

[54] **QUADRAPHONIC RECORDING SYSTEM**

3,632,886 1/1972 Scheiber..... 179/100.4 ST

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[57] **ABSTRACT**

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Apparatus for recording four separate channels of sound information on the two tracks of a stereophonic disc record for subsequent reproduction on four loudspeakers to give the listener the illusion of sound coming from four separate sources. Two of the four signals are applied to respective ones of the two tracks, as in stereophonic recording, with the third and fourth channels superimposed thereon by applying equal in-phase portions of the third signal to both of the tracks and by applying equal portions of the fourth signal to both of the tracks with a differential phase angle between the two portions which remains substantially constant throughout the audio frequency range of interest. The fourth signal applied in this way causes the stylus of a stereophonic cutter to describe an ellipse having major vertical and horizontal axes disposed normal and parallel, respectively, to the surface of the record.

Related U.S. Application Data

[62] Division of Ser. No. 44,196, June 8, 1970, Pat. No. 3,708,631.

[52] U.S. Cl. 179/100.4 ST, 179/1 GQ,
179/100.1 TD, 100.4 C

[51] Int. Cl. G11b 3/74

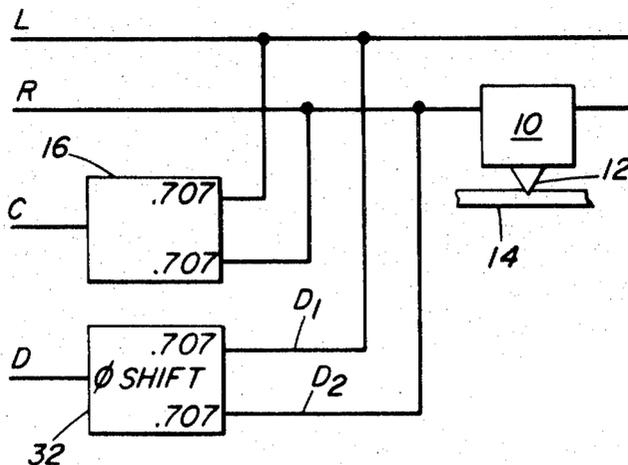
[58] Field of Search..... 179/100.4 ST, 100.4 C,
179/100.1 TD, 1 G, 1 GQ, 15 BT

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2 Claims, 9 Drawing Figures



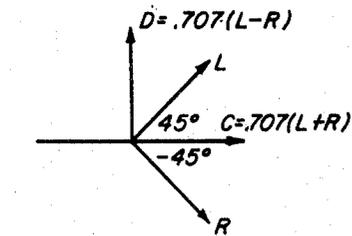
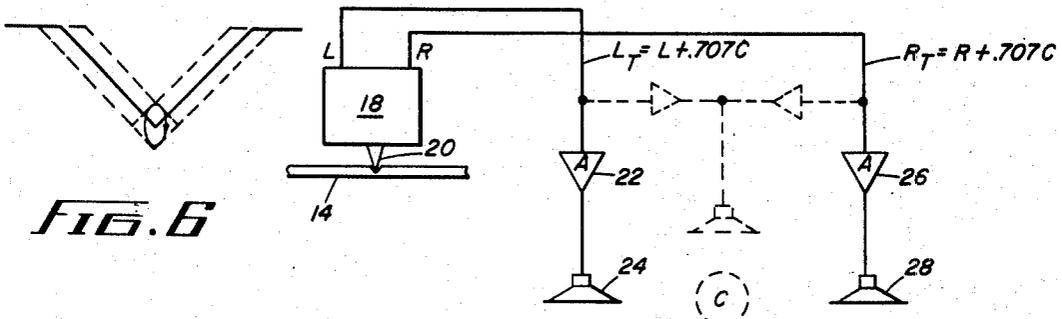
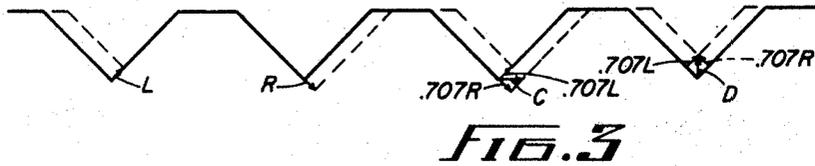
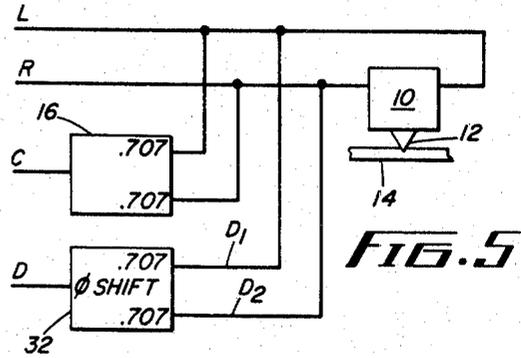
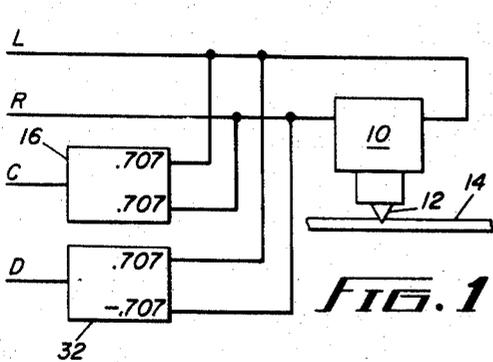
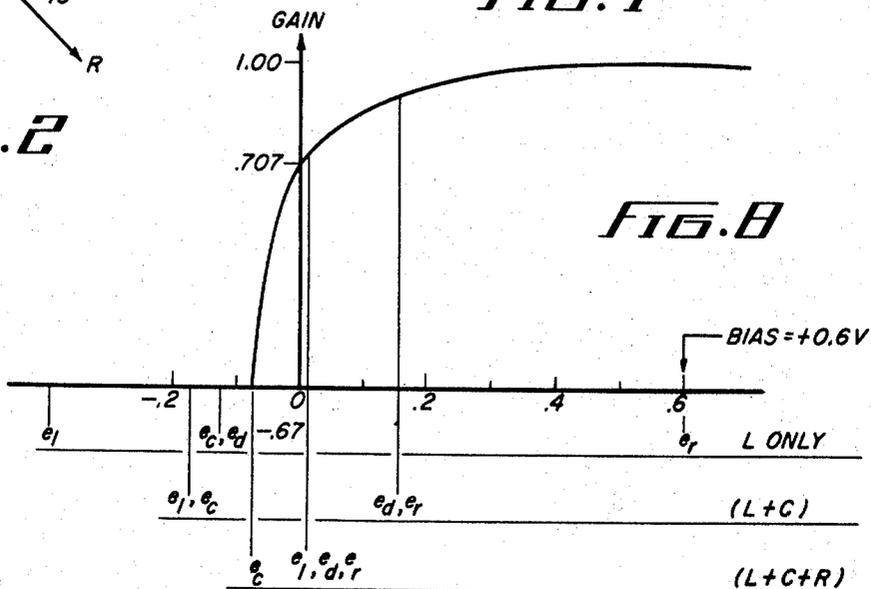


