiers. B+ (245 volts) and all ground and filament connections are also made available through the plug-in connector. Note that the detecting diodes are again identified as type 1N542's indicating they are a matched pair of 1N541's specially selected for their uniform forward and backward conducting and nonconducting characteristics. This circuit represents the straightforward matrix approach.

PILOT RADIO CIRCUITS

Pilot Radio Corporation's Models 100 and 200 represent a rather interesting departure from the normal circuitry encountered thus far. Although both circuits still employ the time-division switching approach, a separation control in the form of corrective partial matrixing is no longer needed. Because of the additional circuits, the manufacturer maintains that no external controls (or internal ones, for that matter) are needed. The novel demodulating circuitry of both models can be seen in Fig. 7-16, a schematic of the Pilot Model 100 multiplexer.

Model 100

Let's dispense with the more conventional part of the circuit first. V1B amplifies the 19-kc pilot carrier signal derived from the tuner or receiver output. The pilot is then selectively tuned in the grid circuit of V2A. The amplified 19 kc is further tuned by the plate transformer of V2A and resistively coupled (for isolation) to synchronize the 19-kc oscillator, V2B. The oscil-

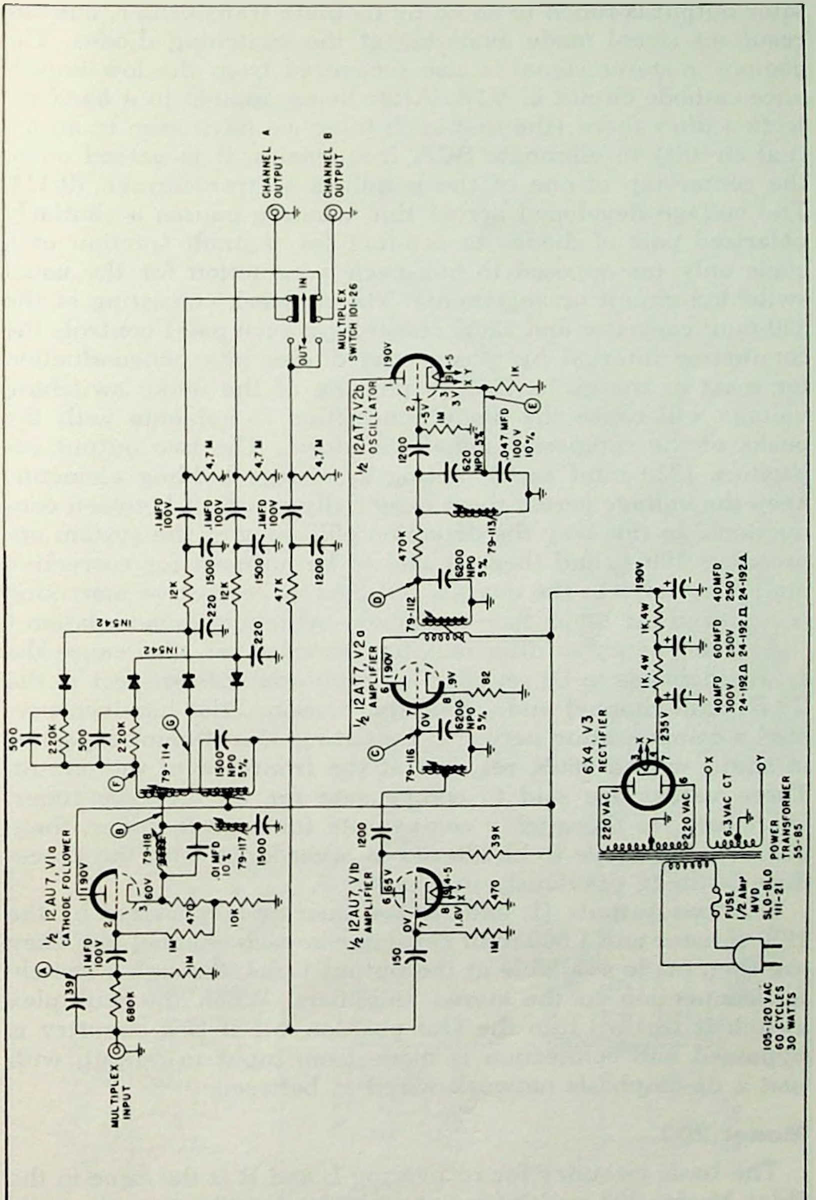


Fig. 7-16. Pilot Radio Model 100 adapter circuit.

lator output is tuned to 38 kc by its plate transformer, and the resultant signal made available at the switching diodes. The composite stereo signal is also recovered from the low-impedance cathode circuit of V1A. After being applied to a *band rejection filter* there (the first such filter we have seen in an actual circuit) to eliminate SCA frequencies, it is passed on to the center tap of one of the windings of transformer 79-114. The voltage developed across this winding causes a similarly polarized pair of diodes to conduct for a small fraction of a cycle only (as opposed