IRE TRANSACTIONS

ON BROADCASTING

PGBC-14 NOVEMBER, 1959

TABLE OF CONTENTS

PUBLISHED BY THE

PROFESSIONAL GROUP ON BROADCASTING

IRE PROFESSIONAL GROUP ON BROADCASTING

The Professional Group on Broadcasting is an organization, within the framework of the IRE, of members with principal professional interest in Broadcasting. All members of the IRE are eligible for membership in the Group and will receive all Group publications upon payment of prescribed fee.

Annual Fee: $2.00

IRE TRANSACTIONS®

on Broadcasting

Published by the Institute of Radio Engineers, Inc., for the Professional Group on Broadcasting at 1 East 79th Street, New York 21, New York. Responsibility for the contents rests upon the authors, and not upon the IRE, the Group or its members. Individual copies available for sale to IRE-PGBC members at $0.60, to IRE members at $0.90 and to nonmembers at $1.80

F O R E W O R D

STEREOPHONIC broadcasting is the newest and one of the most controversial problems the broadcast engineer has had to face. For that reason, the Administrative Committee of the PGBC decided to devote this issue of Transactions to the subject.

At the outset, it must be made clear that the papers presented here by no means represent a complete treatment of the subject. There are many systems proposed and it would be impossible to print papers on all of them at this time. Yet, because of the tremendous current interest of the broadcasters, it seemed expedient to give them what space we could.

It is recognized full well that the final NSRC recommendations may or may not resemble any of the methods discussed herein. This issue can only whet the appetite. There will be much more published on stereo in the IRE Transactions on Broadcasting as elsewhere.

William L. Hughes
Editor

OPTIMIZED COMPATIBLE AM STEREO BROADCAST SYSTEM

H. B. Collins, Jr. and D. T. Webb
Philco Corporation
Philadelphia, Pennsylvania

Summary--A two-channel multiplex system for compatible AM broadcasting is described. System objectives including compatibility, service area, and program quality are discussed. Three different methods of creating the equivalent transmitted signal are reported, and conversion of present-day monaural stations to stereo by each method is indicated. The design of receivers for recovering the two stereo tracks is examined showing the signals derived by various means of detection. Emphasis is placed on a design resulting in a reliable, minimum cost receiver. Field test equipment and results are briefly considered and the level of performance that can be obtained from the system is stated.

Introduction

This paper presents a system of compatible AM stereo broadcasting and describes the associated receiving equipment. This system is considered to be optimum. At first, the mere suggestion of developing a bandwidth-limited stereo system such as would be encountered by employing a standard AM channel might sound unattractive. It was found, however, that stereo added proportionally more realism and listening pleasure to AM than to a full fidelity system. Early tests were conducted by simulating AM broadcast bandwidth limitations on a stereo audio system.

The importance and timeliness of stereo can be judged by the rapid growth of interest in better music and the increased sales of stereo records, tapes, and hi-fi phonograph systems. For example, in the first quarter of 1958, no phonograph units (in the medium to high-price bracket) were stereo-equipped; by the first quarter of 1959, 88 per cent were stereo-equipped. Stereo on AM radio is potentially a means for bringing this new service to a large audience since it performs equally well in either the home or automobile. Addition of an AM stereo medium should not only increase general interest in stereo, but should also stimulate the broadcast industry.

The thought of AM stereo is not new; in fact, some proposals of similar systems have been made both in this country and abroad. The system proposed in this paper is the result of recognizing several factors such as that stereo has subjective value in AM broadcasting and that monophonic AM sideband information is redundant. It remained only to devise an efficient system for multiplexing stereo signals.

Objectives

In devising a method for producing stereo for AM broadcasting, one must keep certain basic objectives in mind. First, any proposal must be compatible with present-day practice in the broadcast field. The conventional single-channel radio should be able to reproduce a stereo transmission with as good quality as the present-day monaural broadcasts; i.e., the radio should produce sound proportional to the sum of the stereo sources. The broadcast band is already crowded, and any increase in channel width would be impractical. Therefore, the present station channel assignment should be preserved. From the broadcasters' point of view, a stereo transmission should not cause any loss of audience. The compatibility objective fulfills part of this requirement. However, it was further desired to cause no deterioration of signal-to-noise ratio (station area coverage) when transmitting stereo.

Of course, the stereo performance should be commercially acceptable and the terminal equipment, especially at the receiving end, should be of such simplicity as to be reliable and inexpensive. The final goal was a system that could easily transfer from monaural to stereo program material. A signal that fulfills these objectives will be described in detail from several points of view in order to demonstrate the characteristics of the proposed AM stereo broadcast system.

Signal Processing

Assume that it is necessary to transmit two audio tracks at a single carrier frequency in such a manner that the two tracks are independent of each other. One track may be used to amplitude modulate a carrier in the normal amplitude modulation method. The second audio track may be used to modulate a second carrier in the same manner. If these carriers are of equal amplitudes and of the same frequency, but are placed in a phase relationship so as to be in quadrature with each other, they may be added together without crosstalk. That is to say, such a signal, after being transmitted, may be separated at the receiving terminals into its two original audio tracks, each track being independent of the other. Figure 1 is a vector representation of such a signal. Track A carrier is shown at 33 1/3 per cent modulation with an audio sinusoid from track A. Likewise, track B carrier is 100 per cent modulated by audio from track B. These carriers are of equal amplitude and the same frequency, but are in quadrature. The addition of these two modulated carriers will produce a resultant modulated carrier that is varying in both phase and amplitude.

The mathematical description of this resultant signal may be expressed as follows:

Track A modulated carrier =
$$C(1+m_a \cos\omega_a t)\cos(\omega_o t+\pi/4)$$

Track B modulated carrier =
$$C(1+m_b \cos\omega_b t)\cos(\omega_o t-\pi/4)$$

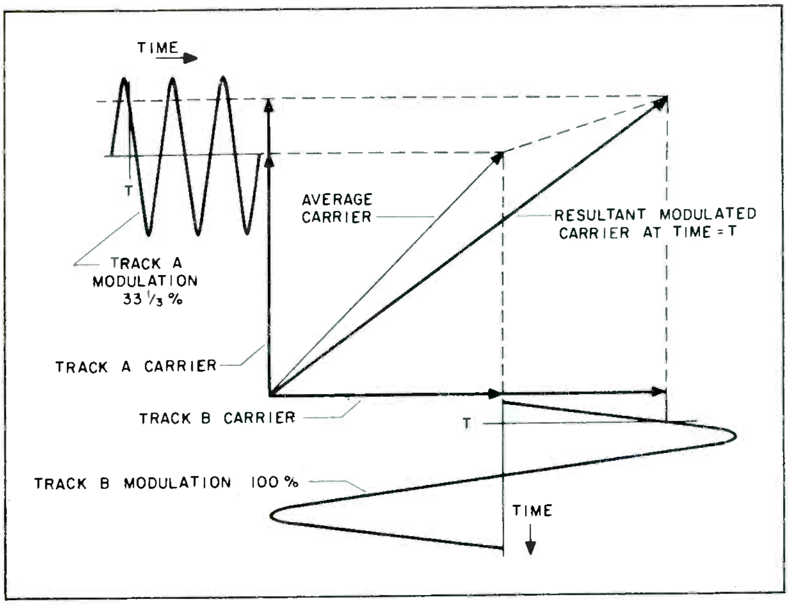

Fig. 1 - Separately modulated quadrature carriers of equal amplitude.

Where C = carrier peak amplitude
m_a = modulation index of A track,
m_b = modulation index of B track,
ω_a = angular frequency of A track,
ω_b = angular frequency of B track, and
ω_o = angular frequency of rf carrier.

The sum of these two is the resultant modulated carrier $= \sqrt{2}\ C \cos \omega_o t$

$$+ \frac{m_b C}{2} \cos\left[\omega_o t + (\omega_b t - \pi/4)\right]$$

$$+ \frac{m_a C}{2} \cos\left[\omega_o t + (\omega_a t + \pi/4)\right]$$

$$+ \frac{m_b C}{2} \cos\left[\omega_o t - (\omega_b t + \pi/4)\right]$$

$$+ \frac{m_a C}{2} \cos\left[\omega_o t - (\omega_a t - \pi/4)\right].$$

Modulation Methods

Figure 2 shows the resultant signal that is generated for different input signals. In Fig. 2a the same signal is used in each audio track to 50 per cent modulate the quadrature carriers. The result is a 50 per cent amplitude modulated carrier having no phase modulation. In other words, a signal that contains no stereo also contains no phase modulation. Figure 2b demonstrates the effect of modulating only one track by 50 per cent. Note that the resultant carrier varies in phase in a nonsymmetrical manner about the phase of the unmodulated resultant carrier. Also notice that the locus of the tip of the resultant carrier follows a straight line.

Figure 2c shows one track with 70 per cent modulation and the other with 30 per cent modulation. Again the locus of the tip of the resultant carrier follows a straight line because the audio modulations are in phase with each other. The phase modulation is still nonsymmetrical about the position of the unmodulated resultant carrier. The resultant amplitude modulation is proportional to the sum of the two audio input signals. A block diagram of the transmitter that will generate a stereo signal in this manner is shown in Fig. 3.

These figures suggest another way of producing a signal that is identical to the modulation of two full quadrature carriers. It may be noted that the amplitude of the resultant modulated carrier, when projected to the axis of the resultant unmodulated carrier, yields a modulation envelope proportional to the sum of the two audio tracks. Also, the phase deviation may be produced by the addition of a component in quadrature with the unmodulated resultant carrier and of an amplitude proportional to the difference of the two audio tracks. This is shown in vector form in Fig. 4. The audio modulation in each track is identical to that used in Fig. 1, and the resultant modulated carrier is identical in each case as it should be. The difference modulated quadrature signal is of the suppressed carrier type.

When no stereo is present, there is no suppressed quadrature carrier present and thus no phase modulation. Note that for the case of one track only being modulated, the sideband power in the sum-modulated carrier is identical to that in the difference-modulated quadrature suppressed carrier. A block diagram of this type of transmitter is shown in Fig. 5.

