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Bauer et al.

[11] **3,708,631**

[45] **Jan. 2, 1973**

[54] **QUADRAPHONIC REPRODUCING SYSTEM WITH GAIN CONTROL**

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[73] Assignee: **Columbia Broadcasting System, Inc.**

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[21] Appl. No.: **44,196**

[52] **U.S. Cl.**.......**179/100.1 TD**, 179/1 G, 179/15 BT, 179/100.4 ST

[51] **Int. Cl.**.........**G11b 3/74**, H04h 5/00, H03g 3/24

[58] **Field of Search**....179/100.4 ST, 100.1 TD, 1 G, 179/1 GA, 1 GP, 15 BT; 330/124, 127, 129, 131, 134, 135, 138, 140, 141

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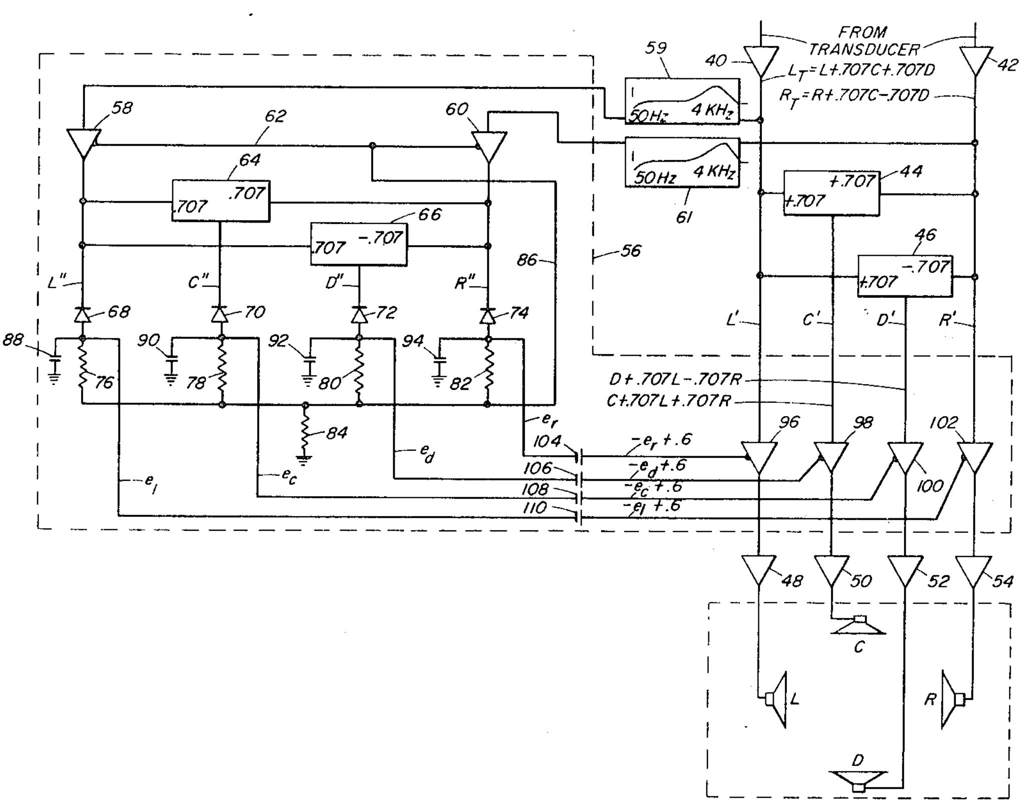
[57] **ABSTRACT**

Method and apparatus for reproducing information recorded or transmitted as four separate channels on a medium having only two independent tracks and presenting it on four loudspeakers so as to give the listener the illusion of sound coming from a corresponding number of separate sources of sound. The two tracks may be provided by any one of several available two-track systems, such as two-track tape, the stereomultiplex system of broadcasting, or a stereophonic disc record, on which two of the four channels are applied as usual, with the third and fourth channels superimposed thereon by respectively applying equal portions of them in-phase and out-of-phase to the two tracks in accordance with known practice. The reproducing apparatus includes transducer means for recovering the composite signals from the two tracks, circuit means for deriving the four channels by appropriately adding and subtracting components of the composite signals, four separate loudspeakers, and control circuitry which recognizes the channel or channels having the dominant signal and which controls the instantaneous amplitudes of signals delivered to the four loudspeakers in a manner to give a substantially perfect illusion of four separate independent sources of sound.