to half-cycle conduction for the usual switching-circuit arrangement). The network consisting of the 500-mmf capacitor and 220K resistor (in each pair) controls the conduction interval by biasing the diodes into nonconduction for most of the cycle. Proper phasing of the 38-kc switching voltage will cause the diode conduction to coincide with the peaks of the composite signal waveform. The two output capacitors (220 mmf each), acting as charge holding elements, keep the voltage across them essentially constant between conductions. In this way the detection efficiency of the system approaches 100%, and there is said to be no need for corrective matrixing later in the circuit. (Of course, corrective matrixing is required for other than inefficient switching demodulation.)

Any deficiency in discriminator response can also cause the L−R sidebands to be reduced in amplitude with respect to the L+R main-channel audio. For this reason, Pilot has incorporated a compensating network, consisting of a 39-mmf capacitor in shunt with a 680K resistor, at the front-end of the circuit. These values are said to compensate for the average tuner. Obviously, to thoroughly compensate for a given tuner, these values might have to be altered in accordance with the procedure outlined previously in Chapter 5.

The two outputs (L and R) are then de-emphasized by the 12K resistor and 1,500-mmf capacitor in each channel leg. They are then made available at the output jacks, through a switch, for connection to the stereo amplifiers. When the multiplex switch is thrown into the Out position, all of this circuitry is bypassed and connection is made from input to output, with just a de-emphasis network wired in between.

Model 200

The basic circuitry for recovering L and R is the same in the Pilot Model 200 multiplexer, as can be seen in the schematic of Fig. 7-17. A photograph of this controlless unit is shown in Fig. 7-18. Automatic switching, actuated by the presence or absence of a pilot subcarrier, bypasses the multiplex subcarrier