FIG. 2

FIG. 4



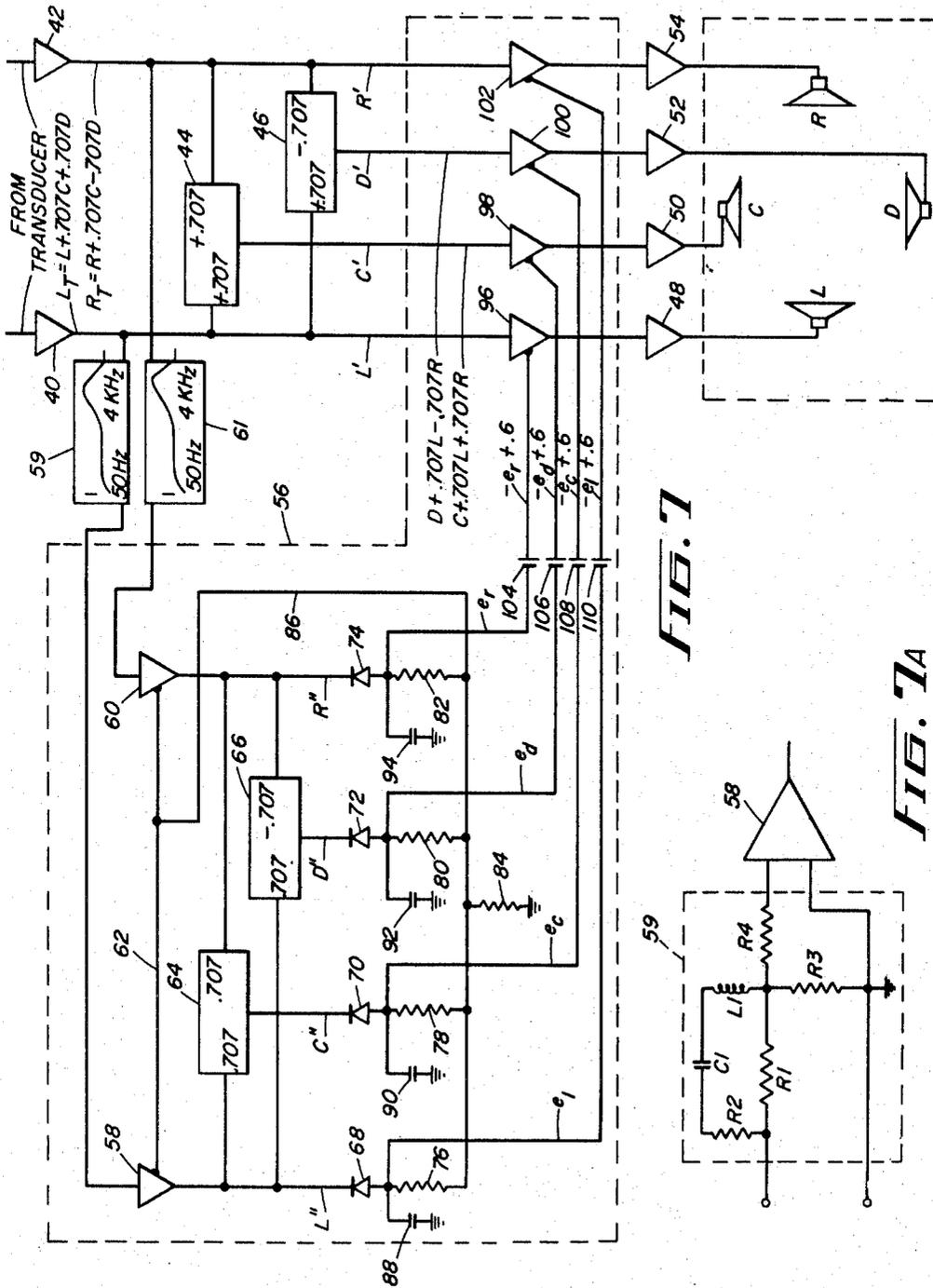


FIG. 7

FIG. 7A

ENCODER FOR MULTI-CHANNEL SOUND SYSTEM

This is a division of application Ser. No. 44,196 filed June 8, 1970, now U.S. Pat. No. 3,708,631.