The mathematical description of this resultant signal may be expressed as follows:

Sum signal modulated carrier =

$$\sqrt{2}\, C(1 + m_\Sigma \cos \omega_\Sigma t) \cos \omega_0 t \; .$$

Difference signal suppressed carrier =

$$\sqrt{2}\, Cm_\Delta \cos\omega_\Delta t \cos(\omega_0 t + \pi/2) \; .$$

The sum of these two is the resultant modulated carrier $= \sqrt{2}\, C \cos \omega_0 t$

$$+ \sqrt{2}\, \frac{m_\Sigma C}{2} \cos(\omega_0 t + \omega_\Sigma t)$$

$$+ \sqrt{2}\, \frac{m_\Delta C}{2} \cos\left[\omega_0 t + (\omega_\Delta t + \pi/2)\right]$$

$$+ \sqrt{2}\, \frac{m_\Sigma C}{2} \cos(\omega_0 t - \omega_\Sigma t)$$

$$+ \sqrt{2}\, \frac{m_\Delta C}{2} \cos\left[\omega_0 t - (\omega_\Delta t - \pi/2)\right] \; .$$

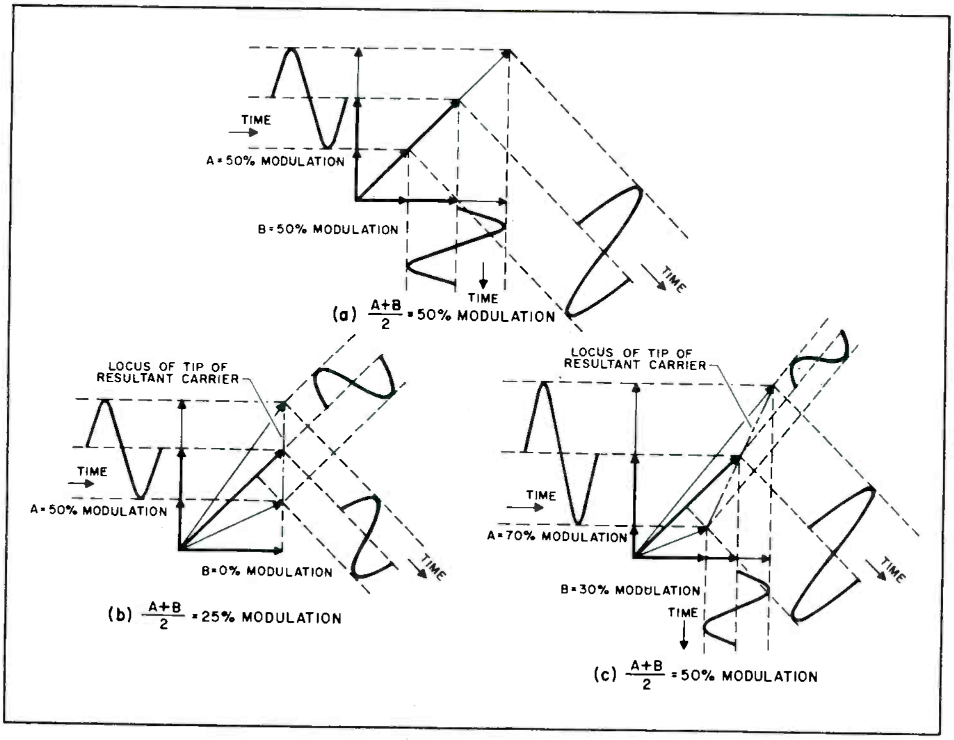

Fig. 2 - Results of various input signal conditions.

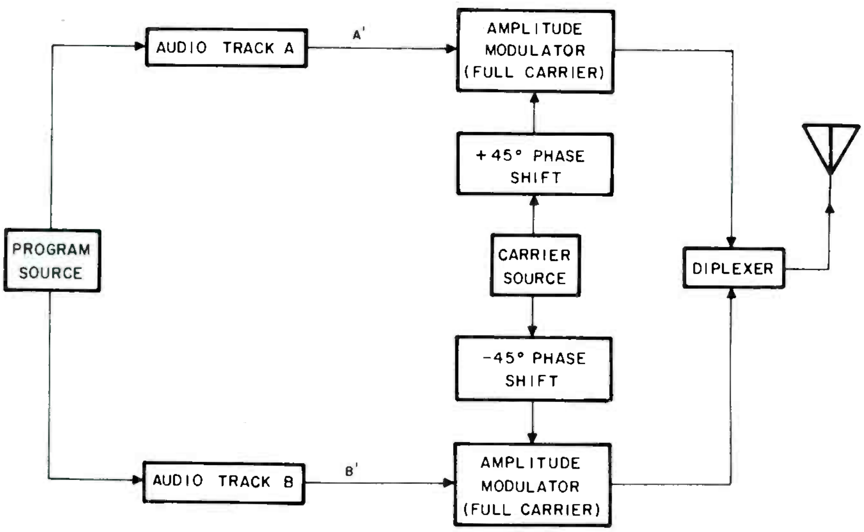

Fig. 3 - Stereo transmitter; two full carrier modulators.

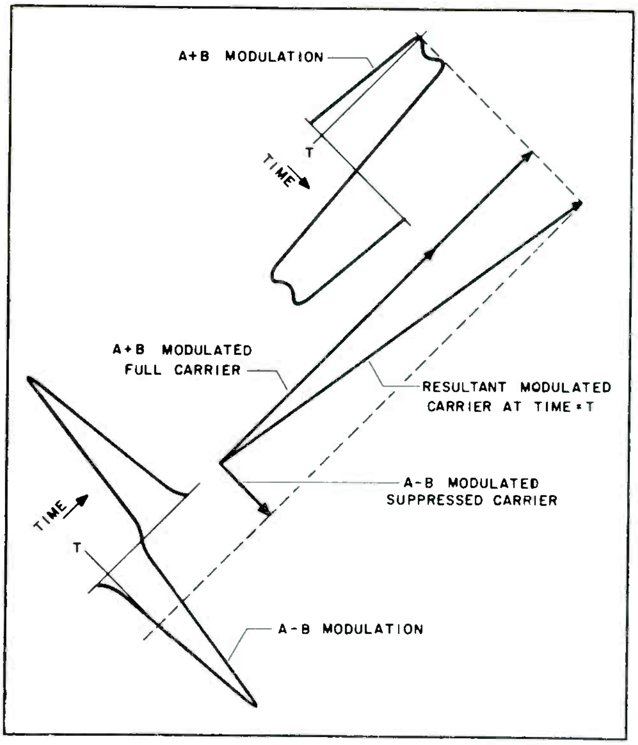

Fig. 4 - Sum modulated full carrier and difference modulated suppressed quadrature carrier.

Note that this is identical to the previous result if:

$$m_\Sigma \cos \omega_\Sigma t = \frac{m_a}{2} \cos\omega_a t + \frac{m_b}{2} \cos\omega_b t$$

$$m_\Delta \cos \omega_\Delta t = \frac{m_a}{2} \cos\omega_a t - \frac{m_b}{2} \cos\omega_b t .$$

This is proof of the identity of these two methods of generating the AM stereo broadcast. Each way of looking at the signal makeup has its own advantages when discussing the characteristics of this stereo system.

A third method of generating the same stereo broadcast signal will now be described. This method proves to be the simplest means of converting existing transmitters to stereo operation. Equipment for the conversion of the 50 kw transmitter of WABC was constructed on the following principles.

The AM stereo signal may be broken into a phase-modulated carrier which is, in turn, amplitude-modulated. The primary component for phase modulation is the difference signal, while for amplitude modulation it is the sum signal. However, an analysis of the signal shows that these modulation components contain some higher order terms from each audio track.

One method of generating the desired phase and amplitude signals is to build a low level set of modulators and combiners operating on the quadrature carrier principles described above. If the stereo output signal is envelope-detected, the proper signal for amplitude modulation is obtained. This signal may be applied directly to the modulator of an existing monaural transmitter. If the low level stereo signal is also passed through a limiter so as to remove all the amplitude modulation, an rf carrier that is properly phase-modulated may be obtained. This phase-modulated carrier may now be used to supply rf excitation to the existing monaural transmitter ahead of the modulator stage.

When proper equalization of the time delay between the rf and the audio paths is obtained so

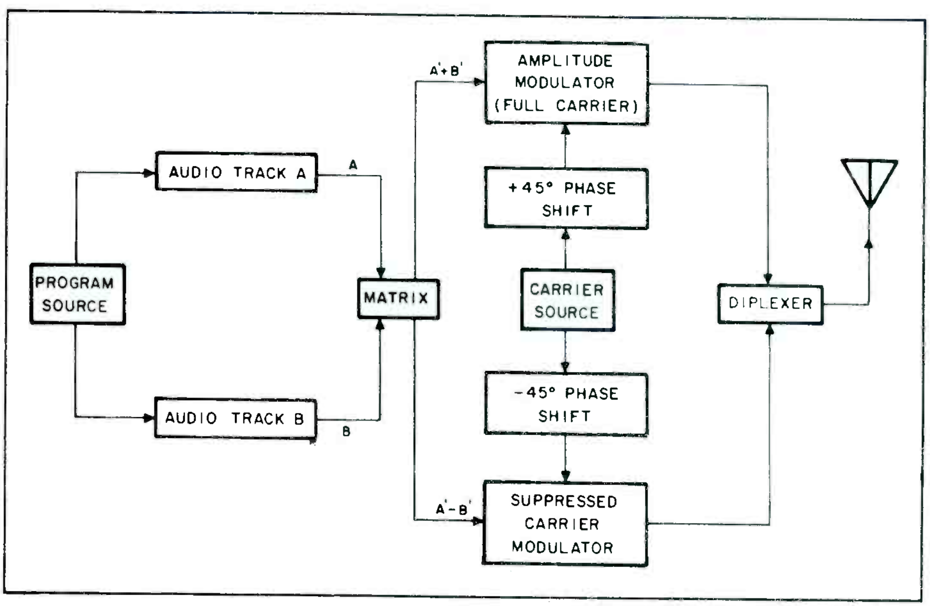

Fig. 5 - Stereo transmitter; one full carrier and one suppressed carrier modulator.

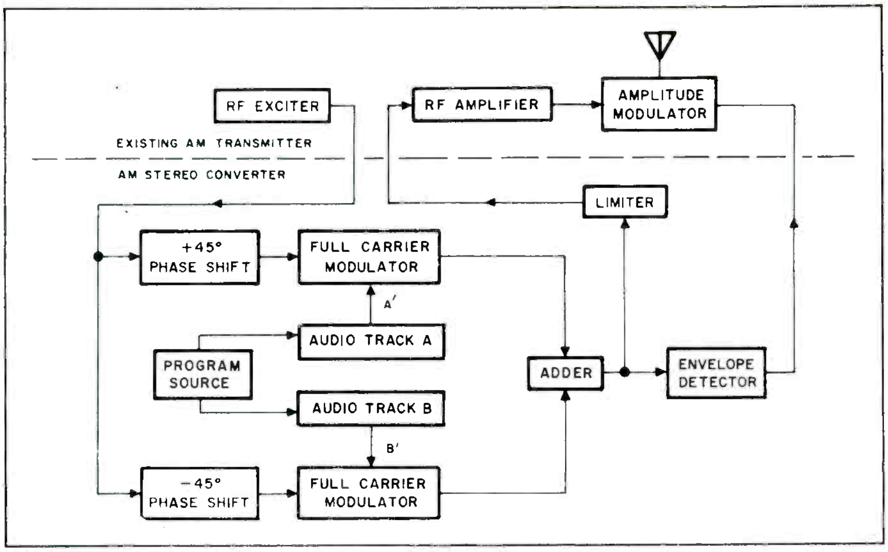

Fig. 6 - Stereo transmitter; phase modulated rf and amplitude modulation converter.

that the phase modulation matches the amplitude modulation, the correct stereo signal will be generated. It should also be noted that, by the use of double heterodyne methods, such a piece of conversion equipment may be made to operate at any frequency in the broadcast band. It is this type of transmitter conversion unit that has been employed at WABC. A simplified block diagram is shown in Fig. 6.

A mathematical description of the phase and amplitude method of modulation may be made. The resultant transmitter output may be expressed as:

$$\text{Output signal} = C(1 + m)\cos(\omega_0 t + \phi)$$

where m = correct envelope modulation and
ϕ = phase deviation of carrier after limiting.