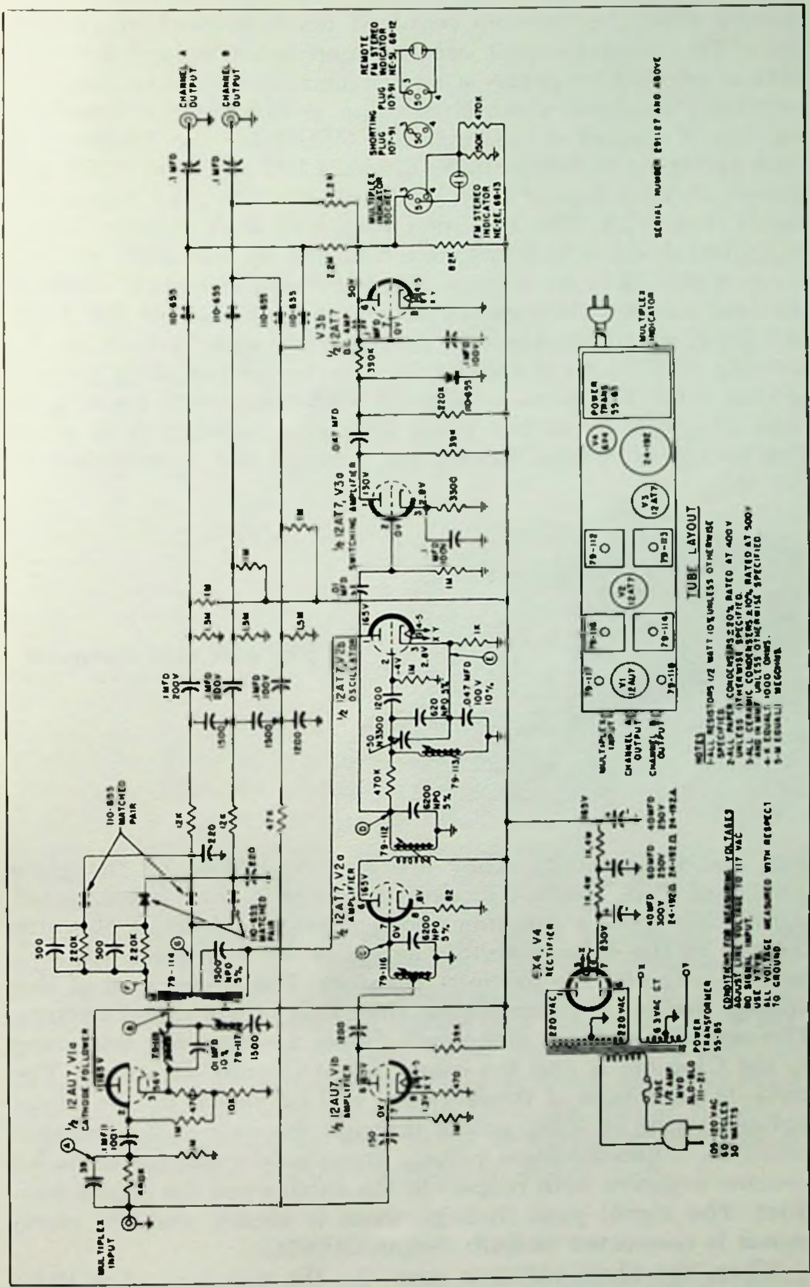


Fig. 7-17. Pilot Radio Model 200 adapter circuit.

circuits when they are not required (as in monophonic reception). This special circuit uses no mechanical relays and consists of two major parts—a switch control amplifier, and the switching elements themselves. The switch control amplifier consists of a stage of amplification (V3A) for the 19,000-cycle pilot carrier, a forward-biased diode, a DC amplifier V3B, and a neon stereo indicator lamp. V3A receives the pilot subcarrier signal from V2A. The amplified output of V3A appears across the shunt diode, which is forward-biased by the 220K resistor connected to B+. As soon as the 19,000-cycle current exceeds the bias current through the diode, the junction of the 220K and 390K resistors starts to go negative with respect to the cathode of V3B. As it does, it tends to lower the plate current of that tube. The plate voltage of V3B rises until there is 60 volts or more across the neon indicator, causing it to glow. The switching diodes themselves, seen in the output-channel

Fig. 7-18. Pilot Radio Model 200 adapter.

legs, are silicon units. They are also controlled by the plate voltage of V3B. The two diodes connected in the "direct mono" signal path have a common anode potential of about 80 volts, created by the voltage-divider action of the 1-megohm (to B+) and 1.5-megohm (to ground) resistors. The impedance of this voltage divider is much higher than that of the signal circuits. The cathodes of these mono path diodes are separate, one going to the Channel-A and the other to the Channel-B output. The cathode potentials of these diodes will be varied by the plate-voltage swing of V3B, acting through the pair of 2.2-megohm resistors. When the plate voltage drops to 50 volts, the cathodes become negative with respect to the anodes and the diodes conduct. The signal path through them is closed, and the mono signal is connected to both outputs.

