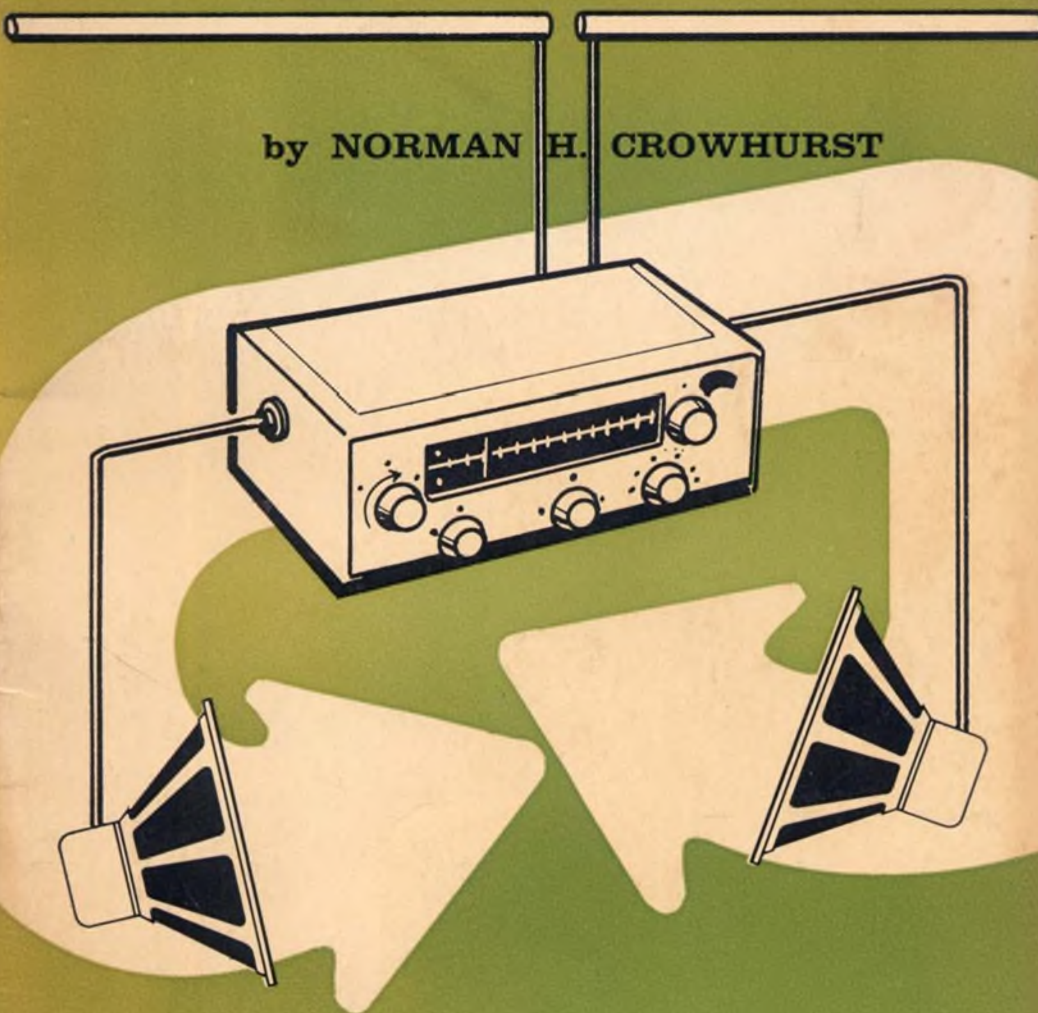
No. 282

FM STEREO multiplexing

by NORMAN H. CROWHURST



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fm STEREO MULTIPLEXING

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preface

Writing a first book on this subject was not easy, because no one had had any experience in its application on which the author could draw. When he started to collect material for it, releases had already been issued, claiming that this or that company was going into production with a new adapter using revolutionary new principles developed by its engineers. . . . . . But technical information was "not ready for release".

Contacting various engineers directly, the author received many invitations to "come and see what we are doing", in lieu of supplying information. This resulted in what proved to be a series of mutual "brain-picking" sessions, from which evolved several of the new adapter units that attempted to fulfill the claims already made by the publicity releases!

After a week or so of such intensive work, the author found himself rapidly accepted as an undisputed authority on the subject, which effectively overcame the slight hesitation he had felt in undertaking the preparation of this book. Actually, the problems were not so much difficult as they were unfamiliar.

It is true that still only a few people "know the answers". That is why this book is needed. There is a tremendous demand for an explanation of what stereo multiplex is, and how it relates to the more familiar aspects of high fidelity. The new system has resulted in a number of approaches to the design of adapters and other items of equipment. These too, need explaining, so their relative efficacy can be more effectively judged.

Thousands of technicians are going to find themselves suddenly confronted with stereo multiplex, its installation, alignment, servicing, etc., and they need something to set them on the right road. At this juncture there is no wide background of experience in the field, but providing them with an understanding of what they have to cope with can be a big help to them in finding their own way.

That, in essence, is what this book does. Because the subject is so new, this first edition must in some respects be somewhat of a shot in the dark. But repeatedly during its preparation, the author realized that stereo multiplex is just another application of basic audio, in which he has many years of experience behind him. So, by the time the manuscript went to the publisher, it was quite an enlightened shot! And we believe that time will show that there is not much, if anything, left out.

NORMAN H. CROWHURST

New York, N.Y.
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1 stereo: requirements for broadcasting ■

Stereo is the latest development in high fidelity reproduced sound. The objective of all this work is realism. Achieving uniform response to all the audible frequencies was the first requirement recognized. Amplifiers and systems, including radio transmitters (FM), in which deviation from uniformity in response was audibly negligible, had been available for a long time. But realism was still lacking.

Waveform distortion — due to nonlinear effects that introduce spurious frequencies, rather than nonuniformity between those that should be there — was the next target for improvement. For a long while, the best systems had had extremely small departure from uniformity of frequency response and almost nonexistent waveform distortion. Reproduction and its attendant realism should have been almost perfect, but it wasn't; you could still tell the difference between reproduced music and the real thing, quite easily.

The next step was stereo. The trouble had been the fact that all the sounds came from one loudspeaker, or system of loudspeakers, whereas original program music comes from many different sound sources, whether these be different instruments in an orchestra, different voices in a choir, or background sounds due to reverberation, accompanying a single speaking voice or singer. The difference was the lack of directional perception to individual sounds in reproduced program.

This fact had been known for many years, but it had not been possible to do much about it. As early as the Paris Exposition of 1881, the realism conveyed by two channels of sound via the extremely primitive microphones and headphones of the day was demonstrated. In the early '30s Bell Labs proved, with somewhat more modern equipment, just what was required, in terms of sound conveyed by two or more channels.

But it was one thing to show this with the facilities available to Bell Labs, and quite another to make it available for the general public to use and hear. The first attempt in this general area came from the motion picture industry. The production of Walt Disney's first experiment, *Fantasia,* followed the original Bell Labs experiments as closely as practical development would allow. It failed to get the appreciation it deserved, however, because too few of the public were able to hear it in all its glory. It required an elaborate installation in theaters and as so many cinemas where it was shown lacked this equipment, and showed only a single-track version of the film, all point of making it was lost. The film's announcement that this represented a new adventure in entertainment had no impact without the comprehensive sound presentation that should have accompanied it.

More than a decade, and a World War, went by, before Cinerama took up the ball, and applied the latest work in optics and multichannel sound recording to start the wide-screen presentation. This system also required such extensive changes to the theater used for its presentation that it was put beyond the reach of the average local theater-goer.

CinemaScope, with only four tracks of sound, on magnetic tracks instead of optical, and anamorphic projection of a single picture instead of three synchronized pictures, as needed by Cinerama, brought the "poor man's Cinerama" into most local cinemas.

All kinds of wide-screen technique, with slight variations in the sound tracks accompanying them, followed CinemaScope, with Hollywood's usual superlative claims for each. Certainly, *good* stereophonic sound could enhance the presentation of a movie. The minimum number of channels used in any system was three, for left, center and right presentation on the screen.

The multiple tracks used for theater presentation were really needed because the sound *was* reproduced in a theater, rather than in a room nearer the size of an average livingroom. The multiple channels helped overcome the audible inconsistencies of scenes supposedly occupying average living areas, but actually presented in an auditorium the size of a movie theater.

Work on multiple-channel presentation (stereo) in the home, using average-sized livingrooms, showed that there was much less, if anything, to be gained by using more than two channels. With two-channel presentation in an average-sized livingroom, much better realism is possible than with the older monophonic, single-channel, high-fidelity system.

The accepted form of stereo for home use employs two channels of sound, one for left and one for right. These are fed to two loudspeakers, usually situated in front of the listening position, and to the listener's left and right respectively (Fig. 1-1). Standard recordings or transmissions consist of two channels of program, intended for feeding to such left and right loudspeakers.

The advent of tape recorders that really enthusiastic high fidelity followers could afford, using tapes recorded with two tracks instead of

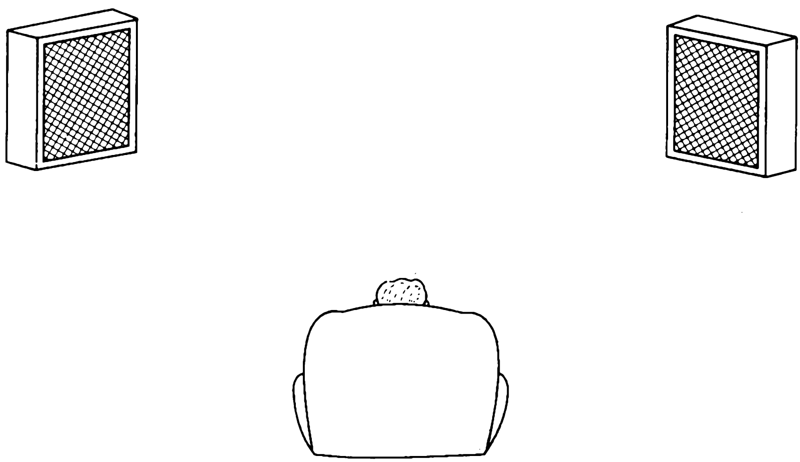


Fig. 1-1. The accepted way of placing loudspeakers and listening to stereo.

the earlier single track, got stereo a hearing in the more exclusive circles. Tape was too costly at that stage to compete with disc phonograph records for wider acceptance. To make the competition more difficult, the quality of phonograph records had made tremendous advances since the advent of microgroove long-playing records in 1948.

Much development work went on in phonograph recording techniques, to try and get two channels recorded in the single groove. Early attempts were pathetic. But finally, effort was rewarded with quality that came close to that of monophonic records of the time — close enough to give hope that they could be perfected to be equal and ultimately better than the monophonic variety. So the 45/45 stereo disc came into being, in 1958.

Back in the '20s, the big question was which entertainment medium for the home would survive, radio or the phonograph. Then came television. Radio had to rely more and more on disc jockeys for most of their programs, so that question was answered! Both were mediums for conveying high-fidelity program, one over space, the other over time — and they could be combined.

Just as getting two channels into one groove on disc had its problems, so did getting two sound channels into one radio transmission. The objectives were the same.

BALANCE

The most obvious requirement for stereo — especially when it is missing — is balance between the sound from both channels. If the left channel is the least bit louder than the right, all the sound seems to come from the left, and you might as well have monophonic. When this condition exists, you have to put your ear right up close to the softer loudspeaker to be sure it is working *at all.*

SEPARATION

Stereo produces its improvement over mono by allowing sounds that originate from separate instruments to sound that way, instead of sounding jumbled into a melange. Although the sound for each instrument may come partially over each channel, this property of separation of individual sounds can only be properly rendered if the stereo channels are kept *really separate.* Sound intended for reproduction by the left loudspeaker must not leak into the right loudspeaker, or vice versa.

FULL FIDELITY IN EACH CHANNEL

Almost as bad as lack of balance or separation in destroying intended stereo effect, is difference in the fidelity with which the separate channels are reproduced. To gain its full advantage as an improvement over mono, each channel should have not less than the full fidelity previously expected from the one channel of a good monophonic system.

Finally, any system that somehow gets two channels into accommodation previously only intended for one, has distortion problems of its own, quite different from those fought against to get high quality mono. Not only might each individual channel get distorted, or one channel leak into the other, or lack of balance exist between them, but there are various interactions that can make new distortions that never bothered a single channel by itself.

So the evolution of a system suitable for handling stereo for radio transmission, as earlier for recording, has quite a number of requirements to meet: (1) balance between channels; (2) separation between channels; (3) maximum fidelity in each channel; (4) freedom from distortions peculiar to a two- or multichannel system.

2 experimental systems

Because the broadcast frequency band had the established bandwidth allocation that limited its fidelity to an upper audio frequency of about 4000 cycles, the first step toward providing high fidelity radio transmission was the establishment of the FM type transmission, up in the 100 mc region.

Two things led to the use of FM in this frequency range. First was the fact that AM (amplitude modulation) showed progressively poorer dynamic range, due to greater atmospheric background noise, as the carrier frequency went up. Extending the audio frequencies used, from 4000 cycles to, say 15,000 cycles, only aggravated this situation. Second was the fact that the use of a much higher carrier frequency made it impractical to put frequency allocations anywhere near as close together as in the broadcast band.

In the broadcast band, with carrier frequencies around 1 mc, adjacent carrier allocations may be 10 kc apart. Going up 100 times in carrier frequency, to the region of 100 mc, adjacent frequency allocations have been set up 200 kc apart, which is still only one-fifth the fractional, or percentage, bandwidth used in the broadcast band. At the same time, it is five or six times the width needed for AM transmission of audio frequencies.

Where FM (frequency modulation) scores over AM is that this "extra space" is used, instead of being thrown away (Fig. 2-1). By allowing the audio to change the frequency, instead of the amplitude, of the transmitted carrier frequency, the *range* of frequency used can be expanded (Fig. 2-2). Although audio frequencies up to only 15 kc (15,000 cycles) may be used, they can modulate the carrier frequency by as much as 100 kc (half the 200 kc spacing between adjacent frequency allocations).

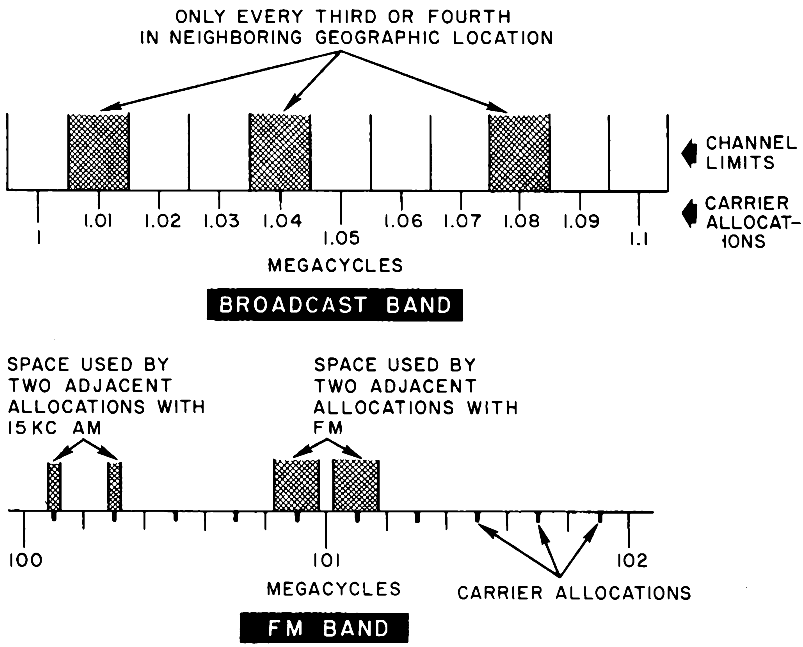


Fig. 2-1. Comparison of channel spacing in broadcast and FM bands to show why frequency modulation can effect such an improvement in the FM band, but not in the broadcast band.

Actually, 75 kc maximum deviation is allowed, to provide a margin of safety from interference between adjacent frequency allocation transmissions.

When 15 kc of audio occupied 15 kc on either side of the carrier, with a "spare" space of about 100 kc (limited by the sharpness of tuning), an FM tuner would pick up 30 kc of program with 200 kc of atmospheric noise. But frequency modulation enables the same tuner to pick up as much bandwidth of program as it does of noise, with resulting reduction in the effect of the noise in the reproduction.

So, after early experiments with it, FM was quickly established as the medium for high-fidelity program transmission. When television became

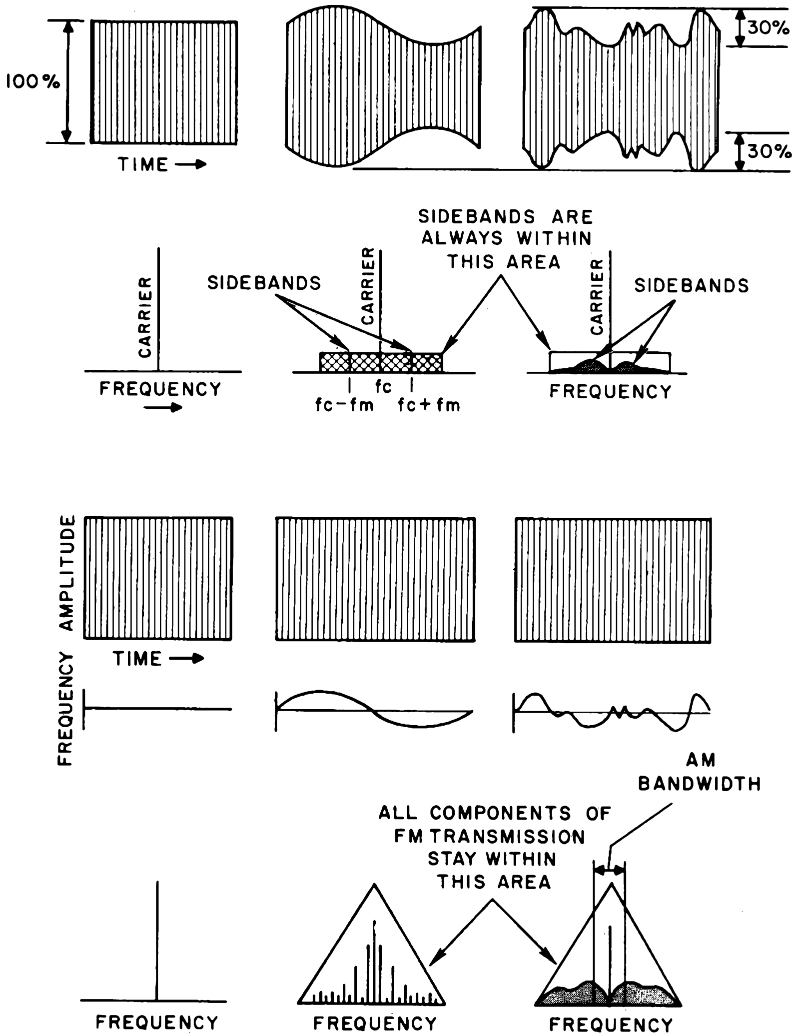


Fig. 2-2. Comparison of different forms of modulation: top row shows the radiated envelope for an AM transmission; second row shows the radio-frequency spectrum for the envelopes in the top row; third row is the envelope for FM transmissions (amplitude does not change); fourth row shows the frequency fluctuation of the carrier for the third row; fifth row shows the radio-frequency spectrum for the envelopes represented in the third and fourth rows. The left column shows an unmodulated carrier; the middle column a carrier modulated with a single frequency; the right column represents a typical composite modulation.

popular, sponsors found it a better medium for selling their products than FM radio, so interest waned and many stations went off the air for economic reasons. A few kept broadcasting because they maintained FM as one of several facilities owned by the same station. AM radio remained popular with sponsors, because through it they could reach people riding in autos, which TV could not.