This may be expanded as:

$$\text{Output signal} = C(1 + m)\cos\phi\cos\omega_0 t$$

$$+ C(1 + m)\sin\phi\cos(\omega_0 t + \pi/2)\ .$$

From a vector diagram, one may write the expression for the phase modulation as:

$$\phi = \cos^{-1}\frac{1 + \frac{A+B}{2}}{\sqrt{\left(1 + \frac{A+B}{2}\right)^2 + \left(\frac{A-B}{2}\right)^2}}$$

$$= \sin^{-1}\frac{\frac{A-B}{2}}{\sqrt{\left(1 + \frac{A+B}{2}\right)^2 + \left(\frac{A-B}{2}\right)^2}}$$

where $A = m_a\cos\omega_a t$ = audio of track A and

$B = m_b\cos\omega_b t$ = audio of track B

so

$$\frac{A+B}{2} = m_\Sigma\cos\omega_\Sigma t$$

and

$$\frac{A-B}{2} = m_\Delta\cos\omega_\Delta t\ .$$

From the same diagram it may be shown that:

$$(1 + m) = \sqrt{2}\sqrt{\left(1 + \frac{A+B}{2}\right)^2 + \left(\frac{A-B}{2}\right)^2}\ .$$

Making these substitutions, one may write:

$$\text{Output signal} = \sqrt{2}\,C\left(1 + \frac{A+B}{2}\right)\cos\omega_0 t$$

$$+ \sqrt{2}\,C\left(\frac{A-B}{2}\right)\cos(\omega_0 t + \pi/2)$$

which may be seen to be identical to one of the previous expressions for the transmitted stereo signal. This demonstrates the need for delay equalization of the audio and limited rf paths in this type of transmitter in order to eliminate the generation of higher order harmonics. It also shows that all three methods of making up the signal yield identical results.

Various modulated waveforms are shown in Fig. 7. The time base is taken at carrier rate, showing about 2 1/2 carrier cycles. Figure 7a shows the effect on the carrier of 50 per cent amplitude modulation. There is no phase modulation so a crossover occurs along the horizontal axis. Figure 7b shows the effect of 50 per cent modulation of difference signal only and clearly demonstrates the phase modulation of the carrier about the average carrier phase. The crossovers appear at the positive and negative peaks of the carrier sinusoid. The phase deviation is ± 26 1/2

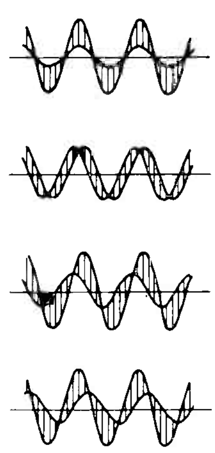

Fig. 7 - Modulated rf waveforms.

degrees. Figure 7c demonstrates the resulting phase and amplitude modulation when applying full audio to the B track only. Note that the cross-overs occur at points leading the peaks of the carrier by 45 degrees. In Fig. 7d, where only A track modulation is used, it may be seen that the null points are 45 degrees lagging the carrier peaks. Note that a null in one track corresponds to a maximum in the other. Herein lies the ability to separate the two tracks, one independent of the other.

In general, the two audio tracks are handled as two completely separate tracks. There is one exception to this statement, however. The frequencies below 300 cps are common to both tracks since, as it has been established, there is no subjective stereophonic effect at very low frequencies.

The audio matrix should be such as to produce the following desired effects at the receiver output. If the audio input to both tracks is identical and constant full amplitude at all audio frequencies, then the output of the receiver speakers should be identical and constant full amplitude at all frequencies. If, however, only one track is supplied with a constant full level at all frequencies, then the combined receiver speaker outputs should be constant, but at half amplitude. At the high frequencies, all the sound should come from one audio channel. But at low frequencies, all the sound should be equally divided between the two channels.

If two full carrier modulators, as shown in Figs. 3 and 6 are used, then full sound level at the source must produce 100 per cent modulation of each carrier. If one track is now turned off, the modulator associated with that channel will be 50 per cent modulated at low frequencies and 0 per cent modulated at high frequencies. The other modulator, for these input conditions, will be 50 per cent modulated at low frequencies and 100 per cent modulated at high frequencies. At low frequencies the resultant carrier will clearly be 50 per cent modulated, but at high frequencies only the average resultant will be 50 per cent modulated. This may be seen by noting the projections from the maximum and minimum instantaneous values of carrier onto the axis of the average carrier.

If, as in Fig. 5, the combination of full carrier and suppressed carrier modulators is used, then full sound level at the source will produce a 100 per cent modulated full carrier and zero suppressed carrier at all frequencies. Now, when one channel is cut off, the full carrier will be modulated 50 per cent at all frequencies. The suppressed carrier, however, will be modulated 0 per cent at low frequencies and 50 per cent at high frequencies.

Figure 8a shows a block diagram of an audio matrix that may be used with the transmitter as shown in Figs. 3 and 6. If the transmitter is the type shown in Fig. 5, then the corresponding audio matrix is diagrammed in Fig. 8b. All the filters indicated are single time constant rc networks with a -3 db cutoff at 300 cps.

Stereo Signal Characteristics

The AM stereo signal as described above has a number of characteristics that warrant further discussion. One of the objectives was to generate a signal that would produce stereo within the existing broadcast channel bandwidth. It is emphasized that a signal containing both AM and PM of the character described has sideband distribution as in a monaural AM broadcast. The maximum sideband frequencies generated become obvious upon inspection of the full quadrature carrier description of the signal. Since it has been shown that the other methods of generating the signal are identical, then any statement made concerning one system is applicable to any other.

If a full carrier is amplitude modulated in the conventional manner, its maximum sideband frequency is proportional to the maximum frequency of modulation. Thus, the spread of sideband energy generated in either quadrature carrier may be controlled by limiting the audio frequencies in each track. The stereo signal is now produced by the linear addition of these two amplitude modulated signals. It is a fundamental theory of network analysis that a linear addition of any number of signals cannot cause the generation of any new frequency components. If the signal generated by each track separately modulating each quadrature carrier is held within channel limits, then the sum of these two signals will also lie within these limits. The AM stereo signal will not cause any additional sidebands outside of a monaural broadcast channel.

Another characteristic of the stereo signal is the sideband power. Each track contains equal sideband energy when modulated to the same depth. Likewise, if the sum and difference audio signals are of equal magnitudes, then the sum and difference sidebands will contain equal energies. Thus the signal-to-noise ratio of one track as compared to the other, or of the monaural as

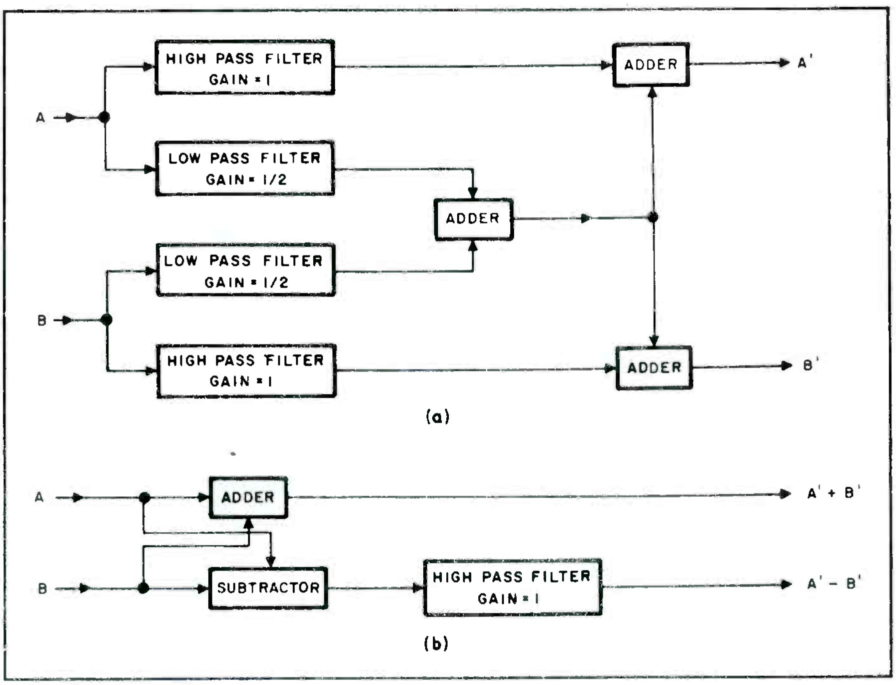

Fig. 8a - Audio matrix for two full carrier modulators.
b - Audio matrix for a full carrier and a suppressed carrier modulator combination.

compared to the stereo portion of the transmission, will be equal. As may be seen from the description of the signal makeup, the transmitter will operate at full carrier and modulation levels. Thus a station broadcasting a stereo signal sustains no loss of coverage.

It has been shown that the average carrier is modulated with the undistorted sum of the two audio tracks. However, when phase modulation occurs the envelope as detected by the usual receiver-type diode detector will contain some distortion terms. Note that the transmitted signal may be written as:

$$\text{Resultant carrier} = \sqrt{2}\,C\cos\omega_0 t\,(1 + m_\Sigma\cos\omega_\Sigma t) + \sqrt{2}\,C\sin\omega_0 t\left[m_\Delta\sin(\omega_\Delta t - \pi/2)\right] .$$

For small values of m_Δ, the second term may be neglected. The output of an envelope detector will then be:

$$\text{Voltage} = \sqrt{2C^2(1 + m_\Sigma\cos\omega_\Sigma t)^2} = \sqrt{2}\,C + m_\Sigma\sqrt{2}\,C\cos\omega_\Sigma t .$$

For large values of m_Δ, the second term may not be neglected and will cause an output of higher order harmonics.

Maximum harmonic distortion will be generated when the phase deviation is greatest. This occurs for full signal level in one track and zero signal in the other. The distortion is demonstrated in Fig. 9 where the amplitude extremes of the resultant carrier are shown for an assumed quadrature carrier amplitude of unity. This amplitude varies from 1 to $\sqrt{5}$, its quiescent point being $\sqrt{2}$.

This means that the negative peak of modulation reaches a value 0.41 units below the unmodulated carrier level and the positive peak reaches a value 0.82 units above the unmodulated carrier level. This inequality of positive and negative peaks indicates the distortion. In the case shown, the total harmonic distortion may be as great as 16 2/3 per cent. Figure 10 is the waveform of such a signal using an audio time base. The distortion may be clearly seen.

Although measurable under laboratory test conditions, this sustained modulation condition has not been found to exist during tests using a wide variety of program material. When signal statistics are applied to test conditions, one finds that specific signal conditions are transitory in nature and make the recognition of this distortion of the carrier envelope extremely difficult. There are a number of reasons why such distortion is not noticeable:

1. The signal is so made up that the frequencies below 300 cps are common to both audio tracks. This means that no phase modulation of the resultant carrier, and hence no distortion, can exist at low frequencies.

2. The distortion is all of the form of even-order harmonic type, primarily second harmonic. Since most AM broadcast receivers limit their audio response to 7.5 kc or less, the harmonics generated by any frequency above 3.75 kc will not be heard.

3. Due to the phase characteristic of the filter producing common low frequencies, the locus of the resultant carrier at mid-range audio frequencies will form an elipse instead of a straight line. It may be shown that this constitutes a reduction of distortion.