When the plate voltages rises (in the presence of a 19-kc stereo subcarrier pilot), the cathodes of the mono diodes become positive with respect to the anodes and the diodes are cut off. The mono signal is then disconnected from the output.

The remaining pair of switching diodes (appearing in the stereo channels) are connected in reverse polarity to that of the mono diodes. They conduct when the mono diodes are open and vice versa.

SHERWOOD

A3MX

The Sherwood Laboratories circuit to be examined is the two-tube Model A3MX adapter (Fig. 7-19), which derives its power and input signals from various late-model Sherwood tuners and receivers. The triode section of the 6EA8 serves as a composite signal amplifier, being fed from pin 2 of the adapter plug. The entire composite signal is extracted from the cathode circuit of this stage for application to the diode switching bridge. Just to make certain the 19-kc pilot carrier at the output will not be of sufficient amplitude to create any problems, a series-resonant trap is wired across the 5K "composite signal level" potentiometer in the cathode circuit.

The plate circuit of the first stage feeds a tuned-primary–tuned-secondary 19-kc transformer. Its secondary is actually the local-oscillator coil, connected in the familiar Hartley configuration. The oscillator stage is the pentode section of the 6EA8 tube. The tuned-plate circuit doubles the frequency and thereby provides the necessary 38-kc switching voltage, which is applied to opposite corners of the demodulating diode bridge circuit. The "almost-left" and "almost-right" outputs of the bridge are fed to a pair of triode amplifiers (12AU7A). Their cathode circuits provide the necessary amount of corrective (L−R) cross coupling to afford optimum separation in the outputs. There also is a balance control in this cathode circuit to insure equal outputs. More importantly, it sets the condition whereby a separation-control setting that results in "pure left" will also result in "pure right." The plate outputs are then passed through the familiar twin-T 38-kc notch filters to eliminate any residual switching voltage from the output signals.

The Model A3MX as well as its self-powered companion adapter, the Model S3MX, are pictured in Fig. 7-20.

ZENITH RADIO 12H26

As a final circuit analysis, let's now examine the circuitry of the Zenith Radio 12H26 chassis, since it includes all the