THE BEGINNING OF STEREO BROADCASTING

Stations that retained, along with their audiences, an interest in high-fidelity transmission, experimented with stereo transmission. Having more than one facility, this was comparatively easy to do. They had an FM and an AM transmitter, often broadcasting identical program. To make it stereo they could, by special arrangement, make one carry the left and the other the right channel of stereo.

As in the early professional experiments, like those at the Paris Exposition and later at Bell Labs, the first objective with system demonstrations was to find out "if stereo worked". Then to show someone else that it did. Listeners were alerted to what was being done, and advised to bring an AM set, such as the kitchen radio, alongside their FM receiver or tuner. They were told how to adjust levels for balance, and then an experimental transmission would follow. It worked, and many stations and listeners were quite enthusiastic.

But only enthusiasts would go to all that trouble. For stereo to be successful as a medium, it wanted something easier than that. Also these AM-FM transmissions had serious limitations. Only quite close to the two transmitters (which would not always be at the same location) would transmission level on both channels stay constant enough to enjoy stereo without constant level checks. Further out from the transmitters, one or other of the transmissions might not be receivable at all, so the effect was completely lost to these listeners.

Also, true balance could never be achieved, because the two channels could never have full fidelity. One could have full fidelity (the FM), but the other (the AM) was limited to about 4000 cycles — a serious limitation.

However, these demonstrations were enough to show enthusiasts that stereo worked, and to let them hope for something better. Only a real enthusiast would appreciate this, because only an enthusiast would go to the trouble of setting up and keeping the proper balance (as well as such a system was capable of) between channels.

In some cities a network might own more than one FM station, or by mutual agreement, two stations might operate together for an experimental transmission. This eliminated the quality disparity between channels, for listeners well within the service area of both. But the method still required careful balancing — which only the real enthusiast was prepared to trouble with — and was limited to relatively few large cities where more than one FM channel was available and they could work together. Even then, stereo was not an economic proposition, because normally the two stations would be broadcasting separate programs, reaching different audiences; they had to sacrifice this during stereo broadcasts.

Just as the need to popularize stereo recordings was a disc with both channels in one groove, so radio needed a way of transmitting both channels over one radio-frequency allocation, not two. Incidentally, an earlier attempt with discs had some success, using two grooves instead of one. This was put out by Emory Cook, and called the *Binaural Record.* The outer half of the disc carried the one channel and the inner half the other. A special "double-headed" pickup played both grooves at once, one for each channel.

The Cook record was good, but not easy to use. Now that we have 45/45 records, with two channels in one groove, Cook makes his stereo records the new way.

MULTIPLEX

A way of getting extra channels of program over a single-carrier radio transmission has long been known; it is called *multiplex.* If you have two

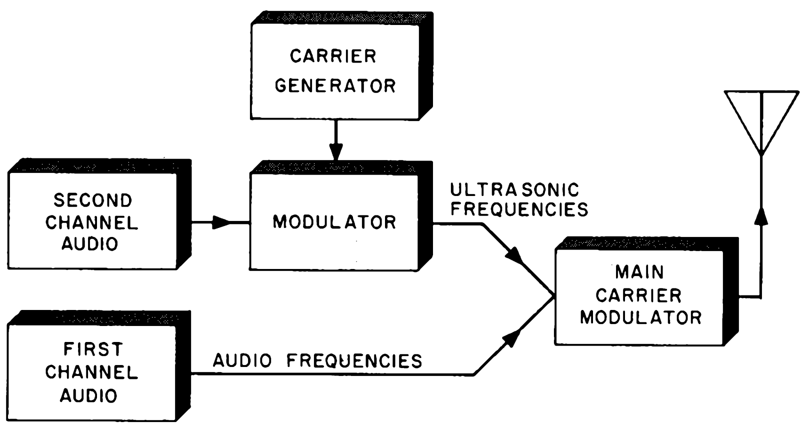


Fig. 2-3. The basic system for multiplexing as developed for multichannel telephony transmission (simplified).

channels to send, you use one of them to modulate a "subcarrier" (Fig. 2-3). This is a frequency not much above the audio range — just far enough above not to interfere with the other channel. Then the two, the simple channel and the modulated subcarrier, are used to modulate the main carrier, just as if it were all one program. At the receiving end, filters separate the first (unmodulated) channel from the one on the subcarrier, and then the subcarrier is demodulated to retrieve the second channel.

The first experiment followed just that method. One channel, say left, modulated the main carrier directly, while the other, in this case the right, modulated the subcarrier and then joined the left in modulating the main carrier. For some reason, both modulations were FM. The right channel FM modulated the subcarrier, which was added to the left channel, and then the combination used to FM modulate the transmitted carrier. In the radiated signal, the right channel was an FM modulation, FM modulating the final carrier.

The problem was that all the frequency modulation, or deviation, had to be kept within the allocated 75 kc per station. If the subcarrier used up half the main carrier's deviation, the left channel only had the other half to use.

There was another problem. Due to the economic situation in which FM stations found themselves, the FCC (Federal Communications Commission) had allowed licenses for FM transmitters to lease subcarriers to private services, such as background music for stores and offices, while retaining public transmission in their main channel. If they now use the same station for multiplexed stereo, of any kind, these stations will need two multiplex subcarriers, one private, one public.

Use of two subcarriers would further restrict available deviation, or modulation, to be used by each. Fidelity would finish up 'way down, as compared with previous mono transmissions. To try and salvage some fidelity, some stations are proposing to use very small deviation for the subcarrier channel, and keep as much as possible for the main channel. This method, however, ends up with one of the main limitations suffered by AM-FM transmissions — a serious quality disparity between left and right.

Quite a number of proposals were put forward for different ways of combining the left and right stereo programs on the one transmitted channel. Some proposals attached exaggerated importance to the provision of room for one, or even more, private subcarriers in addition to one for stereo. Most of them were based on experiments that seemed to prove that certain degradations in the stereo channels, left and right, were completely unnoticeable. Some proposals were very concerned with maintaining what they called compatibility with the AM-FM system, so people could continue listening to that system, or change over to the new, without loss of quality either way.

The fact was, of course, that AM-FM stereo had a built-in loss of quality due to the limitations of the AM channel. Virtually every system that aimed for compatibility with it, also included limitation of the new system in the same way.

MATRIXING

Murray Crosby was responsible for introducing the matrixing idea, which achieves a different and quite useful form of compatibility, and at the same time assures uniform quality between left and right, as well as

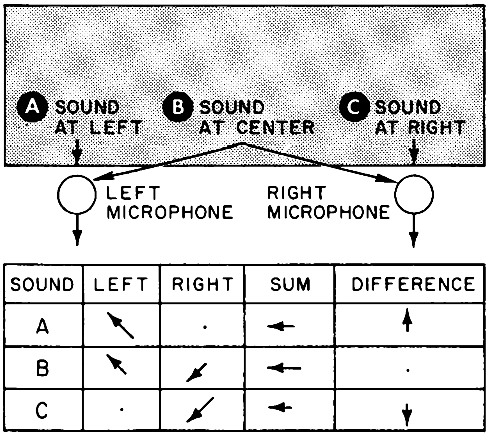


Fig. 2-4. The principle of matrixing illustrated. On discs, movement at opposing 45° angles represents left and right, horizontal (lateral) movement represents center signal. Lateral and vertical represent sum and difference. The significance of this is shown for three representative sound sources, showing that *either* left and right or sum and difference can convey the whole sound picture, and mean the same thing.

good control of balance. His form of matrixing takes the two channels and mixes them in the same way a monophonic pickup mixes the two channels on a stereo disc. At the same time another mix is made similar to the vertical component of movement on a stereo disc (Fig. 2-4).

The first mix, corresponding to lateral movement on a stereo disc, is the *sum* of left and right, and corresponds with what you would hear in a monophonic program. The second mix, corresponding to vertical movement on a stereo disc, is the *difference* between left and right, and is the essential information to convert monophonic into stereophonic.

Such a matrix is relatively simple (Fig. 2-5). By modulating the sum, or monophonic mix directly on the main carrier, and modulating the subcarrier with the difference or stereophonic mix, the listener without the proper adapter or stereo receiver can hear the program as quite normal monophonic sound. The addition of a relatively simple adapter would enable him to convert to stereo.

To maintain maximum quality stereo, in terms of all its qualities — balance, separation, fidelity in each channel — Crosby planned to use an

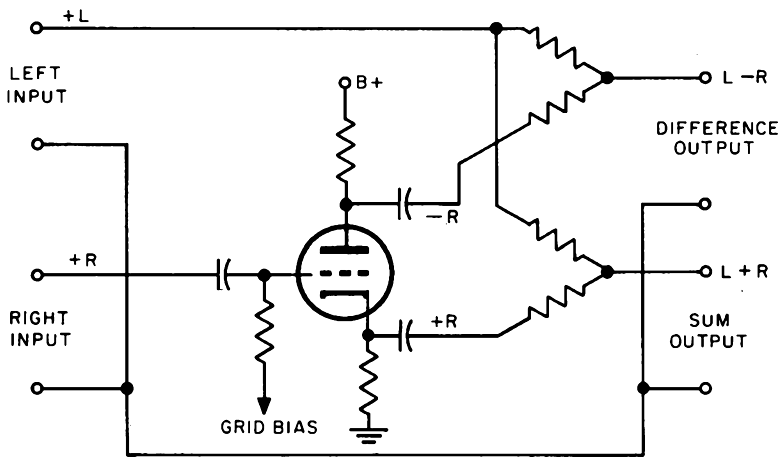


Fig. 2-5. A simple tube circuit for achieving matrixing, to convert left and right to sum and difference; the reverse process is identical.

FM modulated subcarrier with fairly wide deviation. Half the available main carrier deviation was to be used for the sum channel and half for the subcarrier with its difference FM modulated on it.

COMPATIBILITY

Notice the different forms of compatibility involved: the Crosby method maintained compatibility between monophonic and stereophonic reception, which is really the important one, so people who have not "gone stereo" do not suffer; what many others were after, in addition to this, was compatibility with earlier, experimental methods of transmitting stereo.

One of these proposed a different matrix at the transmitter. Instead of taking the sum and difference of left and right, both transmission channels were to carry a form of difference mix. If we write L and R for the original left and right channels, the simple (Crosby) matrix gives

L + R for the sum, and L — R for the difference. The Burden proposal was to use mixes of 2L — R, and 2R — L.

The argument for this was that the greater intensity of L in left, and greater intensity of R in right, would maintain the stereo effect when received stereophonically. But people with either an ordinary FM or an ordinary AM set (which would carry the same as the subcarrier on the FM) would get a mix of L and R that would be acceptable.

TIME-DELAY MATRIX

Where this happens to be a necessary feature of a transmission, there is a better way to do it, which was developed by Bell Labs, and tried experimentally with a TV transmission. The TV sound carried one

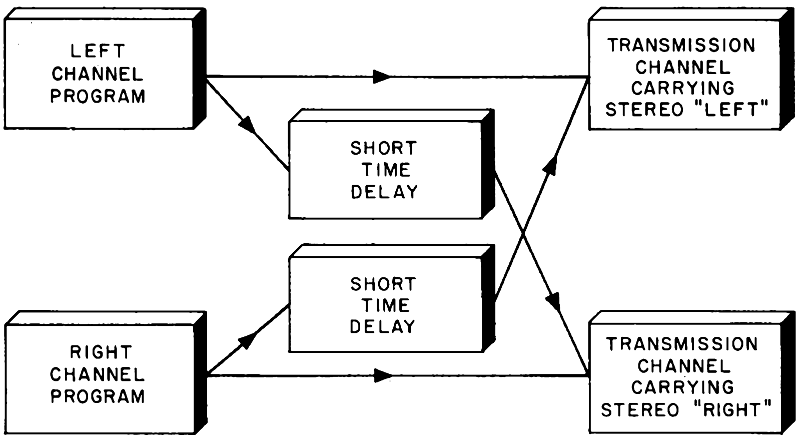


Fig. 2-6. The time-delay matrixing used by Bell Labs for an experimental stereo transmission, using two completely separate channels, where either should convey acceptable monophonic by itself.

channel, and a radio transmission, which could be either AM or FM (or both), carried the other (Fig. 2-6). One channel (say, the TV sound) carried the left program with the right program delayed a few milliseconds. The other carried the right program with the left program delayed an equal number of milliseconds.

The short, few milliseconds delay was not enough to be noticeable on either transmitted channel by itself. It was equivalent to moving a single monophonic microphone a little away from center, one way or the other (Fig. 2-7). But when the two channels were connected to loudspeakers placed in left and right positions, the stereo effect appeared.

The time difference made it appear that left program originated with the left loudspeakers, while the left program coming from the right loudspeaker would be like a reverberant reflection of the same sound, and vice versa. If a large time delay were used, the effect would be like cross

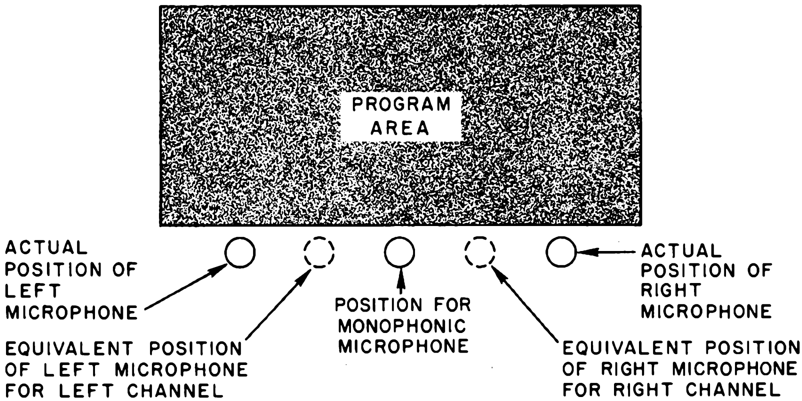


Fig. 2-7. How the matrixing method of Fig. 2-6 affects the effective microphone positioning.

echos. But by using a short delay—no more than occurs in sound reflected from the walls of a normal livingroom, the hearing does not detect it as a separate sound at all; it merges with room reflection effects.

With such experimental transmissions, everyone had full quality; those who only had the TV set, or those who only had a radio set, each received a perfectly satisfactory monophonic transmission; those who put the two together got complete stereo with their TV picture.

Some people saw the Burden proposal as doing essentially the same thing by different means; actually it was simple cross-mixing, a degradation of stereo separation to a mere 6 db, after record makers, pickup manufacturers and others had struggled to get separation figures of 20 or 30 db and higher.

There were also those who suggested the time-delay matrix as an alternative to simple sum and difference, because of its greater versatility. But this was never the intention. If stereo is to be broadcast over one channel, the time delay method is pointless. Only in the absence of multiplexing, where two separate radio channels have to be used, and which may be received separately or together, does the time-delay matrix have any advantage or usefulness.

3 the final standard

Choosing the final standard was no easy job. Responsible organizations set up investigating committees to find the facts about each proposed system, its theoretical possibilities and its practical performance in experimental field tests. After the study of all these facts, which were quite voluminous, and some work of their own, the FCC handed down the final decision.

TIME-DIVISION MULTIPLEX

Another idea had been put forward: the use of time-division multiplex. In data transmission, this method is used to send many kinds of information over a single channel, continuously, but not quite "at once" (Fig. 3-1). The individual pieces of information are transmitted in very fast, repeated sequence. Applied to stereo, the idea was to chop up the left and right program channels at an ultrasonic frequency and then transmit them alternately over the one channel.

If the data-transmission pulse sequence method were used, the high-frequency components of the pulse frequency would be extremely high — at least 75 kc, which would limit the deviation ratio of FM transmission to 1, and virtually destroy the advantage of using FM instead of AM at these transmitter frequencies (around 100 mc). Also the receiver's "decoder" would need very precise synchronizing if the pulses were to be put back in their correct left and right channels. For these reasons, most people dismissed the idea of time-division multiplex for stereo.

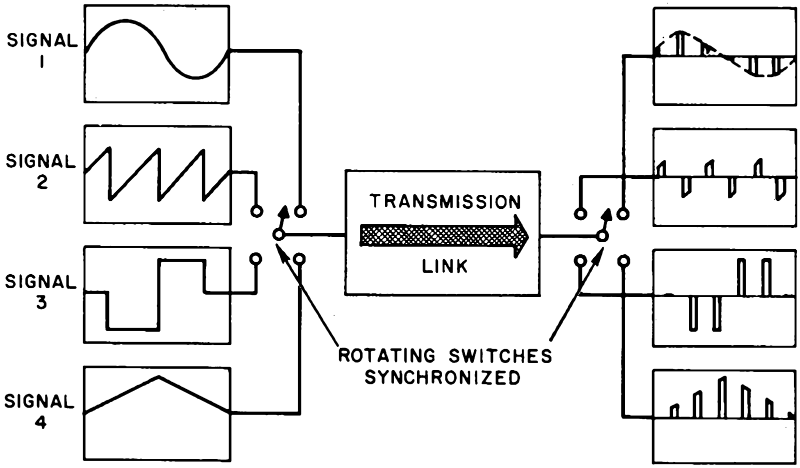


Fig. 3-1. Basic principle of time-division multiplexing.

AM SUBCARRIER

Each of the proposed systems discussed in the previous chapter used an FM-modulated subcarrier for the multiplexed channel, probably because FM has come to be associated with better quality sound than AM. But the real reason for FM's superiority rests in the carrier frequency used — in the 100 mc region. Some unitiated people have suggested using FM in the broadcast range (535 kc to 1.6 mc), but an understanding of the subject shows it to be quite impractical because of the bandwidth limitation.

FM gains in the 100 mc band, only because a much larger bandwidth is available, that AM cannot use. When we come to use multiplex subcarriers, the subcarrier frequency is only in the low kilocycles — well below 75 kc at best. So why use FM here? It is true that FM subcarriers have been used for private music transmission services, but their fidelity is low; they use a small-amplitude subcarrier, with small deviation and correspondingly limited fidelity.

There is no reason for stereo to follow the same method, unless we want to limit its quality, which we do not. On this basis it might seem logical to use an AM subcarrier. In FM modulation of the main carrier, the subcarrier represents a minimum requirement of two sidebands, separated by its own frequency from the main carrier, and an allocation of the total deviation that is proportional to its magniude (Fig. 3-2).