4. As the audio track separation is decreased, so the distortion is decreased. It is obvious that on a statistical basis, program material will very seldom contain all its information in only one track, particularly for any length of time and at full level.

5. The average energy content at high frequencies is considerably less than at low frequencies. Accordingly, mid-range and high-frequency distortion will be small since high modulation percentage will almost never be found in normal program material.

Taking all these factors into account, and particularly items 1, 4, and 5, one finds that such distortion is not of consequence in the proposed AM stereo system, and full AM receiver compatibility has been verified by field tests.

Receiver Design Goals

What, then, are the design goals of an AM stereo radio receiver? First, the receiver should be capable of reproducing either a monaural or a stereo signal with adequate fidelity and noise performance. Adequate performance means that the stereo radio should be comparable with conventional home radios that fall in an equivalent price bracket. The stereo receiver should not require any extraordinary limits on overload or drift of the circuits. The most difficult goal to achieve is to make a stereo radio that has the same tuning characteristics as a conventional home radio. It should not produce any annoying noises or squeals when tuned and should not require critical adjustment to obtain a stereo effect. Finally, it should be possible to build a stereo radio for a reasonable cost. Taking these factors into consideration, a description of a stereo radio follows.

The stereo receiver design selected for field testing consisted of a conventional table model home radio to which a stereo adapter was added. Thus, a two unit receiver resulted, as shown in Fig. 11, in which the rf-IF and one audio track amplifier were contained in one unit, while the stereo detector and the other audio track amplifier were enclosed in a companion unit with a 10 foot cable connecting the two. The units are designed to be placed about six or eight feet apart for the best stereo effect. The cost of the adapter is comparable to the cost of the receiver.

The table model radio had no major circuit alterations or retunings. A plug was provided on the back to feed ac power and IF signal through the cable to the adapter, and to return one audio track to the receiver. The ac power was taken from the radio after the power switch so that this switch would control both units. The IF signal was supplied from a capacity tap on the primary of the final double-tuned IF transformer. This permitted a low impedance drive for the cable so as to prevent IF detuning when the adapter was plugged in. The final connection was audio returned to the top of the volume control from the adapter. The envelope detector was disconnected from this point in the receiver, but was still used to supply agc to the set. These constituted the only alterations to the radio.

The adapter must be capable of taking the composite IF signal and breaking it down into the

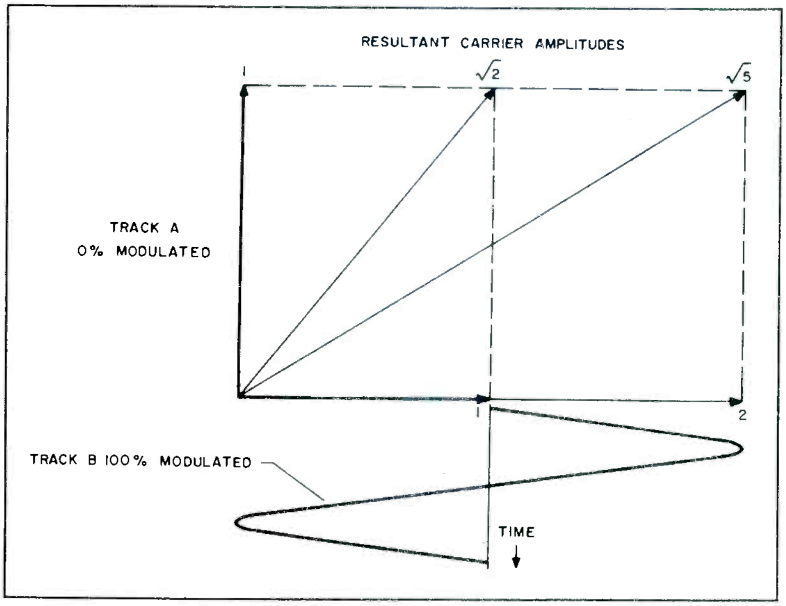

Fig. 9 - Signal conditions causing maximum distortion.

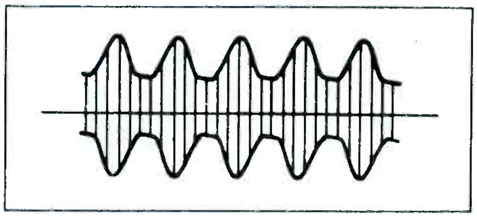

Fig. 10 - RF envelope showing one track modulation.

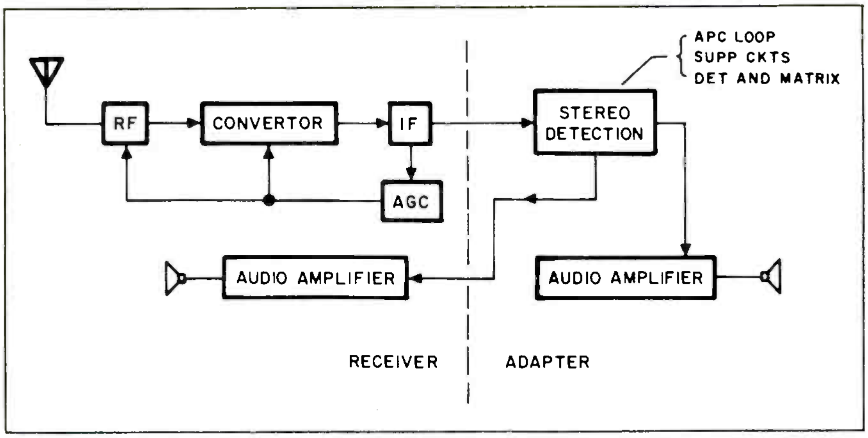

Fig. 11 - Receiver-adapter functional breakdown.

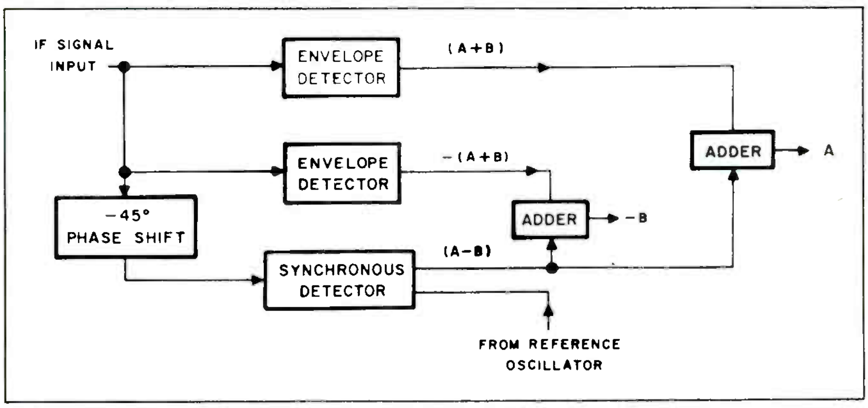

Fig. 12 - Audio detection and matrix.

two separate audio tracks on a stereo signal. It must also be capable of supplying the same signal to each audio amplifier when receiving a monaural signal. Two basic types of detectors are available; i.e., envelope and synchronous detectors. The envelope detectors may be made to recover the sum signal in either polarity, A+B or -(A+B). If one envelope detector of each polarity is used and a synchronous detector supplies the difference signal, A-B, then by means of simple addition the individual A and B tracks may be recovered. This was the basic form of the detector system employed as shown in Fig. 12.

The process of synchronous detection requires the mixing of the modulated carrier with an unmodulated carrier of identical frequencies and in some fixed phase relation with each other. Such a detector, when applied to the AM stereo signal, may be used to recover the audio components that are in phase with the reference cw carrier, at the exclusion of the components in quadrature with this reference carrier.

Since the difference signal is to be detected by means of a synchronous detector, a source of unmodulated IF carrier must be supplied. Such a source can be obtained from a standard automatic phase control servo loop operating on a reference oscillator. (See Fig. 13.) The apc loop contains a phase comparator whose output is dependent on the phase difference between the IF signal and the reference oscillator inputs. The phase comparator output, after proper filtering, controls a reactance tube placed across the reference oscillator tank circuit. Thus, the reference oscillator becomes locked in frequency and phase with the average IF carrier and may then be used for the purposes of synchronous detection.

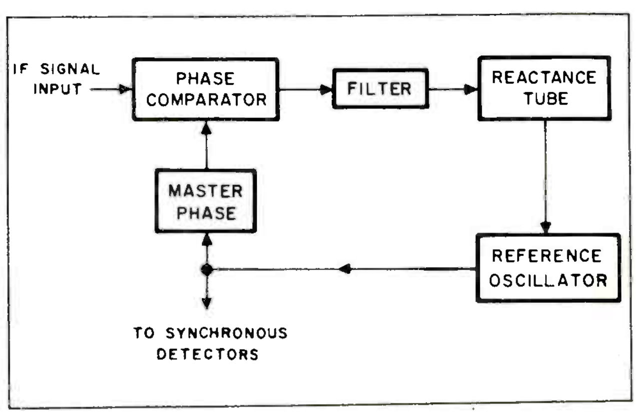

Fig. 13 - Automatic phase control loop.

The apc loop has a hold-in range of at least ±10 kc. This range is dependent only on the product of the sensitivities of the reactance tube and the phase comparator. It is felt that this range is great enough to hold the reference oscillator in lock over the range of the receiver local oscillator drift as well as to keep the static phase error down to ±7 degrees. This small a static phase error is required to hold at least 18 db separation between the audio tracks. The maximum static phase error is proportional to the ratio of hold-in to pull-in ranges.

The pull-in range is dependent on the filter in the output of the phase comparator. The filter used gives a pull-in range of ±1 1/2 kc. This range is kept as large as possible in the interest of producing an easily handled tuning characteristic. A narrow pull-in range means that the receiver must be tuned very slowly in order to lock on a signal. A wide pull-in range means that the output from the phase comparator will follow low audio-frequency, phase modulation of the carrier and thus stereo separation will be lost. This action will also cause low-frequency audio distortion. Since the transmitted signal contains no stereo information below 300 cps, a compromise filter may be designed. Making use of a double time constant filter helps increase the pull-in range without causing distortion.

The synchronous detector used to procure the difference audio signal will generate an annoying squeal when tuning the receiver before the reference oscillator has a chance to become locked. The squeal is the beat note between the reference oscillator and the incoming IF signal. If no suppression circuit is used, this beat will pass through the two audio amplifiers at full strength, producing a very undesirable tuning characteristic. If a dc bias can be used to cut off the difference synchronous detector whenever the system is out of lock, then the squeal can be eliminated from appearing at the audio outputs. This bias may be obtained from detection of the beat signal when the beat signal is separated from the audio. To remove the audio component from the beat note, a cancellation scheme is used. The cancellation scheme consists of obtaining a sum signal, A+B, from a synchronous detector which will also produce beat when the receiver is out of lock. This signal, when added to the output of the envelope detector of the opposite polarity, -(A+B), will produce a beat note that is independent of audio and may thus be detected. This process is shown in Fig. 14.

The matrix of sum signals from the envelope detector and the synchronous detector will produce a resultant signal consisting of the total harmonic distortion produced by envelope detection. The ratio of beat to this leftover audio component is about 6:1 under worst conditions. This signal may be applied to a tube used as a switching device, which is so biased as to conduct only when beat is present. The output of this switch tube is supplied to a peak detector which in turn supplies a dc control bias to the grid of the difference synchronous detector.

