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[54] QUADRAPHONIC REPRODUCING SYSTEM WITH GAIN RIDING LOGIC

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Related U.S. Application Data

[63] Continuation of Ser. No. 81,858, Oct. 19, 1970, abandoned.

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

[51] **Int. Cl.** **H04r 5/00**

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

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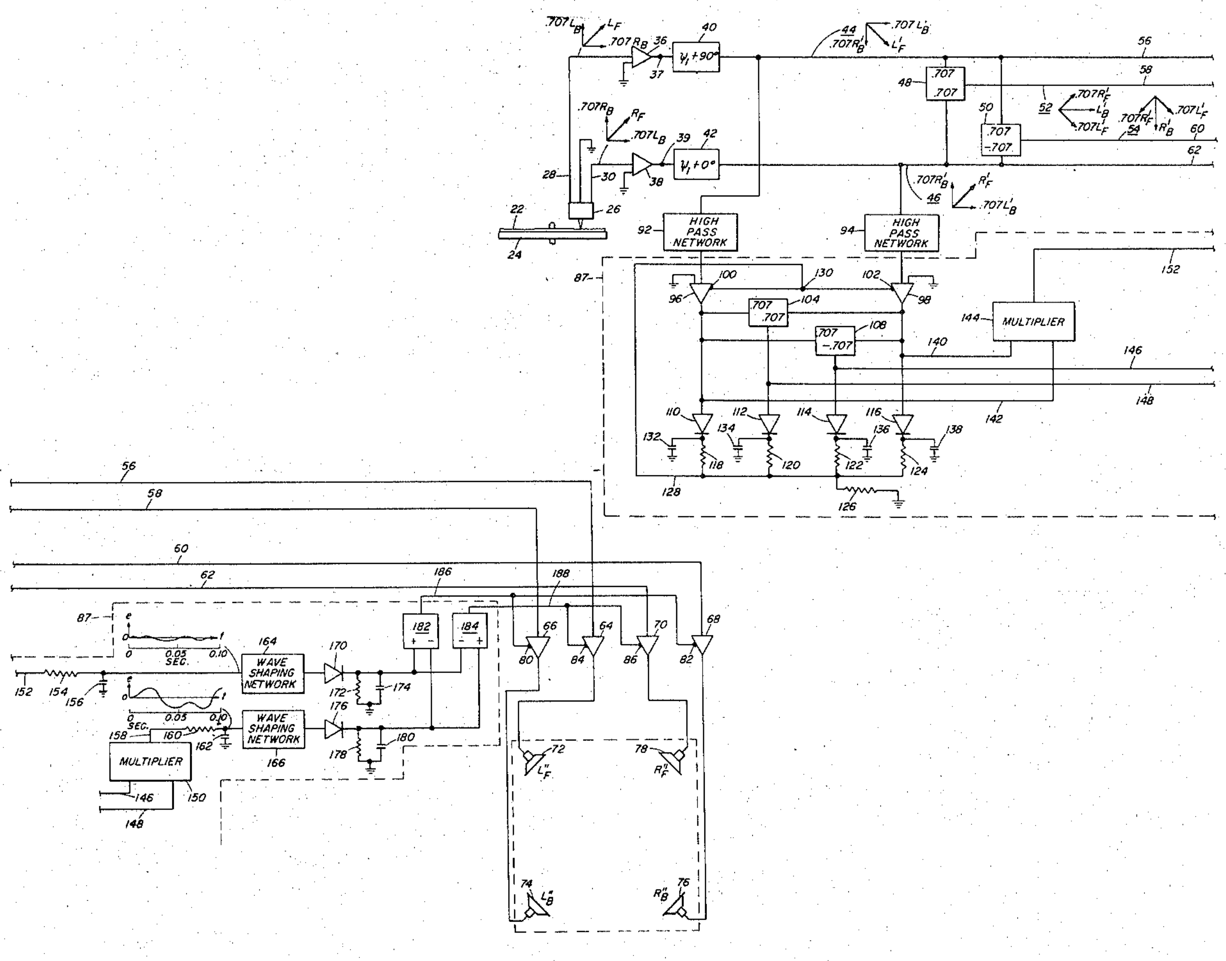
[57] **ABSTRACT**

Apparatus for reproducing four separate channels of information on a medium having only two independent tracks and presenting it on four loudspeakers to give a listener the illusion of sound coming from a corresponding number of separate sources. The realism of the illusion is enhanced by a decoding system which accepts the two outputs from the medium, which may be a disc record, one from each track, separates them into four independent channels each carrying predominantly the information contained in the originally recorded sound channels, and, utilizing cross-correlation techniques, derives a pair of control signals for controlling the gains of amplifiers associated with the four loudspeakers. The use of cross-correlation improves the separation of the four independent channels, particularly the generally "front" from the generally "back" signals, thereby to enhance the realism of four-channel simulation.