The disadvantage of amplitude modulation as a subcarrier component of composite frequency modulation, is that the biggest single element is the carrier, which is unchanging. The sidebands can each reach only 50% or less of the subcarrier amplitude (Fig. 3-3). This seriously limits the

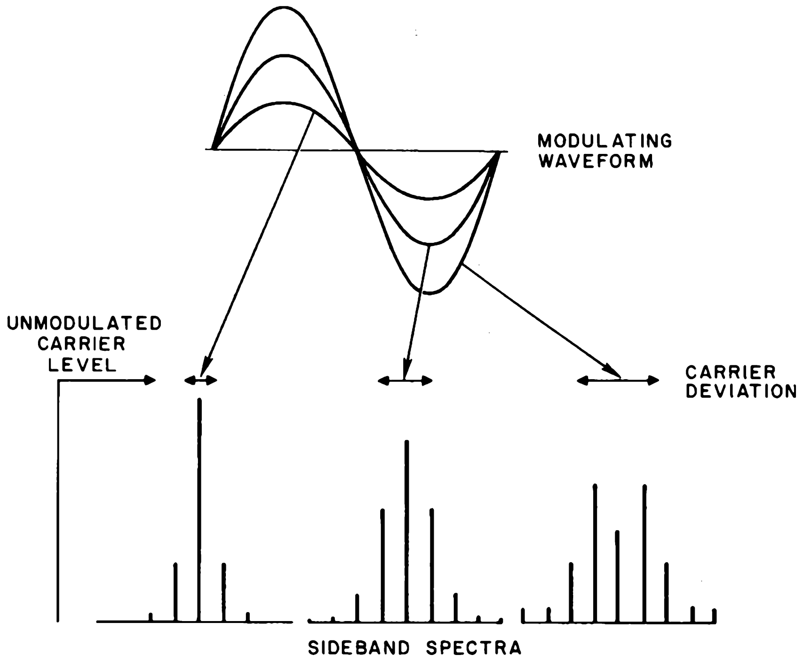


Fig. 3-2. How changing the amplitude of a single frequency, FM-modulating a carrier, affects the deviation and sideband content; notice that the carrier level diminishes, so the total energy radiated on carrier and sidebands is constant; also notice the progressive spread in magnitude and number of sidebands needed for just this one frequency.

relative magnitude of the subcarrier modulation, and thus restricts the dynamic range (signal-to-noise ratio) of the difference channel.

If the subcarrier can be suppressed and only its sidebands used as part of the composite radiated modulation, the level can be improved by about four times, on the difference channel.

MERGING IDEAS

At the same time as one company was working on a system using this approach, another company was working with the time-division multiplex idea. If the switch, instead of being made an on-off device, or a succession of more or less square pulses, is a sinusoidal variation from left to right at a predetermined frequency, the additional frequencies produced by multiplexing are only the sidebands of the "switching" frequency pro-

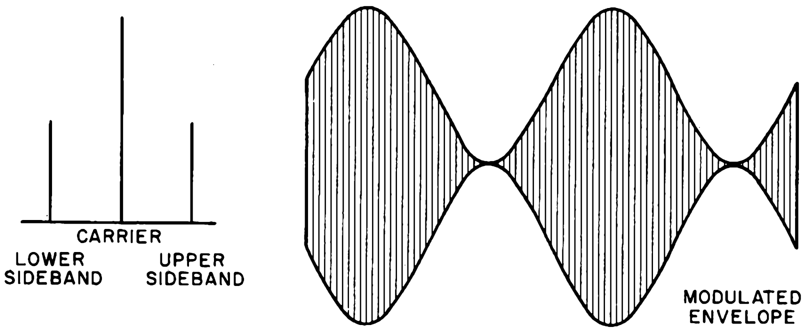


Fig. 3-3. Limitation of AM modulation: the envelope is that of a single frequency amplitude-modulating a carrier to its theoretical maximum of 100%; the spectrum shows the frequency content of this envelope. Practical modulations seldom go to 100% and consist of many frequencies, instead of just one, so the magnitude of sidebands is always a long way below that of the carrier.

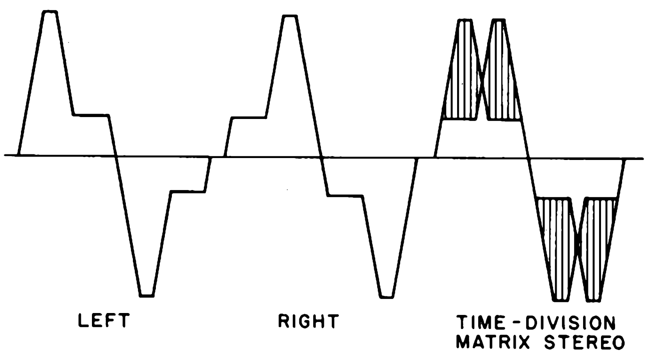


Fig. 3-4. Principle of time-division matrixing applied to stereo. Here, some rather impractical waveforms have been chosen, to make the effect easier to trace. Only where the left and right waveforms differ, is there any subcarrier present.

duced by a modulation consisting of the difference between left and right (Fig. 3-4).

In other words, time-division multiplex, done this way, is the same as suppressed-carrier amplitude-modulated multiplex, using sum and difference. So two ways of doing it result in the same transmitted "signal" (Fig. 3-5). Just one more thing needs attention.

SYNCHRONIZATION

If you regard the signal as suppressed carrier multiplex, the carrier has to be reinserted at the receiver before the difference channel can be de-

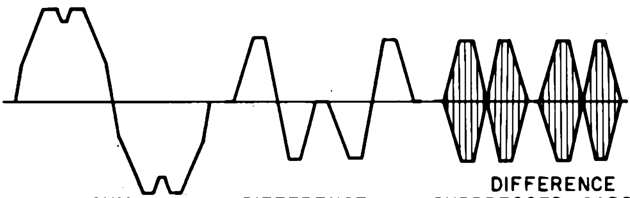


Fig. 3-5. The same thing as Fig. 3-4, regarded as suppressed-carrier modulation of difference signal. The extreme left and right waveforms here, added together, yield one identical to the right waveform of Fig. 3-4.

modulated. If you regard it as time-division multiplex, the switching at the receiver needs synchronizing, so left always gets switched to left, and right to right. The receiver needs something to synchronize its subcarrier, or switching rate, according to which way you view it, with the corresponding action at the transmitter.

Could a small component of the switching frequency, or subcarrier, be transmitted along with the modulation, so a frequency-selective circuit in the receiver could filter it out and amplify it? The trouble with this idea is that this control frequency would be mixed up with most of the modulation (sideband) frequencies. The receiver might sometimes follow one of the sidebands representing a low audio-frequency difference component, and thus have the left and right channel trading places at an audio frequency!

PILOT FREQUENCY

To safeguard against this possibility the method adopted is to send a pilot frequency exactly half that of the subcarrier or switching frequency. Then all the receiver has to do is double it. A few figures will show how this avoids trouble.

The sum signal contains frequencies from the lowest audio (which does not matter, exactly) up to 15,000 cycles. The switching frequency is 38 kc. This is not transmitted. But its sidebands, due to difference information from a low frequency up to 15,000 cycles, results in fluctuating sidebands of from 23 to 53 kc. The half-frequency of 19 kc fits nicely in the slot between 15 kc, which is the top of the sum modulation, and 23 kc, which is the lower limit of the difference sidebands (Fig. 3-6).

Even if there were small-amplitude bits of frequencies above 15,000 cycles in both sum and difference channels, their magnitude will always

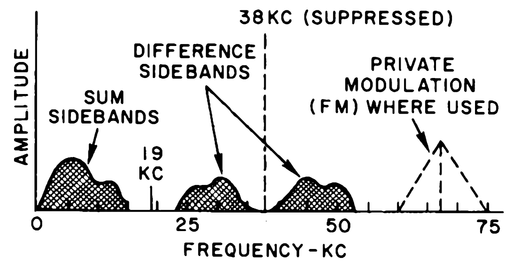


Fig. 3-6. The frequency spectrum of the modulation used for transmitting stereo by the new system.

be very small, and therefore not likely to interfere with proper separation of the 19 kc pilot in the way the sidebands nearer to 38 kc would, all the time. To put in the FCC standards: the pilot of 19 kc is transmitted so as to modulate the main carrier between 8% and 10%; the 38 kc subcarrier must be suppressed to less than 1%.

REASONS FOR CHOICE

Many people felt that the Crosby system, with its intent to provide maximum fidelity in every respect on stereo, with compatibility to monophonic reception, was the best choice. But experimental work with the Crosby system showed that while it might have the *potential* for the best system, the tuners actually tried did no better than with the system eventually accepted. In two important respects, the system accepted had advantages. Figure 3-7 shows the relative features here discussed.

Because the monophonic (sum) component of the Crosby transmission is 6 db below that on normal monophonic transmissions, reception without a stereo tuner is correspondingly degraded. In strong signal areas, this does not matter, but it does restrict the effective service radius of a station at its allocated operating power.

The accepted system reduces the monophonic component of a stereo transmission by less than 1 db, so that restriction of service area by the changeover is very much less. Against this, the stereo (difference) component is degraded more by the accepted system than it would have been by the Crosby system. However, this does not compensate for the advantage the other way. People outside the monophonic radius after the change will not be brought inside by trying to receive stereophonically instead of monophonically. If they cannot receive monophonically, they cannot receive it at all.

Admittedly ability to receive the best quality stereo may be confined to a slightly smaller circle with the accepted system than it would have been with the Crosby system, but this difference is not so great as the exclusion from service altogether would have been — in the opposite direction. Where signal strength is adequate for good stereo reception, the practical

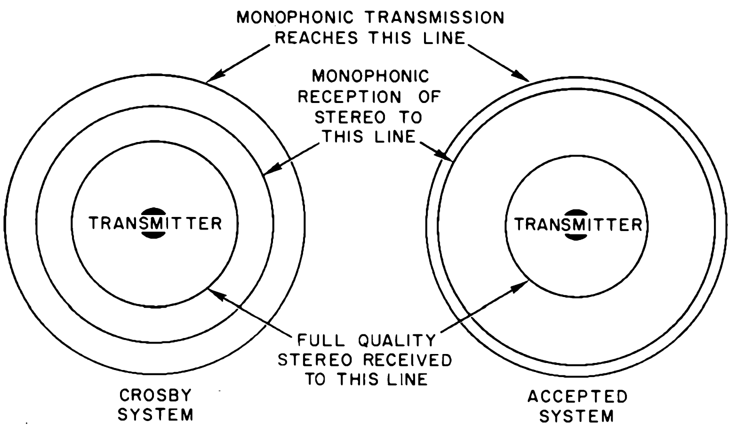


Fig. 3-7. Comparison of the service area provided by the two systems finally considered.

differences between reception by the two systems are so small as to be negligible, with the best equipment.

The accepted system has the advantage that passable quality reception can be had with relatively low-cost adapters or receivers.

PRIVACY OR PIRACY

The second important advantage of the accepted system is the compatibility with private service subcarrier use, where this is licensed. Because the two subcarriers use quite different forms of modulation, as well as different subcarrier frequencies, privacy of the privately subscribed one is maintained. This does not mean they cannot *interfere* with one another, but it does mean the stereo adapter cannot be used for "pirating" the private service without paying the proper subscription.

If the private subcarrier frequencies get into the stereo adapter, they will cause unintelligible noise. So the frequencies do have to be filtered out. But removal of the filters, or failure to filter these frequencies out, does not enable a stereo adapter to "receive" what is legally a private transmission.

SIMILARITIES WITH DISC

There is a strong similarity between the new stereo multiplex radio system and the 45/45 system used for disc recording of stereo. The basic reference in the disc system applies left and right at opposing 45° angles. But it can also be regarded as lateral recording of sum with vertical recording of difference.

In the new stereo multiplex system, we can regard the basic concept as being time-division multiplex, accepting left and right program directly. Alternatively, regarding the same thing as suppressed-carrier multiplex, the direct modulation is sum and the suppressed-carrier modulation is difference. This analogy is useful in understanding some of the distortions that can occur in this kind of system.

DIFFERENT METHODS OF DOING IT

Because the theory of the system can be explained in various ways — at least two — the practical program signal can be generated in various ways, too. At least three have been proposed:

1. A time-division multiplex, employing an electronic switch synchronized by the pilot frequency, followed by a low-pass filter that "cleans off" the higher-frequency components (sidebands of 114 kc and higher odd multiples of 38 kc).
2. Combination of the 38 kc switching or subcarrier frequency with left and right program, in opposite phases, in a pair of multiplier-type mixers. The opposing phase results in zero 38 kc output when left and right are identical.
3. Matrixing from left to right, to get the sum and difference, then using the difference in a balanced modulator to get suppressed-carrier modulation of the 38 kc subcarrier.

At the receiving end, we have a similar choice of procedures to obtain the stereo program from the received signal. Notice that the method used does not change the nature of the final signal transmitted or received in any way. Although we might refer to three (or maybe more — others may yet be invented) systems of transmission or reception, the basic system — the form of the modulation actually transmitted — is the same in all of them, and satisfies the specification laid down by the FCC.

4 typical receiver circuits ■

The fact that the accepted system could be described in more than one way, promised plenty of variety in approach to receiver or adapter design. This has proved true, beyond original expectations. Circuits can range from the almost ridiculously simple to the extremely complex, with similar range in possible performance — but not necessarily any connection!

BASIC CIRCUITS

There seem to be three basic forms of circuit, each of which may have any number of more detailed variations. Beside the beam-switching and matrixing forms, there is an envelope-demodulation form, which may eventually prove to be the most popular. If we take the basic transmitted signal, Fig. 4-1 part (a), and reinsert the subcarrier in correct phase and at sufficient magnitude, the top of the envelope is the left signal, while the bottom is the right signal, Fig. 4-1 part (b).

By using relatively simple demodulation that follows one edge of this envelope, the left and right outputs can be retrieved separately.

Whichever method is used, the 19-kc pilot signal needs isolating, and the subcarrier must be regenerated in correct phase, synchronized by this pilot. A problem some designers have encountered is avoiding spurious control because the 19 kc circuit picks up second-harmonic distortion of any 9.5 kc program component present (or nearby frequency).

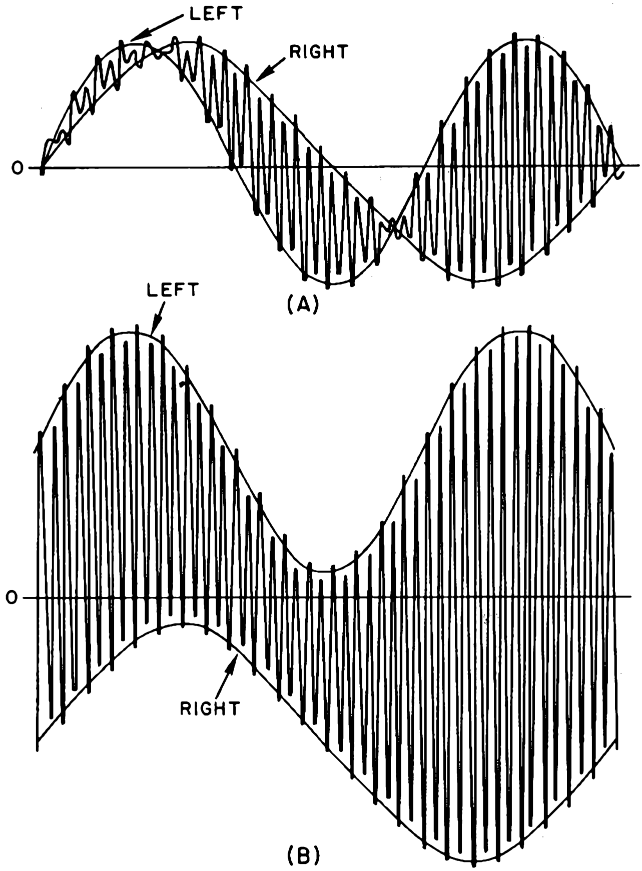


Fig. 4-1. The principle of envelope detection: (a) a composite signal, as received at the output of an FM tuner; (b) with 38-kc regenerated subcarrier added in the correct phase, the top of the envelope is left and the bottom right. The irregular up-and-down effect of alternate peaks is due to the 19-kc pilot, which can be removed by filtering.

Two main approaches are used to minimize this: (1) use only very low-distortion circuits to amplify before the 19-kc signal is separated from the remaining audio (50 to 75,000 cycles); (2) position the first stage of separation (a 19-kc tuned circuit) before any amplification is applied to the main signal. A third method, where the distortion is not adequately low, has been to use a 9.5-kc rejection filter at the input to the amplifier feeding the 19-kc filter.

Regeneration of the subcarrier can use at least three methods: (1)

frequency doubling of the pilot with further amplification (Fig. 4-2); (2) synchronized oscillator at 19 kc, followed by a frequency doubler (Fig. 4-3); and (3) synchronized oscillator at 38 kc, synchronized by 19 kc, on alternate cycles (Fig. 4-4).

The matrixing method involves separation of the L + R signal (from 50 to 15,000 cycles) from the L — R subcarrier modulation (from 23,000 to 53,000 cycles), before the subcarrier is reinserted in the latter. This requires a low-pass filter at 15 kc, and a high-pass filter at 23 kc

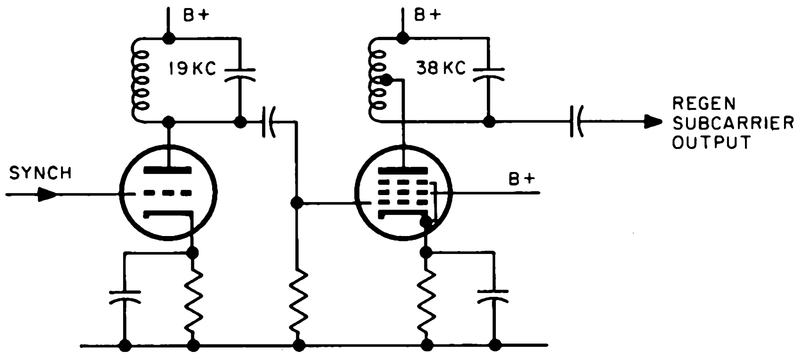


Fig. 4-2. Simple 19-kc filter and frequency-doubler circuit for regenerating the subcarrier.

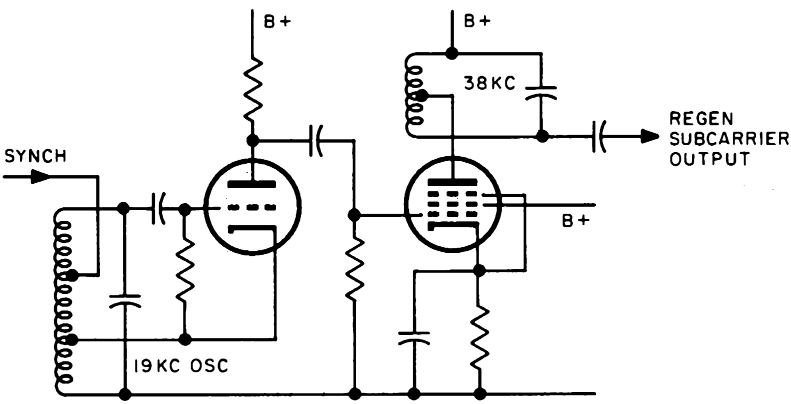


Fig. 4-3. Circuit using 19-kc oscillator synchronized by incoming pilot, followed by doubler.

(which will usually be a 23-53 kc bandpass filter, to reject any SCA* subcarrier as well).

For the output of the L + R and L — R to matrix correctly to L and R, they must maintain both their correct amplitude and phase, as at the

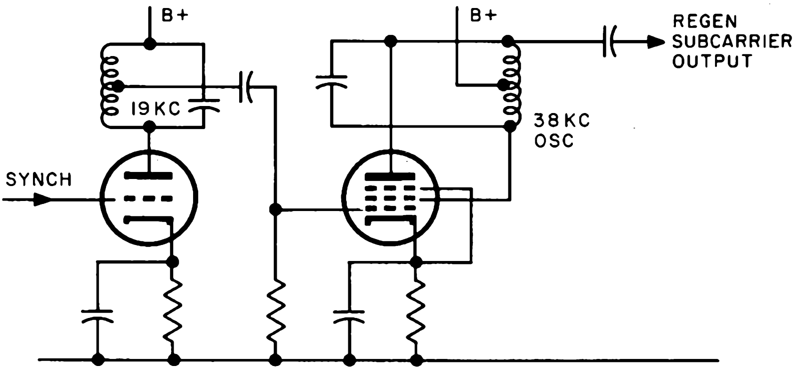


Fig. 4-4. Circuit using 38-kc oscillator synchronized by 19-kc pilot, after isolation by filter.

transmitter. This requires that both demodulator (L — R) and filters have the same time delay in both circuits, *at all frequencies.*

In a high-pass or bandpass filter, a uniform time delay is impossible. There are two ways to circumvent this difficulty. In the first, a compromise adjustment is used. The subcarrier phase is reinserted at a phase that minimizes distortion and allows maximum separation at the high frequencies — say, from 8000 to 15,000 cycles. The matrix is then slightly offset to yield maximum separation at the lower frequencies, because the phase adjustment disturbs the balance.