17 Claims, 12 Drawing Figures



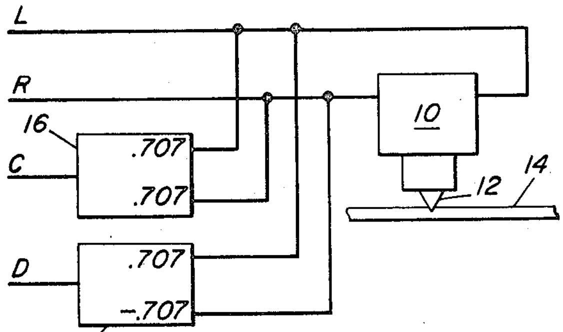


FIG. 1

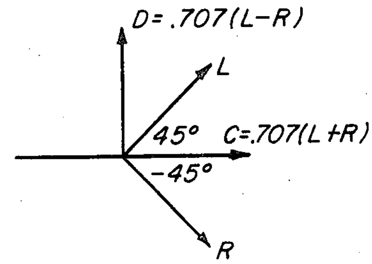


FIG. 2

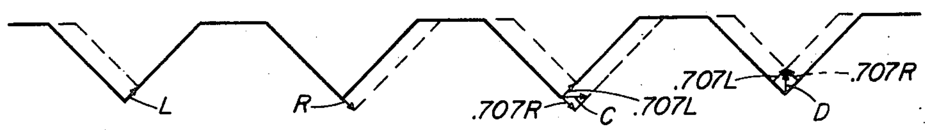


FIG. 3

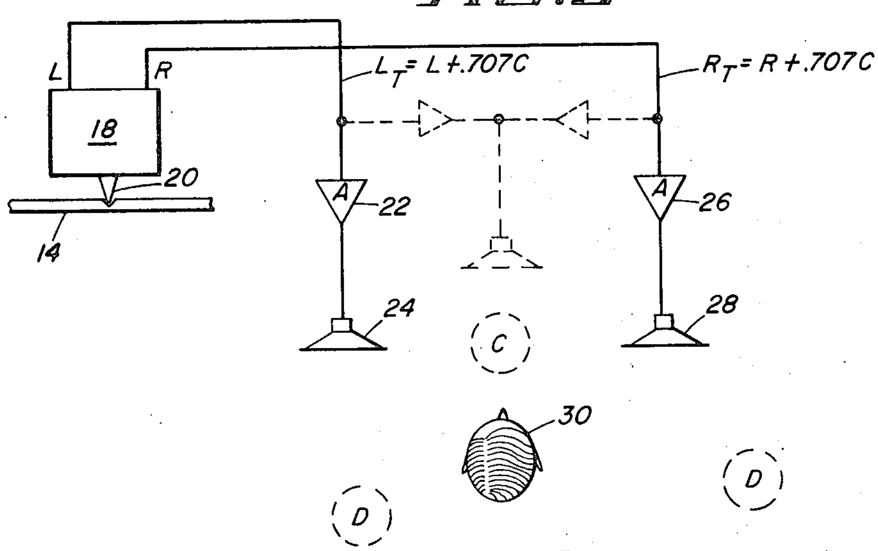


FIG. 4

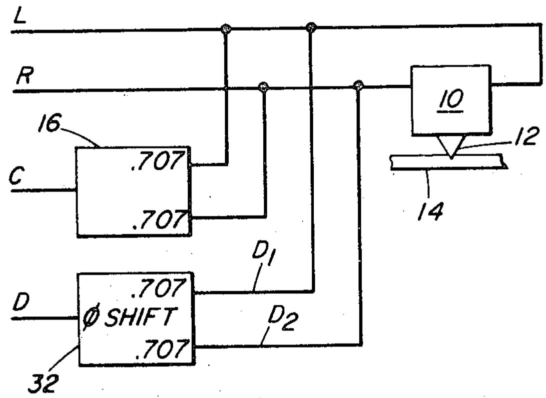


FIG. 5

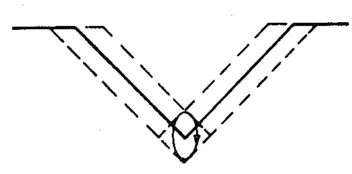


FIG. 6

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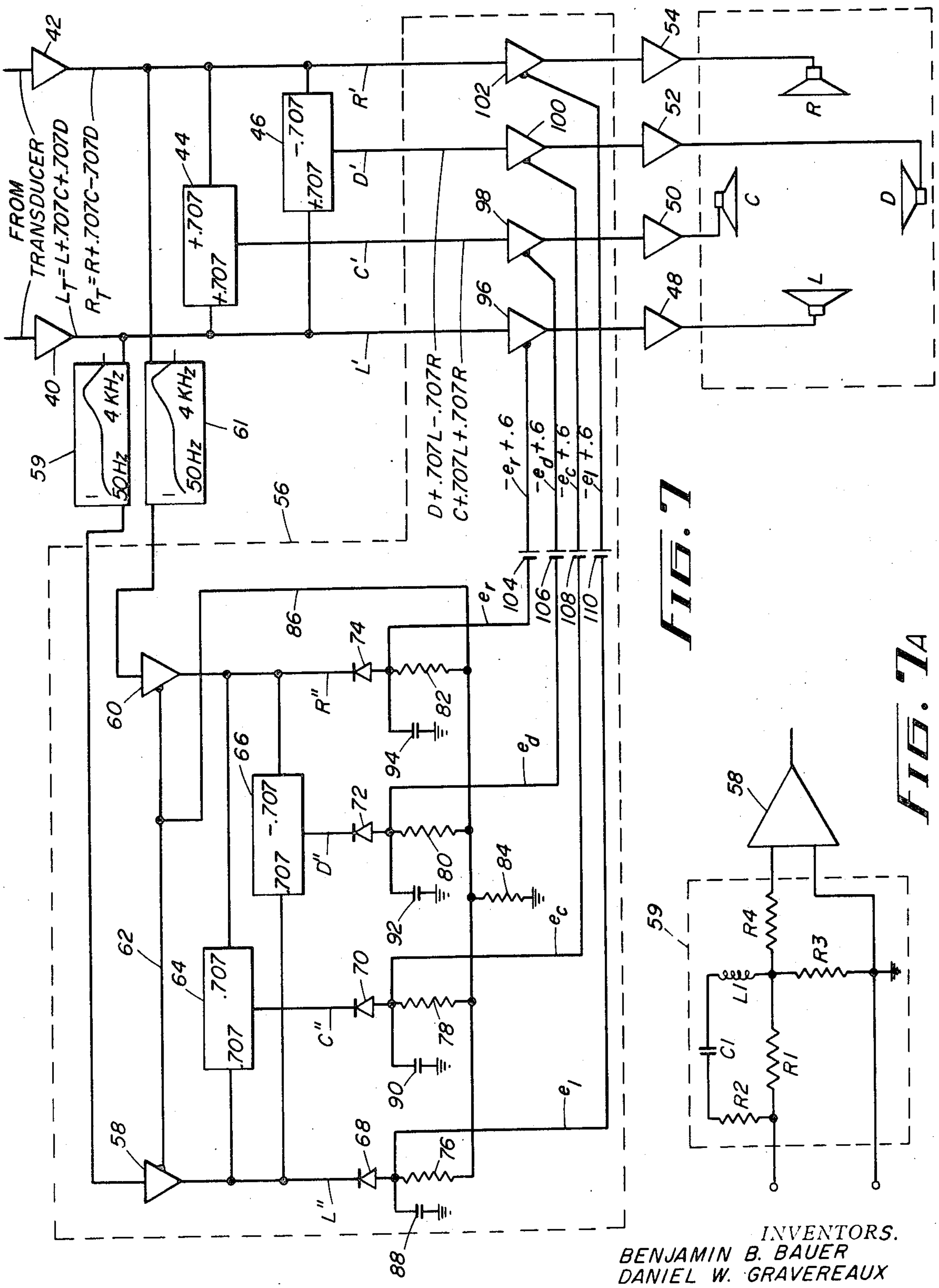