14 Claims, 5 Drawing Figures



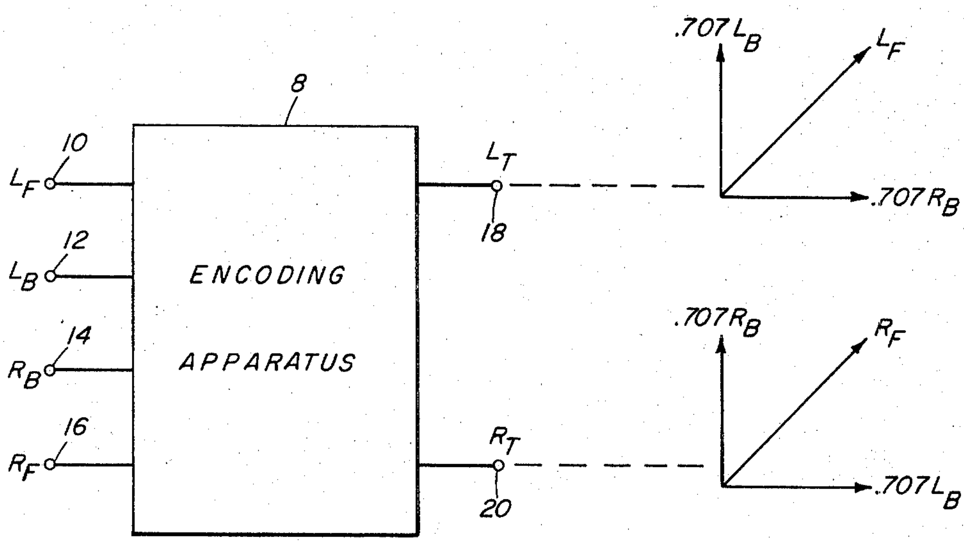


FIG. 1

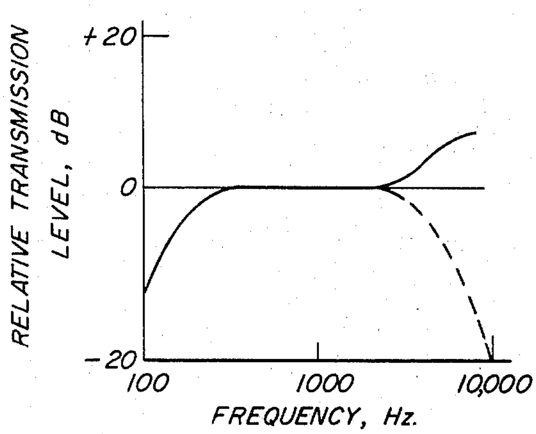


FIG. 3

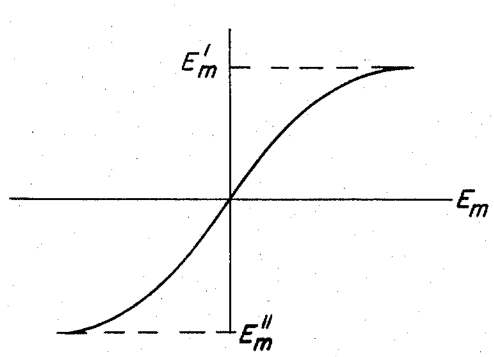


FIG. 4

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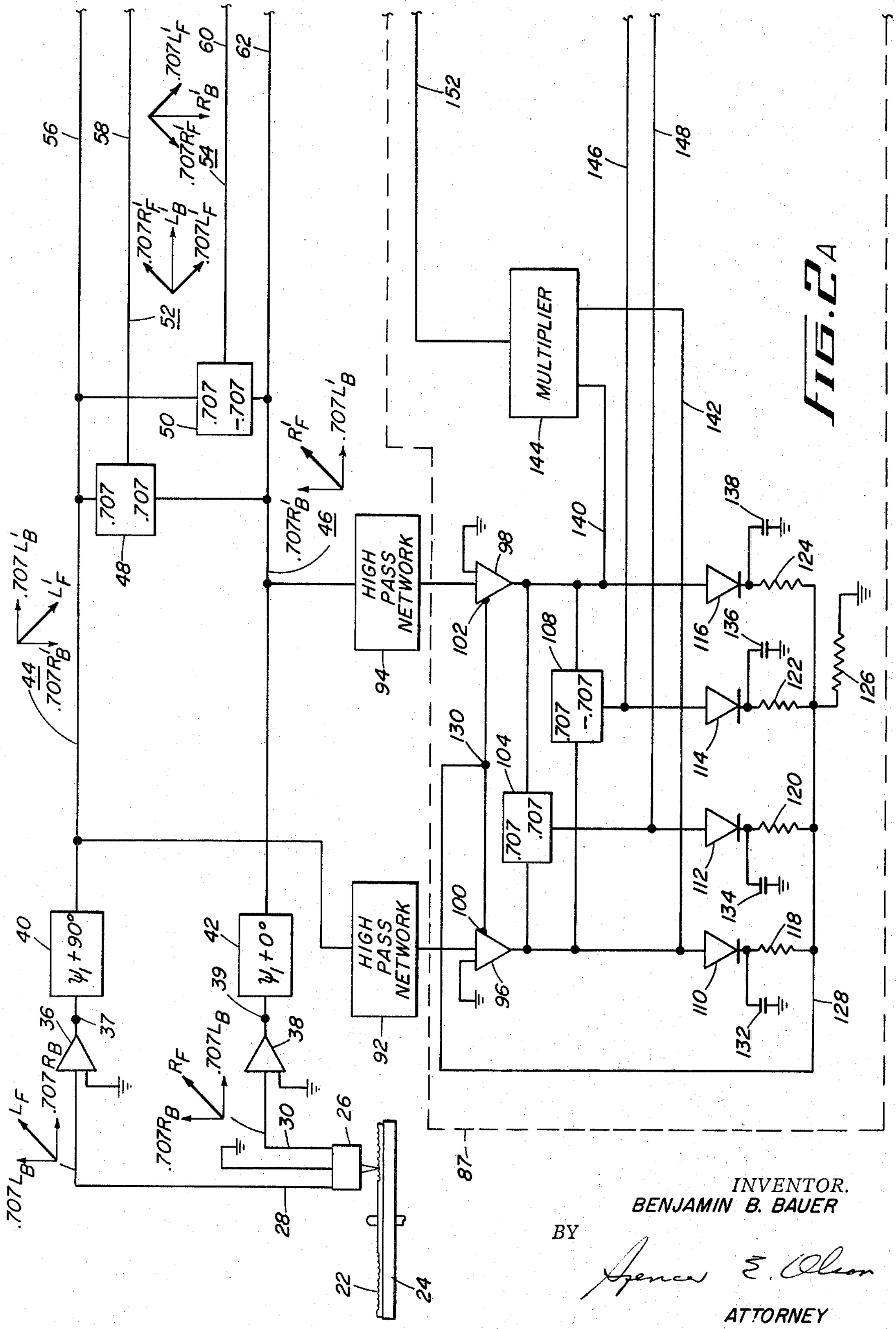