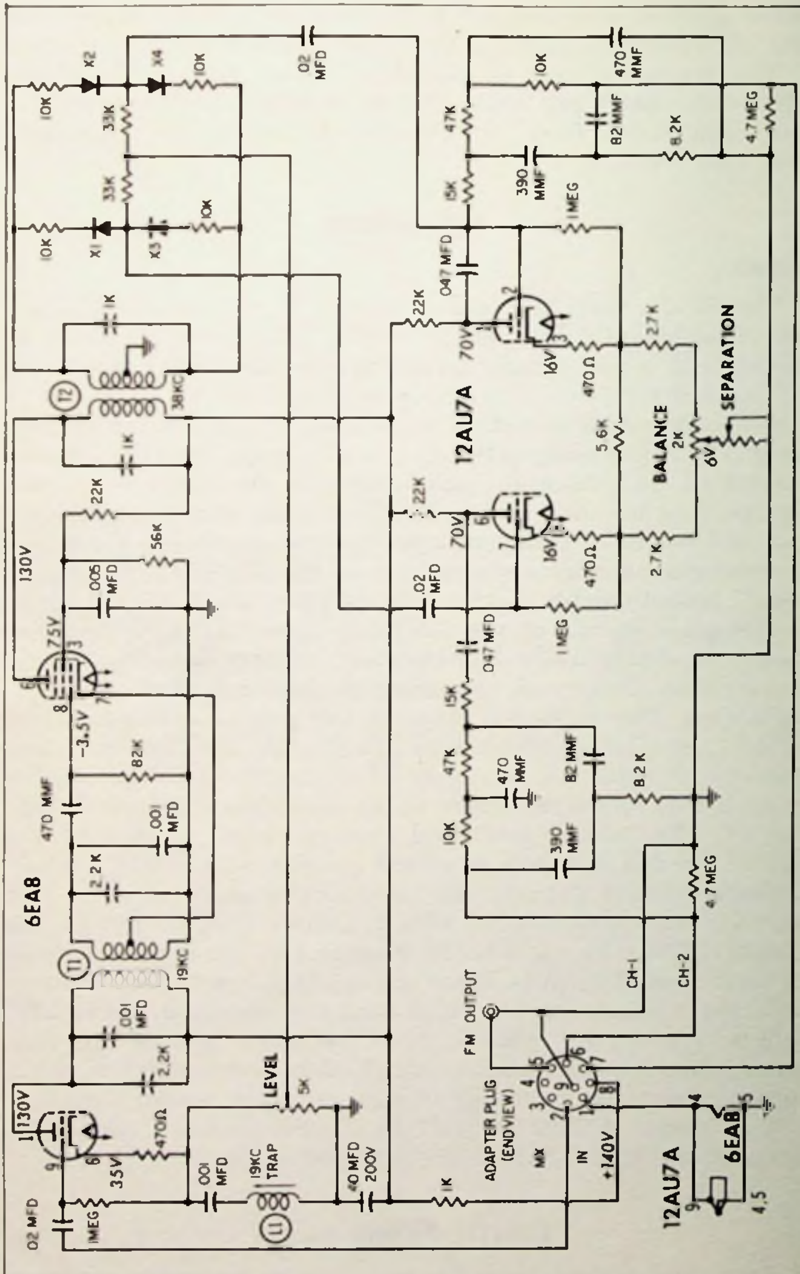


Fig. 7-19. Sherwood A3MX adapter circuit.

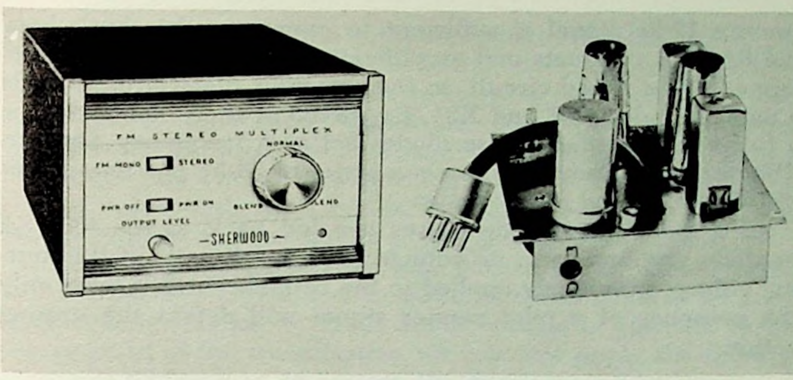


Fig. 7-20. Sherwood FM S3MX and A3MX stereo adapters.

features common to all multiplex chassis in that company's line of products. Fig. 7-21 is a partial schematic of the multiplex circuitry showing only the 19-kc pilot amplifier and frequency doubler. Since this circuit uses some new approaches, let's examine it in somewhat more detail than usual.

The 19-kc pilot signal taken from the ratio-detector output of the tuner portion (not shown) is fed to a tuned-grid–tuned-plate amplifier. To prevent operation of the circuit except when stereo is received, this tube (half of a 6AW8A) is muted in the absence of an adequate 19-kc signal. The muting voltage is obtained from B+, through 25K potentiometer R2, and applied to the cathode of the 6AW8A pentode section. When the in-