FIG. 7

FIG. 7A

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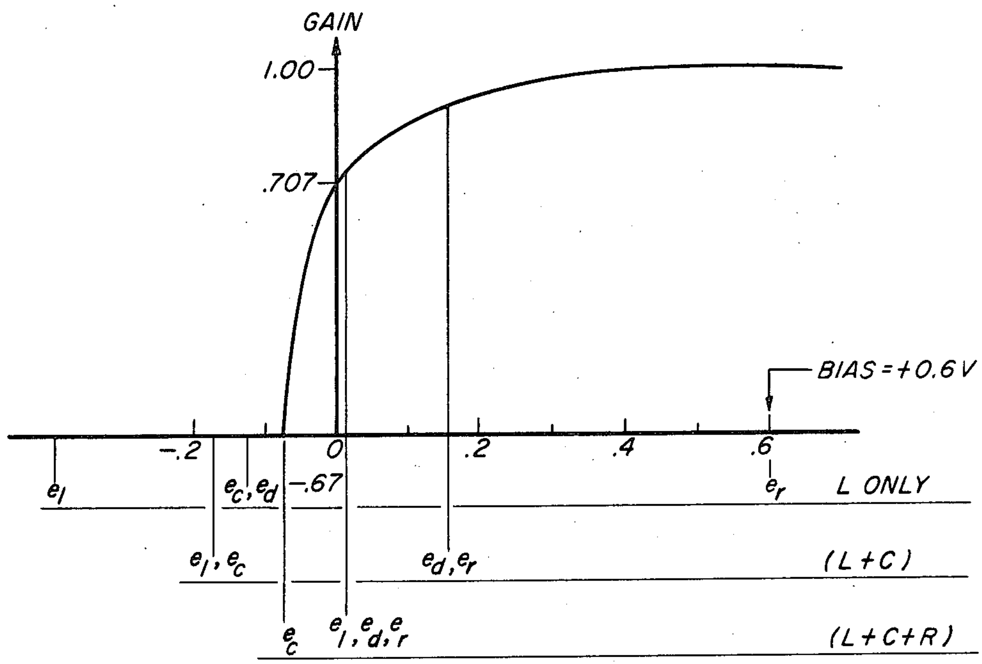