FIG. 2A

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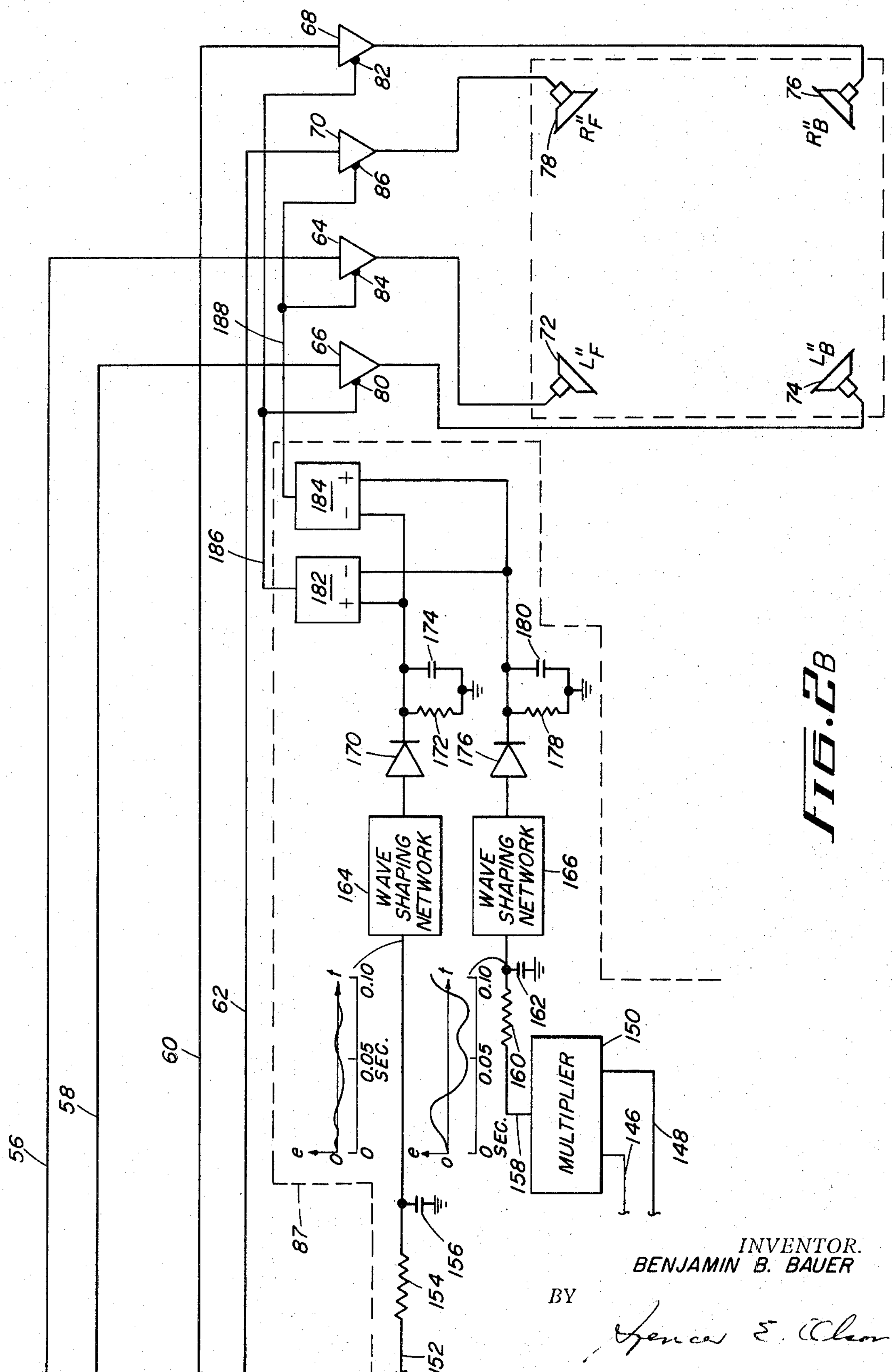

FIG. 2B

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QUADRAPHONIC REPRODUCING SYSTEM WITH GAIN RIDING LOGIC

This is a continuation, of application Ser. No. 81,858, filed Oct. 19, 1970, now abandoned.

BACKGROUND OF THE INVENTION

This invention relates to apparatus for recording and reproducing four separate channels of information on a medium having only two independent tracks, and more particularly to apparatus for reproducing such information and presenting it on four loudspeakers to give the listener the illusion of sound coming from a corresponding number of separate sources. The two tracks may be provided by any one of several available two-track systems, such as two-track tape, or a stereophonic disc record. In a system of this kind described in U.S. Application Ser. No. 44,224 filed June 8, 1970 entitled "Quadruphonic Recording and Reporducing System" and assigned to the assignee of the present invention, now abandoned in favor of continuation application Ser. No. 251,544, filed Apr. 21, 1972 which, in turn, has been abandoned in favor of continuation-in-part application Ser. No. 328,814 filed Mar. 10, 1973, which has now been abandoned in favor of continuation-in-part application Ser. No. 384,334 filed July 31, 1973, in which a stereophonic record, which may be in the form of disc or tape, etc., is used as the two-track medium, there are recorded on the left and right channels signals to be presented on the "left front" and "right front" loudspeakers, respectively, together with signals on both channels identified with "left back" and "right back" loudspeakers at 90° out-of-phase relative to each other, with the "left back" signal leading in the "left" channel and the "right back" signal leading in the "right" channel. Also described in the aforementioned co-pending application is a decoding system which accepts the two outputs from the disc record, one from each track, and by appropriate electronic manipulation separates them into a simulation of four independent channels, for presentation on four separate loudspeakers, each carrying predominantly the information contained in the originally recorded sound channels with attenuated information from other channels. The present invention is directed to an improved decoding apparatus of the general type described in the aforementioned co-pending application.

To better understand the nature of the problem to be overcome by the improved decoding apparatus of the present invention, the technique and apparatus for recording the four channels of information on a two-track stereophonic record, described in detail in the aforementioned application, will be briefly described with reference to FIG. 1 of the accompanying drawings. In this figure, the encoding apparatus is diagrammatically illustrated as a block **8** which includes a number of all-pass networks (characterized as Ψ-networks in the aforementioned application) which provide uniform transmission of all the important frequencies of the audio spectrum while at the same ime providing a fixed differential phaseshift between the various networks so that a fixed phase difference relationship exists between the output voltages. An understanding of the details of these all-pass circuits being unnecessary for an understanding of the present invention, and, moreover, since they are described in detail in the aforementioned application, only the function of the encoding apparatus, as it applies to the present invention, will be described. The encoding apparatus includes four input terminals **10**, **12**, **14** and **16** for receiving four separate signals which are designated left front (L_F), left back (L_F), right back (R_B) and right front (R_F), respectively. These designations signify the locations in a listening area of the four loudspeakers on which the signals are intended for ultimate presentation. The characteristics of the all-pass networks are such that if input voltages L_F, L_B, R_B and R_F of the same amplitude and frequency are applied in succession to their respective input terminals, the two output voltages L_T and R_T respectively appearing at output terminals **18** and **20** display the amplitude and phase relationship, referred to the input signals (phase-shifted by frequency dependent reference phase angle Ψ) shown by the phasor diagrams adjacent the respective output terminals. It is demonstrated in the aforementioned application that if the resulting voltages L_T and R_T are recorded on a two-track medium, such as a stereophonic disc record, the resulting record is fully compatible with conventional stereophonic, as well as monophonic, players, with all the sounds appearing in full quality and amplitude over the two-channel stereophonic system.