Although careful selection and alignment can achieve a good compromise, it *is always* a compromise. Theoretically, distortion and separation can never be perfect throughout the audio range. The second method is better from this standpoint. It eliminates the need for a high-pass or bandpass filter in the L — R channel.

By returning the detector load circuits to the output of the low-pass filter, instead of ground, no high-pass filter is needed. If the detection delay, plus any delay in the filter used to reject the SCA subcarrier is made equal to the delay in the low-pass filter, and losses in both are equalized, perfect matrixing and optimum separation at all frequencies is assured (Fig. 4-5) provided the subcarrier phasing is correct.

**Subsidiary Communications Authority.*

Every circuit must take some steps to reject the SCA channel. There are two, almost opposite, approaches to this. The first was the incorporation into low-pass filter action of deliberate 67 kc rejection (Fig. 4-6).

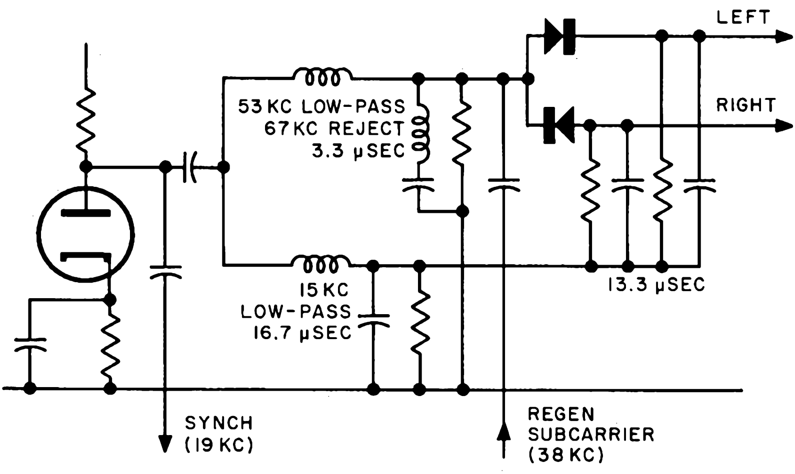


Fig. 4-5. Matrixing circuit that avoids the use of a bandpass or high-pass filter.

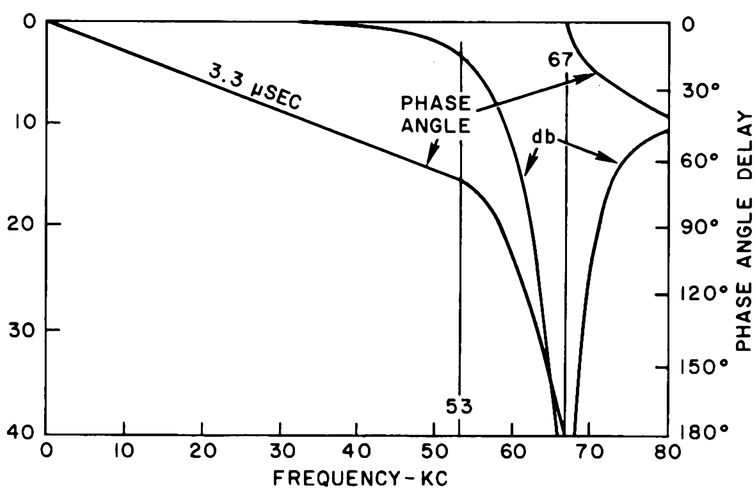


Fig. 4-6. One way to remove the SCA subcarrier is to use a phase-linear filter with 53-kc low-pass and 67-kc reject characteristics; circuit configuration is in upper part of Fig. 4-5, response is shown here.

This will considerably minimize interference, and is certainly adequate if SCA and stereo are not transmitted simultaneously — only at separate times.

The newer approach takes advantage of the simple fact that two different kinds of modulation are used. If the 38-kc subcarrier is correctly inserted, and the 67-kc subcarrier in the composite has no amplitude modulation, then by the simple expedient of leaving out all filters and using very linear detection, the SCA subcarrier has no effect. This approach seems to fit best with the envelope-demodulation method, which requires no filters for any purposes of its own.

Whichever kind of circuit is used, de-emphasis is needed somewhere. It is also advisable, if a tuner or adapter is ever to feed a tape recorder, to have filters at the output to remove the 38-kc subcarrier components so they cannot beat with a bias oscillator and cause "birdies".

Sometimes these functions are combined, because the de-emphasis requirement is already in that direction. An asymmetrical twin-T does both jobs admirably without requiring close-tolerance components.

REFINEMENTS

Different circuits add certain refinements. It has been found that a high input-noise level affects the subcarrier modulation to a greater extent

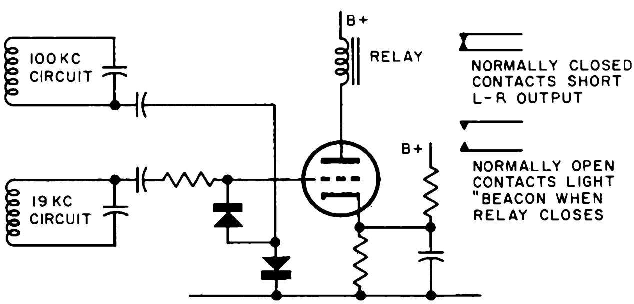


Fig. 4-7. Basic circuit of Fisher Stereo Beacon.

than the L + R modulation, because it occupies twice the bandwidth. Putting a noise filter in the circuit so that it reduces L — R noise only will improve the signal-to-noise ratio on weak reception, at a loss in stereo effect, but not overall quality.

Absence of stereo transmission — no pilot or subcarrier — leaves the adapter "straining" to find what is not there. The selective circuits will amplify any noise components in the general range of 19 kc, and produce a noise output in the L — R channel that is dominantly low-frequency, being near to the subcarrier frequency. For this reason, it is well to "kill" the L — R channel when there is no stereo, or if the reception is too weak to allow proper stereo to be reconstituted. This has been done in some circuits by a variety of ways.

The Fisher unit was the first circuit to do this, with a relay called the *Stereo Beacon* (Fig. 4-7). This is activated — or not — according to a combination of two voltages: the first is a rectified output from the 19-kc filter, which provides a positive voltage; the second is a rectified output of a band of frequencies around 100 kc (above the program range, both stereo and SCA) — which provides a negative voltage. These two voltages are combined and applied to the grid of a tube normally cut off.

Only when there is 19 kc without high-frequency noise does the relay operate, to allow the stereo to operate. If there is no 19 kc, or if the 19 kc is accompanied by high-frequency noise — as when tuning between stations, or when reception is too weak — the relay drops out, shorting the output from the L — R section.

Other circuits have been evolved with the object of achieving similar operation without the need for a mechanical relay.

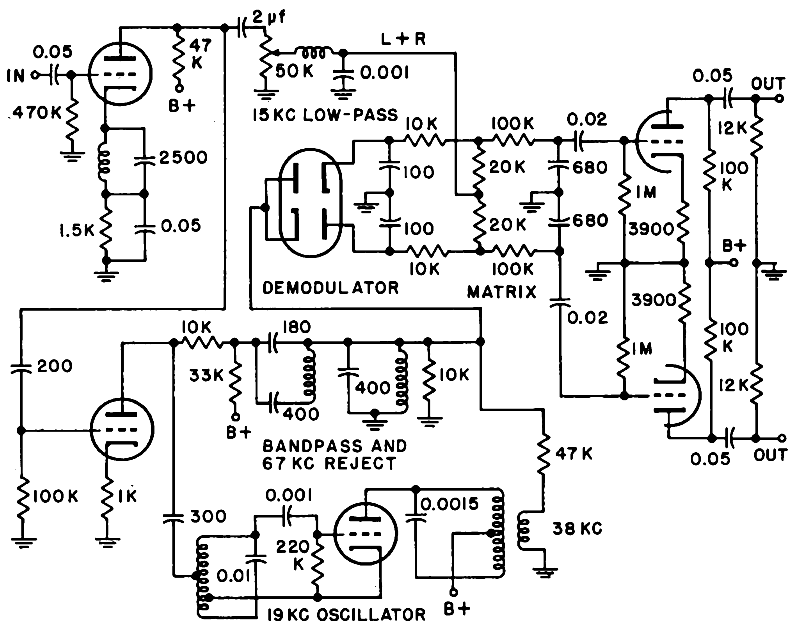


Fig. 4-8. The Knight multiplex adapter uses a full complement of filters and matrixing; oscillator is at 19 kc, with doubling in plate circuit. *Courtesy Knight Electronics Corp.*

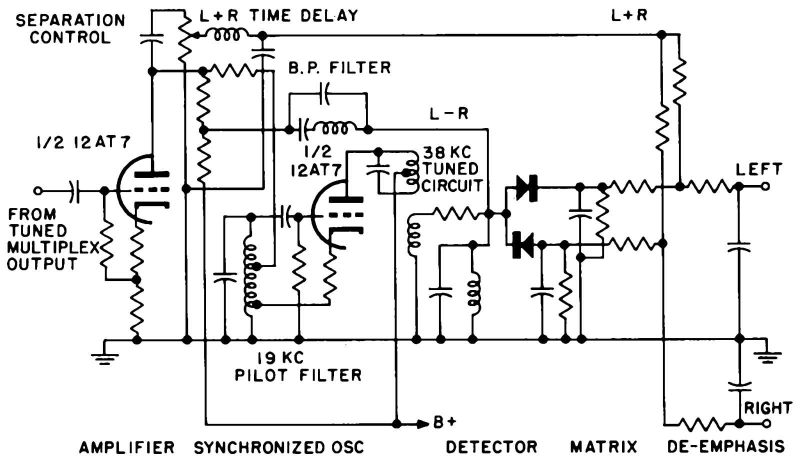


Fig. 4-9. One of the General Electric suggested schematics, using synchronized 19-kc oscillator and matrixing, with 15-kc low-pass filter, 23 to 53-kc bandpass with 67-kc reject filter. *Courtesy General Electric Company.*

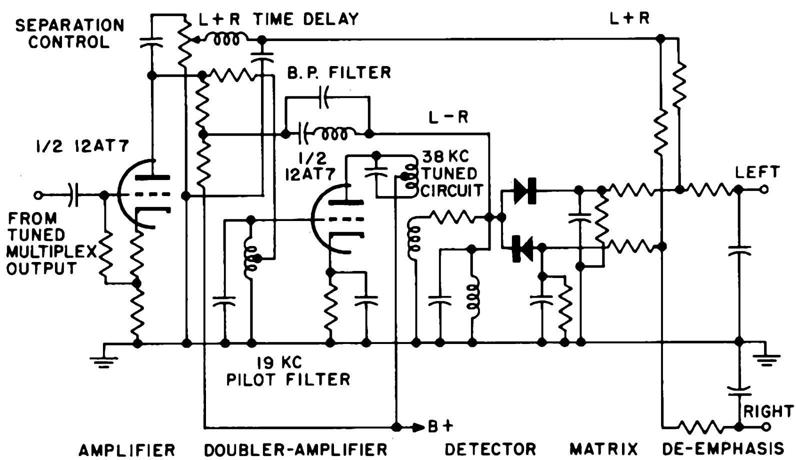


Fig. 4-10. A slightly different GE circuit using a doubler instead of a synchronized oscillator. *Courtesy General Electric Company.*

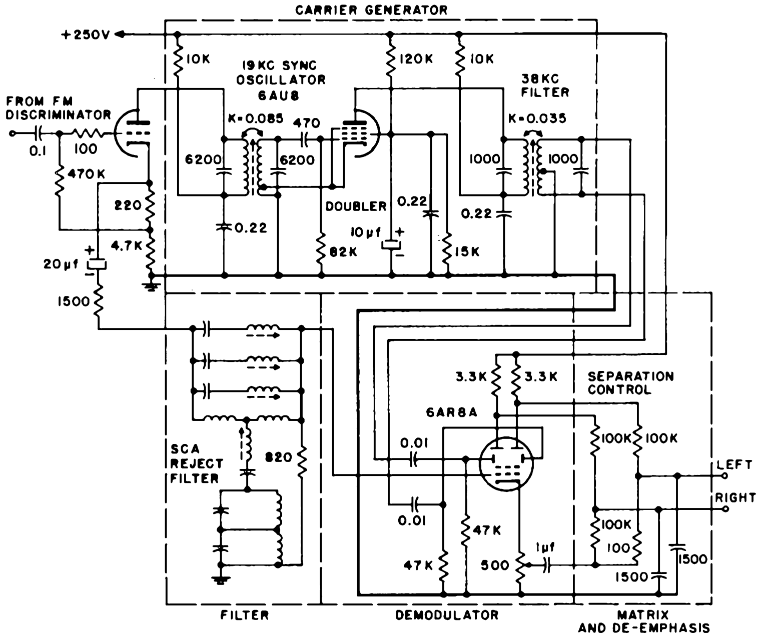


Fig. 4-11. The original suggested circuit from Zenith, using a beam-switching tube for demodulator. Later circuits use a simpler filter than the Bode type shown here, but are otherwise similar. *Zenith Radio Corp.*

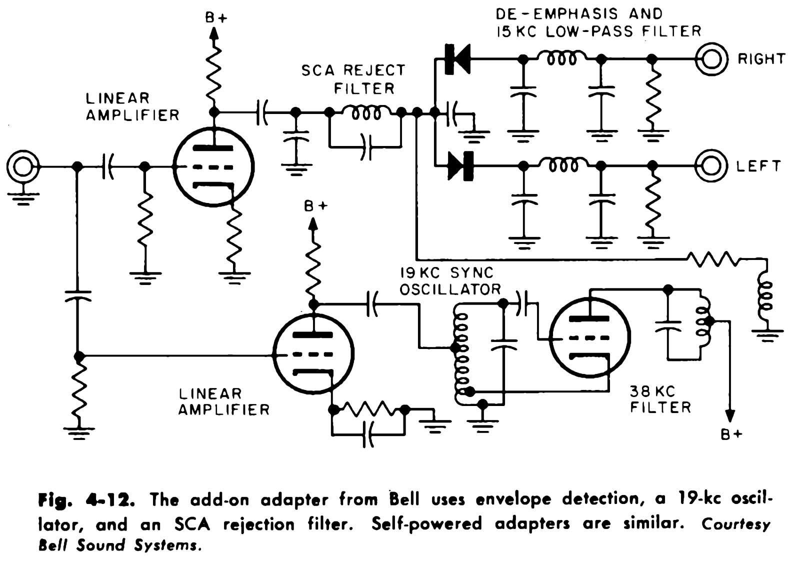


Fig. 4-12. The add-on adapter from Bell uses envelope detection, a 19-kc oscillator, and an SCA rejection filter. Self-powered adapters are similar. *Courtesy Bell Sound Systems.*

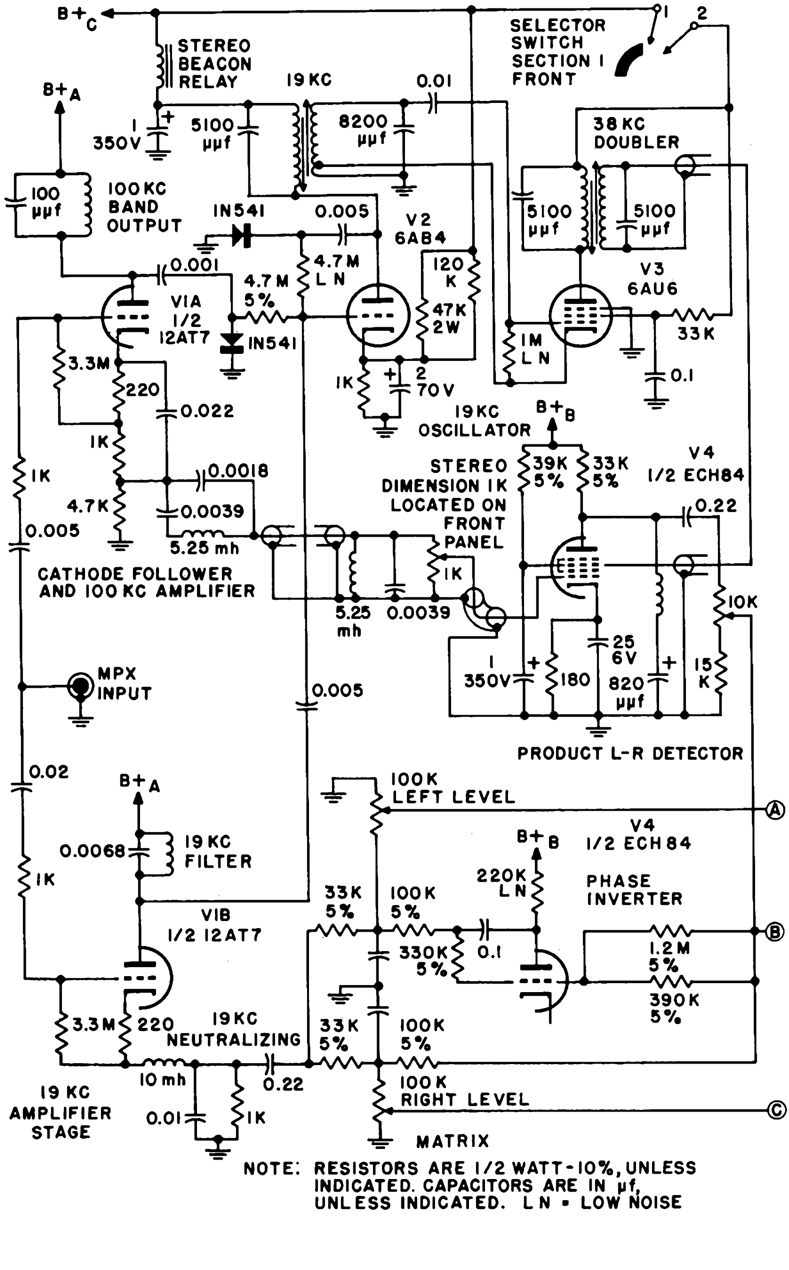
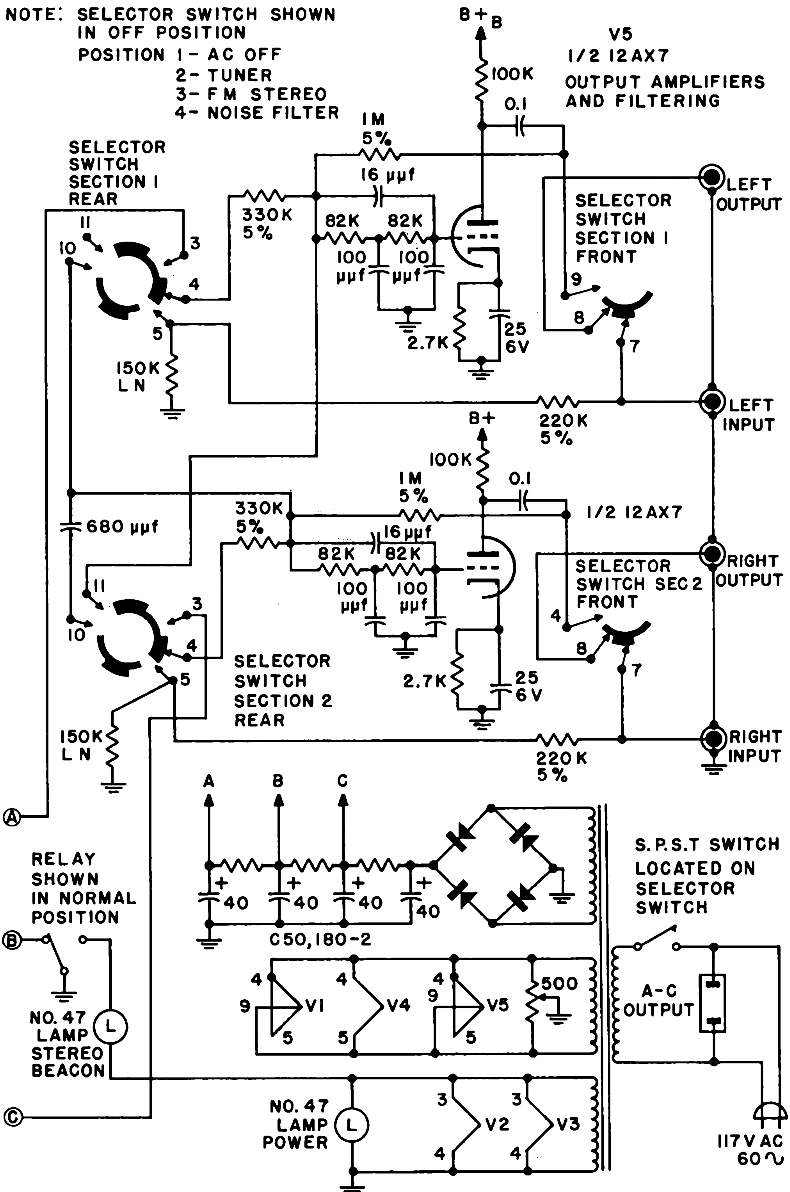