FIG. 8

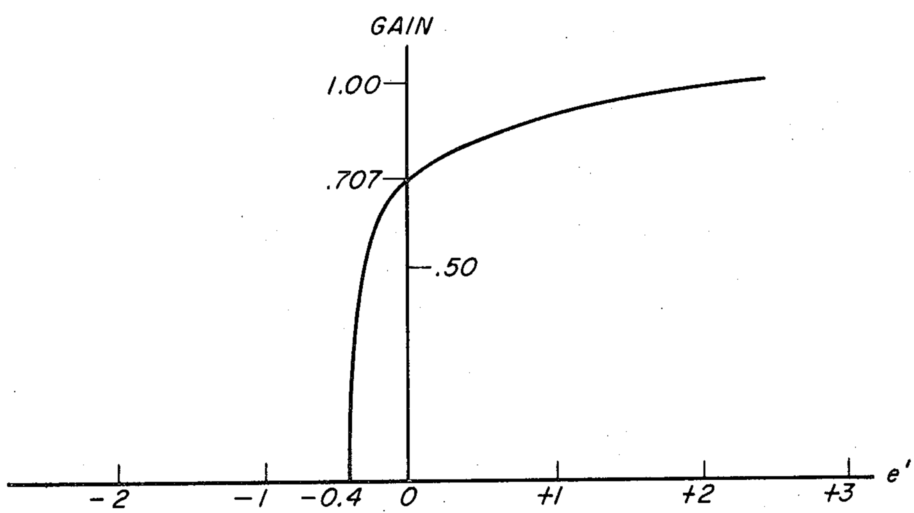

FIG. 10

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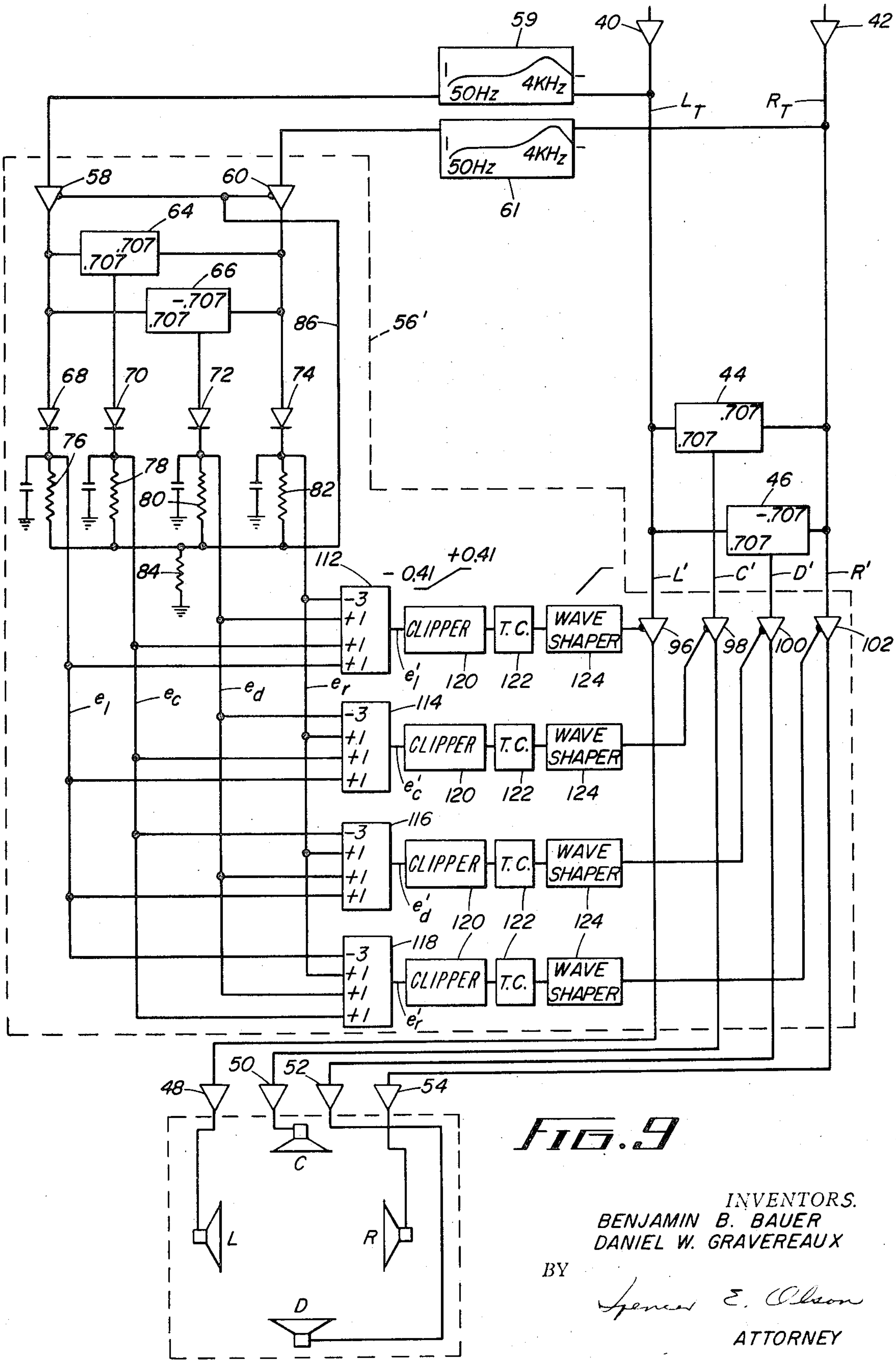