It has been found that if the sounds are represented predominantly by a set of two musically related and similar, albeit incoherent, signals L_F and R_F, the decoder described in the aforementioned application not only turns "on" the "front" loudspeakers, as it should, but also tends to some degree to turn "on" the "back" loudspeakers, obviously an undesirable result. It is the primary object of the present invention, therefore, to provide a decoder which gives better separation during reproduction between the generally "front" sounds, represented by the aforementioned L_F and R_F voltages, and the "back" sounds represented by the L_B and R_B voltages.

SUMMARY OF THE INVENTION

Briefly, the foregoing object is attained by providing means in the decoder of the reproducing apparatus for controlling the gains of the amplifiers associated with the four separate loudspeakers with signals derived by appropriate cross correlation of signals closely resembling the four independent channels being presented on the loudspeakers. More specifically, the two signals predominantly carrying left front (L_F) and right front (R_F) sound information are multiplied and the product integrated, and the two signals predominantly containing left back (L_B) and right back (R_B) information are multiplied together and their product integrated. Because of the relative phase relationships of the voltages representing the four signals, the two signals delivered by the integrators correlate differently, particularly when the integration is carried out over a time period corresponding to approximately 150 Hz. The signals delivered by the integrators are rectified and applied in parallel, but with opposite polarity, to a pair of subtracting circuits that are operative to produce a pair of gain control signals. The signal delivered by one of the subtracting circuits controls in unison the gains of the gain control amplifiers associated with the "left front" and "right front" loudspeakers, and the signal delivered by the other subtracting circuit is applied in parallel to the gain control amplifiers associated with "left back" and "right back" loudspeakers to control them in unison. The relative phases of the voltages representing the independent channels together with the use of cross-correlation techniques improves the separation between the "front" and "back" sounds to enhance the realism of four-channel simulation. The improved decoder is adaptable for operation in any encoder-decoder system in which four signals are encoded for recording on a two-track medium and the decoder decodes them for display over four loudspeakers in a manner to cause the predominant information in the four signals, together with attenuated information from others of the signals, to be displayed over corresponding loudspeakers.

BRIEF DESCRIPTION OF THE DRAWING

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

FIG. 1 is a schematic diagram of encoding apparatus to which reference has already been made;

FIGS. 2A and 2B together constitute a schematic diagram of decoder and playback apparatus embodying the invention;

FIG. 3 is a curve illustrating the transmission characteristics as a function of frequency of wave-shaping networks embodied in the system of FIG. 2; and

FIG. 4 is a curve illustrating the charactereistics of other wave-shaping networks embodied in the system of FIG. 2.

DESCRIPTION OF THE PREFERRED EMBODIMENT

The decoder of the present invention is similar in many respects to the decoder described in applicant's aforementioned U.S. Application Ser. No. 384,334, certain details of which, in turn, are described in U.S. Application Ser. No. 44,196, also filed on June 8, 1970 in the names of Benjamin B. Bauer and Daniel W. Gravereaux, now U.S. Pat. No. 3,708,631 and also assigned to the assignee of the present application. The disclosures of these applications are incorporated herein by reference to the extent necessary for understanding the operation of certain aspects of the herein described decoder; however, the description to follow is believed to be sufficiently complete to enable ones skilled in the art to understand this operation without recourse to the co-pending applications.

Referring now to FIG. 2, the decoder of the present invention is shown in the context of reproducing information recorded on a disc record 22 mounted for rotation on a turntable 24 in the usual manner. The record is played with a stereophonic phonograph pickup 26 of conventional design having two output circuits 28 and 30 which respectively carry the output signals transduced from the "left" and "right" channels of the record. The "left" and "right" channel voltages are portrayed adjacent their respective output circuits, and it will be seen from comparison with FIG. 1 that these voltages correspond to the voltages L_T and R_T produced by the encoding apparatus of FIG. 1. Should it be necessary in the replication of the voltages L_T and R_T to amplify the signals, they may be applied to respective amplifiers 36 and 38, which preferably include the necessary amplitude-frequency modifying networks for restoring the frequency balance that may have been used in recording the record. It will be evident that the apparatus described thus far is completely conventional and well understood by ones skilled in the art.

Essential to the decoding of the outputs L_T and R_T to provide a simulation of the original sounds is the shifting of the signal from the "left" channel by 90° relative to the signal on the "right" channel. To this end, the signals from amplifiers 36 and 38 appearing at their respective output terminals 37 and 39 are respectively applied to corresponding all-pass phase-shift networks 40 and 42, the latter providing a reference (lagging) phase shift of Ψ_1, and network 40 providing a lagging phase shift of $\Psi_1 + 90°$. The value of Ψ_1 (which generally is a function of frequency) may be the same as the phase shift used in the encoding apparatus 8 (FIG. 1), or it may be different, the principal requirement being that the reference phase angle Ψ_1 be the same in both of networks 40 and 42. The phase-shift networks 40 and 42 transform their respective input signals L_T and R_T into two new signals L_T' and R_T' which are very nearly equal to the signals L_T and R_T except that they are displaced at 90° with respect to each other. This is illustrated by the phasor diagrams 44 and 46 in which the phasors of the R_T' signal are in the same relative position as in the R_T signal, but the vectors of the L_T' signal are all rotated 90° clockwise. Since the reference phase shifts Ψ_1 are the same in both networks, their relative effects have been disregarded in the phasor diagrams. It will be observed that these two voltages contain predominant L_F' and R_F' voltages, with the former also including 0.707 of each of L_B' and R_B', and the latter further including 0.707 of each of R_B' and L_B'.

The signals delivered by the all-pass networks 40 and 42 are further processed to derive two additional voltages containing predominant L_B' and R_B' information. The former is derived by summing the two signals from networks 40 and 42, each after multiplication by the factor 0.707, in a summing circuit 48 of known configuration. A signal containing predominant "right back" information is derived in a similar way by subtracting one of the signals from the other again after multiplication of each by the factor 0.707. As illustrated by the phasor diagrams 52 and 54, the voltages produced by the operation of circuits 48 and 50 now contain predominant L_B' and R_B' information.

The four voltages represented by phasor diagrams 44, 46, 52 and 54 are applied over conductors 56, 58, 60 and 62, respectively, to corresponding gain control amplifiers 64, 66, 68 and 70 and thence to corresponding loudspeakers 72, 74, 76 and 78. If necessary in a given system application, additional amplifiers (not shown) may be provided between the gain control amplifiers and their respective loudspeakers. The loudspeakers are positioned in a listening area (represented by the dotted line enclosure) with the loudspeaker carrying predominantly the signal L_F' at the left front corner, the loudspeaker carrying predominantly the R_F' at the right front corner of the listening area, the loudspeaker predominantly carrying the signal L_B' at the left back corner, and the loudspeaker predominantly carrying the signal R_B' at the right back corner of the listening area.