As in all such devices, a switching transient is generated as the receiver is tuned rapidly through a station. Holding the time constant on the output filter of the squelch detector to as low a value as possible will help to make the transient less noticeable. No matter how short this time constant is, however, the fact remains that the process of turning off the synchronous detector will also cause a transient which will generate a short audible noise.

The performance of the stereo receiver as described above is comparable in sensitivity, selectivity, and fidelity to the receiver as used without the adapter. In tuning, the output signal has the same sound characteristics as in a conventional set, except that tuning to obtain stereo is slightly more critical. Locking on a weak station that is adjacent to a strong one poses no problem. However, fading on a weak signal may cause the set to drop out of lock if the set is left in a mistuned condition such as to cause a strain on the apc servo. In general, it was felt that this field-test stereo receiver met the design goals reasonably well. The finished units are shown in Fig. 15.

Construction of a stereo receiver that eliminates the need for squelching action and the residual transient when tuning through a carrier

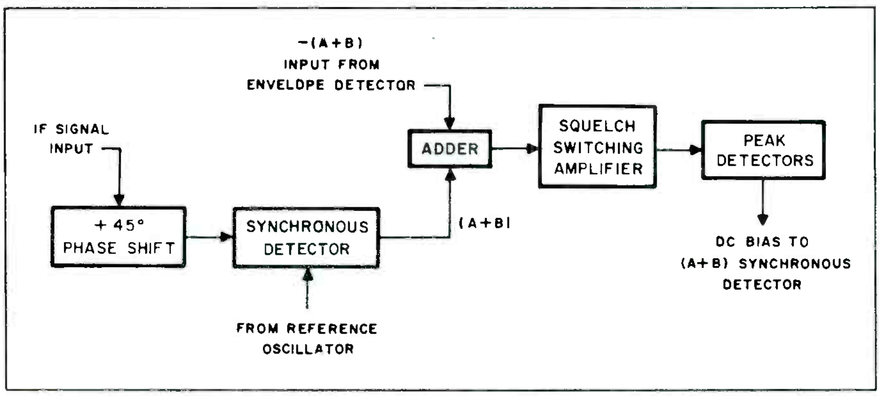

Fig. 14 - Suppression circuits.

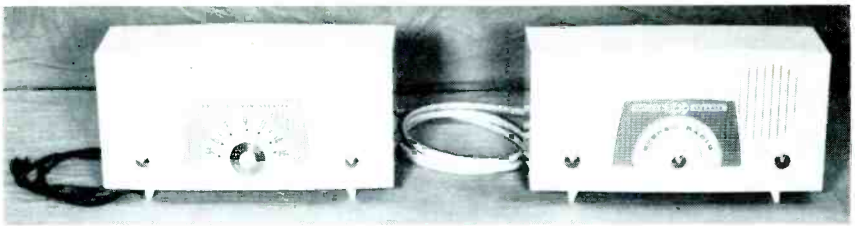

Fig. 15 - AM stereo receiver showing home radio on left and stereo adapter on right.

is highly desirable. To accomplish this, it is necessary to find a type of detector not involving a beat process such as the synchronous detector. A unique and compact circuit that may be used for stereo detection will now be presented. This type of receiver has been constructed and has several advantages in both design and handling characteristics. In brief, the improved stereo radio consists of two slope detectors and a differential amplifier.

The demodulator is a variation of the Travis type discriminator. Two envelope detectors are connected in such a manner that their outputs will track each other on an AM signal. The diodes are driven from two single-tuned circuits, one of which is tuned higher than the IF frequency and the other tuned low. The audio output from the slope detectors is used to drive a dual triode differential amplifier as shown in Fig. 16. If there is no carrier phase deviation (i.e., a monaural signal), then the output of the slope detectors will consist of two audio signals in phase with each other and following the envelope of the input signal.

Since each grid of the differential amplifier receives an identical signal, then the common cathode voltage will represent the amplitude variations of the transmitted signal. This is the sum of the two audio tracks, A+B, in the case of a stereo transmission. The sum signal will not appear across the plate loads of the differential amplifier as long as the cathode load is large with respect to the plate load, since the tubes act as cathode followers with little change in plate current when a signal is applied to the grid.

Phase modulation of the carrier will cause a frequency shift and, therefore, the outputs of the detectors will no longer be identical. One grid of the differential amplifier will be driven positive while the other is driven negative. The resultant current through the common cathode impedance will be constant, and thus no difference signal output will result. The individual tube currents, however, will be proportional to the phase deviation and in opposite polarity. In this case, the phase modulation is proportional to the derivative of the difference signal, A-B.

Thus the plate loads of each section of the differential amplifier are connected to integrators which will produce A-B and -(A-B). If these difference signals are added to the correct amplitude of the A+B signal appearing at the common cathode, then separate A and B signals may be obtained directly. This presents a simple method of detection and matrix of the AM stereo transmission without the use of beat type detectors.

The frequency deviation of the carrier when a difference-signal modulation exists may be determined from the expression for the instantaneous phase of the resultant stereo carrier. The phase may be written as:

$$\beta = \tan^{-1} \frac{1 + m_a \cos \omega_a t}{1 + m_b \cos \omega_b t} .$$

For a monaural signal, when $m_a = m_b$ and $\omega_a = \omega_b$, then $\beta = 45$ degrees. The maximum phase deviation occurs when $\omega_a = \omega_b$ and $m_a = -m_b = 1$. Then $\beta = 45$ degrees ± 45 degrees. In terms of maximum frequency deviation this may be written as:

$$\Delta\omega_{o\ max} = \frac{d\beta}{dt} = \frac{\pi}{4}\omega_{max}$$

where ω_{max} = highest audio frequency used for modulation. This also demonstrates the fact that the output voltage of a slope detector will be proportional to the derivative of the phase modulation of the carrier.

It should be noted that the A+B signal and both the difference signals when derived in this manner will contain some distortion products which cancel when properly matrixed. On a monaural signal, no distortion will occur and the identical outputs will be obtained from both A and B channels.

To obtain the desired audio outputs, it is necessary for the IF carrier to fall at the crossover points of the slope detectors. Should this condition not be satisfied, then the audio recovered from one slope detector will be too large, while the other output will be too small. In the case of a stereo signal, the result would be the loss of track separation due to the addition of sum and difference signals of improper amplitudes. Since no track separation exists below a modulation frequency of 300 cps, due to the nature of the transmitted signal, the IF carrier need not fall closer than 300 cps away from the detector crossover. However, more accurate tuning than the customer could perform would still be required and therefore an automatic tuning aid is needed. Such an aid may be obtained with ease from the slope detectors.

In series with each of the tuned circuits used to drive the envelope detectors is an additional dc load, used to produce a detector output of the opposite polarity from the first-mentioned loads. By adding loads of opposite polarity, a discriminator response may be obtained, and the output voltage will be a function of the input frequency.

If the resonant point of each tuned circuit is symmetrical about the center frequency (of the IF), then the slope about the crossover point is the average slope of the two responses. Since these responses are symmetrical and have the same slope at the crossover frequency, then the resultant slope is the slope at that point of a single response. It may be shown that the discriminator sensitivity may be increased by decreasing the pole separation (to a limited point) or the capacity, or the damping factor. The actual discriminator sensitivity in volts/cps may be calculated or measured.

The sensitivity of this discriminator will be derived. The output voltage from one detector as a function of frequency is proportional to the slope of the absolute magnitude of the impedance which may be written as:

$$\left.\frac{d/z/}{d\omega}\right/_{n\sigma} = \frac{n}{(1 + n^2)^{3/2}} \cdot \frac{1}{\sigma^2} \cdot \frac{1}{2C}$$

where $/z/$ = absolute value of impedance of single tuned circuit
ω = angular frequency
C = total capacity in single tuned circuit
σ = damping factor = $\frac{\omega_o}{2Q}$
n = any value from 0 to ∞.

Let the amount of detuning = δ = $n\sigma$. Then

$$\left.\frac{d/z/}{d\omega}\right/_{\delta} = \frac{\delta}{2C(\sigma^2 + \delta^2)^{3/2}} .$$

Maximum slope occurs at:

$$\delta = \frac{1}{\sqrt{2}}\sigma .$$

The crossover point of this discriminator may be varied by controlling the signal level supplied to each of the tuned circuits. This is done by driving the circuits from the outputs of a differential amplifier whose gain is controlled by the discriminator output as shown in Fig. 16. The gain ratio required for any given amount of detuning from center frequency may be expressed as:

$$k = \sqrt{\frac{1 + \left(\frac{\omega_o - x}{\sigma}\right)^2}{1 + \left(\frac{\omega_o + x}{\sigma}\right)^2}}$$

where $k = A_1/A_2$ = gain ratio of differential amplifier
σ = damping factor of tuned circuits
ω_o = center frequency
x = amount of movement of crossover point.

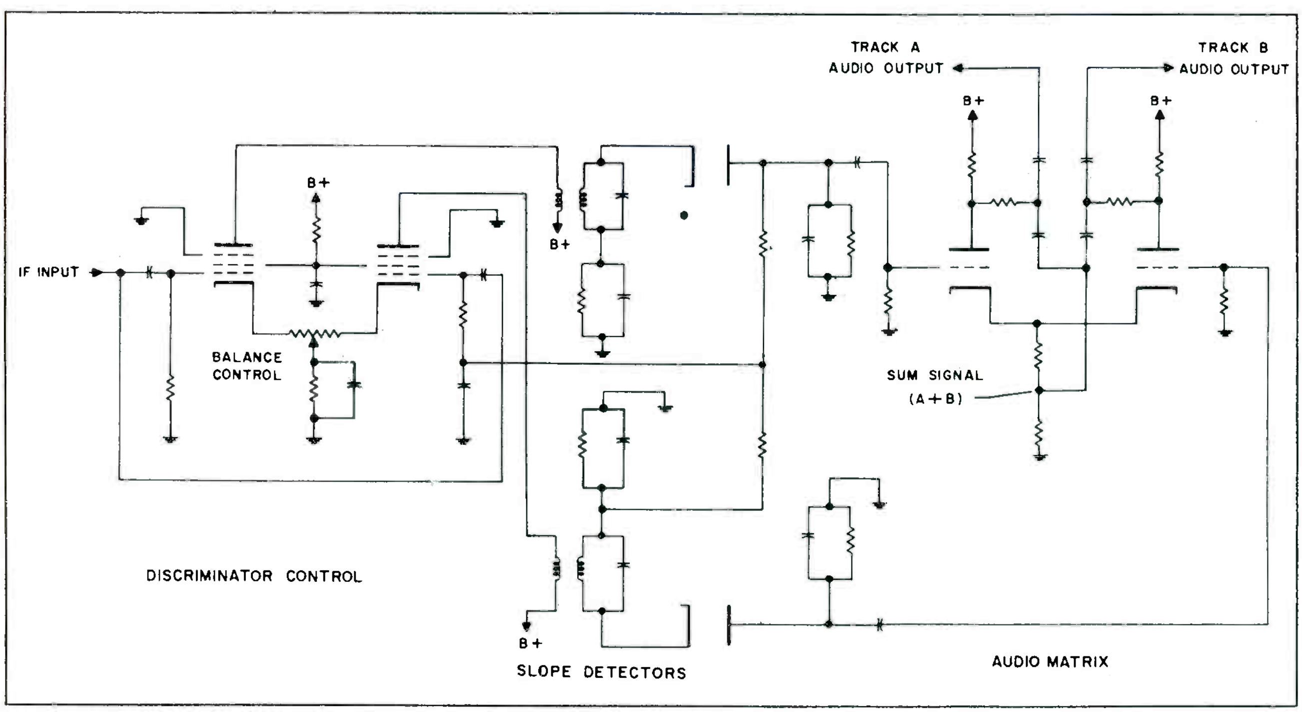

Fig. 16 - Schematic of slope detector type of AM stereo receiver.