Fig. 4-13. The Fisher adapter introduces some noteworthy features, most notably the product detector and the circuitry associated with the *Stereo Beacon*, which



not only indicates when stereo is being received, but changes the connection from mono to stereo. *Courtesy Fisher Radio Corp.*

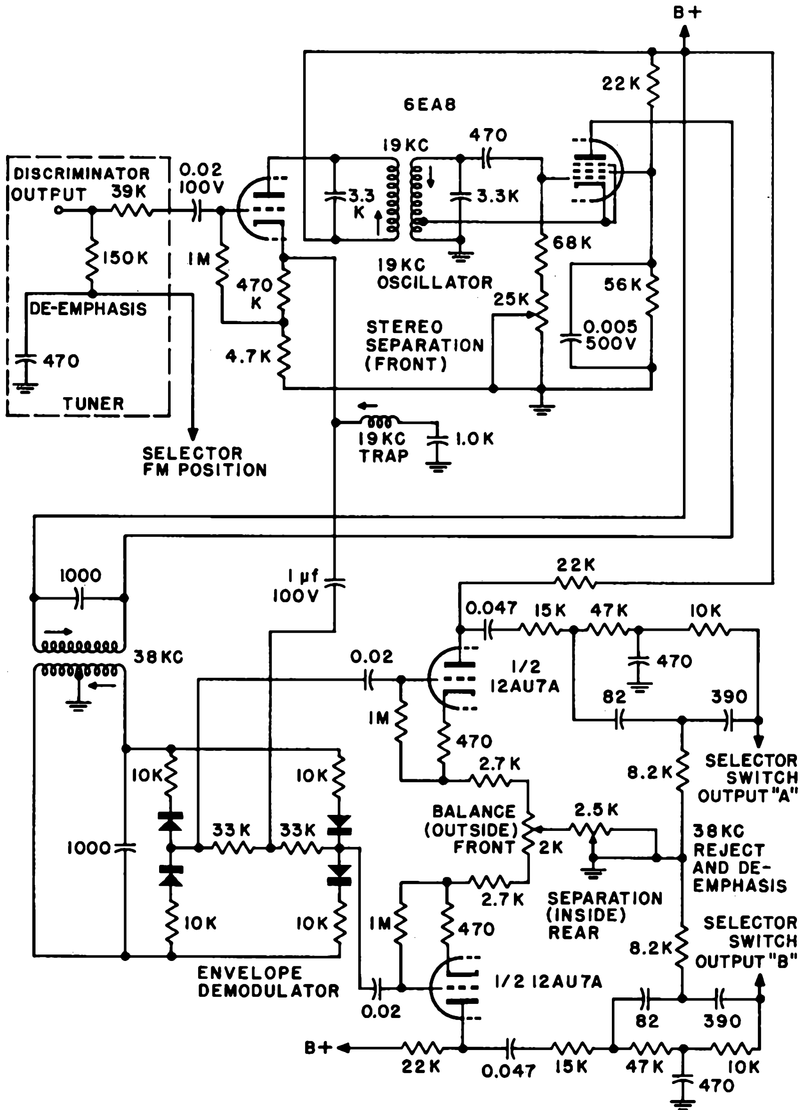


Fig. 4-14. The Sherwood adapter uses a synchronized 19-kc oscillator and frequency doubler, followed by bridge-type envelope detection, with no 67-kc filter at all. It relies on linearity of amplitude detection to eliminate the frequency-modulated 67-kc carrier, when present. Output uses combined twin-T de-emphasis and 38-kc reject filters. The same circuit is used for the adapter and for the multiplex portion of the new Sherwood tuners. *Courtesy Sherwood Electronic Labs. Inc.*

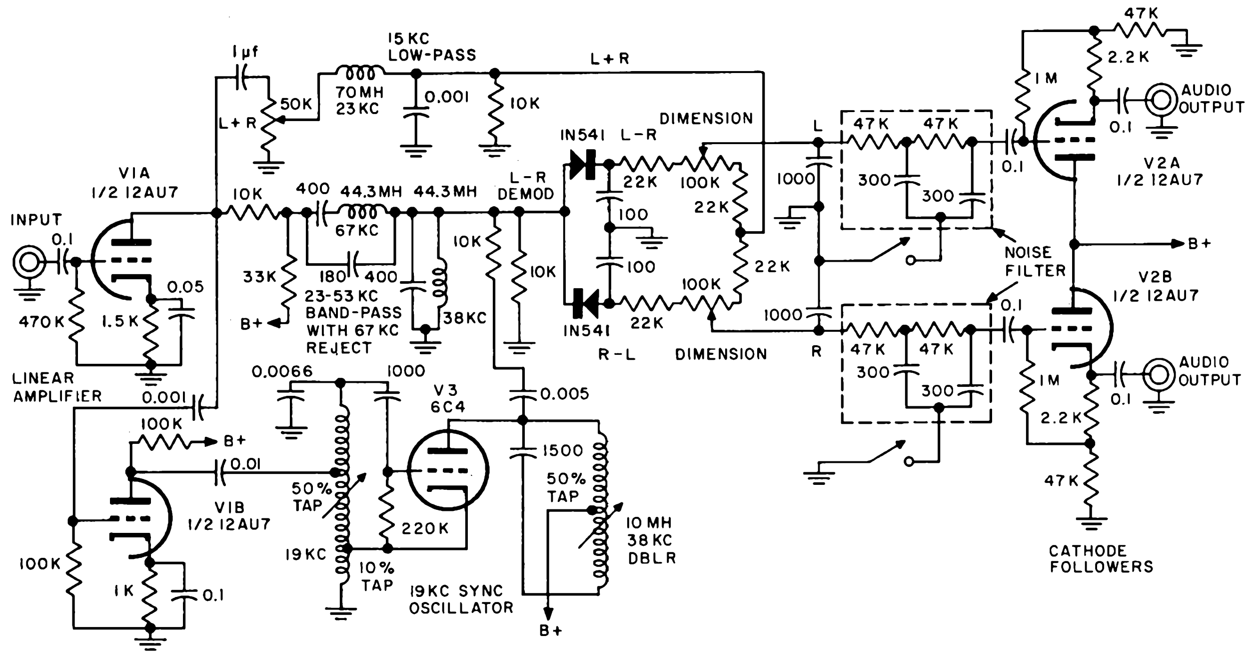


Fig. 4-15. The Crosby adapter is similar to the circuit of Fig. 4-9, except for the method of coupling the regenerated subcarrier into the circuit, and the additional switchable noise filters and cathode followers. *Courtesy Crosby Electronics, Inc.*

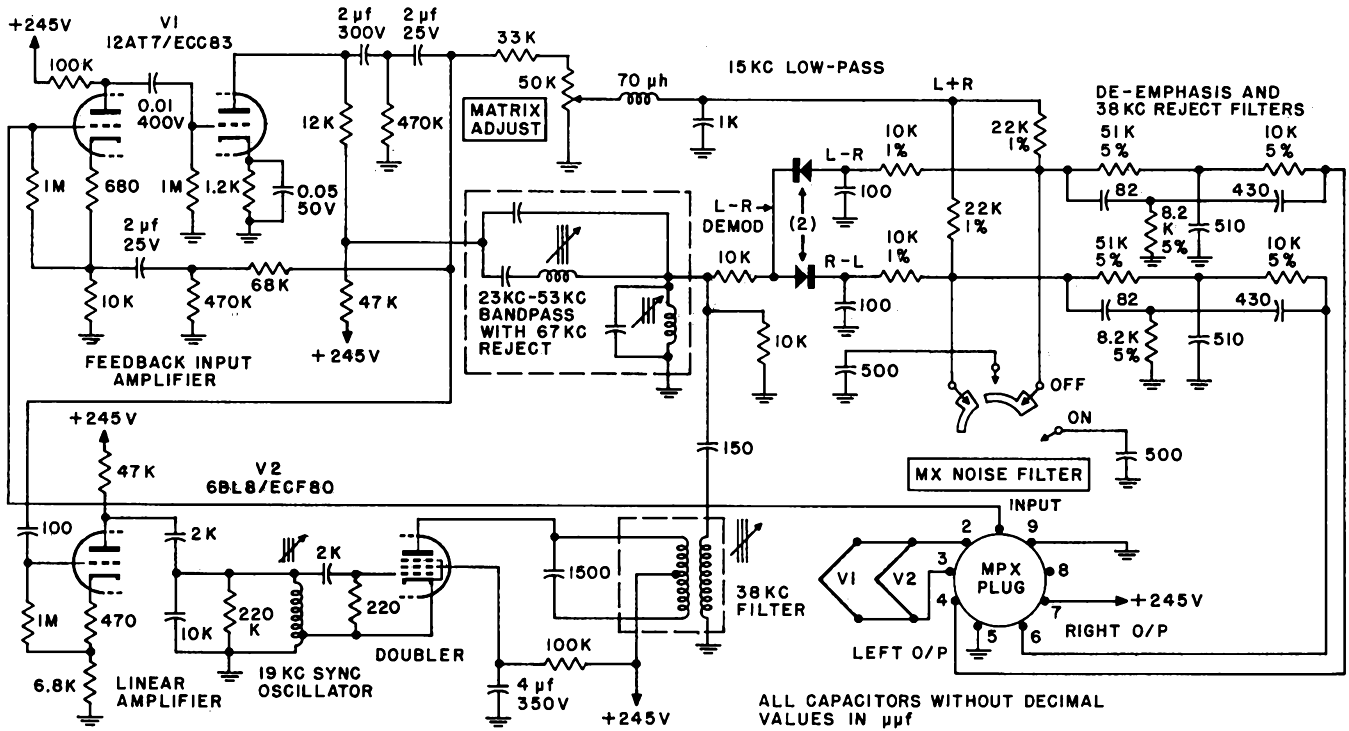


Fig. 4-16. The Harman-Kardon adapter uses a two-stage-with-feedback amplifier at the input, and the asymmetrical twin-T filters at the output. Otherwise it is similar to the circuit of Fig. 4-9. *Courtesy Harman-Kardon, Inc.*

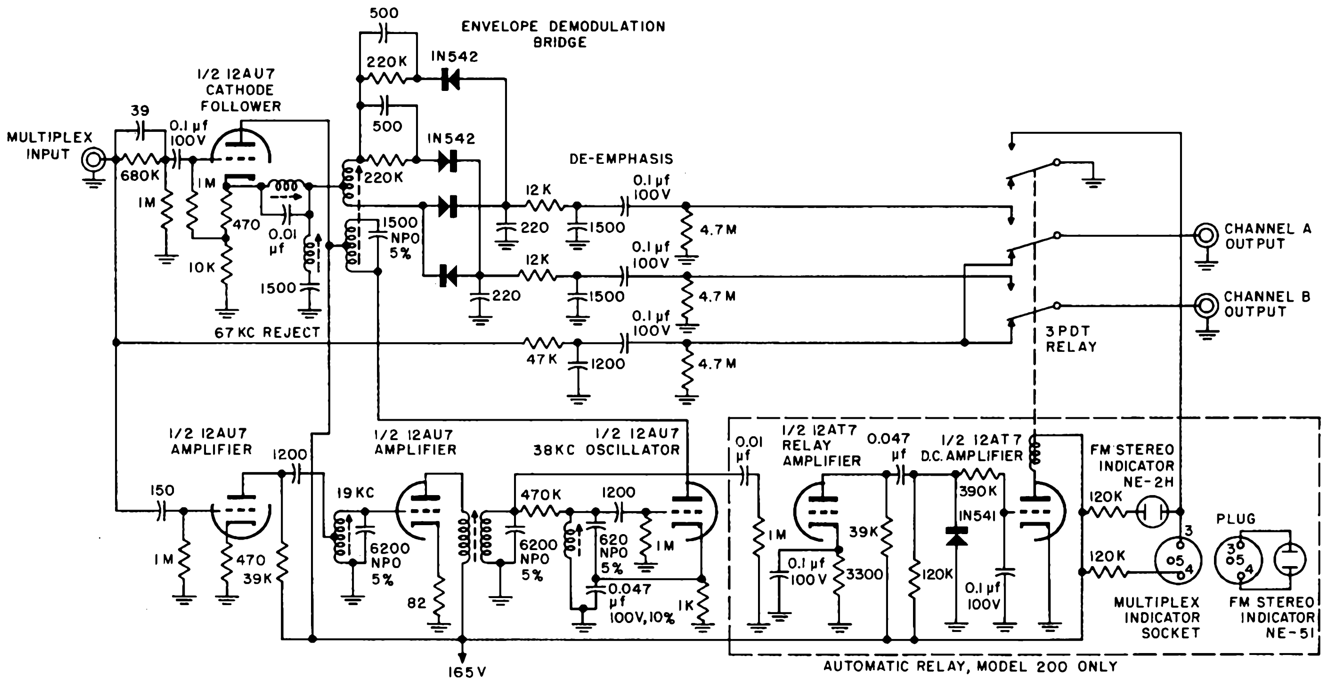


Fig. 4-17. Pilot Radio adapters use envelope detection with special transformer coupling to reinsert the subcarrier, and a 38-kc synchon-ized oscillator. A double-tuned 67-kc reject filter is employed. The Model 200, shown here, incorporates automatic switching when stereo is received; the Model 100 omits the portion enclosed in dashed lines and uses a slide switch instead of the relay contacts. *Courtesy Pilot Radio Corp.*

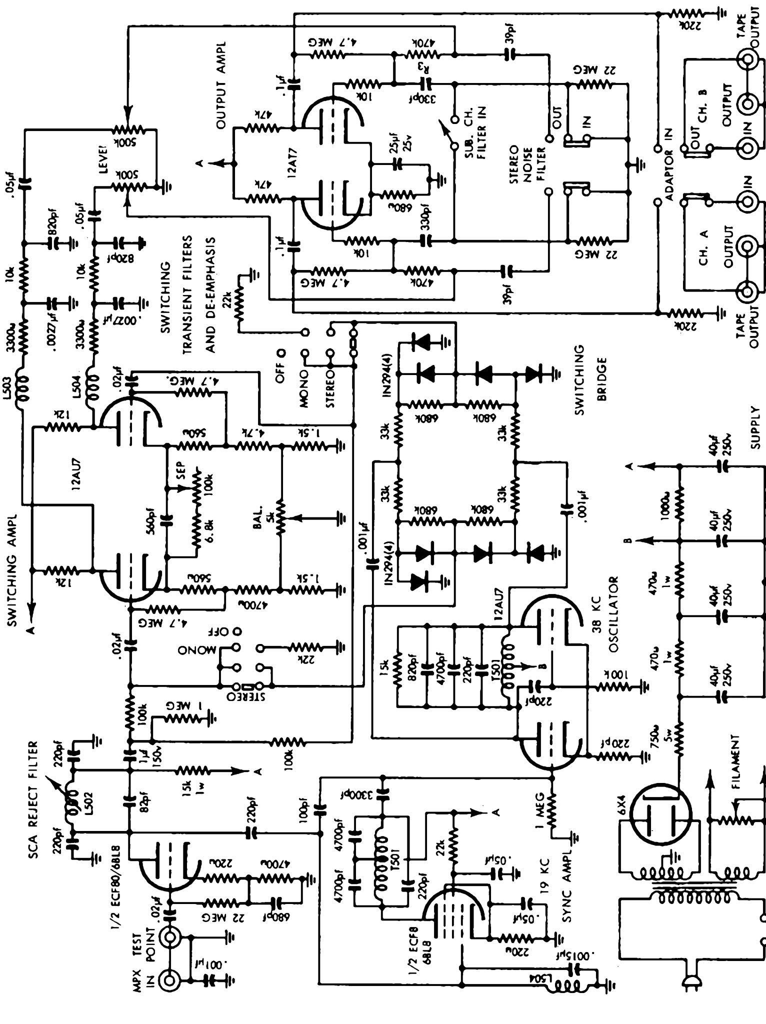


Fig. 4-18. The H.H. Scott type 335 adapter uses a method of operation basically similar to that of Fig. 4-11, but without a beam-switching tube; instead it uses a bridge employing eight switching diodes, whose function it is to short (ac) to ground the grids of the switching amplifier during alternate half cycles of the subcarrier. This circuit uses a 38-kc oscillator for subcarrier regeneration, with more complete filtering to isolate the 19-kc pilot than most. *Courtesy AUDIO and H.H. Scott.*