The arrangement thus far described is similar to the decoding apparatus described in the aforementioned co-pending applications and as mentioned therein produces a good similitude of the original four channel sound even if the gain control amplifiers are maintained in a quiescent, or normal gain state, or even if they are replaced with ordinary linear amplifiers. This result obtains because the predominant voltage in each channel is 3dB higher in level than the other two voltages, and corresponds to the signal originally intended for reproduction at the particular location. However, the realism is enhanced in the systems described in the previous applications by providing a logic and switching circuit which senses whether or not a particular channel should be increased or decreased in gain so as to emphasize those sounds which momentarily predominate in particular channels. While the action of the logic employed in the systems of the earlier applications greatly enhances the simulation of four independent channels, under certain conditions of operation there tends to be a deterioration in the simulation. The present invention is concerned with an improved form of logic which overcomes the shortcomings of the previous circuits and consistently achieves the enhanced realism of four channel simulation.

To better understand the problem to which the present invention is directed, it is convenient to assume that the two signals encoded on the disc record consist of only two signal voltages, L_F and R_F, which are shown by heavier lines than the 3dB-down components in the phasor diagrams presented in FIG. 2. Further in the analysis to follow the two signals are considered to be musically similar and in compass, but mathematically they are incoherent. Examples of such signals would result from two persons preforming a two-part musical selection in unison; while the melody or melody and accompaniment would be in proper tempo and compass, the mathematical correlation, consisting of multiplication and integration over a period of time equal to the reciprocal of the lowest frequency, would tend to be reduced to zero. Only in the case of the two signals being identical would the correlation take on a finite value. Now, it is noted that when the signals have passed through networks **40** and **42**, respectively, they appear as signals L_F' and R_F' at 90° with respect to their previous positions, so that even if they were originally indentical, after phase shifting they would no longer show coherency. The reason for this is that the product of a sine and cosine function over one cycle is precisely zero.

After the voltages L_F' and R_F' in phasor diagrams **44** and **46** are acted upon by summing networks **48** and **50** they appear as 0.707 R_F' and 0.707 L_F' in both of phasor diagrams **52** and **54**. Now, it should be noted, if the voltages L_F' and R_F' are completely independent, occurring singly, the magnitude of the value of the phasors in diagrams **44** and **46** would be 3dB greater than the magnitude of the corresponding signals in the phasor diagrams **52** and **54** and thus readily identifiable as to magnitude by logic circuitry of the aforementioned applications so as to cause the appropriate gain control amplifier to turn "on" or "off" as the case may be. However, should the voltages L_F' and R_F' be in compass and appear almost simultaneously, as they tend to do during a performance by two musicians playing different parts of the same musical selection, then the R.M.S. value of the sum of the phasors **52** and **54** is the same as the value of the corresponding phasors in diagrams **44** and **46.** In this circumstance, the logic of the system described in U.S. Pat. No. 3,708,631 becomes confused and tends to turn all loudspeakers "on" whereas what is desired in that only the "front" loudspeakers **72** and **78** be turned "on."

It will have been observed from the foregoing example, in which only the voltages L_F' and R_F' are present, that while the voltages in the phasor diagrams 44 and **46** are uncorrelated, the corresponding voltages in the phasors **52** and **54**, namely, $0.707R_F'$ and $0.707L_F'$ are individually either positively or negatively correlated. This can be seen from the fact that the arrow labeled $0.707R_F'$ in diagram **52** is parallel to, but in the opposite direction, to the corresponding arrow in diagram **54**; consequently, upon multiplication and integration the output of these two phasors would be a finite negative number. Although the arrow labeled $0.707L_F'$ in phasor diagram **52** is also parallel to the corresponding arrow in diagram **54**, it points in the same direction and, accordingly, multiplication and integration of these two signals would produce a finite positive number. It will be appreciated that if both the signals L_F' and R_F' had equal long-term average magnitudes, a long-term integration would reduce to zero, but this situation almost never occurs for short-term integration because even if the musicians are playing the same instrument in unison there always tends to exist lapses or differences in pitch, tempo, attack, decay, etc. As a result, each of individual phasors $0.707R_F'$ and $0.707L_F'$ integrated over the short-term would tend to take on its negative or positive finite value, while the corresponding phasors L_F' and R_F', being incoherent even over the short-term, tend to take on a zero value, even for short-term integration, provided the integration extends at least over a period equal to the inverse of the lowest frequency being analyzed. The improved logic now to be described is based upon the recognition of the just-described properties of the signals and application of the principles to a practicable implementation.

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 **87** in such a manner that a listener is given a substantially perfect illusion of four separate independent sources of sound. The two signals represented by phasor diagrams **44** and **46** delivered by all-pass networks **40** and **42** are additionally applied to a pair of similar high-pass networks **92** and **94**, respectively, which are designed to cut off all information below approximately 150 Hz. While the cutoff frequency may be altered over a very wide range without seriously affecting the operation of the circuit to follow, it has been observed that very little directional information is contained at frequencies below roughly 150 Hz, and that by eliminating all frequencies below this value permits the short-term integration in the analyzer (to be described) to be carried out over an interval of only approximately 6.7 milliseconds. The transmission characteristics of the networks **92** and **94** are preferably as shown in FIG. **3**, being substantially flat over a frequency range between about 150 Hz and 2,000 Hz, and increasing at this point by about 10dB for the higher frequencies. Thus, the two networks exhibit transmission characteristics which resemble the equal loudness contour of the human ear at moderate loudness level and over the audio range of interest, except for an added sharp cutoff below approximately 150 Hz. A set of equal loudness contours are illustrated and described in an article by applicant and Emil Torick entitled "Researches in Loudness Measurement,"

IEEE TRANSACTIONS ON AUDIO AND ELECTRO-ACOUSTICS, Vol. AU-14, No. 3, pp. 141–151, 1966. The function of networks **92** and **94** is to so shape the signals delivered to the circuitry to follow that the signal switching logic will place the respective L_B, L_F, L_B, R_F and R_B signals in their proper channels on the basis of their relative loudness, rather than their energy content. For example, the weighting curves of the networks would preclude a low frequency, but high energy, signal produced by a drum from incorrectly switching a higher frequency, lower energy, signal produced by a piccolo, for example. While networks having the characteristics illustrated in FIG. **3** are preferred, most of the advantages of the invention may be achieved without increasing the transmission of the network at the higher end of the frequency band, or, indeed, it may even be attenuated above approximately 10,000 Hz as shown by the dashed-line in FIG. **3.**