FIG. 9

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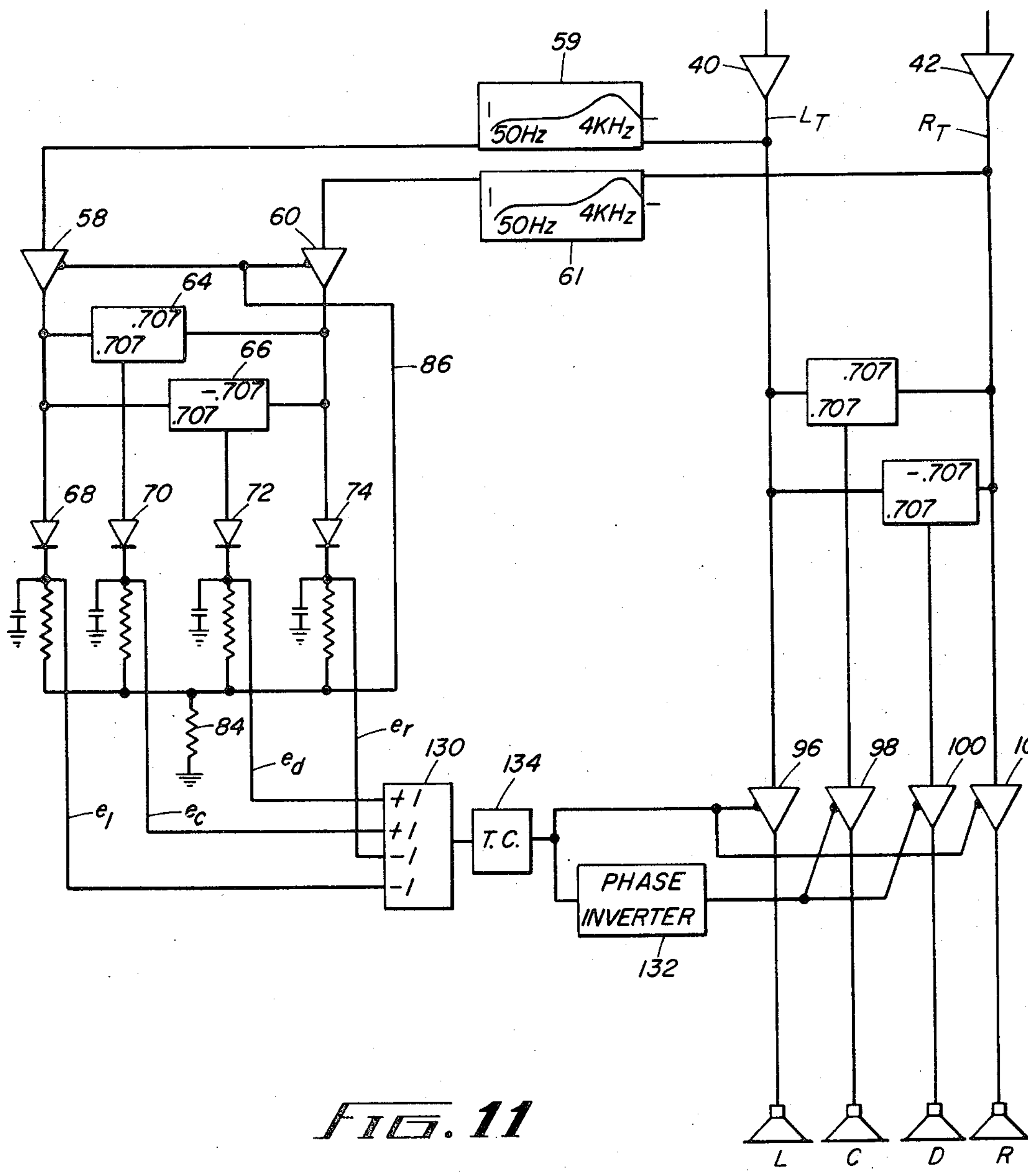

FIG. 11

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QUADRAPHONIC REPRODUCING SYSTEM WITH GAIN CONTROL

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 copending application of William S. Bachman, Ser. No. 40510 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 copending 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;

FIG. **9** is a block diagram of a preferred embodiment of the playback apparatus;

FIG. **10** is a curve showing the transfer characteristics of the logic circuitry of FIG. **9**; and

FIG. **11** is a block diagram of still another alternative embodiment of the invention.

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 ($\vec{L} + \vec{R}$), 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 preceptible 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 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 8 db.

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, 0.707 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 his 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 frequencies (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 R1, in parallel with a series branch containing a lower valued resistor R2, a capacitor C1 and an inductor L1, followed by a small resistor R3 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 R1, 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:

R1 = 5.1 Kohms
R2 = 1.6 Kohms
R3 = 91 ohms
R4 = 1.0 Kohms
C1 = 0.01 μf.
L1 = 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_1, 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_1, 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_1=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_1=0.760, e_c=0.760, e_d=0.440 and e_r=0.440.

TABLE I

Signal		e_1	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_1=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_1, 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_1, e_c, e_d and e_r to the gain control amplifiers as follows: e_1 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_1, 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** cuases 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_1, 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.

While the system of FIG. **7** is sound in principle, it suffers the disadvantage that gain control amplifiers **58** and **60** must maintain extremely close control of the voltages developed across resistors **76–82** because of the narrow range of discrimination between the "off" and "on" conditions protrayed by the control characteristic of FIG. **8**. For this reason, it has been found advantageous to employ a differential discriminator circuit, shown in FIG. **9**. The right-hand portion of this circuit is identical to the corresponding portion of the FIG. **7** circuit, and like parts are identified with like reference numerals. The logic portion **56′** differs in that the rectifiers **68, 70, 72** and **74** are reversed in polarity so that the e_1, e_c, e_d and e_r voltages developed across resistors **76, 78, 80** and **82**, respectively, are positive. Additionally, rather than directly applying these voltages to the gain control amplifiers, they are differentially added in adding elements **112, 114, 116** and **118** which are designed to weight the predominant voltage negatively by a factor of three while the remaining three voltages are added positively with a factor of one. For example, to the element **112** are applied the predominant e_r signal developed scross resistor **82**, which is added negatively by a factor of three, along with the e_1, e_c and e_d voltages, each of which are added with a weighting factor of +1. The predominant voltages applied to adding elements **114, 116, 118** are e_d, e_c and e_1, respectively. The adding elements may be of the form described on page 42 of "Applications Manual for Operational Amplifiers" published in 1968 by Nimrod Press, Boston, for Philbrick/Nexus Research, a Teldyne Company, Dedham, Massachusetts. The adding elements provide respective output voltages e'_1, e'_c, e'_d and e'_r, the values of which for the various combinations of signals discussed earlier appear in Table II below. These values are, of course, derived by applying the above weighting factors to the relative voltage values presented in Table I.