BACKGROUND OF THE INVENTION

There is an increasing interest in multiple-channel recording and reproduction because of the variety of sounds and music forms that can be achieved thereby due to the well known phenomenon that the quality of music reproduction is enhanced when the number of reproduction channels increases. In the early days of the phonograph, only single channel or monaural recording was used, and as early as 40 to 50 years ago, investigators realized the value of recording and transmitting two separate channels of information, which in modern parlance is known as binaural or stereophonic sound. However, even two channels of information are not considered sufficient for full illusion of reality. For example, when a listener is placed in front of a symphony orchestra he hears sounds arriving from many different directions and from a variety of instruments, as well as reflections from the walls and ceiling, which gives him an accustomed illusion of space perspective. However, when reproduction is accomplished by utilizing only two channels it is difficult, if not impossible, to produce true reality with respect to spatial perspective. Early experiments have demonstrated that a minimum of three independent channels are needed to convey a satisfactory illusion of reality in the reproduction of orchestral music.

The modern stereophonic phonograph is capable of recording, or encoding, modulation along two separate channels, which geometrically are at 90° to each other and at 45° to the disc surface. It is usual practice to include a third channel by "matrixing" or adding it as a phantom channel to the other two, which causes it to be recorded as lateral modulation parallel to the record surface. Oftentimes, to obtain special effects, some of the channels are applied to the tracks in phase opposition, in a manner exemplified by test records Models STR 110, 111 and 120 produced and distributed by CBS Laboratories, a Division of the Assignee of this invention. Upon reproduction, the third (or central) channel appears on the two loudspeakers of the stereophonic phonograph, and an observer placed centrally between the loudspeakers perceives the illusion of the third channel being located between the other two. The fourth, or vertical, channel when reproduced on a conventional two-loudspeaker stereophonic phonograph gives the illusion of "spread" sound. Although there have been attempts to reproduce the third or center channel on a separate loudspeaker, the results have not been entirely satisfactory, and most stereophonic systems, even though many stereo records carry a "center" channel, employ only two loudspeakers.

In the co-pending application of William S. Bachman, Ser. No. 40,510 filed May 26, 1970, now abandoned in favor of continuation-in-part application Ser. No. 164,675 filed July 21, 1971, and assigned to the assignee of the present invention, there is described a system for providing third and fourth playback channels to otherwise two-channel systems by feeding third and fourth loudspeakers with signals respectively representing the sum and difference between the left and right channel signals. The left and right loudspeakers may be

located, for example, on opposite sides of a listening area, with the loudspeakers for the two virtual channels positioned at opposite ends of the listening area. Each loudspeaker displays the particular information fed to its channel accompanied by half-power signals from its adjacent channels. This system provides a pseudo-four-channel effect, but does not give complete illusion of each channel appearing independently on its corresponding loudspeaker.

If a record as described above is played on a monophonic phonograph, the vertically recorded channel will not be reproduced. It is desirable, of course, that such "four-channel" records be compatible with the older monophonic and stereophonic phonographs, because of the large numbers in current use. In other words, it is desirable that when the new medium is played on a monophonic or stereophonic phonograph, all channels recorded on the multi-channeled disc be heard with the loudspeaker system of the old phonograph.

SUMMARY OF THE INVENTION

A principal object of the present invention is to provide a method and apparatus for reproducing and separately presenting on independent loudspeakers four channels of information recorded as described above on an otherwise two-track record medium, such as a stereophonic disc record, a two-track tape system having separate recording and reproducing heads for each track, or the stereo-multiplex broadcasting system which provides for transmission of two independent channels or "tracks" of information, such that the listener experiences the illusion of listening to a corresponding number of separate sources of sound.

Another object of the invention is to provide a more realistic illusion of four separate channels than is afforded by the system described in the aforementioned co-pending application.

The invention is applicable to any of the aforementioned presently available two-track systems of recording and/or transmission on which two of the four separate channels of information are applied in the usual manner, with the other two channels superimposed on the two tracks by applying equal portions of them in phase and relatively shifted in phase, respectively. As applied to a 45°-45° stereophonic disc record, two of the channels are recorded on the two separate tracks provided by the walls of the groove, a third channel is recorded by applying equal portions of the signal in phase to the "left" and "right" channels of the stereophonic cutter to produce lateral modulation of the groove, and the fourth channel is recorded by applying it in equal amounts, but displaced in phase, to the "left" and "right" terminals of the cutter to produce vertical modulation of the record groove. In order that the vertical modulation have a horizontal component to which the older monophonic and stereophonic phonographs will be sensitive, the fourth signal, rather than being split into two equal signals which are applied 180° out-of-phase, may be applied through a phase-shift network which produces two signals displaced in phase from each other to cause the cutter stylus to execute an elliptical motion rather than the purely up and down motion produced by a difference signal.