After shaping by the networks **92** and **94** the signals are applied to a pair of gain control amplifiers **96** and **98,** the gains of which are made to closely track each other by application of a common control signal to their respective gain control terminals **100** and **102.** The two outputs of amplifiers **96** and **98** are separated into four separate outputs by networks **104** and **108** in the same manner as the four outputs are produced by circuits **48** and **50.** 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 **92** and **94**), by the action of gain control amplifiers **96** and **98.** To achieve this constant output level, the four signals are rectified by rectifiers **110, 112, 114** and **116,** respectively, and summed by isolating resistors **118, 120, 122** and **124** to develop a sum signal across a common resistor **126** connected between a bus **128** common with the resistors and ground. The voltage developed across resistor **126** is applied over conductor **128** to the junction **130** between the control electrodes **100** and **102** of gain control amplifiers **96** and **98,** respectively, which are operative in responsive thereto to keep the average voltage across resistor **126** substantially constant. The gain control action is enhanced by connecting four capacitors **132, 134, 136** and **138** across the resistors **118, 120, 122** and **124,** 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 **118, 120, 122,** and **124** constant because the voltage across the relatively smaller resistor **126** is the sum of the four rectified voltages.

While half-wave rectifiers are illustrated, and the circuit will function satisfactorily with half-wave rectification, it should be understood that full-wave rectification is preferable and is used in the preferred embodiment. While the just-described gain control function is not absolutely essential for the proper functioning of the balance of the logic, applicants have found that it provides somewhat better performance and its use is preferred.

An important aspect of the invention is that the signals delivered by amplifiers **96** and **98** and the networks **104** and **108** (whether or not subjected to automatic gain control) are cross-correlated, as by multiplication in a particular way and subsequent integration. More particularly, the output signals from amplifiers **96** and **98** are applied via conductors **142** and **140,** respectively, to a first multiplying network **144,** and the outputs from summing networks **104** and **108** are applied via conductors **148** and **146,** respectively, to a second multiplier **150.** The output of multiplier **144** is coupled over conductor **152** to an R-C integrating circuit consisting, for example, of resistor **154** and capacitor **156** and the output of multiplier **150** is applied over conductor **158** to a similar integrating cirucit consisting of resistor **160** and capacitor **162.** The values of resistors **154** and **160** and capacitors **156** and **162** are so related that the integrators have a turnover point at approximately the cutoff point of networks **92** and **94,** namely, at approximately 150 Hz, so as to provide integration of all signals at 150 Hz and above.

The function of the multipliers and associated integrators will be evident from a comparison of the outputs of the two integrators for the example previously discussed; i.e., when the input signal consists of only a "front left" signal, L_F, and a "right front" signal, R_F. It is seen from the waveform associated with the output of integrator **154–156,** which handles the product of the uncorrelated voltages L_F' and R_F' is very nearly zero; it is not precisely zero only because the integrator is not a perfect integrator. The output of integrator **160–162,** on the other hand, varies plus and minus depending upon the synchronization of the parts played by the two musicians. Over a long period of time the output of integrator **160–162** is also zero, but if the integration is carried out for about 6.7 milliseconds, corresponding to 150 Hz, the instantaneous high frequency perturbations will be eliminated, leaving, however, the output corresponding to the interplay of the two signals.

As an alternative to the RC integrators, suitable low-pass filters with rejection above about 150 Hz. may be used. The use of such filters would eliminate the actual A.C. signals and their simultaneous products, and would transmit only the low frequency control signal corresponding to the correlation function.

The outputs of integrators **154–156** and **160–162** are respectively applied to a pair of similar wave shaping networks **164** and **166,** the purpose of which is to limit the excursion of the output voltages from the integrators to a reasonable figure. The transmission characteristic of the networks may take a variety of shapes, but in general, circuits having the characteristic depicted in FIG. **4** would be used. The output voltage maxima E_m' and E_m'' of the shaping network are adjusted to levels suitable for use with the electrical elements to follow. The output voltages from the shaping networks **164** and **166** are rectified by corresponding rectifiers **170** and **176** to charge capacitors **174** and **180,** respectively. The charge on the capacitors is allowed to discharge through resistors **172** and **178,** respectively, The values of the resistors and capacitors are so chosen that the discharge time constant is of the order of 100 milliseconds, it being understood, however, that the choice of time constant may vary over a very large range without significantly affecting the performance of the decoder. While the rectifiers **170** and **176** are shown as half-wave rectifiers, it will be understood that full-wave rectifiers may be used and are generally preferred.

The outputs of rectifiers **170** and **176** are used to control the gain of the gain control amplifiers which precede the four loudspeakers **72, 74, 76** and **78.** To this end, the outputs of the rectifiers are applied to a first subtracting circuit **182**, in plus-minus relationship, and to a second subtracting network **184**, in a minus-plus relationship. The output of network **182** is applied in parallel to the gain control terminals **80** and **82** of gain control amplifiers **66** and **68**, respectively, while the output of network **184** is applied in parallel to the gain control elements **84** and **86** of amplifiers **64** and **70**, respectively.

The performance of the decoder will now be analyzed by again assuming that the input consists of the same input voltages L_F and R_F as before. The output of rectifier **170**, depicted by the associated wave form, is near zero, whereas the output of rectifier **176**, as is apparent from the depicted wave form, initially has a positive output. Consequently, the output of summing circuit **182** is negative, and since it is applied as a control signal to gain control amplifiers **66** and **68**, their gain is diminished, or, depending upon the adjustment of the gain, may be completely cut off. At the same time, the output of network **184** would be positive, thereby enhancing the gains of amplifiers **64** and **70**. Therefore, the two signals L_F' and R_F' appear only at the corresponding loudspeakers **72** and **78**, positioned at the left front and right front corners of the listening area, respectively, as signals L_F'' and R_F''.

Conversely, if the incoming signals delivered by the pickup **26** were composed of only the "back" signals, $0.707L_B$ and $0.707R_B$, then the output of rectifier **176** would be at or near zero, while a significant voltage would appear at the output of rectifier **170**. Under these conditions, the output of summing device **182** would be positive, while the output of summing device **184** would be negative, thereby causing the gain of amplifiers **66** and **68** to be increased and the gain of amplifiers **64** and **70** to be diminished, or completely cut off, which, of course should be the result if the input "back" signals are to appear as signals L_B'' and R_B''.