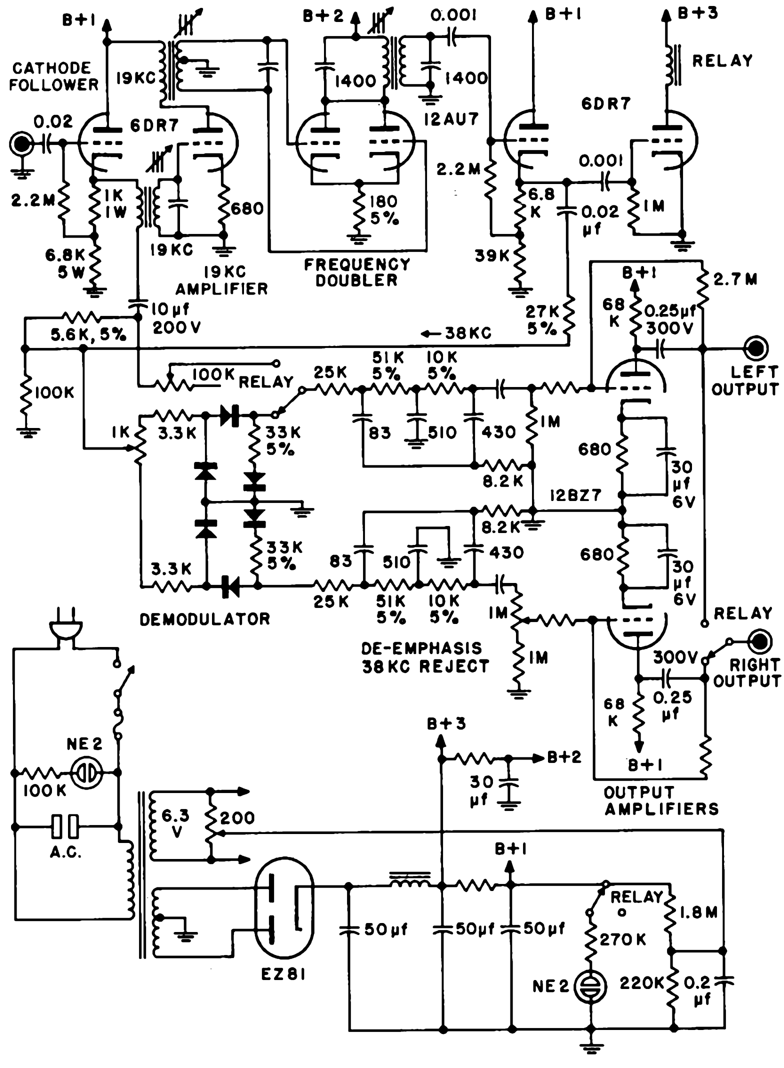


Fig. 4-19. The Lafayette LT-30 adapter uses some variations: the cathode follower feeds the 19-kc selection circuit and the composite audio with series separation, and is followed by twin-triode frequency doubling, producing close to perfect 38-kc regeneration; envelope detection with twin-T filters and a low-gain output stage complete the active part of the circuit. It also includes a relay to bypass the envelope detector in the absence of a 19-kc pilot, with attenuation equalization. *Courtesy Lafayette Radio Electronics Corp.*

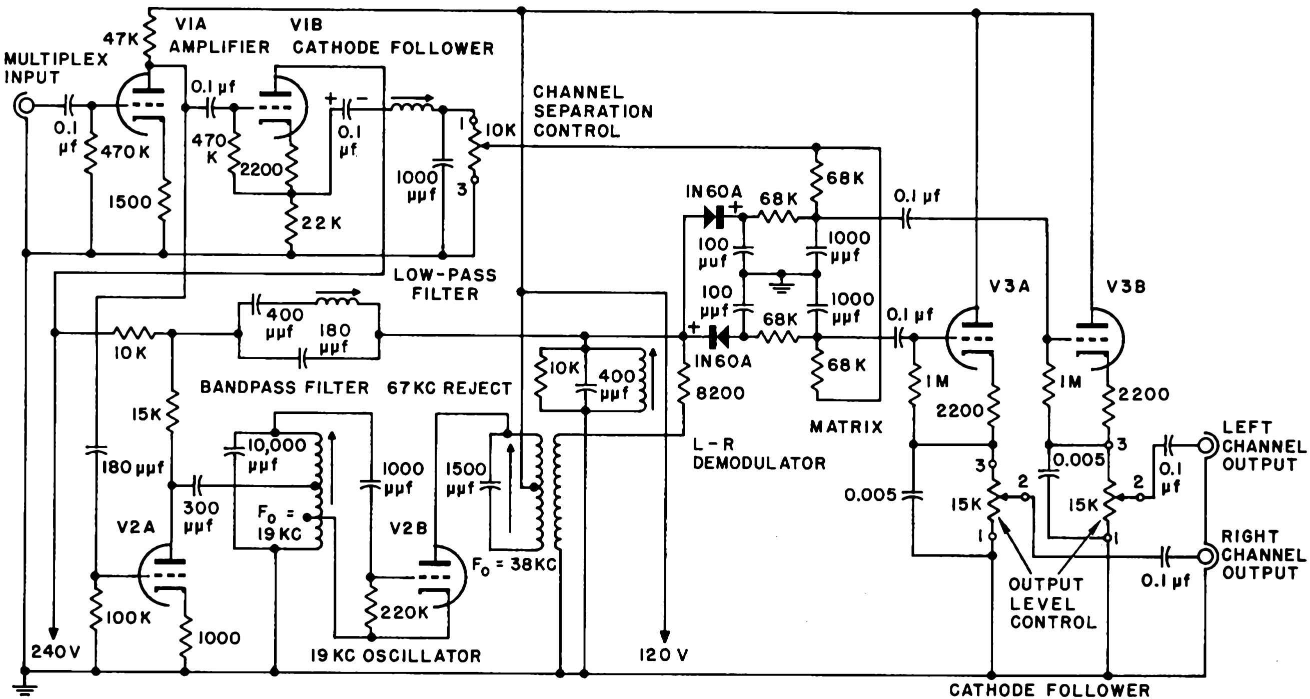


Fig. 4-20. The Heath adapter, AC-11, uses an amplifier stage for composite, separate further amplification for 19-kc pilot, 19-kc oscillator and doubler. It uses matrixing with full filtering, and cathode-follower outputs. *Courtesy Heath Company.*

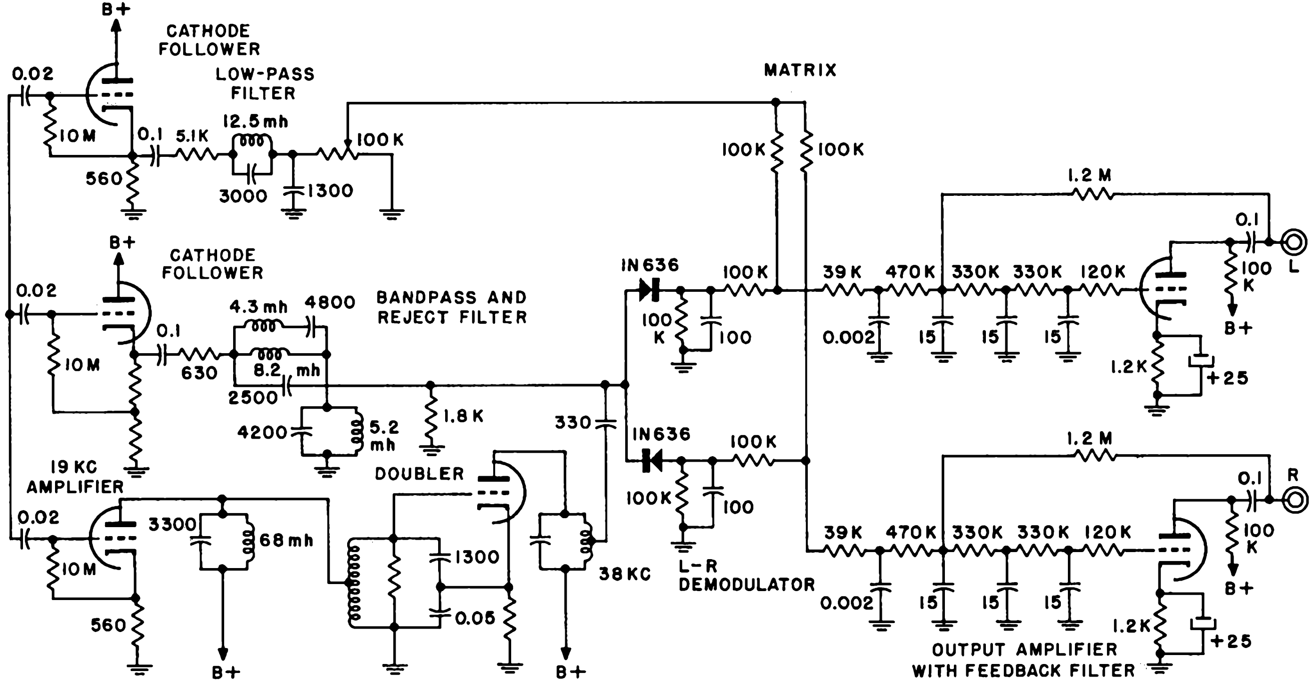
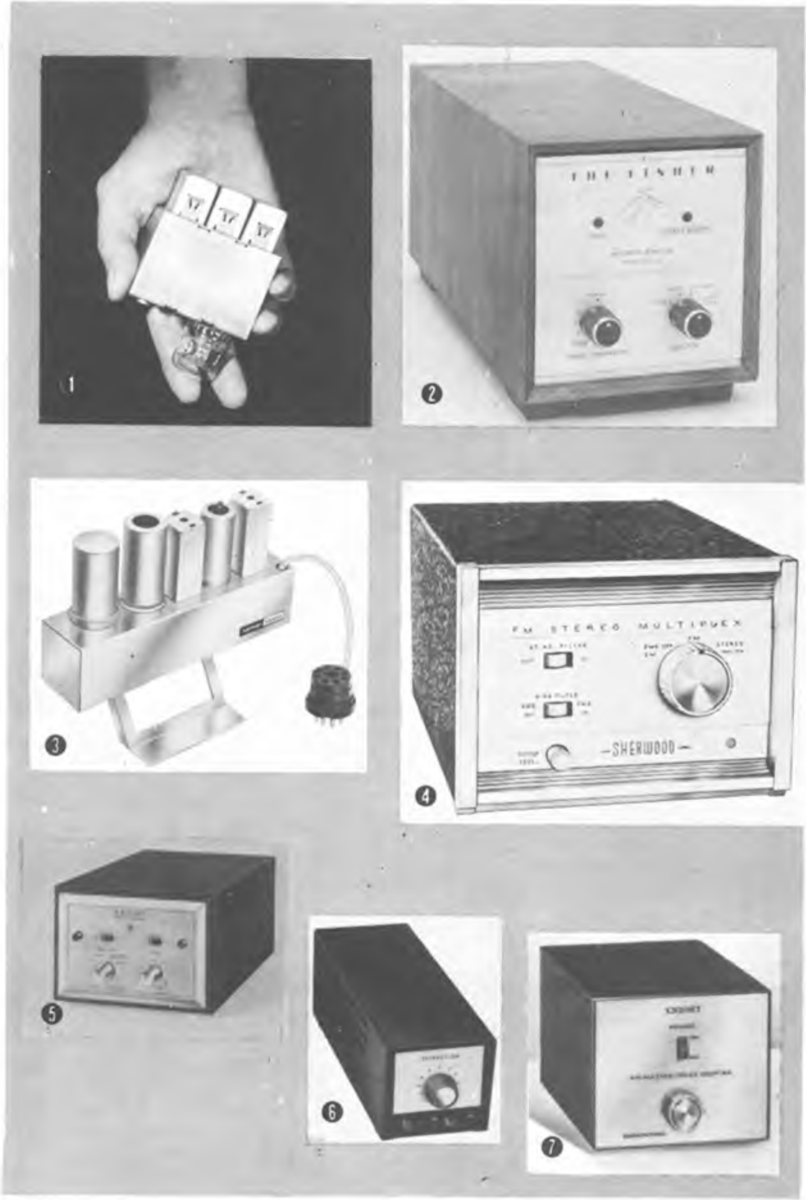


Fig. 4-21. The Pacotronics adapter uses separate cathode followers for L + R and L — R sidebands, with amplifier and doubler stage for subcarrier regeneration. It uses matrixing with full filtering, and output stages with feedback filtering. *Courtesy PACO Electronics Co., Inc.*



Typical Adapters: (1) Bell MX-1; (2) Fisher MPX-100; (3) Harman-Kardon MX-600; (4) Sherwood separate adapter; (5) H.H. Scott 335; (6) Heathkit AC-11; (7) Knight KN-MX.

5 installation and conversion problems

The overall installation or hook-up picture is the same with any adapter and most units will come with a schematic to show how to hook them up, either in a new system, or to convert an existing monophonic system to stereo. The basic procedure consists of breaking the audio connection at the tuner output, inserting the adapter, and using two channels of amplification and loudspeakers from the output of the adapter (Fig. 5-1).

Some of the tuners issued within the last year or so before multiplex was officially approved, were provided with a multiplex output; in this case, the input to the adapter just connects to this point, and there should be no need to get inside the tuner at all. Where no such provision is made in the tuner, the take-off point for the adapter must (a) have no de-emphasis, and (b) be such that connection of the adapter does not cause distortion.

Connecting the adapter directly to the detector load (Fig. 5-2) should avoid the de-emphasis, but in some circuits there will still be some roll-off in frequency above 15 kc, and below 53 kc, if the de-emphasis R and C are left connected; so it will be well to disconnect the de-emphasis altogether. Even then, the detector circuit may roll-off itself too soon, or the addition of the adapter may cause distortion, in which case the tuner circuit values may need changing, or the tuner may need realigning, which we discuss in Chapters 6 and 7.

While simple installation, following the directions issued with some of the less expensive adapters, may result in sound the owner hopes is

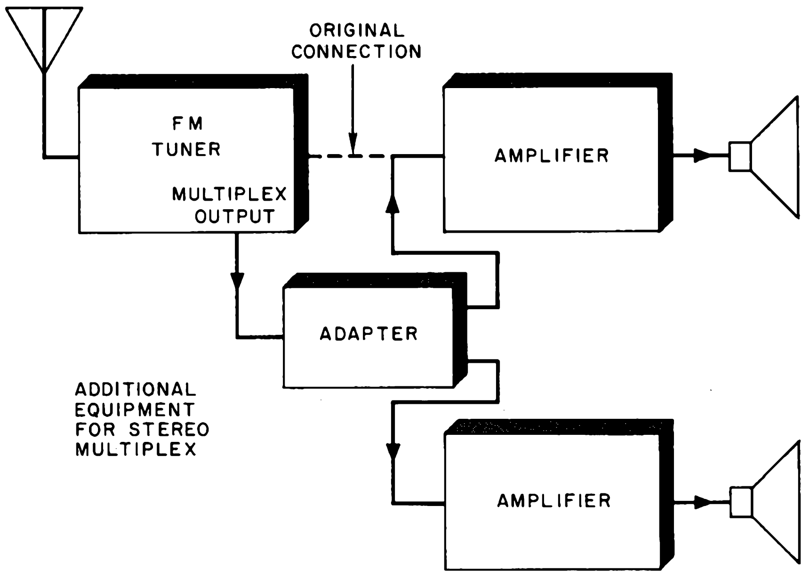


Fig. 5-1. Changes needed to convert to stereo multiplex; if system already has stereo, only the adapter is needed in addition to existing equipment.

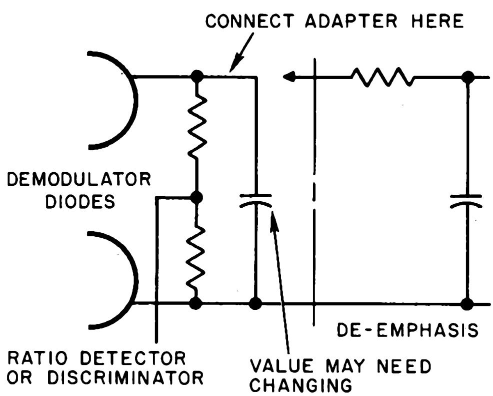


Fig. 5-2. How to connect an adapter to a tuner when no multiplex output is provided in the tuner circuit.

stereo, it may be no nearer the real thing than some things we see on a TV screen are to a perfect picture!

ANTENNA INSTALLATION

Just as poor antenna installation can cause "ghosts" in TV picture reception, it can also spoil stereo reception of FM. With monophonic FM, although deviation up to the same frequency — 75 kc either side of the carrier — is received, the modulating frequencies fall off at a much lower frequency, usually 15 kc. With stereo, the range of frequencies extends up to 53 kc. If there is an SCA transmission (for privately subscribed background music) on the same station, there are also frequencies in the range from 60 to 75 kc.

Some of the ghosts that cause multiple-image reflections on a TV screen can also upset the sideband pattern in an FM transmission, resulting in distortion. Very few, if any of these distortions would appear in demodulated frequencies below 15 kc in the range used for monophonic transmission. They may, however, appear in the sidebands of the subcarrier frequency; in the SCA frequencies where these are used (but this will not directly affect stereo); and as interference between the frequencies

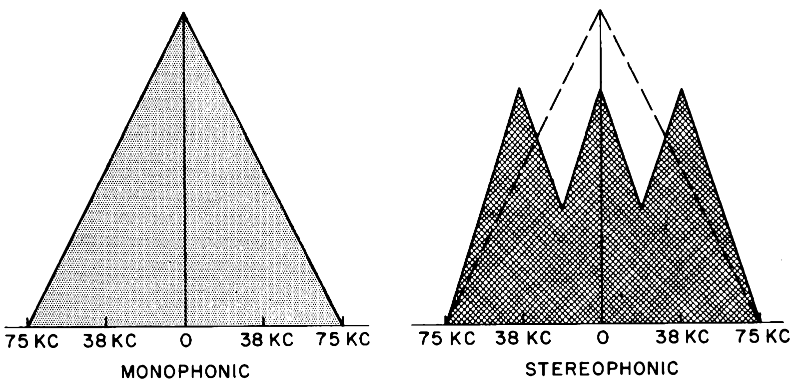


Fig. 5-3. Change in distribution of energy in average FM-transmission sidebands from monophonic to stereophonic.

used by the stereo and SCA subcarriers, which will show up as distortion on stereo.

All in all, there is good reason to avoid ghost reflections in FM reception when you want good stereo. Unfortunately they are not as easy to eliminate as on TV — you cannot watch them as you rotate an antenna. But the method is the same; you do get rid of the trouble by orienting the antenna, and you use an antenna that is good enough to eliminate reflections, properly matched to the imput circuit of the tuner. It must also be placed so that it will do so effectively.

To facilitate elimination of reflections, if the adapter has a point at which the sum (mono) is separated from pilot and subcarrier sidebands, shorting, or disconnecting the main (sum) channel and listening to or looking at (on an oscilloscope) the stereo output, will show up the reflections much more clearly. This is a good reason for preferring the matrix-type receiver or adapter circuit.

The difficulty is that any distortion received may be due to either ghosts or wrong phasing of the regenerated subcarrier. While adjustment of either will produce a null, or minimum distortion point, this is not a guarantee that true minimum has been achieved.

For example, distortion due to a ghost or reflected signal, may be minimized by adjustment of the phasing control on the reinserted carrier (how ever this is achieved in individual models), by what is really an incorrect phase setting. The distortion will be less at this incorrect setting than it would be at the correct setting, while the ghost is there, but it would be even lower if the ghost were eliminated.

Similarly, it is possible that maneuvering the antenna orientation will produce a minimum distortion when the phase adjustment is wrong, but this must be by choosing a deliberate ghost, so to speak, with much more distortion than there should be, were everything correctly aligned.

MATCHING ADAPTER TO TUNER

This is a temporary problem, but a very important one for the first year or two multiplex is operating. Later, the problem will disappear because adapters will no longer be made, only complete tuners with the multiplex features built-in. Meanwhile we have the problem of converting existing systems to stereo by the addition of an adapter.

As with any matching problem, the basic concepts are simple; both levels and impedances need to be right. But the troubles that incorrect matching can cause are in some instances a little more difficult to eliminate with this one.

The impedance of the tuner needs to be such that loading it with the adapter does not produce any kind of distortion. Such distortion can happen for a variety of reasons, and take a variety of forms.