TABLE II

Signal	e_1'	e_c'	e_d'	e_r'
L	+2.414	−0.414	−0.414	−1.586
(L+C)	+0.640	+0.640	−0.640	−0.640
(L+C+R)	0	+0.414	−0.414	0
(L+C+R+D)	0	0	0	0

It is readily apparent from Table II that the discrimination range has been greatly increased by the differential adding elements, and make it possible to utilize the gain control curve illustrated by FIG. **10**. In this case, the amplifiers, which may, for example, take the form of Motorola's monolithic four-quadrant multiplier

Model MC1495L, the characteristics of which are described in the specification sheet therefor published in March 1969, have unity gain when the control signal has a value of +2.4 volts, and a gain of 0.707 (3 db down) when the control voltage is equal to zero. The charactersitic then falls off rapidly such that the amplifiers are cut off when the control voltage has a value of −0.414 volt. Analysis of the kind applied to the gain control characteristic of FIG. **8** will show that the correct amplifiers will be turned "on" in response to the predominant sounds present in the system at any given time. The gain control charactersitic of FIG. **10** being considerably less critical with respect to cutoff than the gain control curve of FIG. **8**, the system of FIG. **9** is considerably more stable. Moreover, since the voltages e'_1, e'_c, e'_d and e'_r are differential voltages, slight deviations from perfect in the gain control characteristics of amplifiers **58** and **60** are less critical to the performance of the system than in the circuit of FIG. **7**.

The control signals developed by the adding elements **112–118** are each applied through suitable wave shaping circuitry to the gain control element of a respective gain control amplifier **96–102**. Circuitry for establishing the gain control characteristic of FIG. **10** can be obtained in a variety of ways, but in an embodiment which has been successfully operated, this circuitry consists of a clipper **120** having the illustrated characteristic, with clipping action at −0.4 volt and +0.4 volt, followed by a time constant circuit **122**, which, in turn, is followed by a wave shaper **124** having a transfer characteristic as shown. This transfer characteristic may be provided with a conventional limiter circuit, or by an arbitrary function fitting circuit of the type described on page 52 of the aforementioned "Applications Manual for Operational Amplifiers." These circuits co-act to provide an amplification control charactersitic as illustrated in FIG. **10** for controlling the gain of amplifiers **94–102**. The circuit is designed to have a rapid attack, of the order of 0.1 millisecond, and a slower decay time in the order of 0.4 second. It is to be understood, however, that these are only typical values, and both are subject to a range of values without departing from the spirit of the invention.

It will be evident from the foregoing that in the circuits of FIGS. **7** and **9** all four channels operated independently. That is, at any given instant only one loudspeaker might be on, with the remainder off or at reduced gain. By the relatively simple modification shown in FIG. **11**, it is possible to switch two of the channels together, in response to a predominant signal in one of them; similarly, the remaining two channels may be switched in unison, again in response to the presence of a predominant signal in either. For example, the gain control amplifiers **96** and **102** respectively controlling the gain of the signals applied to the L and R loudspeakers may be subjected to the same control signal, and amplifiers **98** and **100** controlling loudspeakers C and D, respectively, similarly controlled by the same signal. The L and R loudspeakers may be placed in diagonally opposite corners of a listening area and the C and D loudspeakers placed in the other opposite corners of the area whereby the speakers at opposite corners go up and down together.

The control signals are derived in the same manner as in the circuits of FIGS. **7** and **9** (and a description thereof will not here be repeated) except that only one summing element **130**, which is similar in form and function to summing elements **112–118** except that it has different weighting factors, and a phase inverter **132** are needed to develop the signals for application to gain control amplifiers **96–102**. The e_1 and e_r voltages derived from the rectifiers are applied to the summing element with a weighting factor of −1, and the e_c and e_d voltages are applied with a weighting factor of +1. It will be evident that if either the R or L channels are alone present, it will produce a control signal, which may be modified by a suitable time constant circuit **134**, for application to amplifiers **96** and **102** to turn "on" both the L and R loudspeakers. But, since only one of these channels is "on," sound will come out of the corresponding loudspeaker. Similarly, if both the L and R channels are simultaneously present, with no signal in the C and D channels, the L and R loudspeakers will again be turned on, each reproducing its respective signal, and thus simulating a pair of independent channels.

Similarly, the presence of either of channels C and D only produces a control signal, which by reason of the +1 weighting, must be inverted to give the proper polarity to turn amplifiers **98** and **100** on. At the same time, the non-inverted signal applied to amplifiers **96** and **102** turns them off or substantially lowers their gains. It follows that if both channels C and D are present, the same two loudspeakers, namely, C and D, will be turned on and the other two turned off. Since each of loudspeakers C or D reproduces only its appropriate channel, the result is a good simulation of two additional channels.