It will be recognized by ones skilled in the art that it is not essential that the shaping networks **164** and **166** be placed immediately following the integrators, but may be placced at any appropriate location elsewhere in the circuit, for example, immediately following the summing networks **182** and **184**, respectively, or may be eliminated by provision of suitable gain characteristics in the gain control amplifiers **64**, **66**, **68** and **70**.

As a further refinement, but not essential to the operation of the switching logic of the invention, it may be desirable, as taught in applicant's co-pending application Ser. No. 384,334, to connect in circuit, in advance of one or all of applifiers **64**, **66**, **68** and **70**, additional Ψ_1 networks for the purpose of placing the acoustical output phasors of loudspeakers **72**, **74**, **76** and **78** in an optimum phase position; or the amplifiers **64** and **70** may derive their outputs directly from the terminals **37** and **39** respectively, instead of deriving them from the Ψ-networks **40** and **42**.

It will be evident from the foregoing that there is provided a system for reproducing and presenting on four indepdent loudspeakers four channels of information carried as two composite signals on a two-track medium. An important aspect of the decoding apparatus which gives it its capability of creating a substantially perfect illusion of sound proceeding from four separate sources is the concept of sensing, using cross-correlation techniques, which channels have the predominant signal and switching to those channels, while at the same time attenuating the signals in the other channels, to give the illusion of four separate channels of information.

While the designations L_F, R_F, L_B and R_B have been used in the specification and claims to identify signals corresponding to particular loudspeaker locations and a particular matrixing scheme, it is to be understood that this has been done as a convenience in describing the operation of the invention in a particular arrangement of matrix and loudspeakers, and not in a limiting sense. That is, it will be recognized by ones skilled in the art that the described principles can be utilized in reproducing apparatus employing an arrangement in which four loudspeakers are positioned at locations which may not be accurately characterized as "left front," "right front," etc. and supplied from a decoder based on any suitable matrix scheme, and that in this case signal designations corresponding to the locations of the loudspeakers would be used.

I claim:

1. In apparatus for reproducing on four separate loudspeakers four separate channels of program information designated L_f, R_f, L_b and R_b contained in first and second composite signals respectively including L_f and R_f as its predominant signal component and each including sub-dominant signal components L_b and R_b, and including signal-combining networks operative to derive from said first and second composite signals a third composite signal including a dominant signal L_b and sub-dominant signals L_f and R_f and a fourth composite signal including a dominant signal R_b and sub-dominant signals L_f and R_f, and means for applying said first, second, third and fourth composite signals to respective first, second, third and fourth gain control amplifiers connected to corresponding ones of said loudspeakers, a control circuit for enhancing the realism of the four channel sound produced by said apparatus, said control circuit comprising:

first and second phase shift networks operative to position said first and second composite signals in phase-quadrature prior to application thereof to said signal-combining networks,

first circuit means connected to receive said quadrature-related first and second composite signals and operative to produce first and second alternating current signals respectively representing the cross-correlation of said first and second and of said third and fourth composite signals over a predetermined time interval,

first and second rectifying and time constant means respectively connected to receive and operative to convert said first and second alternating current signals to first and second unidirectional signals,

second circuit means operative to subtract said first and second unidirectional signals from each other to produce first and second control signals, and

means for applying said first control signal to said first and second gain control amplifiers and for applying said second control signal to said third and fourth gain control amplifiers.

2. Apparatus according to claim **1**, wherein said first circuit means includes:

means operative in response to said quadrature-related first and second composite signals to produce first, second, third and fourth auxiliary signals substantially corresponding to respective ones of said first, second, third and fourth composite signals,

first and second multiplying circuits operative continuously to multiply said first and second and said third and fourth auxiliary signals, respectively, and

first and second integrating means respectively operative to integrate the output signals from said first and second multiplying circuits over said predetermined time interval, thereby to produce integrated products of said first and second and of said third and fourth auxiliary signals.

3. Apparatus according to claim 1 wherein said predetermined time interval is approximately 6.7 milliseconds.

4. Apparatus for decoding first and second composite signals respectively containing predominant signals L_f and R_f to the extent they are present, and each containing two sub-dominant signal components L_b and R_b to the extent they are present, with the L_b and R_b components in one of said composite signals at a predetermined phase angle relative to the L_b and R_b components in the other of said composite signals, and with the L_b and R_b components in one of said composite signals leading and lagging, respectively, the L_b and R_b components in the other of said composite signals, said apparatus comprising:

all-pass phase-shifting means operative to shift the phase of one of said composite signals relative to the other by said predetermined phase angle to position the L_b and R_b components in one composite signal substantially in phase coincidence or in phase opposition with the corresponding components in the other composite signal;

combining networks operative to derive from the relatively phase-shifted composite signals from said phase-shifting means third and fourth composite signals respectively containing predominant signal components L_b $_{and\ Rb}$;

first, second, third and fourth gain control means connected to receive composite signals respectively containing predominant signal components L_f, R_f, L_b and R_b; and

control circuit means connected to receive the relatively phase-shifted composite signals from said phase-shifting means and operative to produce a first unidirectional signal of one polarity or of opposite polarity when said L_b and R_b signals instantaneously appear therein in phase coincidence or in phase opposition, respectively, and to produce a second unidirectional signal of one polarity or of opposite polarity when said L_f and R_f signals instantaneously appear therein in phase coincidence or in phase opposition, respectively, subtracting circuit means operative to subtract said first and second unidirectional signals from each other to produce first and second control signals, and means for applying said first control signal to said first and second gain control means and for applying said second control signal to said third and fourth gain control means to control the gains thereof.

5. Apparatus according to claim 4 wherein said control circuit means includes

circuit means operative in response to the relatively phase-shifted signals from said phase-shifting means to produce first and second alternating current signals respectively representing the cross-correlation of said first and second and of said third and fourth composite signals over a predetermined time interval, and

first and second rectifying means respectively connected to receive and operative to convert said first and second alternating current signals to said first and said second unidirectional signals, respectively.

6. Apparatus according to claim 5 wherein said circuit means comprises

first and second multiplying circuits respectively connected to receive and operative to continuously multiply signals substantially corresponding to said first and second composite signals and signals substantially corresponding to said third and fourth composite signals, and

first and second integrating means respectively operative to integrate the output signals from said first and second multiplying circuits over said predetermined time interval.