Most tuners are designed to handle the full 75 kc deviation, but only with modulation frequencies up to 15 kc. This means that the amplitudes of sidebands up near the 75 kc limits are very much less than those for sidebands below 15 kc. When stereo comes in, these proportions are changed (Fig. 5-3). Sidebands in the region of the subcarrier frequency (38 kc) can be as large as those below 15 kc.

First question is whether the tuner can handle this change in relative amplitude distribution without distortion either in the intermediate frequencies or in the demodulator. There seems to have been some thought that this aspect will be more evident with Foster-Seeley discriminators than with ratio detectors, but actually it is a matter of correct design and alignment with either circuit.

The response of the tuner from antenna input, through the intermediate frequencies and limiter(s) must not only be linear over the whole 150 to 200 kc bandwidth (flat-topped), but it must also be capable

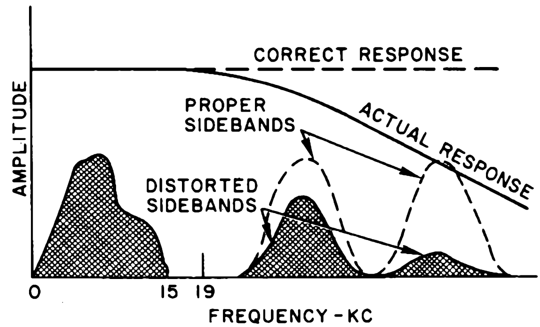


Fig. 5-4. Effect of undue roll-off that would be acceptable for high-quality monophonic, but not for stereo.

of handling the relative amplitude of component sidebands in a stereo broadcast as well as it did the relative amplitudes in a monophonic broadcast.

Next, having settled that little detail, we hope successfully, the discriminator or ratio detector must have a linear response, unaffected by instantaneous sideband amplitude, over the full bandwidth.

With monophonic reception, deficiency in either of these requirements could, to a large extent, be offset by a readjustment in the other circuit. If the discriminator or ratio detector were not quite linear, slight "fudging" of the i-f's could minimize monophonic distortion. But not on stereo.

Now, assuming the tuner checks out for stereo, we still have to add the multiplex adapter. First, the audio stage of the tuner may have a roll-off above 15 kc, even without the normal de-emphasis. Unless something can be done about it, this will cause distortion on stereo because of change in the relative magnitude of the pilot and lower and upper subcarrier sidebands, in that order (Fig. 5-4).

Even assuming the tuner output is flat and undistorted, it needs to be phase linear. Actually, if it is flat to at least 53 kc without compensation, and will handle the stereo signal successfully, it is a fair bet that the phase linearity is not too bad. Any deviation there may be can be reasonably well corrected in the phase adjustment of the adapter, if the tuner is this good.

Assuming the tuner itself checks out, the input leads to the adapter, or the input impedance of the adapter itself, may impose loading on the tuner that could destroy either its linearity of demodulaion of certain components, or its linearity of audio frequency output (here, we take audio to mean the whole stereo complex of frequencies, from low audio to 53 kc).

If all these things check, and the output level from the tuner is such that the adapter level adjustments can handle it (there is no limiter in this type of adapter, as there would have been in the Crosby system) then we are in business, except for the final trims.

MATCHING ADAPTER OR TUNER TO STEREO SYSTEM

At first sight this may seem no more difficult than matching a preamplifier to a power amplifier: just a matter of having levels and impedances right. But there is one important additional feature: the possible presence of ultrasonic residue frequencies.

These are not audible, so long as they do not cause trouble by interacting in some way with something else to produce a further by-product that is audible. An amplifier with unduly extended high-frequency response but inadequate handling capacity in the ultrasonic range (which it never needed before) may cause distortion, with resulting intermodulation products some of which will be audible. Most likely they will show as twitters or whistles.

If you want to *record* your received stereo broadcast, there is an additional hazard: possible interaction between these ultrasonic frequencies and the tape recorder's bias and/or erase oscillator. This will give rise to "birdies" any time there is stereo information.

The best remedy for either of these problems is filtering to remove the offending unwanted frequencies after stereo demodulation and matrixing, at the output of the adapter or stereo tuner. Some tuners will incorporate such filtering as either a standard or an optional feature. But good filtering is expensive, so many may not include it, you do, however, definitely need such filtering.

If this chapter makes the installation of stereo multiplex look complicated, it is because there really are a number of things that might not be right, and it may not always be easy to correct them. However, it is probable too that many installations will be made without encountering any of these problems. By pointing out as many as possible of the things that can go wrong, we hope to minimize the chance that you may come across one that "wasn't in the book".

6 alignment and performance checks

As we mentioned in the preface, the first edition of a book such as this must try to be an enlightened "shot in the dark". As this is being written, some test equipment is just being finalized and put on the market by one or two manufacturers. Very few stations have actually begun to broadcast multiplex, nothing has been said about providing test transmissions to help with alignment and performance checks.

In the absence of transmissions designed for the purpose, it will be difficult to align one of these adapters without a special test generator. First we will outline what needs alignment, and how failure of proper alignment can affect performance or to what degree it can be compensated. Then we will suggest ways to do these things, according to facilities that may or may not be available.

LOCAL SUBCARRIER REGENERATION

Whichever type of adapter or tuner circuit is used, it must employ some means to regenerate the subcarrier. Not only must the subcarrier stay accurately in synchronism with the original carrier of the transmitted subcarrier sidebands, it must also be very precisely *in phase* with it, within 3° for undistorted playback.

If the subcarrier is regenerated by frequency multiplying from the pilot 19 kc transmitted, phase adjustment will be relatively simple, being determined once and for all by the precise values in the 38-kc-tuned circuit, and to a lesser extent by the values in the filter that separates the 19-kc signal from everything else (Fig. 6-1).

One reason for not using the frequency-multiplying technique in some adapters is the possibility that components of the low frequency (sum signal) may slightly modulate the 19-kc output, as well as that of the 38-kc sidebands. If the reinserted 38 kc picks up a second-order amplification of this modulation, the stereo-matrix output could have considerably more distortion than with a locked oscillator. This represents the decision of several manufacturers at this stage of the art, but it is a situation that may change with more highly-developed techniques.

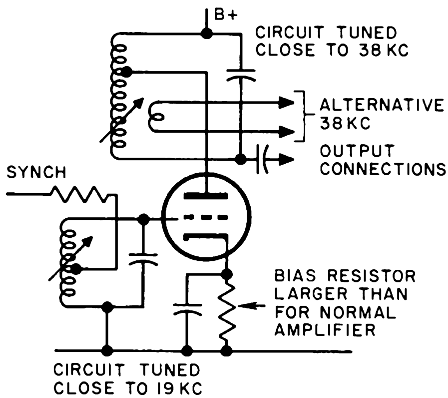


Fig. 6-1. Basic circuit arrangement for frequency doubler without an oscillator. For method of phase adjustment, see text.

Where a locked oscillator is used, the local oscillator may be either 19 or 38 kc. As mentioned in Chapter 4, different advantages are claimed for each. The important thing in alignment procedure is to know which frequency is being used.

Where a 19-kc oscillator is used, it will use a tube operating under conditions that give its amplification characteristic considerable curvature, so the plate current has a strong second harmonic that can be resonated and used. Phasing will have to be controlled by the precise way the synchronizing signal of 19 kc locks the 19-kc oscillator (Fig. 6-2). This can be adjusted in one of two ways: by changing the strength or phase of the injected synchronizing input; or by changing the free-running frequency of the oscillator. The latter method works because change in its free-running frequency, without going far enough to pull it out of sync, will make it pull forward or backward against the sync lock. To a lesser degree, the 38-kc circuit will trim phase angle at the output.

Where the oscillator frequency is 38 kc, its free-running frequency has to be much more precise than with the 19-kc type oscillator. The sync will only correct the frequency once every two cycles, so if the free-running frequency is appreciably off, the two cycles will be unequal,

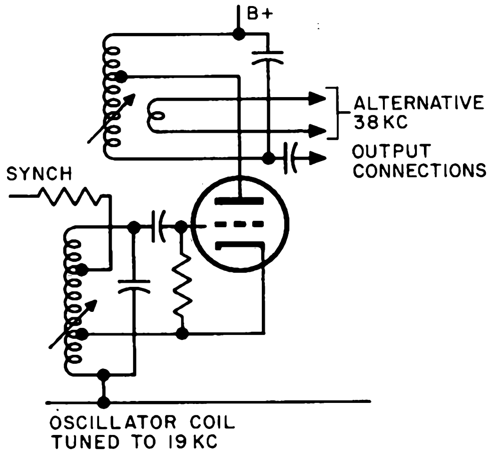


Fig. 6-2. Basic circuit arrangement for locked pilot-frequency oscillator.

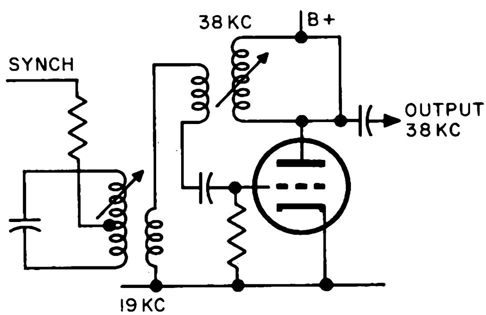


Fig. 6-3. One form of basic circuit for locked subcarrier-frequency oscillator.

causing distortion, even though the average frequency over the two-cycle period is correct. In this case, very little phase control can be achieved by adjusting the free-running frequency (Fig. 6-3). The only feasible method is by adjusting the phase of the 19-kc sync signal. This is achieved by adjusting the exact resonant frequency of the filter that separates the 19 kc. Making the resonant frequency slightly high advances the phase of the sync signal; making it low, retards the phase.

With either kind of receiver, phase adjustment of the regenerated carrier is important, but the effects of incorrect adjustment differ. With a beam-switching circuit, or any similar circuit that does not separate

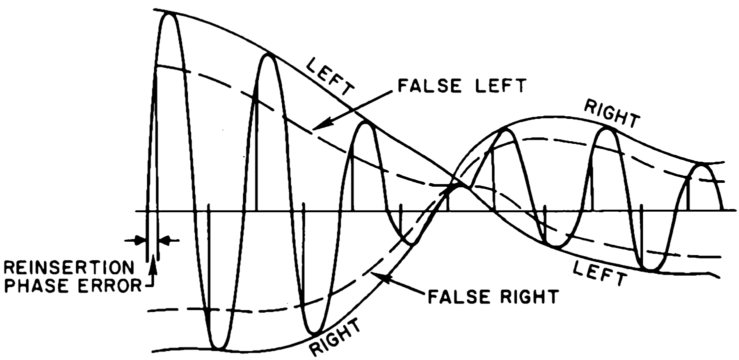


Fig. 6-4. The effect of incorrect subcarrier phasing in a circuit using beam-switching demodulation: dashed-line curves show incorrect output due to phase shift indicated in subcarrier phase.

sum from difference, the main effect of incorrect phase adjustment is loss of separation (Fig. 6-4). Phase can be adjusted with either a test transmission or a signal generator when left only and right only are used alternately. Check for a null in the right output when left only is received, and for a null in the left output when right only is received.

With a circuit involving matrixing, where sum and difference are separated, phasing can affect both separation and distortion. But with this kind of circuit a control called *dimension* will also affect separation, so the phase adjustment should be used to minimize distortion, leaving the dimension control (a matrixing adjustment) to obtain correct separation (Fig. 6-5).

The best way to do this is with a reliable signal generator in which the phase and magnitude of the various components is carefully controlled and checked to conform to FCC standard, and the modulation is low in distortion. Switch off the sum channel component (this can easily be done at the generator) and measure distortion in the demodulated subcarrier output. Align phase to minimize this. Now switch on the sum component and transmit audio on left or right only; use the dimension control to null the other (right or left respectively) output (Fig. 6-6).

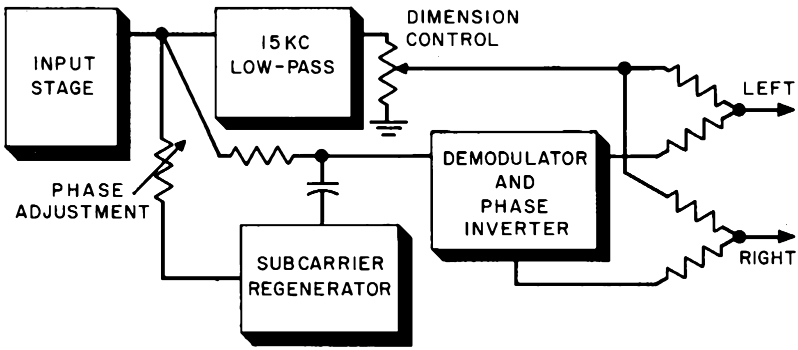


Fig. 6-5. Partial (block) schematic of receiver using matrixing, showing possible locations of phase adjustment and dimension control, for separate adjustment of distortion and stereo separation.

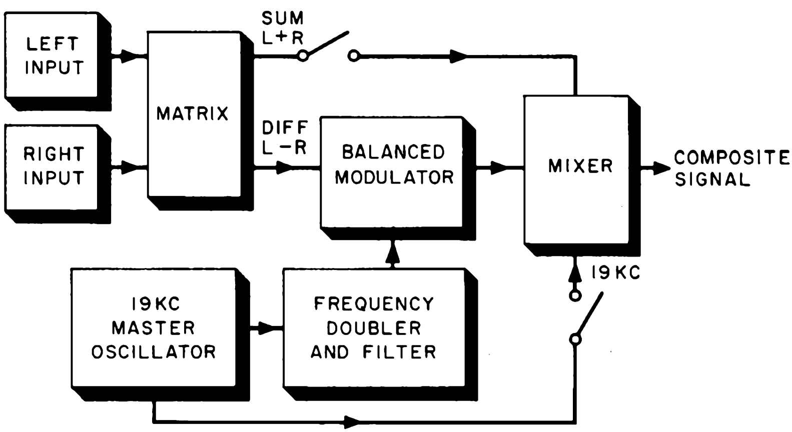


Fig. 6-6. Block schematic of a stereo-multiplex signal generator, showing how sum signal can be separately "killed" to facilitate checking distortion in subcarrier channel of receiver.

FILTERS

Some adapters have several filters, some hardly any. Some use adjustable filters, some use preadjusted units that cannot be changed, only replaced, if they are wrong. The main thing is to know what any individual filter should do, how to check it, and adjust it if necessary, or possible.

Two varieties of low-pass filter will be encountered: one with a roll-off at 53 kc, the other with a roll-off at 15 kc; the first is there to reject demodulated frequencies associated with a private SCA subcarrier (Fig. 6-7); the second, is to separate the sum signal from everything else (Fig. 6-8).

The 53-kc low-pass filter may include a high-rejection of 67 kc, by m-derivation in its design. If no transmissions with SCA subcarriers are

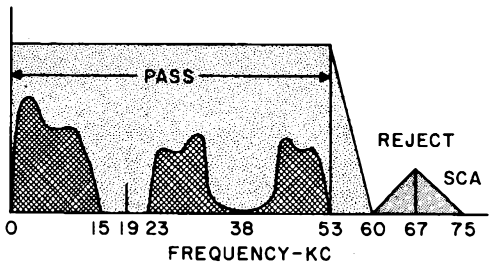


Fig. 6-7. Function of 53-kc low-pass filter is to separate the stereo program complex below 53-kc, from any SCA transmission above 60-kc.

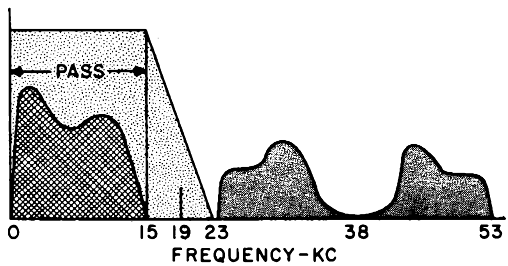


Fig. 6-8. Function of 15-kc low-pass filter is to separate the sum signal from the difference sidebands.

receivable in an area, such a filter is not strictly necessary, and the adapter may even perform better without it. But if SCA subcarriers are present, lack of such a filter will result in a splash-like noise (not an intelligible reproduction of the SCA program).

Another type of filter some adapters will use is a bandpass filter, with a pass range of from 23 to 53 kc. This combines the function of eliminating the SCA subcarrier (above the 53 kc end) with separation of the stereo subcarrier modulation from the sum signal.

At the output, in addition to regular de-emphasis, some adapters will include either a pair of low-pass filters rolling-off at 15 kc, or 38-kc rejection filters. They may also be composite filters, combining low-pass with rejection action.

Proper performance of the first 15-kc filter mentioned, or of the 53-kc low-pass, is essential to good quality stereo with proper separation. The output filters are not so critical, so long as they serve their primary purpose of removing unwanted frequencies above 15 kc, and particularly in the region of 38 kc.

To separate sum signal, the 15-kc filter must have minimum phase shift below 15 kc, which means its response must be flat up to almost 15 kc, before it starts to roll-off. If the filter boosts at all before it rolls-off, or if it rolls-off too droopily, distortion and separation will suffer badly in the higher frequencies.

The 53-kc low-pass filter needs to be as nearly as possible phase-linear up to 53 kc, for the same reason. If it is not, the effect will be similar. The bandpass filter cannot possibly be phase-linear, so a circuit that uses it has basically limited performance.

For economy reasons not all filters used, especially in lower-cost adapters, will be as phase-linear as they might be. If their response is reasonably flat over the range they are intended to cover, it can be assumed their performance is as good as it was intended to be in the design. If phase linearity is not adequate for really high-quality reproduction, phase adjustment of the regenerated subcarrier can minimize the overall distortion effect.

The way to check phase linearity of the overall filter system is to check separation and distortion at different audio frequencies, which will require either the broadcasting of individual tones of different frequencies alternately over left only and right only, or a signal generator that can allow this to be done on the bench. Such generators are going into production, but, initially at least, they are expensive. If minimum distortion and maximum separation require change in phase adjustment for different audio frequencies, one or more of the filters suffers from phase nonlinearity.

STEREO REGENERATION

The subcarrier regeneration and filter circuits prepare the incoming, demodulated signal for the circuit that re-forms the stereo left and right channels. This may be a beam-switching or balanced-modulator type, or it may be a subcarrier demodulator and matrixing network.

If everything is functioning correctly, the beam-switching circuit will give well-separated and balanced left and right. Lack of balance must be due to lack of balance in external circuits, or to imbalance in the switching (38-kc subcarrier) signal. Failure of an element in a balanced demodulator circuit will usually completely disrupt proper stereo restoration.

Separation and balance in the subcarrier demodulator and matrixing-type circuit depend on accuracy of the circuit components in the matrixing section. If a dimension control is provided, change of setting will alter separation — it is supposed to — but balance should be preserved.

Demodulator and matrixing circuits can readily be checked as separate entities. The beam-switching and balanced demodulator circuits can only be checked as a complete entity, using test transmissions, or special signal generators.

DE-EMPHASIS

Here, both the main tuner output and the de-emphasis at the adapter outputs should be checked. The main tuner should not have any de-

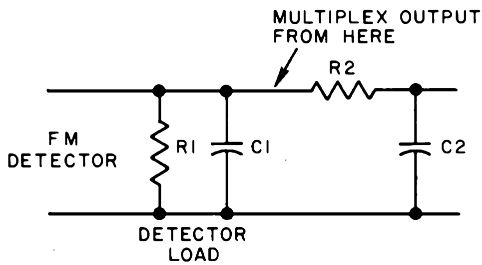


Fig. 6-9. Output circuit of typical FM tuner demodulator, to show relevant values: R1 and C1 are the detector load; they must allow flat, distortionless detection of frequencies up to 75 kc, where previously 15 kc was adequate; R2 and C2 are the de-emphasis components; it may be sufficient to connect to the input end of R2, but these components may produce a loading effect that drags the response down, in which case they should be removed.

emphasis in its feed to the adapter. Its response should be flat, to 53 kc at least. If there is de-emphasis, it needs removing at this point (Fig. 6-9).