Should signals be present simultaneously in any two adjacent channels, the control signal turns out to be zero, and since the phase inversion in this case has no effect, all four amplifiers are turned on, but at reduced gain; i.e., 3 db down.

It will be evident from the foregoing that applicants have provided systems for reproducing and presenting on four independent loudspeakers four channels of information carried as two composite signals on a two-track system. An important aspect of the reproduction apparatus which gives it its capability of creating a substantially perfect illusion of sound proceeding from four separate sources is the concept of sensing which channel has the preponderant signal and switching to that channel, while at the same time attenuating the signals in the other channels, in response to the transients of musical sounds, to give the illusion of four separate channels of information.

An important objective of the invention has been achieved in the embodiments shown in FIGS. **7** and **9**, in which each of the four loudspeakers control circuits is controlled depending on the dominance of the respective signal, and entirely independently of the other three control circuits. Thus, it is possible to turn "on" one circuit at a time. This is not the case with the embodiment in FIG. **11**, which, while possessing the virtue of greater simplicity does not exhibit as precise a degree of control as the other two circuits.

It is again emphasized that although the stereophonic disc record has been selected as the medium with which to explain the recording and playback techniques, it is to be understood that the playback method and apparatus responds in the same manner to similar composite signals regardless of the medium from which they are transduced.

We claim:

1. In apparatus for reproducing on four separate loudspeakers adapted to be arranged around a listener four individual audio information signals contained in first and second composite signals each of which includes a combination of at least three of said audio information signals with preselected phase and amplitude relationships, the combination comprising:

first and second input terminals to which said first and second composite signals are respectively applied,

a first signal combining network connected to said input terminals and operative to derive by combining the signals in said composite signals with preselected amplitude and phase relationships four output signals predominantly containing signals representing the information in respective ones of said four individual signals each accompanied by lower level signals representing information contained in others of said individual signals,

signal coupling means including first, second, third and fourth variable gain amplifiers for separately coupling one of said output signals to a respective one of said four loudspeakers, and

gain control means for varying the gain of said variable gain amplifiers so as to selectively amplify the output signals or signals instantaneously predominant relative to the remaining output signals, said gain control means including

control circuit means connected to said first and second input terminals and operative to derive from said first and second composite signals four auxiliary signals containing audio information substantially corresponding to the audio information contained in corresponding ones of said four output signals, and

means for continuously comparing the instantaneous amplitudes of said auxiliary signals.

2. A method of reproducing four individual audio information signals which are contained in first and second composite signals each of which includes a combination of at least three of said audio information signals with preselected phase and amplitude relationships, comprising the steps of

deriving said four output signals from said first and second composite signals by

deriving one pair of output signals by combining said first and second composite signals with preselected phase and amplitude relationships wherein different phase relationships are used to derive each signal of said pair of signals, and

deriving the remaining output signals from respective ones of said first and second composite signals, whereby a different desired one of said four audio information signals is predominant in each of said four output signals,

deriving from said first and second composite signals four auxiliary signals containing audio information substantially corresponding to the audio information contained in corresponding ones of said four output signals,

comparing said auxiliary signals to produce control signals, and

controlling the gain of at least some of said four output signals depending on a preselected comparison of said four auxiliary signals.

3. In apparatus for reproducing on at least three sound reproduction devices an equal number of individual audio information signals contained in first and second composite signals each of which includes a combination of at least three of said audio information signals with preselected amplitude and phase relationships, the combination comprising,

signal combining means to which said first and second composite signals are applied and operative to produce output signals equal in number to the number of individual audio information signals contained in said composite signals and each predominantly containing signals representing the information in a respective one of said individual audio information signals, and

signal coupling means for coupling said output signals to respective ones of said sound reproduction devices, said signal coupling means including

control signal generating means connected to receive said first and second composite signals and operative to derive therefrom auxiliary signals of substantially constant amplitude regardless of the amplitudes of said composite signals,

means for combining said auxiliary signals to produce control signals corresponding to said audio information signals and at any given instant to produce one or more control signals corresponding to the predominant audio information signal or signals then present in said composite signals, and

amplitude-modifying means connected to receive said control signals and operative in response thereto to enhance the amplitude of the output signal or signals which instantaneously contain predominant audio information signals relative to the other output signals.

4. Apparatus according to claim 3 wherein said amplitude modifying means provides a signal transmission function which remains at a normal level when substantially all of the audio information signals contained in said composite signals have approximately the same amplitude.

5. In apparatus for reproducing on four separate loudspeakers adapted to be arranged around a listener four separate channels of program information contained in first and second composite signals respectively containing first and second component signals representing the program information in first and second ones of said channels, and each containing component signals representing the program information in third and fourth ones of said channels, the combination comprising:

first and second input terminals for receiving said first and second composite signals, respectively,

a first signal combining network connected to said input terminals and operative to derive from said composite signals four output signals predominantly containing signals representing the information in respective ones of said four channels each accompanied by lower level signals representing information contained in others of said channels,

signal coupling means including first, second, third and fourth variable gain amplifier means for receiving and separately coupling a respective one of said four output signals to a respective one of said four loudspeakers, and

control circuit means having a pair of input terminals connected to respective ones of said first and second input terminals and four output terminals connected to respective ones of said variable gain amplifier means, said control circuit means being operative in response to said first and second composite signals to produce four auxiliary signals containing program information substantially corresponding to the program information contained in respective ones of said four output signals and to detect the predominant signal or signals instantaneously present in said auxiliary signals and to produce control signals for varying the gain of said variable gain amplifier means so as to increase separation between adjacent loudspeakers.