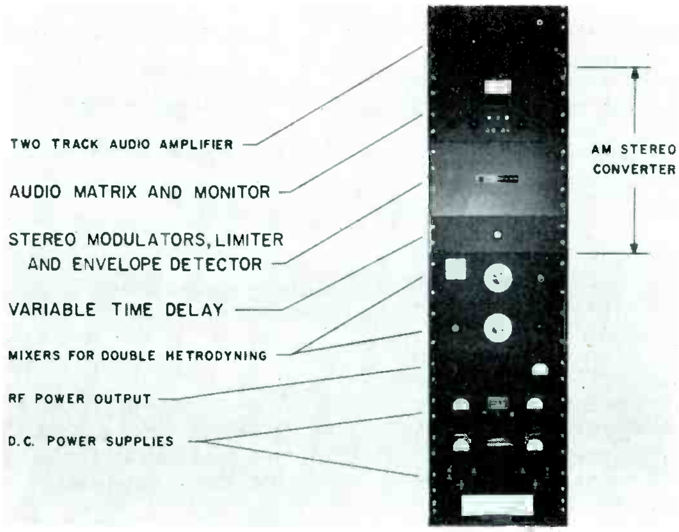

Fig. 17

Note that if $x = 0$, then $k = 1$.

It has been assumed that

$$\sigma = \frac{\omega_{o1}}{2Q_1} = \frac{\omega_{o2}}{2Q_2}$$

and

$$\frac{\omega_{o1} + \omega_{o2}}{2} = \omega_a$$

where ω_{o1} = angular frequency of resonance of one tuned circuit of Q_1
ω_{o2} = angular frequency of resonance of other tuned circuit of Q_2
ω_a = crossover frequency when $k = 1$.

The control sensitivity is a function of the gain of the differential amplifier and may be determined by measuring the slope of the curve of crossover frequency as a function of control bias. This sensitivity may be expressed in dimensions of cps/volt.

The servo loop gain may also be calculated by forming the product of the control sensitivity and the discriminator sensitivity.

$$G = S_D \times S_C$$

where S_D = discriminator sensitivity in volts/cps
S_C = control sensitivity in cps/volt.

Once the loop gain is known, the crossover error for a given amount of receiver detuning may be determined:

$$x = \frac{\Delta}{1 + G}$$

where x = error in cps between crossover and incoming frequency
Δ = amount of detuning in cps of incoming frequency from center frequency of IF.

The discriminator response is essentially linear over a bandwidth of 10 kc which is sufficient to allow for 1 or 2 kc detuning of the receiver. The servo loop gain is sufficient to hold the crossover point close enough to the incoming signal frequency to produce good track separation for these tuning conditions.

The gains of the differential amplifier vary linearly over a range of ± 0.7 volt about the quiescent point. This is sufficient to vary the gain ratio, k, from 1.0 to 2.7. The dc gain of the differential amplifier is 1/2. A difference in g_m of either tube will cause a large difference in gain between the two tubes. To allow for the use of tubes with different g_m's, the cathode impedance may be varied so as to adjust the gains of each amplifier to be equal at the IF.

The tuning characteristic of this receiver is almost identical to that of a conventional home radio. While tuning through a station, the sound intensity from one speaker predominates upon approaching correct tuning. When tuned to center frequency, the sound level in each speaker is the same. If the signal is stereophonic, the sound appears to spread between the speakers. As tuning is continuted in the same direction, the sound from the other speaker predominates. There is no indication of pulling into or out of a locked condition. The tuning effect is smooth and not unusually critical for adjustment to either a monaural or a stereo broadcast. It is felt that this characteristic satifies the original receiver objectives and leads to a practical design.

Field Tests

Field tests were conducted in conjunction with WABC in New York. The terminal equipment used at the transmitter site will be described in detail in a forthcoming paper, but, as stated previously, the phase and amplitude concept was used to convert WABC transmitter to stereo. Figure 17 shows the conversion unit. All measurements indicated the transmitter performed as predicted. Numerous stereo receivers of the adapter concept were available for field evaluation of the "on the air" signal.

The main receiver field test location was in Philadelphia some 75 miles airline from WABC. At this distance, stereo reproduction was, as predicted, quite excellent. The stereo receivers were equipped with loop antennas which enabled them to be orientated for a null to check weak signal conditions. Good stereo was received under fringe conditions. There was no detectable loss in signal-to-noise ratio between stereo and monaural broadcasts.

Near locations reported no increased spurious outputs from the transmitter when broadcasting stereo and very good stereo reception under strong signal conditions. Studio program lines were used for some of the experimental testing with no noticeable loss of stereo separation. Monaural receivers reported no degradation of fidelity, range, or signal-to-noise ratio in fringe locations. Strong signal reports were excellent.

In conclusion, it is felt due to the excellent field results, all of the original objectives of the transmission and receiving systems have been met. Information from the experimental field test has been made available to the FCC and the NSRC. Technical progress has been such that AM stereo should become a reality in the near future.

Acknowledgment

The authors would like to express acknowledgment of the work done by a number of other people on the AM stereo broadcast project. At Philco Corporation the original suggestion for the project came from W. P. Boothroyd. Guidance in various areas came from R. C. Moore and E. M. Creamer. Many other people helped construct and test equipment. All field testing was accomplished with the full cooperation of F. Marx and R. M. Morris of the American Broadcasting Company and N. Hagmann and his staff at WABC. The authors are very grateful to all these people.

A COMPATIBLE STEREOPHONIC SOUND SYSTEM*

Floyd K. Becker
Bell Telephone Laboratories,
Murray Hill, N. J.

The "new sound" of high fidelity stereophonic sound is far from new. At least as early as 1881 a demonstration of an electrical transmission of binaural sound, a close cousin of stereophonic sound, was made at the Paris Opera. That demonstration used two widely spaced microphones connected to a binaural headset. In 1925 the New Haven radio station, WPAJ, made binaural broadcasts by employing two separate AM transmitters on different wavelengths. Two standard studio microphones placed 7 inches apart originated the binaural signals.

In 1933 the Bell Telephone Laboratories culminated a series of auditory perspective tests in a 3-channel system demonstration. Under the auspices of the National Academy of Sciences a concert of the Philadelphia Orchestra was transmitted from the Academy of Music in Philadelphia to Constitution Hall in Washington. The orchestra was conducted by Associate Conductor Smallens while Dr. Stokowski, the Director, manipulated electrical controls from a box in the rear of Constitution Hall.

Three microphones were placed before the orchestra, one on each side and one in the center at about 20 feet in front of and 10 feet above the first row of instruments of the orchestra. The microphone outputs were transmitted to Constitution Hall in Washington by three separate electrical circuits tailored for the occasion. At Constitution Hall these transmission lines were connected to power amplifiers. The associated loudspeakers were placed on stage in positions corresponding to the microphones in the Academy of Music in Philadelphia. The comments of those who heard this reproduced concert proclaimed the development of a system with possibilities for even greater emotional appeal than that obtained when listening to the orchestra "live". Much of this reaction was undoubtedly due to the enhanced volume range.

In 1936 the Bell Telephone Laboratories produced disk recordings of 2-channel stereo. In 1939 the N. V. Phillips Co. of the Netherlands experimented with stereophonic sound reproduction in large auditoriums. In 1941 the Bell Telephone Laboratories demonstrated sound motion picture recording using 3-channel stereo. This chronology of early stereophonic sound is by no means complete. It only attempts to place a few of the salient efforts at stereophonic sound transmission and/or reproduction.

Radio broadcasting of stereophonic sound programs by two separate channels became popular in 1952-4. In various experimental arrangements, the two channels required have been selected from different combinations of AM, FM and TV. The listener spacing appropriate recievers properly in his home. Results have been sufficiently favorable that more broadcasters are considering offering stereophonic sound programs. Many such programs have been originated on the national networks.

The major obstacle to vastly increased use of this type of stereophonic broadcasting, however, is the person who listens with only one receiver. If the broadcaster tries for the full stereophonic effect, the sound the single channel listener hears comes from only one of two widely spaced microphone pickups, and he misses a portion of the program. The effect in many cases is similar to listening to one-half of an orchestra or to one side of a two-way conversation. What the single channel listener does receive is poorly balanced, because of the microphone placement in relation to the sound sources. The broadcaster, in order to protect investment and sponsor revenue, has had to dilute the stereophonic effect in order to preserve satisfactory reception for the single channel listeners.

Most of the effort to produce a compatible stereophonic sound system has been directed at single channel systems. These systems generally comprise a frequency or time multiplexing of the stereophonic channels on a single carrier. Most of these systems are, indeed, compatible with present day single channel receivers but require additional equipment - other than AM and FM receivers - to produce stereophonic sound.

One solution to the 2-channel problem is gained through the use of a compatibility circuit developed at the Bell Telephone Laboratories. This circuit, equally adaptable for a 2- or 3-channel system, depends for success

*Reprinted from AUDIO, May, 1959

upon a psychoacoustic phenomenon known as the Precedence Effect.

Before discussing this effect it would be well to review the principles of sound localization. The localization of a sound source with respect to the observer requires 3 coordinates: the radial distance, the altitude angle and the azimuth angle. Man's auditory perception of distance seems to be primarily governed by the loudness and ratio of direct to reverberant sound. There is little or no altitude perception. Azimuth localization is extremely good. Accuracies of about 2 degrees are average. The mechanisms for azimuth detection are (1) phase differences between sound waves at the two ears; (2) differences in the time of arrival of transient sounds; (3) differences in intensity at the two ears due to shadowing by the head, which also has a frequency dependence and will result in an interaural quality difference. Azimuth localization of pure tones is possible only in areas approximating free-space. In reverberant rooms, even those with optimum reverberation time for music reproduction, the standing wave patterns destroy the sense of directivity. Hence the arrival time an d intensity differences of transients assume the predominant roles in determining azimuth in ordinary listening environments.

It is possible to trade loudness differences for arrival time differences. An observer seated before two inphase loudspeakers which are separated by several feet will gain the impression of a single, centrally located source if the two loudspeakers have the same loudness. If the loudness of one speaker increases while the other correspondingly decreases, the apparent source will shift toward the more intense loudspeaker. The amount of the apparent shift depends upon the sound intensity at the ears. If the sound levels are the same in both speakers, but a time delay is introduced in one source, the apparent source will shift toward the undelayed speaker. Time delays as short as 250 microseconds produce considerable shifts in the apparent source. These time delays are of the same magnitude as the time difference of transient arrivals at the two ears for a sound source to the observer's side. When the time delay in the one speaker system is increased to 2 or 3 milliseconds the delayed source must be intensified until it is 10 times more powerful than the undelayed source before the observer will detect it to be of the same loudness. This condition holds while the delay is increased to about 35 milliseconds. In the neighborhood of 35 milliseconds time delay the observer begins to detect a distinct echo. It is the reaction of the azimuth localization mechanism in the region of 1 to 30 milliseconds that is called the Precedence Effect. In this region, the localization of a sound source is determined by the direction of the first arriving sound. The later arriving echoes are virtually disregarded. This may at first seem an amazing phenomenon but a closer examination discloses that it is at least an every day experience. In the average indoor environment, the bulk of the sound power reaching a listener arrives by way of reflections or echoes. Yet, in this same environment the listener has no difficulty localizing the sound source.