The proper place for de-emphasis is at the output from the adapter, whichever type is used. This can be checked by ascertaining that the 3-db point on both left and right channels, is at 2120 cycles, going into a 6-db per octave slope, with 7 db loss at 4240 cycles, 12.25 db at 8480 cycles, and 18 db at 16,960 cycles (Fig. 6-10). If filters are used as well, that last frequency will not need checking.

OPTIMIZING PERFORMANCE

Although we have dealt with each stage and its adjustment separately, it is evident that overall performance depends on the combination of everything being right. If separation is lacking, it can be due to several

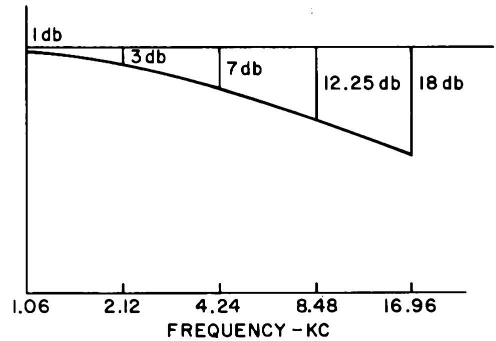


Fig. 6-10. The proper de-emphasis curve may be checked at the frequencies indicated here.

causes. Several adjustments may improve it individually, but only the right combination of adjustments can optimize it completely.

If there is distortion, it may be possible to reduce this by a similar combination of adjustments, but if the circuit uses filters that are not adequately phase-linear, the improvement possible can be severely limited.

7 general troubleshooting procedure

Troubleshooting stereo multiplex follows the same general principles used for any r-f or a-f troubleshooting, but involves some previously unfamiliar techniques in dealing with subcarriers and matrixing. Until more stations have been on the air, and decisions have been made as to the extent to which test transmissions can help, it is not certain how useful "live" transmissions will be as a troubleshooting or alignment aid.

Whatever facilities transmissions may provide for testing, it is certain they will be inadequate as a consistent service "tool". The primary purpose of such transmissions will be for program and entertainment; test transmissions will at best occupy only brief intervals. So the service man will need a test generator to do most of his work.

For service purposes, a simplified type of generator, such as that announced by General Techniques Inc., serves better than the more elaborate types (Fig. 7-1). This generator can simulate quite accurately the composite audio for either left only or right only, using either a pure tone, to enable distortion to be checked as well as separation, or program, (e.g., from a monophonic phonograph) so the full range of reproduction and separation can be checked.

Additionally, it includes a reactance modulator, which enables the whole demodulator section of the FM tuner, along with the multiplex adapter, to be checked. After the tuner has been aligned as closely as possible with a regular sweep generator, the multiplex generator can be connected to an early i-f stage and this signal used to make final adjustments to the i-f and demodulator tuning, as well as the adapter, to get maximum separation and minimum separation in each channel. Figures 7-2 and 7-3 show methods of checking adapter alignment and overall performance on multiplex, respectively.

As with other electronic equipment, a number of faults can occur that may be identified by the way the system misbehaves. Just as loss of vertical sync in TV results in rolling, loss of horizontal sync in tearing, and so on, each defect in a multiplex system will have its characteristic symptoms. From this diagnosis, the area of trouble is narrowed and the fault can be pinpointed and remedied. We will take the possible troubles according to the defect that causes them, describe the symptoms and then where to look for the trouble and how to make the remedy.

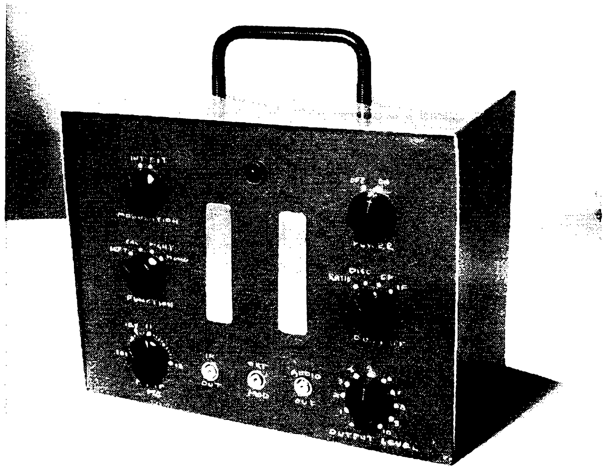


Fig. 7-1. The service-type stereo multiplex checker announced by General Techniques Inc. *Courtesy General Techniques, Inc.*

LOSS OF SYNC

In multiplex, loss of sync means the subcarrier drifts in the general region of 38 kc without being locked to the suppressed carrier frequency. As a result, the upper audio frequencies will beat back and forth between the left and right channels, usually at a frequency corresponding with one of the lower audio frequencies present. The effect is similar to severe intermodulation distortion, producing a "gargling" effect on the reproduction.

If the 38-kc oscillator happens to be close in frequency to its true

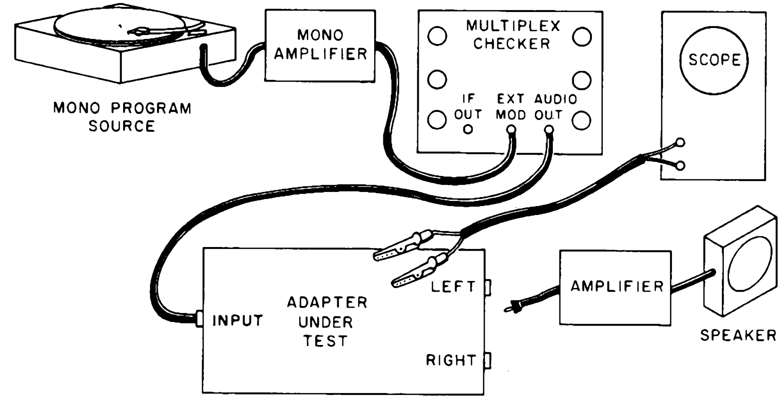


Fig. 7-2. Set-up for checking an adapter with audio. Procedure is as follows:

1. Connect 'scope to audio output, or adapter input, set modulation to *internal,* function switch to *cal,* and output switch to *ratio* or *disc* according to type of FM-tuner detector the adapter is intended to work with; check that display is correct and adjust controls if necessary.
2. Connect 'scope and amplifier to left output of adapter, set function switch to *left* and check waveform and tone; set function switch to *right* and check separation (should be almost negligible, undistorted output).
3. Connect 'scope and amplifier to right adapter, and reverse procedure of (2).
4. Switch to *external mod.* and repeat checks (2) and (3); also check in mono position of function switch, to see that both channel outputs are equal and undistorted.

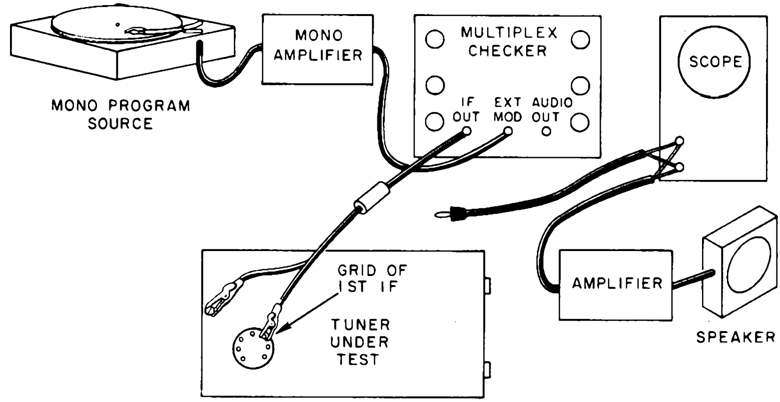


Fig. 7-3. Set-up for checking a complete tuher with multiplex. Procedure is the same as in Fig. 7-2, except that for step (1) 'scope and amplifier are connected to the audio output of the multiplex checker, and the output switch is set to i-f; the FM I-F control should also be tuned to find the operating i-f frequency of the tuner.

value, the drift may be much slower, so that a warbling effect is achieved, due to left and right channels exchanging places at a slow rate.

With either of these defects, the cause is failure of the 19-kc pilot to provide synchronization. Assuming the 19-kc pilot is received — which it must be unless the transmission is defective — the fault lies in that part of the adapter circuit that handles the 19-kc pilot. This is always a tuned circuit, in some cases more than one.

Failure of a component in one of these tuned circuits, or of the stage connected with them, will result in loss of sync. If a tuned circuit gets sufficiently off-tune, this can cause a similar effect, but in this case, severe phase-shifting effects should have been experienced before actual loss of sync.

LOSS OF SUBCARRIER

In a sense, this is a worse defect than loss of sync, but its effect will depend on the kind of circuit used. In a matrixing circuit, loss of subcarrier means the demodulated L — R signal will be severely (100%)

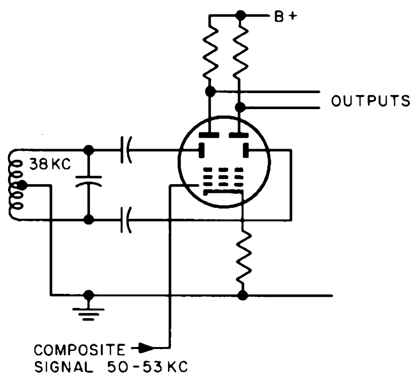


Fig. 7-4 Failure of the 38-kc regenerated subcarrier in a beam-switching circuit will result in complete loss of separation — monophonic; both outputs will be the same because the beam-switching electrodes have steady (zero) voltage.

distorted, so the overall output will sound distorted. In a beam-switching circuit where the regenerated subcarrier is applied separately from other signal components, to the beam-switching plates of a tube, or to diode bridges that serve a similar function, loss of the subcarrier merely results in loss of any L — R signal there is, and the reproduction sounds monophonic (Fig. 7-4).

In this case, the cause depends on the type of circuit. If no oscillators are used, the most probable cause of trouble is failure of the frequency doubler. Where either a 19-kc or a 38-kc oscillator is used, the most probable cause is failure of the oscillator — the tube or one of its associated components.

PHASE-SHIFTED SUBCARRIER

Changing the phase of the reinserted subcarrier relative to the received sidebands, alters the effective form of modulation from amplitude to a limited frequency or phase modulation (if taken far enough). So incorrect phase will result in reduced output from the L — R detector of a matrix-type receiver, with consequent loss of separation.

In the beam-switching type receiver, the effect is basically the same: time that should be entirely devoted to left channel is partially devoted to right, and vice versa. The overall effect is loss of separation.

Incorrect phase adjustment may affect some audio modulation frequencies more than others, or result in the generation of spurious frequencies at the L — R demodulator, either of which will give the effect of distortion in the reproduction, in addition to loss of left-to-right separation.

Fortunately, or unfortunately, according to viewpoint or the particular adapter in use, some of the defects caused by incorrect phase can be minimized by adjustment of the matrixing (dimension) control, or the efficiency balance in a beam-switching circuit. This means that an adjustment that may achieve an apparent maximum separation may result in more than necessary distortion. For optimum performance, both subcarrier phasing and matrix or efficiency balance need to be optimized.

MATRIX INCORRECT

Under this heading we also consider incorrect setting of the efficiency balance in a beam-switching circuit, which serves the same purpose. This results in unequal magnitudes of L + R and L — R signals, so proper matrixing to restore L and R completely separate cannot occur. This usually results only in loss of separation.

In circuits where the phase-linearity of the filters used is poor, the best operating condition will usually be achieved by successive adjustment. First, set the matrixing or efficiency balance adjustment to give best separation at low frequencies; then set the phase adjustment to achieve best separation at higher frequencies; then retrim the matrixing or efficiency balance for maximum low-frequency separation; keep repeating these adjustments until you are satisfied that overall separation is optimized.

The service generator is invaluable for making these adjustments. It is a good idea to set up a single loudspeaker and associated oscilloscope so either left or right output can be listened to and looked at in quick succession (Fig. 7-5). Separation should be tested both ways as a check.

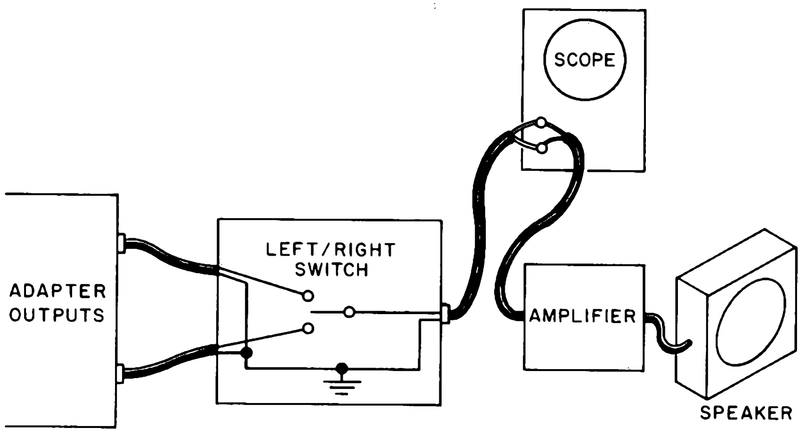


Fig. 7-5. A switch-box can help in quick checking of left and right outputs.

First set the generator to give a "left only" signal, either single tone or program; listen, and look at the left output to check the quality of that channel; then look at and listen to the right-channel output, which should be silent. Most important is that its level is very much lower than that of the left output, and what signal there is should be undistorted. Adjustments should usually be made to achieve minimum output in the right channel with the generator giving "left only" input.

Repeat the test with the generator set to give "right only", and reversing the order of connections at the output: check the quality of right, and for minimum "breakthrough" on left.

These same tests will also be useful for finalizing the alignment of the FM-tuner stages ahead of the adapter. Slight distortions in the linearity of the i-f's or the discriminator (or ratio detector) characteristic may not be enough to cause distortion to the L + R channel, but they may cause interference with the L — R products in the subcarrier channel.

DEFECTIVE FILTERS

Filters do not often go defective, but such a fault is as likely as some other "rare ones". A filter can become faulty in one of two ways: (1) it can fail to pass what it should — in fact, it can fail to pass anything at all; (2) it can pass frequencies that it should not pass.

The symptoms of such failure need working out according to the function the filter is intended to serve. A few examples are given to illustrate, but there will be many more possibilities that cannot be covered in detail here.

A 15-kc low-pass filter may be used in one of two places: (1) to isolate L + R from the subcarrier sidebands before detection and matrixing; (2) to eliminate unwanted high-frequency products at the output.

Failure to isolate L + R from subcarrier sidebands will result in distortion and lack of separation. If such a filter fails by not passing anything, there will be no L + R, and the program will be thin and reedy; it will also lack separation and possess distortion.

If the rejection part of a 53-kc low-pass filter intended to eliminate SCA subcarrier, fails, this may not show up on test, and will only cause trouble if a local station happens to transmit background music at the same time as a stereo program. This is a local condition, about which you will probably know. As soon as the trouble is reported, you need to check

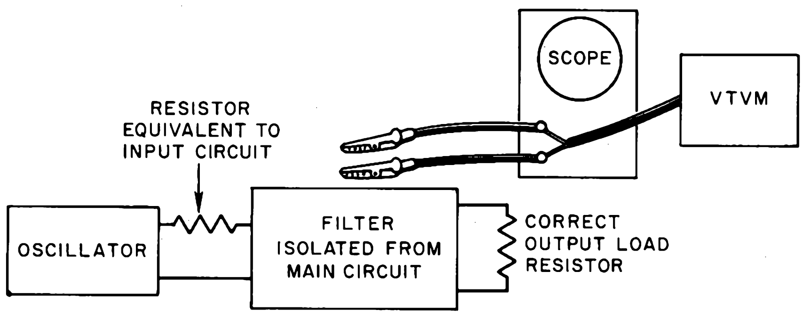


Fig. 7-6. Checking the performance of a filter: input is compared with output at each frequency. If the impedance used is high, the connections should be low-capacitance shielded cable.

the rejection properties of filters intended for that purpose. This requires an audio oscillator with an output including frequencies up to 100 kc.

Failure of a 15-kc low-pass filter at the output may not be noticed if you do not try to record your stereo program. If such a filter fails, it will allow the unwanted frequencies to make "birdies" with the bias oscillator of the tape recorder.

The only way to check the performance of any filter is to isolate it from its circuit and test it under simulated conditions (Fig. 7-6). In the range of frequencies the filter is intended to pass, shorting its input terminal to its output terminal should not materially increase the output. At the cutoff frequency, the output should progressively decrease.

Failure to pass anything can be due to an open connection in a series element of the filter (Fig. 7-7) or to a short-circuit in a shunt element.

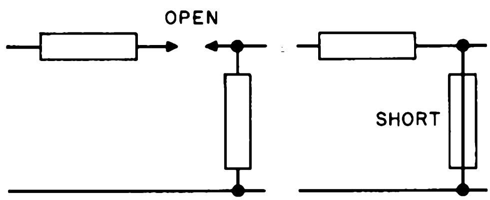


Fig. 7-7. Faults in a filter that can result in loss of any transmission.

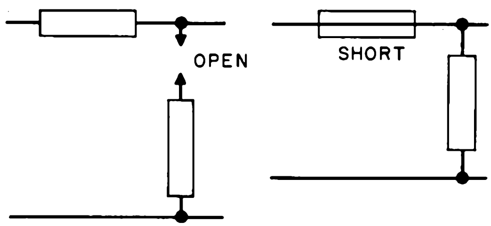


Fig. 7-8. Faults in a filter that can result in loss of desired rejection.

Failure to reject frequencies that should not be passed may be due to an open connection in a shunt element of the filter (Fig. 7-8) or to a short-circuit in a series element. Keeping these simple facts in mind will usually enable a filter fault to be traced, without needing a complicated knowledge of filter action.

Undoubtedly, as with other types of equipment, certain faults will prove prevalent with certain models, and service technicians will develop the kind of experience that says, "If such and such symptoms develop in model XYZ, the trouble is a disconnected caboodle". Till then, they will have to "play it by ear" and use the general detective principles outlined.

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NORMAN H. CROWHURST

Audio Consultant

In terms of appropriateness, perhaps no one is more suited to write an authoritative book on FM stereo multiplexing than Mr. Norman Crowhurst. Long a recognized authority in the field of audio engineering and high fidelity, Mr. Crowhurst has "lived" with the audio field since its inception as a distinct branch of electronics. Mr. Crowhurst was educated in London, where he graduated as an electrical engineer in 1935, with distinction in all subjects. This was followed by several years as a lecturer in engineering technology and mathematics in two of London's leading technical colleges, and several years as Chief Engineer of the Tannoy organization, one of the leading companies in the world specializing in audio.

His arrival in the United States in 1953 was warmly received by the audio engineering field, and he was immediately accepted as an outstanding consulting engineer. He has continued in this capacity, and has worked with several of the leading audio firms in this country. Mr. Crowhurst liked what he saw here, and became an American citizen in 1960. As a result of his heavy contributions and in recognition of his stature in the field, Mr. Crowhurst has received the distinction of being named a Fellow of the Audio Engineering Society, the leading technical body in the audio field.

When the FCC announced its standardization of FM stereo multiplexing, the editors of John F. Rider Publisher, Inc., could think of no one better qualified to prepare this timely and important book.

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