The information recorded or transmitted on the medium is reproduced by an appropriate transducer to produce two composite signals, a "left" signal which

contains, in addition to the left channel signal, a fraction of the third channel and a similar fraction of the fourth channel, and a "right" signal containing the right channel signal, a fraction of the third signal, and a similar fraction of the fourth signal, the latter, however, being in the negative sense. Four independent signals, in which the original four channels are predominant, but each also containing to a lesser degree portions of two other channels, are derived from the composite signals by appropriately adding and subtracting components of the composite signals. An important aspect of the playback apparatus of the invention is that the instantaneous amplitudes of the four independent signals delivered to four corresponding loudspeakers are automatically controlled in response to the signals then present on the four channels so as to give the listener a substantially perfect illusion of four separate independent sources of sound.

BRIEF DESCRIPTION OF THE DRAWING

An understanding of the foregoing and additional aspects of this invention may be gained from a consideration of the following detailed description, taken in conjunction with the accompanying drawings, in which:

FIG. 1 is a schematic diagram of a system for recording four channels of information on a stereophonic record;

FIG. 2 is a vector diagram useful in explaining the motion of the cutter stylus in response to application of left, right, center and difference signals;

FIG. 3 is a cross sectional view of a fragmentary portion of a record showing four record grooves on a greatly enlarged scale, to illustrate the motion of the cutter in response to various signals;

FIG. 4 is a schematic diagram of a prior art stereophonic playback system for providing the illusion of a third channel;

FIG. 5 is a schematic diagram of a system according to the invention for recording four channels on a two-track stereophonic record;

FIG. 6 is a greatly enlarged illustration of a record groove illustrating the effect of applying the "difference" signal to the left and right channels through a phase shift network;

FIG. 7 is a schematic diagram of one form of playback apparatus embodying the invention;

FIG. 7A is a circuit diagram of a transmission network forming part of the system of FIG. 7;

FIG. 8 is a curve showing the transfer characteristic of the logic circuitry of FIG. 7, useful in explaining the operation of the system;

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Although, as noted above, the invention is applicable to any of a number of known two-track systems, it will be described in the environment of a 45° - 45° stereophonic disc record. By way of background, the current method of recording stereophonic signals including a third or center channel, and a method of reproducing these signals over a stereophonic two-loudspeaker system will be described with reference to FIGS. 1-4. The currently provided left (L), right (R) and center (C) signals are applied to the two terminals of a stereophonic cutter 10 having a cutting stylus 12 which is adapted to cut a groove in the lacquer of a master disc

14, revolving on a recording turntable (not shown). The C signals is applied through a matrix or signal divider 16 of known configuration resulting in application of portions thereof, equivalent to $0.707C$, to each of the L and R lines in an additive manner. As is well known in the groove cutting art, the tip of the cutter is capable of motions contained within a surface generally perpendicular to the disc in the manner portrayed by the vector diagram of FIG. 2. When a left signal L is applied, the stylus executes motions along the arrow L, which is at an angle of 45° to the horizontal, and when an R signal is applied, the stylus motion is along the arrow R, at an angle of -45° to the horizontal. Application of 0.707 parts of C to each of the L and R lines in an additive manner causes motion of the stylus

along the arrow C, equal in magnitude to $0.707(L + R)$, which is of the same magnitude as either L or R, but directed horizontally. It will be appreciated that instead of applying the L, R and C signals directly to the cutter, as shown in FIG. 1, they may, in keeping with common practice, be first recorded on a two-track master tape recorder and the output of the tape reproducer used to drive the record cutter. Discussion of the difference signal D illustrated in FIGS. 1 and 2 will be deferred until later.

The type of groove modulation resulting from the just-described procedure is shown in FIG. 3. When only the left signal L is applied, the groove is modulated in accordance with the arrow L, which is essentially confined to one wall of the groove. Similarly, when the R signal is applied, the modulation is in the opposite wall of the groove in the direction of the arrow R, which, it will be noted, is perpendicular to the arrow L. Application of equal amounts of the center signal C to the L and R lines causes both walls of the groove to be simultaneously and equally modulated in the directions indicated by the arrows $L = 0.707C$ and $R = 0.707C$, resulting in horizontal or side to side translation indicated by arrow C.

Apparatus for reproducing a stereophonic record carrying L, R and C signals recorded in this manner, schematically illustrated in FIG. 4, includes a stereophonic pickup having a cartridge 18 and a stylus 20 which enters the groove in the record and is actuated by the groove modulation to deliver output voltages on the L and R terminals. If only L signal modulation is present in the groove, an output signal appears only at the L terminal and is amplified by a suitable power amplifier 22 and reproduced by a loudspeaker 24. Similarly, when only R signal modulation is present in the groove, an output voltage appears at only the R terminal of the pickup, which is amplified by power amplifier 26 and applied to its respective loudspeaker 28. When the groove has lateral modulation consisting of the presence of equal amounts of left and right signal, then equal signals, namely, $0.707C$, appear on both the left and right loudspeakers, resulting in the appearance of a phantom source C (shown surrounded by a dashed line circle) midway between loudspeakers 24 and 28. However, this illusion is perceptible only to the centrally located observer 30; when he moves to either side, the C signal is heard over the nearest loudspeaker unless special precautions are made to adjust the directional characteristics of the loudspeakers with respect to the position of the observer.