Now to turn to Figs. 1 and 2 and an explanation of this form of a compatible stereophonic sound system. Referring first to Fig. 1, the circuits between the microphone pickups and their corresponding radio or TV transmitters are cross connected through two delay lines, each with its own buffer amplifier. Because of these cross connections, music or voice signals from the left microphone are transmitted directly to the left loudspeaker in the listener's home, while the same signal is slightly delayed before reaching the speaker to his right. The stereo listener will hear the sound as if it came only from the left loudspeaker because of the Precedence Effect. Conversely, the sound from the right microphone goes direct to the right speaker, but is delayed before reaching the left speaker, and is therefore unheard in the left speaker. Thus, the stereo listener localizes

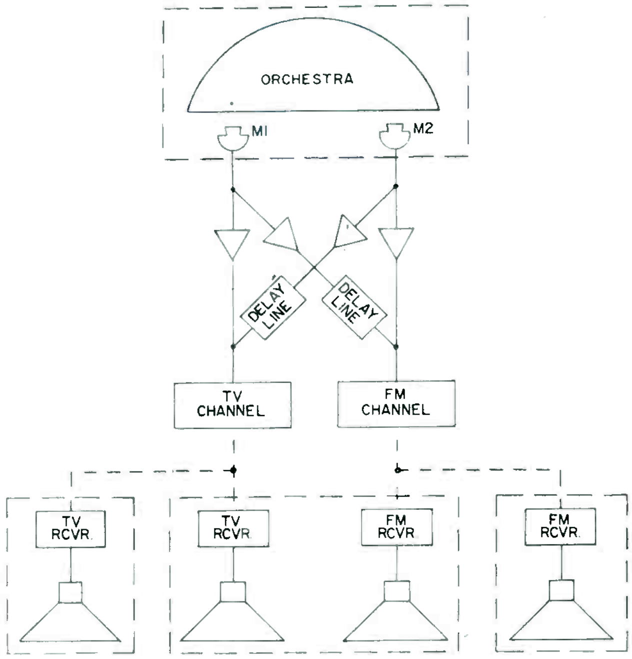

Fig. 1.

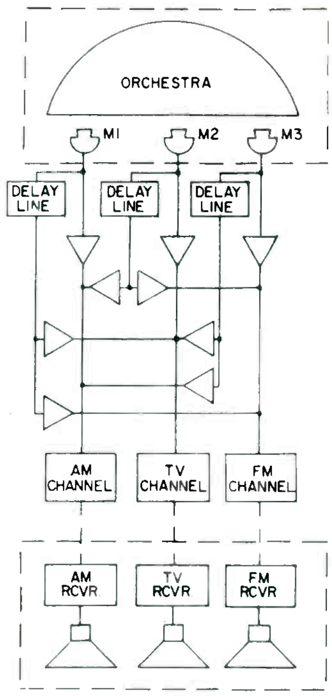

Fig. 2.

the sound he hears as coming direct from each of his two speakers, and full stereophonic effect is maintained.

However, monophonic reception is completely compatible with this, since a listener to each single channel hears the total sound from both microphones in a balanced reproduction. The slight delay of one signal does not affect his reception at all.

The 3-channel system of Fig. 2 operates in a similar manner. The direct signal travelling only in the primary channels while a time delayed replica of the direct signal is added to the alternate channels in order to achieve compatibility.

Results of 2-channel subjective listening tests with musical material indicate a preferred time delay of about 10 milliseconds with the intensity of the delayed signal equal to the direct signal. A different set of parameters appear optimum for speech; e. g., 5 milliseconds delay and the intensity of the delayed signal 3 db. less than the direct signal. A tested compromise of 10 milliseconds time delay and 3 db. attenuation yields very good over-all results. The 8 to 10 db channel separation due to Precedence Effect added to the 3 db intensity difference gives results comparable to a 12 db channel separation system.

It might be of interest here to mention some desirable side effects of these circuits. The literature is full of references to the "hole in the middle" and to the subjective reaction to the reproduction of the music of a full orchestra from a box occupying some 2 or 3 cubic feet.

The employment of the Precedence Effect for channel separation causes the area of the apparent sound sources to seem quite large. The sound no longer emanates from the small speaker cabinets but appears to be produced by large area sound sources. The apparent area of these sources greatly diminishes the "hole in the middle" effect and is more suggestive of the appropriate size of loudspeakers required to reproduce orchestral music.

This development should make it possible for many more broadcasters to offer double or triple channel stereo programming without diluting the stereo effect or penalizing the single channel listener, who will make up the majority of their audience.

NEW DIMENSIONS IN SOUND . . . VIA SINGLE A-M BROADCAST CHANNEL*

H. E. Sweeney
Television-Radio Division
Westinghouse Electric Corp.
Metuchen, New Jersey
and
C. W. Baugh, Jr.
WASP
Westinghouse Electric Corp.
Drexel Hill, Pennsylvania

Stereophonic sound - sound with true spatial dimension - is creating a new interest in both a-m and f-m radio. As with the introduction of high-fidelity equipment a few years ago, stereophonic sound has been largely spurred by recent developments in recording techniques.

Many methods of stereophonic broadcasting are under investigation. Most of the methods that have been tried on the air dual-channel systems - two separate broadcasting channels are required, such as am and fm, fm and fm, or tv and am or fm. In effect, the signals from a right-hand microphone are broadcast on one channel, and the signals from a left-hand microphone are broadcast on the other. When these separate signals are recreated in the am and fm receiving sets, spaced some distance apart, the reproduced sound assumes the depth and direction that it had in the recording stage or studio.

However, these dual-channel systems have two major disadvantages: from the broadcasters standpoint, two outlets are consumed by a single program; and from the listener's standpoint, the systems are not entirely compatible. If the listener tunes on only one channel, he will not receive a truly balanced monophonic signal.

The ideal in a stereophonic broadcast system is to provide a stereophonic signal that is completely compatible and that transmits a minimum of information necessary for good stereo presentation. Complete compatibility means not only that a stereo signal may be heard monophonically on a conventional receiver with the minimum degradation, but that a conventional radio signal may be heard monophonically on a stereo receiver. The Westinghouse system approaches this ideal.

Two separate information channels, necessary for stereophonic transmission and reproduction, are broadcast by simultaneous amplitude and frequency modulation of the standard broadcast band carrier. The amplitude of the carrier is modulated with the sum of the signals from the left and right microphones, (L+R). The frequency of the carrier is modulated with difference between the two signals, (L-R). This difference signal is limited to the frequency range of 300 to 3000 cps and contains the stereophonic information.

A specially designed stereophonic receiver will recombine the (L+R) information at its a-m detector with the (L-R) information at its f-m detector in a simple algebraic manner:

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

The L and R signals are applied to the two speakers.

A conventional receiver, tuned to the center of the band and having only an a-m detector, will reproduce only (L+R), which is balanced monophonic program. Furthermore, a stereophonic receiver, tuned to a conventional radio signal that has no f-m information, will produce identical sounds on both of its speakers.

An interesting additional feature of the Westinghouse a-m stereo system is that two standard a-m radio receivers carefully tuned - one slightly on the low side of the a-m channel and the other slightly on the high side - can also provide separation of the L and R signals, and thus produce stereophonic sound.

The new system is compatible with present Federal Communications Commission's standards for a-m broadcasting. The amplitude modulation is essentially that of the normal broadcast band signal; the essential stereophonic information, supplied by varying the carrier frequency, is within the allotted 10-kc channel bandwidth, thereby preventing interference with adjacent a-m channels.

Westinghouse a-m stereo system specifications

A set of specifications were developed for the proposed a-m/f-m signal:

*Reprinted from Westinghouse Engineer, July 1959, pages 122-125.

1. The signal carrier shall be both amplitude and frequency modulated.

2. The amplitude modulation is predominantly proportional to the algebraic sum of the two stereophonic signals (L-left microphone+ R-right microphone) but includes a smaller signal that is a function of the stereophonic difference signal (L-R). This function is developed by a compensating system.

3. The maximum amplitude modulation shall be limited to 95 percent.

4. The frequency modulation shall be from the components of the stereophonic difference signals between 300 and 3000 cycles per second. The filter cut-off rate shall be 6 decibels per octave. The maximum deviation shall be 3 kilocycles.

5. When only one stereophonic signal exists, the maximum instantaneous amplitude shall occur simultaneously with the maximum instantaneous frequency deviation of the transmitted signal.

Compensating system

Since the average a-m broadcast receiver has frequency-selective circuits (signal amplification is a function of frequency), the frequency modulated carrier produces some amplitude modulation, or "cross-talk" (Fig. 1a). Furthermore, since for a center-tuned receiver, the resulting amplitude modulation will be predominantly the second harmonic of the original frequency modulation, the effect is actually worse than simple cross-talk, because it introduces harmonic distortion, both in stereophonic and monophonic receivers. Therefore, a compensating system is needed to precorrect the transmitted signal for the average receiver i-f passband. Because passbands of receivers vary widely, only partial compensation can be expected in many receivers, but this has been found satisfactory.