7. Apparatus according to claim 6 wherein said predetermined time interval is approximately 6.7 milliseconds.

8. Apparatus according to claim 4 wherein said predetermined phase angle is substantially 90°.

9. In a sound system in which first, second, third and fourth audio information signals are carried on two channels as first and second composite signals respectively containing said first and second audio information signals to the extent they are present, and each containing said third and fourth audio information signals to the extent they are present, with the third and fourth signals in one of said composite signals at a predetermined phase angle relative to the third and fourth signals in the other of said composite signals, apparatus for decoding said composite signals to derive four output signals in which said first, second, third and fourth audio information signals are respectively predominant and to enhance the instantaneously present predominant signal, said apparatus comprising:

phase-shifting means connected to receive said first and second composite signals and operative to shift the phase of one of said composite signals relative to the other by said predetermined phase angle to position the third and fourth audio information signals in one composite signal substantially in phase coincidence or in phase opposition with the corresponding audio information signals in the other composite signal;

combining networks connected to receive and operative to derive from the relatively phase-shifted composite signals from said phase-shifting means third and fourth composite signals respectively predominantly containing said third and fourth audio information signals;

first, second, third and fourth gain control means connected to receive composite signals respectively predominantly containing said first, second, third and fourth audio information signals; and

a control circuit including means connected to receive the relatively phase-shifted composite signals from said phase-shifting means and operative to produce a first unidirectional signal of one polarity or of opposite polarity when said third and fourth signals are instantaneously present therein in phase coincidence or in phase opposition, respectively, and to produce a second unidirectional signal of one polarity or of opposite polarity when said first and second signals are instantaneously present therein in phase or in phase opposition, respectively,

subtracting circuit means operative to subtract said first and second unidirectional signals from each other to produce first and second control signals, and

means for applying said first and second control signals to said first and second gain control means and to said third and fourth gain control means, respectively, to control the gains thereof.

10. In apparatus for reproducing on four sound reproduction devices adapted to be positioned at the left front, right front, left back and right back corners of a listening area a like number of audio information signals contained in first and second composite signals respectively containing predominant left front (L_f) and right front (R_f) signals to the extent they are present, and each containing sub-dominant left back (L_b) and right back (R_b) signals to the extent they are present, with the L_b and R_b components in one of said composite signals at a predetermined phase angle relative to the L_b and R_b components in the other of said composite signals and with the L_b and R_b components in one of said composite signals leading and lagging, respectively, the L_b and R_b components in the other of said composite signals, the combination comprising:

an input circuit to which said first and second composite signals are applied including all-pass phase-shifting means operative to shift the phase of one of said composite signals relative to the other by said predetermined angle to position the L_b and R_b components in one composite signal from said phase-shifting means substantially in phase coincidence or in phase opposition with the corresponding components in the other composite signal from said phase-shifting means;

signal combining means to which said relatively phase-shifted composite signals are applied and operative to produce third and fourth composite signals respectively containing predominant left back (L_b) and right back (R_b) signals; and

signal coupling means for coupling said first, second, third and fourth composite signals to respective ones of said sound reproduction devices, said signal coupling means including

control signal generating means connected to receive the composite signals from said phase-shifting means and operative to produce a first unidirectional signal of one polarity or of opposite polarity if L_b and R_b signals are instantaneously present therein in phase coincidence or in phase opposition, respectively, and to produce a second unidirectional signal of one polarity or of opposite polarity if L_f and R_f signals are instantaneously present in said third and fourth composite signals in phase coincidence or in phase opposition, respectively,

subtracting circuit means operative to subtract said first and second unidirectional signals from each other to produce first and second control signals,

first, second, third and fourth signal amplitude-modifying means connected to receive said first, second, third and fourth composite signals respectively containing predominant signal components L_f, R_f, L_b and R_b, and

means for applying said first and second control signals to said first and second signal amplitude-modifying means and to said third and fourth signal amplitude-modifying means, respectively, to control the signal transmission characteristics thereof.

11. Apparatus according to claim **10** wherein said control signal generating means includes

circuit means operative to derive from the relatively phase-shifted composite signals from said phase-shifting means four auxiliary signals of substantially constant amplitude regardless of the amplitudes of said composite signals and substantially corresponding to respective ones of said first, second, third and fourth composite signals,

circuit means operative in response to said auxiliary composite signals to produce first and second alternating current signals respectively representing the cross-correlation of the two output signals containing L_f and R_f as predominant components and of the two output signals containing L_b and R_b as predominant components, and

rectifying circuit means connected to receive and operative to convert said first and second alternating current signals to said first and second unidirectional signals, respectively.

12. In apparatus for decoding first and second composite signals respectively containing predominant first and second independent audio information signals to the extent they are present and each containing subdominant third and fourth audio information signals to the extent they are present, with the third and fourth signals in one of said composite signals in a preselected phase-angle relationship with the third and fourth signals in the other of said composite signals, to produce first, second, third and fourth composite output signals respectively containing said first, second, third and fourth audio information signals as predominant signals for reproduction on respective sound-reproducing devices and including first, second, third and fourth transmission control means for coupling said first, second, third and fourth composite output signals to their respective sound-reproducing devices, a control circuit for adjusting said transmission control means, said control circuit comprising:

means for shifting the phase of one of said composite signals relative to the other by said preselected phase-angle to position said third and fourth signals in the relatively phase-shifted composite signals either in-phase or in phase opposition,

first circuit means connected to receive said relatively phase-shifted composite signals and operative to produce a first unidirectional control signal when said third and fourth signals appear therein either in phase or in phase opposition,

signal-combining means connected to receive said relatively phase-shifted composite signals and operative to produce fifth and sixth composite signals substantially corresponding in signal content to said third and fourth composite output signals and in which said first and second independent signals, to the extent they are present, appear as subdominant signals either in phase or in phase opposition, second circuit means connected to receive said fifth and sixth composite signals and operative to produce a second unidirectional control signal when said first and second signals appear therein either in phase or in phase opposition, and

means for applying said first control signal to said first and second transmission control means and for applying said second control signal to said third and fourth transmission control means.

13. Apparatus according to claim **12**, wherein said means for applying said control signals to said transmission control means comprises means for subtracting said first and second control signals from each other.

14. Apparatus according to claim **12**, wherein said first circuit means includes

a first multiplying circuit operative continuously to multiply said relatively phase-shifted first and second composite signals, and first integrating means operative to integrate the output signal from said first multiplying circuit, and

wherein said second circuit means includes

a second multiplying circuit operative continuously to multiply said fifth and sixth composite signals, and second integrating means operative to integrate the output signal from said second multiplying circuit.

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