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

means for continuously comparing the relative amplitudes of said four auxiliary signals and operative to produce control signals for varying the gain of the variable gain amplifier means carrying the output signals instantaneously containing said predominant signal or signals.

7. Apparatus according to claim 6 wherein said control circuit means includes means for applying said control signals to the variable gain amplifier means and operable to maintain substantially equal the gains of the variable gain amplifier means associated with the first and the fourth of said loudspeakers and to maintain substantially equal the gains of the variable gain amplifier means associated with the second and the third of said loudspeakers.

8. Apparatus according to claim 6 wherein said comparing means includes means for separately rectifying said four auxiliary signals, and means operative in response to said rectified auxiliary signals to maintain the average value of the sum of said rectified auxiliary signals substantially constant.

9. Apparatus according to claim 8 wherein said comparing means further includes at least one signal weighting network to which all of said rectified auxiliary signals are applied for producing a signal indicative of which of said output signals contains said instantaneously predominant signal or signals.

10. Apparatus according to claim 8 wherein said comparing means further includes a single signal weighting network to which all of said rectified auxiliary signals are applied, said signal weighting network being operative to combine two of said rectified auxiliary signals modified by a given weighting factor with the other two of said rectified auxiliary signals modified by a different weighting factor to produce a control signal indicative of which of said two sets of rectified auxiliary signals contains said instantaneously predominant signals or signals, means for applying said control signal in one phase to the control terminals of said first and fourth variable gain amplifier mean, and means for applying said control signal with opposite phase to the control terminals of said second and third variable gain amplifier means.

11. In apparatus for reproducing on four separate loudspeakers four separate channels of program information contained in first and second composite signals respectively containing first and second component signals representing the program information in first and second ones of said channels, and each containing in-phase portions of a signal representing the program information in a third one of said channels and phase-displaced portions of a signal representing the program information in a fourth one of said channels, the combination comprising:

first and second input terminals to which said first and second composite signals are respectively applied,

first signal transfer means including a pair of summing networks connected to said input terminals and operative to add and subtract selected components of said composite signals to produce four output signals predominantly containing signals representing the program information in respective ones of said four channels each accompanied by signals representing information contained in its adjacent channels,

first, second, third and fourth signal coupling means, each including variable gain amplifier means, for coupling said four output signals to respective ones of four loudspeakers adapted to be arranged around a listener,

control circuit means connected to respective ones of said first and second input terminals and including second signal transfer means operative in response to said first and second composite signals to produce four auxiliary signals containing program information substantially corresponding to the program information contained in respective ones of said four output signals,

signal comparing means operative to continuously compare the relative amplitudes of said four auxiliary signals and to produce control signals indicative of which signal or signals is instantaneously predominant among said four output signals, and

means for coupling said control signals to said variable gain amplifier means and for varying the gain of said variable gain amplifier means so as to increase separation between adjacent loudspeakers.

12. Apparatus according to claim 11 wherein said control signal coupling means includes

a logic circuit operative to momentarily apply a gain-varying control signal to the variable gain amplifier means carrying the output signals corresponding to the instantaneously predominant signal or signals.

13. Apparatus according to claim 12 wherein said control circuit means further includes first and second signal shaping networks connected between said first and second input terminals and said second signal transfer means, said first and second signal shaping networks having like transmission characteristics approximating the equal loudness contour of the human ear at moderate loudness level and over the audio range of interest and operative to modify said first and second composite signals, respectively, according to said transmission characteristics.

14. Apparatus according to claim 12 wherein said second signal transfer means includes first and second gain control amplifiers, each having an input terminal to which said modified first and second composite

signals are respectively applied, a control terminal and an output terminal, means including summing networks connected to the output terminals of said first and second gain control amplifiers for adding and subtracting selected components of said modified composite signals to produce and apply to said signal comparing means said four auxiliary signals, and wherein said signal comparing means includes means for separately rectifying the four auxiliary signals and means for applying a control signal to the control terminal of both said first and second gain control amplifiers of a magnitude to so control their gain as to maintain the average value of the sum of said rectified four auxiliary signals substantially constant.

15. Apparatus according to claim **14** wherein said logic circuit includes at least one signal weighting network to which said rectified four auxiliary signals are applied.

16. Apparatus according to claim **14** wherein said logic circuit includes first, second, third and fourth signal weighting networks to each of which all of said rectified four auxiliary signals are applied, each of said weighting networks being operative to combine a respective one of said rectified auxiliary signals modified by a given weighting factor with the other three rectified auxiliary signals modified by a different weighting factor.

17. Apparatus according to claim **16** wherein said control signal coupling means includes first, second, third and fourth wave-shaping networks respectively connected between said first, second, third and fourth signal weighting networks and respective ones of said variable gain amplifier means.

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