It will be noted that the described three-channel record is "compatible" because the L, R and C signals all

have a horizontal component and thus will be heard when played on a monophonic player, which is sensitive only to lateral modulation, albeit their relative intensities will not be in the balance initially intended by the recording director. In reality, in spite of the introduction of a third channel, the above-described system reproduces only two independent channels of information. The third channel, C, is contained in both the left and right channels and the listener will, therefore, usually hear it reproduced from the loudspeaker nearest to him. This "center" channel may be presented on a separate loudspeaker system, as shown in dotted lines in FIG. 4, and amplifiers are commercially available for this purpose. This permits the observer to perceive the "center" information without having to locate himself equidistant from the left and right speakers.

Reverting to FIGS. 1-3, a fourth channel, D, may be introduced to the two-channel stereophonic system by dividing it into equal parts by a matrix or signal divider 32 and applying them in phase opposition to the left and right channels. As shown in FIG. 2, application of the D signal in this manner causes motion of the stylus in the vertical direction, along the arrow D, to an extent specified as 0.707 times the amount of D contained in the left and right channels subtracted from each other; i.e., $0.707(L - R)$. As seen in FIG. 3, this causes the left and right motions of the stylus to be out-of-phase relative to each other, resulting in up and down motion. When vertical modulation is reproduced by the system of FIG. 4, the loudspeaker cones are driven in opposite directions, resulting in out-of-phase sound pressures applied to the ears of the listener, and since this condition of pressure on the ears does not correspond to any known normal listening condition, the observer is unable to localize the sound. The difference signal D appears at some indefinite point in space, shown as D in a dashed circle, and the listener is unable to locate its whereabouts. Furthermore, some listeners of such out-of-phase sound have complained of a peculiar "pressure in the ears" sensation. This is in part overcome, however, by the system described in the aforementioned Bachman application Ser. No. 164,675 wherein the difference signal, as well as the "center" signal, are reproduced on separate loudspeakers.

To afford better compatibility with monophonic and conventional stereophonic players, while at the same time improving the illusion of four separate channels during playback, the difference signal D is preferably applied in the manner suggested in applicant Bauer's article entitled "Some Techniques Toward Better Stereophonic Perspective," IEEE TRANSACTIONS ON AUDIO, Vol. AU-11, No. 3, May-June, 1963. In keeping therewith, and as is illustrated in FIG. 5, instead of applying the difference signal equally and oppositely to the left and right channels as in the circuit of FIG. 1, the D signal is applied through an acoustical phase shift network 32 which splits the incoming signal into two equal amplitude signals D_1 and D_2 , each containing all of the frequencies of the D signal, but displaced in phase with respect to each other. Relative phase displacements in the range of 110° to 170° have been successfully used, with an angle of 135° being particularly suitable. It can be readily demonstrated that when the two signals are thus displaced relative to each other, the tip of the stylus instead of undergoing a purely up and down motion as shown in FIG. 3, executes the elliptical motion illustrated in FIG. 6. The limits of stylus

motion are shown by the dashed lines and the direction of motion of the ellipse depends on whether D_1 leads D_2 , or vice versa. The important consideration is that the groove has a horizontal component defined by the horizontal width of the ellipse, whereby both monophonic and stereophonic phonographs will reproduce all four signals; that is, the record with four separate channels will be fully compatible with the older playback systems, albeit with monophonic systems the signal D is attenuated by about 8db.

Although the description thus far has been concerned primarily with recording four separate channels of information on a two-track stereophonic record, it will be recognized by ones skilled in the art that similar techniques may be employed to record similar signals on a two-track tape system, for example, or to transmit comparable signals over the known stereo-multiplex system of broadcasting. The description to follow will be directed to apparatus for reproducing the signals in a manner such that the four channels may be separately and independently presented on four different loudspeakers. Except for the transducer means required to derive the signals from the record medium, the playback apparatus of the invention is applicable to all of such known two-track systems.

Referring now to FIG. 7, and considering the playback apparatus as applied to a disc recorded as described above, the recorded signals are derived by a stereophonic pickup of the type illustrated in FIG. 4, and may be applied to the system directly or through a pair of suitable amplifiers 40 and 42. It will be evident from what has been said earlier that the left signal, labeled L_T , is a composite signal including in addition to the L signal, 0.70 of the signal C and 0.707 of the signal D. Similarly, the right signal, R_T , contains the right channel signal R, 0.707 of the signal C, and 0.707 of the signal D, the latter being out-of-phase in the two cases. It is evident that if the outputs of amplifiers 40 and 42 were connected to two respective loudspeakers, the reproduction would be equivalent to that of a conventional two-channel stereophonic system. In the present system, however, the L_T and R_T signals are also applied to a pair of signal dividing circuits 44 and 46 of known configuration which, by appropriate addition and subtraction of components of the composite signals, produce four signals L' , C' , D and R' , respectively containing the four predominant channels L, C, D and R, with each also containing to a lesser degree portions of two other channels. Therefore, the signals delivered to the respective loudspeakers L, C, R and D, after amplification by suitable power amplifiers 48, 50, 52 and 54, respectively, are not composed of the pure information of the corresponding L, C, D and R channels, but rather are diluted with portions of the information from the adjacent channels.