Precompensation can be accomplished with a "notch filter," which has a frequency response that is the inverse of the average receiver passband over the bandwidth to be corrected. Frequency modulation of the oscillator (L-R modulation) causes the signal to sweep across the notch filter response. Hence, the envelope of the signal leaving the notch filter is precorrected so that f-m cross-talk in the a-m channel of the receiver with an average i-f passband is effectively canceled. When L+R is zero, the envelope of the signal at the antenna has the same shape as

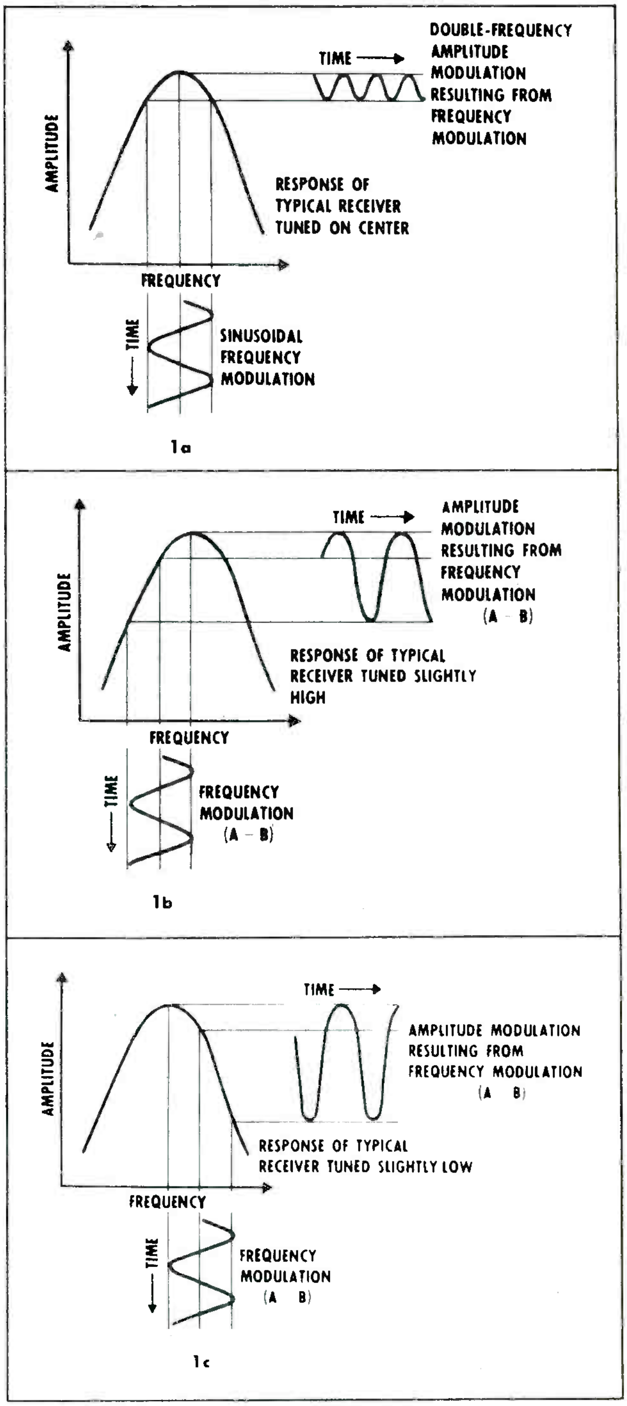

Fig. 1 - (a) Because of the IF response of AM radio receivers, double-frequency amplitude modulation would result from frequency modulation if no precompensator were used at the transmitter. (b) Slope detection enables a standard AM receiver to add or subtract the fm signal from the AM signal. If the receiver is tuned high, the fm signal is added to the AM signal; (c) if the receiver is tuned low, the fm signal is subtracted from the AM signal.

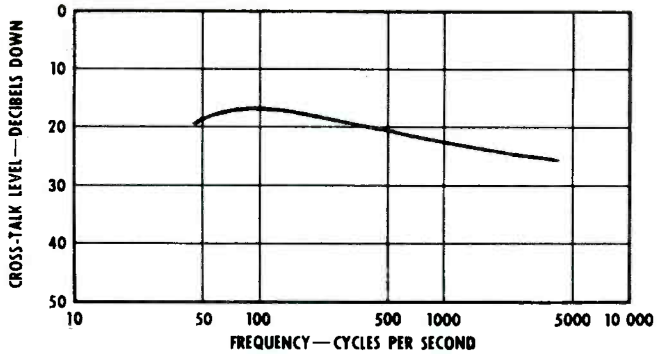

Fig. 2 - Test measurement of the cross-talk of frequency modulation into the AM channel.

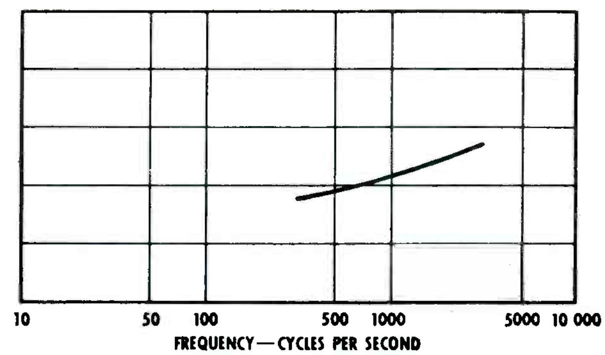

Fig. 3 - Test measurement of the cross-talk of amplitude modulation into the fm channel.

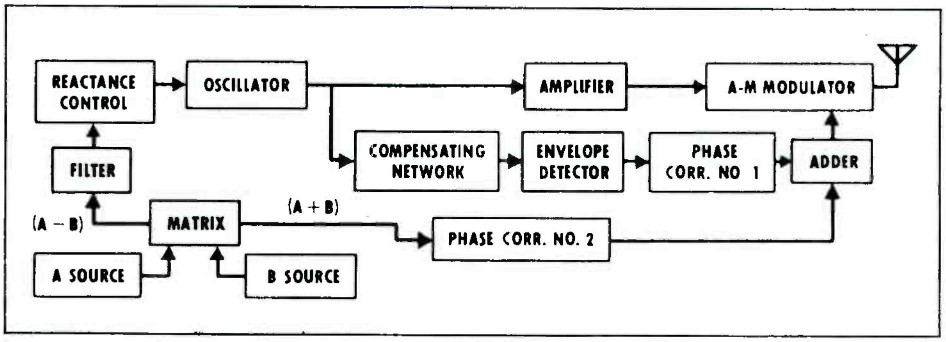

Fig. 4 - Block diagram of AM stereo transmitter.

the envelope of the signal at the output of the notch filter. When L+R is not zero, the envelope at the antennas has approximately the shape of the sum of phase-corrected (L+R) and the phase-corrected envelope of the output of the notch filter.

A phase-corrector network is used to equalize the audio phase shift so that the frequency modulation and the pre-correction envelope modulation are coincident in the radiated signal. A second phase corrector is used to put the (L+R) modulation in time coincidence with the (L-R) modulation in the radiated signal.

A measurement of the cross-talk of the frequency modulation into the a-m channel is shown in Fig. 2. Conversely, a measurement of the cross-talk of amplitude modulation into the f-m channel is shown in Fig. 3. Cross-talk of about 20 db down is usually considered sufficient separation for stereophonic purposes, where the signals in the two channels are not completely distinct.

Transmitter

A block diagram of the complete transmitter is shown in Fig. 4. Neglecting the compensation system, the signal is generated in a straightforward manner. A source of frequency-modulated signal capable of 3-kc deviation at the broadcast frequency is required. This frequency-modulated carrier is the r-f drive for the power amplifier.

The audio frequency bandwidth of the f-m unit (L-R) is limited to 300 to 3000 cps. No changes are necessary in the existing power stages of the transmitter if the frequency response of the amplifier stages is at least 5 kc with good phase response. The amplitude modulation is straightforward, and is no different than in standard practice except for the amplitude limitation of 95 percent, as opposed to a full 100 percent allowed in normal a-m broadcasting. This maximum allowable amplitude modulation is largely determined by economical receiver design. The modulation limit sets the minimum signal level that must be handled by the frequency detection system of the receiver. A maximum transmitter amplitude modulation of 95 percent will not place a severe requirement on the gain required in the f-m channel of the receiver, and is high enough to cause only a small reduction in transmitter coverage.

The frequency response of the amplitude modulation system can be wide as allowed by

FCC standards but should be no narrower than 3 kc for good stereo performance.

The only unusual block in the transmitter diagram is the notch filter, or precompensator. The notch filter can also be placed in the main r-f line at the point where the output of the filter feeds both the envelope detector and the r-f amplifier. This arrangement has the advantage of requiring less delay correction, but puts another unit in the main r-f line.

Monitor equipment is necessary to check the f-m deviation and the compensation system. Both of these functions can be checked with a stable receiver that is aligned to have an "average" passband.

The broadcast transmitter limiters or volume compressors should be modified to limit the L and R signals proportionately. Any gain change in one should be accomplished by the same change in the other to preserve balanced stereo information.

Stereo Receiver

A block diagram of a stereo receiver is shown in Fig. 5. The receiver has a conventional envelope detector for the L+R a-m signal, and an f-m detector for the L-R signal. Several refinements in receiver design have been made. For f-m detection, the signal is first amplitude-limited to a constant value. Because of the 95-percent a-m modulation of the carrier, the amplitude rejection capabilities of the f-m section are extremely important.

For ultimate stereophonic performance, the a-m and f-m signals should be balanced for both weak and strong signals. This requires a much more sensitive automatic gain control system than needed in conventional am receivers.

The master gain control is a ganged pair of potentiometers, one in each audio channel. The balance control is in the L-R channel. When it is set at zero, there is no stereo information. As the (L-R) component is increased, separation of the two channels is effected.

Two systems of matrixing the sum-and-difference signals to recover L and R channels have been tried. Transformer matrixing has the advantage of simplicity. The output windings should deliver balanced L and R signals over the stereo bandwidth of 300 to 3000 cps. A phase splitter triode with a resistive matrix is another method of obtaining a well-balanced output.

Two-receiver method

As previously mentioned, a standard a-m receiver, when tuned to the center of the carrier signal, will produce a balanced monophonic signal, (L+R). However, when a receiver is slightly side tuned, slope detection of the f-m signal (L-R) takes place in combination with amplitude detection of the (L+R) signal. If the receiver is tuned high (Fig. 1b), slope detection will add some of the (L-R) signal to the (L+R) amplitude signal to produce a composite signal predominantly L. A second receiver tuned slightly low (Fig. 1c) reverses the phase of the (L-R) component to produce a signal predominantly R. Using this technique, good quality stereo can be produced from two unmodified broadcast receivers.

Stereo tests

A number of compatible stereo systems have already been announced, and more are being developed. As a result, an all-industry

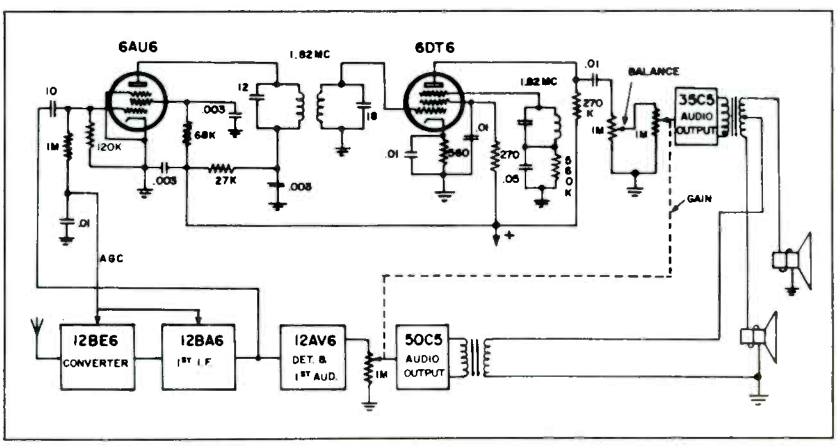

Fig. 5 - Schematic of stereo receiver.

National Stereophonic-Radio Committee has been formed by the Electronic Industries Association. The committee's purpose is to study the various transmission systems and to advise the Federal Communications Commission on the characteristics of each.

Westinghouse pioneer radio station KDKA, in Pittsburgh, has already conducted a number of stereophonic test broadcasts. With FCC permission, KDKA is presently preparing to test the a-m/f-m duplex system described in this article. The results of all these tests are being filed with the FCC.

These latest tests will give both the listener and the broadcaster a chance to evaluate the advantages of the Westinghouse a-m stereo system. Only minor modifications are required in the transmitter to broadcast stereo information. Stereo receivers required for this system are relatively simple and inexpensive.

Stereo reception using two ordinary a-m receivers is also being evaluated. During these tests, no degradation of the program will result for the listener using a single broadcast receiver.