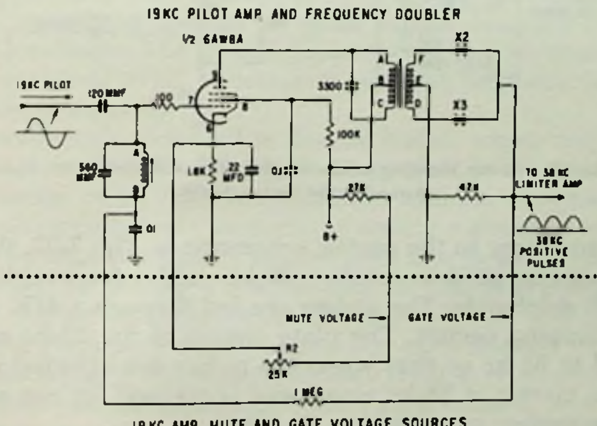


Fig. 7-21. Zenith 12H26 multiplex pilot amplifier and frequency doubler.

coming 19-kc signal is sufficient to overcome this back bias, the 6AW8A conducts and amplification takes place. A center-tapped 19-kc tuned circuit, in the amplifier plate circuit, feeds a pair of diodes (X2 and X3). By virtue of their connection as a full-wave rectifier, these diodes act as a frequency doubler. The output of their junction is a series of 38-kc positive pulses of DC and is used in two ways:

As a gate voltage, the pulses are fed back to the 6AW8A pentode grid and raise its voltage to within 2 volts of the muting voltage previously applied to the cathode. In this way, only the presence of a pilot carrier signal will defeat the muting action.

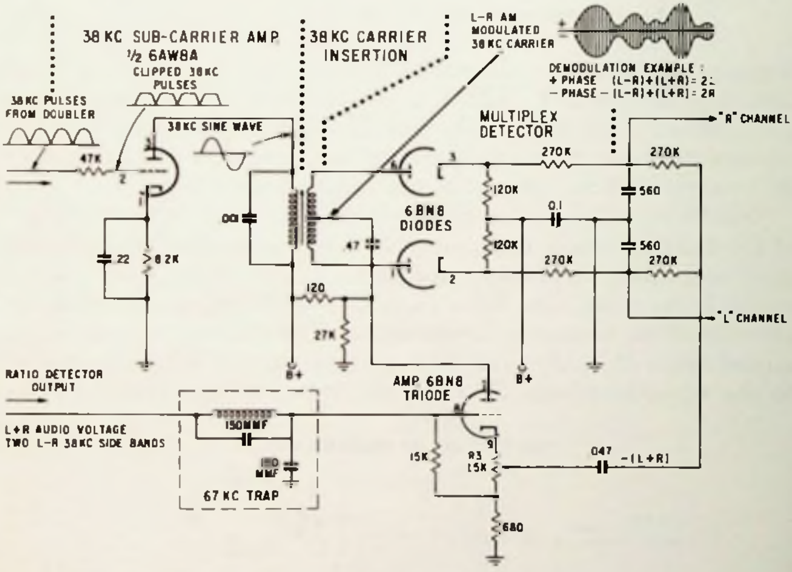


Fig. 7-22. Zenith circuitry showing subcarrier amplifier and reinsertion, L and R signal demodulation and matrixing.

Referring now to the partial schematic in Fig. 7-22, the 38-kc pulses are also used to create a 38-kc carrier for insertion with the L−R sidebands. The pulses are fed through a 47K resistor, where clipping occurs. The plate circuit of the 38-kc amplifier is tuned to 38 kc so that when the pulses are injected into this resonant circuit, a 38-kc sine wave is created for use as a carrier reinsertion, or switching, voltage.

At the ratio-detector output, the L+R signal, two L−R sidebands, and possibly the SCA (background music) carriers are all present. Therefore the total composite signal is fed to the broad-band 67-kc trap, and then to the triode amplifier (6BN8). The plate output of this stage is coupled, through a 0.47-mfd capacitor, to the center tap of the 38-kc carrier-reinsertion transformer. Because the 38-kc carrier discussed previously is also there, the two L−R sidebands will now have their 38-kc carrier reinserted. As a result, the familiar amplitude-modulated envelope appears at the plates of the 6BN8 diodes.