In accordance with the invention, the instantaneous amplitudes of the signals delivered to the four loudspeakers are controlled by logic circuitry contained within the dashed line enclosure 56 in such a manner that a listener is given a substantially perfect illusion of four separate independent sources of sound. This objective is achieved by reason of the character of the music normally reproduced on a phonograph record, aided by a phenomenon known in acoustical science as the "precedence effect." In most musical selections, the individual performers do not play continuously, but rather, produce a constantly varying pattern of attacks,

decays, percussion sounds, etc., which do not occur simultaneously but are interleaved with each other. For example, first the sound of a drum may appear on channel L, followed by the sound of a cymbal on channel R, followed by the voice of a soloist pronouncing various syllables on channel C, etc. If one is able to switch the loudspeaker system in such a way that a loudspeaker is turned "on", or the signal preferentially amplified, each time a particular impulsive or percussive sound is started in its channel, while the remaining loudspeakers are correspondingly turned "off" or attenuated, the listener will have fixed this attention upon the particular sound coming from that loudspeaker, and even if the sound is transferred to another loudspeaker, he will have an illusion of the sound proceeding from the first loudspeaker. This results from the well known "precedence effect" which is based on the observation that when a sound originates from a given loudspeaker, and then gradually is switched into another loudspeaker, the listener continues to hear the sound coming from the given loudspeaker long after it has ceased to be the true source.

Accordingly, the function to be performed by the control logic portion 56 of the playback apparatus is to identify which channel has the strongest signal at any instant in time and to turn that channel "on," or to preferentially increase its gain, while attenuating or turning "off" the remaining channels. As the sound diminishes in the channel first identified and another sound appears on a different channel, the logic circuit rapidly attenuates the gain in the first channel and increases the gain in a different channel. It is useful to think of the action of this logic in terms of the following truth table:

If there are sound signals in the system it logically follows that:

- If L signal = 0, only R signal is present
- If R signal = 0, only L signal is present
- If C signal = 0, only D signal is present
- If D signal = 0, only C signal is present.

It will be observed that this is a form of negative logic; i.e., the lack of sound in channel L, for example, or the diminution thereof, serves to turn on or to enhance the gain of the R channel, etc., in accordance with the truth table.

Turning now to a discussion of the logic system, the outputs from amplifiers 40 and 42 (if provided) are respectively applied to gain control amplifiers 58 and 60, the gains of which are controlled in unison as indicated by the connection 62 therebetween.

For reasons which will be better understood after considering the logic circuit, it is desirable to apply the L_T and R_T signals to amplifiers 58 and 60 through respective signal modifying networks 59 and 61. These two networks are identical and exhibit transmission characteristics which resemble the equal loudness contour of the human ear at moderate loudness level and over the audio range of interest. A set of equal loudness contours are illustrated and described in an article by applicant Bauer and Emil Torick entitled "Researches in Loudness Measurement," IEEE TRANSACTIONS ON AUDIO AND ELECTROACOUSTICS, Vol. AU-14, No. 3, pp. 141-151, 1966. The 70 phon equal loudness contour developed in the study described in this article is shown in blocks 59 and 61 in the inverted, or sensitivity, form. It will be noted that there is a peak in the transmission characteristic at the higher frequen-

cies (at approximately 4 KHz), which may be of the order of 8 db, is essentially constant from approximately 2 KHz down to about 50 Hz at which it exhibits a drop of approximately 5 db. The function of the networks 59 and 61 is to so shape the signals delivered by the transducer to the gain control amplifiers 58 and 60 so that the signal switching logic (the operation of which is about to be described) will place the respective L' , C' , D' and R' signals in their proper channels on the basis of their relative loudness, rather than their energy content. For example, the weighting curves of networks 59 and 61 would preclude the low frequency, but high energy, signal produced by a drum from incorrectly switching the higher frequency, lower energy, signal produced by a piccolo, for example.

While the transmission characteristic exhibited by circuits 59 and 61 may be obtained in a number of ways, a preferred embodiment is shown in FIG. 7A, consisting of a high resistance R_1 , in parallel with a series branch containing a lower valued resistor R_2 , a capacitor C_1 and an inductor L_1 , followed by a small resistor R_3 to ground and a series resistor to amplifier 58 (or amplifier 60). The values of the components in the series branch are selected so that in cooperation with the parallel resistor R_1 , the circuit produces the peak in the transmission characteristic centered at about 4 KHz. Typical values of the components used in the circuit of FIG. 7A are:

- $R_1 = 5.1$ Kohms
- $R_2 = 1.6$ Kohms
- $R_3 = 91$ ohms
- $R_4 = 1.0$ Kohms
- $C_1 = 0.01$ μ f.
- $L_1 = 300$ mh.

It will be understood that the exact shape of the characteristic may be modified by experiment to provide the best results with any particular decoder, or it may be desirable to provide an adjustment to permit the user to adjust the decoder for any particular type of music, or any particular type of listening environment.