During positive half-cycles, the bottom diode will demodulate the positive half of the L−R voltage and develop an L−R audio voltage. Since an L+R audio voltage is also fed to the center point of the transformer, we can now apply the familiar algebraic addition to determine the resultant signal.

$$L-R + L+R = 2L$$

On negative half-cycles, the opposite diode will detect the negative (L−R) half of the envelope. Since the L+R voltage is also being fed to the center point of the transformer, we can add, as follows:

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

As you learned earlier, the above explanation assumes the L+R and L−R components in the composite signal are equal—which they never are. For this reason, a separation control employing some degree of corrective matrixing is used. The desired amount of −(L+R) voltage is taken from the cathode circuit of the 6BN8 triode by R3 and applied to the left- and right-output channels. This is done in such a way that the net L+R voltage is reduced until it is equal to the detected L−R and −(L−R) voltages. The de-emphasis circuit in each channel consists of a pair of 270K resistors in parallel, followed by a 560-mmf roll-off capacitor.

I am extremely indebted to Zenith Radio, whose explanation was paraphrased here. Zenith Radio prefers to describe its circuit in terms of an AM demodulation system, whereas in the book such circuits have been continuously viewed as switching types. The important point is that even the so-called switching circuit may be analyzed successfully in this way—that is, by treating the actual components (L−R sidebands and L+R audio) of the composite signal separately, even though never separated electrically. Therefore you may wish to re-examine some of the other switching circuits, with a view toward proving that they really are AM waveform demodulators and L+R adders after all! Whether they are or not depends on your point of view.

APPENDIX

FCC RULES AND SPECIFICATIONS

The rules and specifications concerning FM stereo broadcasting were adopted by the FCC in four parts.

The first part, new section 3.297 authorizes FM stations to begin stereo broadcasts as of June 1, 1961 and defines notification procedure.

Part two, section 3.310, is amended to include definitions on stereo broadcasting.

Part three, amended section 3.319, defines engineering standards for SCA operations with or without simultaneous stereophonic broadcasting.

The fourth part, new section 3.322, defines stereophonic transmission standards.

PART 2—DEFINITIONS

Crosstalk. An undesired signal occurring in one channel caused by an electrical signal in another channel.

FM Stereophonic Broadcast. The transmission of a stereophonic program by a single FM broadcast station utilizing the main channel and a stereophonic subchannel.

Left (or right) Signal. The electrical output of a microphone or combination of microphones placed so as to convey the intensity, time and location of sounds originating predominantly to the listener's left (or right) of the center of the performing area.

Left (or right) Stereophonic Channel. The left or right signal as electrically reproduced in reception of FM stereophonic broadcasts.

Main Channel. The band of frequencies from 50 to 15,000 cycles which frequency modulate the main carrier.

Pilot Subcarrier. A subcarrier serving as a control signal for use in the reception of FM stereophonic broadcasts.

Stereophonic Separation. The ratio of the electrical signal caused in the right (or left) stereophonic channel to the electrical signal caused in the left (or right) stereophonic channel by the transmission of only a right (or left) signal.

Stereophonic Subcarrier. A subcarrier having a frequency which is the second harmonic of the pilot subcarrier frequency and which is employed in stereophonic broadcasting.

Stereophonic Subchannel. The band of frequencies from 23 to 53 kilocycles containing the stereophonic subcarrier and its associated sidebands.

PART 3—SCA OPERATION

Strict attention should be given to section 3.319(b) which states that the instantaneous frequency of SCA subcarrier(s) shall at all times be within the frequency range 53 to 75 kc. The stereophonic receiver must be designed so SCA signals at 53 kc or higher will not interfere with portions of the stereo signal creating audio products which will be heard at the stereophonic receiver output.