Returning now to the control circuit itself, the outputs of amplifiers 58 and 60 are separated by separating circuits 64 and 66 into four separate outputs L'' , C'' , D'' and R'' in the same manner as the outputs L' , C' , D' and R' are produced by circuits 44 and 46. These two sets of four outputs thus resemble each other in musical content, but the former set is held at a uniform level, despite variations in the dynamic range of the record (as modified by circuits 59 and 61), by the action of gain control amplifiers 58 and 60. To achieve this constant output level, the L'' , C'' , D'' and R'' signals are rectified by rectifiers 68, 70, 72 and 74, respectively, and summed by isolating resistors 76, 78, 80 and 82 to develop a sum signal across a common resistor 84. The voltage developed across resistor 84 is applied over conductor 86 to the gain control lead 62 of amplifiers 58 and 60, which are operative in response thereto to keep the average voltage across resistor 84 substantially constant. The gain control action is enhanced by connecting four capacitors 88, 90, 92 and 94 across resistors 76, 78, 80 and 82, respectively, whereby the rectified voltage represents the envelope of the wave rather than its instantaneous value. The automatic gain control action maintains the sum of the voltages across the resistors 76, 78, 80 and 82 constant because the

voltage across the relatively smaller resistor 84 is the sum of the four rectified voltages.

The action of the gain control circuit can best be understood by consideration of several illustrative examples. Let it be assumed that the system is playing a record which contains a single signal, say a left signal L, the amplitude of which is arbitrarily assigned a value of unity. It follows from the vector diagram of FIG. 2 that the voltages e_l , e_c , e_d and e_r respectively developed across resistor 76, 78, 80 and 82 would then have relative values of 1, 0.707, 0.707, and 0. The sum of these voltages is 2.414, and the gain control amplifiers 58 and 60 are designed to maintain the voltage across resistor 84 at this value regardless of the sound level in channel L. This represents one condition of circuit operation and will be considered later in further detail.

Now let it be assumed that only the C signal is present, in which case the voltages e_l , e_c , e_d and e_r will be 0.707, 1, 0 and 0.707, respectively. It will be noted that the sum of these voltages is also 2.414.

Now, if incoherent signals are simultaneously present on channels L and C (say, signals resembling white noise; that is, signals emanating from two different sources), the voltages corresponding to the sum of these channels will be equal to the square root of the sum of the squares of both voltages. Therefore, the unadjusted sum of the signals L and C will be as shown in the following Table I, namely, $e_l=1.223$, $e_c=1.223$, $e_d=0.707$ and $e_r=0.707$. The sum of these four voltages being 3.860, the automatic gain control amplifiers instantaneously adjust the component voltages so as to total 2.414. Thus, each of these voltages is proportionally reduced by the fraction 2.414/3.860 to give adjusted values of $e_l=0.760$, $e_c=0.760$, $e_d=0.440$ and $e_r=0.440$.

TABLE I

| Signal | e_l | e_c | e_d | e_r |
|----------------------|-------|-------|-------|-------|
| L | 1.000 | .707 | .707 | 0 |
| C | .707 | 1.000 | 0 | .707 |
| (L+C) unadjusted | 1.223 | 1.223 | .707 | .707 |
| (L+C) adjusted | .760 | .760 | .440 | .440 |
| R | 0 | .707 | .707 | 1.000 |
| (L+C+R) unadjusted | 1.223 | 1.414 | 1.223 | 1.223 |
| (L+C+R) adjusted | .557 | .667 | .557 | .557 |
| D | .707 | 0 | 1.000 | .707 |
| (L+C+R+D) unadjusted | 1.414 | 1.414 | 1.414 | 1.414 |
| (L+C+R+D) adjusted | 0.600 | 0.600 | 0.600 | 0.600 |

Consider now the presence of a third channel, say, R. The voltages corresponding to this channel acting alone are $e_l=0$, $e_c=0.707$, $e_d=0.707$ and $e_r=1$. Again, assuming that all three signal are white-noise-like incoherent signals, the resulting unadjusted values of e_l , e_c , e_d and e_r for the (L+C+R) condition are obtained by taking the square root of the sum of the squares of the individual voltages for these three channels, and are shown in Table I as being 1.223, 1.414, 1.223 and 1.223, respectively. These four voltages add up to a total of 5.083, whereby the automatic gain control amplifiers cause these component voltages to immediately reduce by the factor 2.414/5.083, resulting in the adjusted voltages shown in Table I.

If a fourth voltage corresponding to the D channel is now added, it turns out that each of the four summed voltages has a value of 1.414, which, when acted upon by the automatic gain control amplifiers are reduced to a value to total 2.414, with the result that each of the component voltages has a value of 0.6.

Returning for a moment to the right-hand portion of Fig. 7, the four signals L', C', D' and R' are applied to respective gain control amplifiers 96, 98, 100 and 102, and then to four loudspeakers L, C, D, and R, respectively, with intermediate amplification, if necessary, provided by amplifiers 48, 50, 52 and 54. The gain control amplifiers are key elements of the invention in that they control the gain of the signals applied to the respective loudspeakers in accordance with the logic described earlier.

If the aforementioned signals are incoherent, but exhibit definite frequency-like character, then the added peak value in the logic will approximate the sum of the signals, rather than the root-mean-square value. This, however, does not alter the analysis made hereinabove.