PART 4—STEREOPHONIC TRANSMISSION STANDARDS

a. The modulating signal for the main channel shall consist of the sum of the left and right signals.

b. A pilot subcarrier at 19,000 cycles ±2 cycles shall be transmitted that shall frequency modulate the main carrier between the limits of 8 and 10%.

c. The stereophonic subcarrier shall be the second harmonic of the pilot subcarrier and shall cross the time axis with a simultaneous positive slope with each crossing of the time axis by the pilot subcarrier.
d. Amplitude modulation of the stereophonic subcarrier shall be used.
e. The stereophonic subcarrier shall be supressed to a level less than one percent modulation of the main carrier.
f. The stereophonic subcarrier shall be capable of accepting audio frequencies from 50 to 15,000 cycles.
g. The pre-emphasis characteristics of the stereophonic subchannel shall be identical with those of the main channel with respect to phase and amplitude at all frequencies.
h. The modulating signal for the stereophonic subchannel shall be equal to the difference of the left and right signals.
i. The sum of the sidebands resulting from the amplitude modulation of the stereophonic subcarrier shall not cause a peak deviation of the main carrier in excess of 45 percent of total modulation (excluding SCA subcarriers) when only a left (or right) signal exists; simultaneously in the main channel, the deviation when only a left (or right) signal exists shall not exceed 45 percent of total modulation (excluding SCA).
j. Total modulation of the main carrier including pilot subcarrier and SCA subcarriers shall meet the requirements of Section 3.268 with maximum modulation of the main carrier by all SCA subcarriers limited to 10 percent.
k. At the instant when only a positive left signal is applied, the main channel modulation shall cause an upward deviation of the main carrier frequency; and the stereophonic subcarrier and its sidebands signal shall cross the time axis simultaneously and in the same direction.
l. The ratio of peak main channel deviation to peak stereophonic subchannel deviation when only a steady state left (or right) signal exists shall be within plus or minus 3.5 percent of unity for all levels of this signal and all frequencies from 50 to 15,000 cycles.
m. The phase difference between the zero points of the main channel signal and the stereophonic subcarrier sidebands envelope, when only a steady state left (or right) signal exists shall not exceed plus or minus 3 degrees for audio modulating frequencies from 50 to 15,000 cycles.
n. Crosstalk into the main channel caused by a signal in the stereophonic subchannel shall be attenuated at least 40 decibels below 90 percent modulation.
o. Crosstalk into the stereophonic subchannel caused by a signal in the main channel shall be attenuated at least 40 decibels below 90 percent modulation.
p. For required transmitter performance, all of the requirements of Section 3.254 shall apply with the exception that the maximum modulation to be employed is 90 percent (excluding pilot subcarrier) rather than 100 percent.
q. For electrical performance standards of the transmitter and associated equipment, the requirements of Section 3.317 (a), (2), (3), (4), and (5) shall apply to the main channel and stereophonic subchannel alike, except that where 100 percent modulation is referred to, this figure shall include the pilot subcarrier.

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FM Multiplexing for STEREO

by Leonard Feldman

Leonard Feldman, a graduate of the College of the City of New York, is a well-known author in the hi-fi field, having written several books, as well as numerous articles on audio and FM. Formerly a project engineer at Fisher Radio Corp., he is now Director of Engineering for Crosby Teletronics Corp. Mr. Feldman is a member of the IRE and the Audio Engineering Society.

After nine years of development, FM stereo broadcasting is a reality.

FM Multiplexing for Stereo is a guide for technicians, audiophiles, and laymen who want to know the principles of FM stereo multiplexing, the theory and operation of the various receiver circuits, and the details of aligning and servicing these circuits. Discussions of the FM stereo composite signal, converting to FM stereo, multiplex circuits and decoders, and the latest multiplex circuits, introduce the reader to receivers, tuners, and adapters. Two full chapters on servicing and alignment of multiplex circuits are included, in addition to an appendix listing Federal Communications Commission specifications and multiplex demodulation techniques.

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