The requisite control is accomplished by applying the component voltages e_l , e_c , e_d and e_r to the gain control amplifiers as follows: e_l is applied to amplifier 102 to thereby control the R' signal; e_r is applied to amplifier 96 to control the L' signal; e_c is applied to amplifier 100 to control the D' signal; and e_d is applied to amplifier 98 to control the C' signal. For convenience in implementing the above-discussed truth table, the rectifiers 68, 70, 72 and 74 are connected in a negatively conducting fashion so that voltages e_l , e_c , e_d and e_r and the various combinations thereof shown in Table I are negative voltages. Additionally, the control circuits of gain control amplifiers 96, 98, 100 and 102 are positively biased with a relative voltage of 0.6 volts by means of batteries 104, 106, 108 and 110, respectively.

The gain control amplifiers have the control characteristic shown in FIG. 8, which indicates that when a voltage of +0.6 volt is applied to the gain control terminal of the amplifier, its gain is maximum, at a value designated as unity. When the applied voltage is reduced to zero, the gain of the amplifier is decreased to 0.707 of maximum; that is, the gain is down 3 db. The characteristic then falls rapidly such that when the gain control voltage is -0.67 volts, the gain is reduced to zero and the amplifier is turned "off." While the appropriate control characteristic may be obtained in a number of ways, it is conveniently obtained by using an integrated DC amplifier Model CA3000, available from RCA, in the circuit configuration described on page 6 of "RCA Integrated Circuits Application Note" ICAN-5030 printed in Sept. 1967 and then available to the industry.

The action of the logic circuit 56 for the four conditions of L signal only, two signals, such as L + C, three signals, such as L + C + R in combination, and four random incoherent signals L + C + R + D, will now be examined. It will be observed from Table I that when only the L signal is present, $e_r=0$ whereby the +0.6 volt bias turns amplifier 96 (which controls the L' signal) fully "on", whereas negative voltages of 1.0, 0.707 and 0.707 are respectively applied to the amplifiers controlling the R', D' and C' signals. Consequently, amplifiers 102, 100 and 98 are biased beyond cutoff so as to have zero gain, and the L signal, which otherwise would also appear in loudspeakers C and D, appears, only in loudspeaker L. By similar analysis it may be shown that any one signal appearing in any one individual channel will turn on only the gain control amplifier appropriate to that signal.

Considering now the condition when L and C signals are both present, it will be seen from Table I and from

the notations in FIG. 8 that the -0.76 volt applied to amplifiers 100 and 102 exceed the $+0.6$ volt bias thereby causing loudspeakers D and R to be turned "off;" however, the -0.440 volt applied to amplifiers 96 and 98 causes them both to be turned "on," with only a slight reduction from unity gain, whereby the L and C signals appear in their respective loudspeakers L and C.

Similarly, when L, C and R signals are simultaneously present, it will be seen from Table I and the notations on FIG. 8 that e_c is sufficiently negative to cause amplifier 100 (which controls the signal applied to loudspeaker D) to be turned "off" whereas the -0.577 volts applied to the other three gain control amplifiers causes them to remain on, but with their gains reduced by approximately 2.5 db, so that the L', C' and D' signals are reproduced on their respective loudspeakers.

Finally, when L, C, R and D signals are all present in equal amounts, the respective gain control voltages e_l , e_c , e_d and e_r are all 0.6 volts which allows all four amplifiers 96, 98, 100 and 102 to be turned "on" and to apply the signals to their respective loudspeakers. However, it will be seen from the control characteristic that the gain of the amplifiers is reduced by approximately 3 db when all four signals are present. Thus, the total sound energy reproduced by the loudspeakers remains essentially constant regardless of the number of signals present.

It is evident from the foregoing that the logic system is operative to turn on those loudspeakers which correspond to the predominant sounds instantaneously present in the system, thereby accomplishing an important object of the invention. In actual practice, all signals seldom occur simultaneously, but rather there is a constant interplay of the various instruments which turns the loudspeakers on and off in a manner to give a completely natural illusion of four separate sources of sound being present and reproduced over the four loudspeakers.

For best operation, it is preferable that the time constants of the rectifier circuits 68-74 have a very rapid attack time of the order of 0.1 milliseconds and relatively slow decay time of approximately 10 milliseconds. Likewise, the attack time of the gain control amplifiers 96-102 should be extremely rapid—of the order of 0.1 millisecond—while a decay time of the

order of 0.4 second has been found satisfactory. It is to be understood, however, that these attack and decay times may be adjusted between relatively wide limits without seriously impairing the performance of the circuit.

We claim:

1. Apparatus for transforming a multi-channel program including at least first, second and third separate audio program signals into two composite signals, said apparatus comprising,

at least first, second and third input terminals to which said separate first, second and third program signals are respectively applied,

first and second output terminals,

means connected between said first and second input terminals and said first and second output terminals, respectively, operative to transfer substantially equal amplitude proportions of said first and second program signals without relative phase change to said first and second output terminals, respectively, and

means connected between said third input terminal and both said first and second output terminals operative to transfer substantially equal reduced amplitude proportions of said third program signal to both said first and second output terminals and including phase-shifting means operative to provide a differential phase-shift angle between the proportion of said third program signal transferred to said first terminal and the proportion of said third program signal transferred to said second output terminal, which phase-shift angle has a value in the range between 110° and 170° and remains substantially constant over the range of frequencies of said third program signal.

2. Apparatus in accordance with claim 1 further including

a fourth input terminal to which a fourth program signal is applied, and

means connected between said fourth input terminal and both said first and second output terminals operative to transfer substantially equal reduced amplitude in-phase proportions of said fourth program signal to both said first and second output terminals.

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