United States Patent

[11] 3,632,886

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[54] **QUADRASONIC SOUND SYSTEM**
27 Claims, 18 Drawing Figs.

[52] U.S. Cl. .... 179/15 BT, 179/100.4 ST, 179/1 G
[51] Int. Cl. H04h 5/00
[50] Field of Search .... 179/1 G, 1 GP, 1 GA, 100.4 ST, 100.1 TD, 15 ST, 15 MM, 15 BC; 325/36; 350/118, 119

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Assistant Examiner—Tom D'Amico
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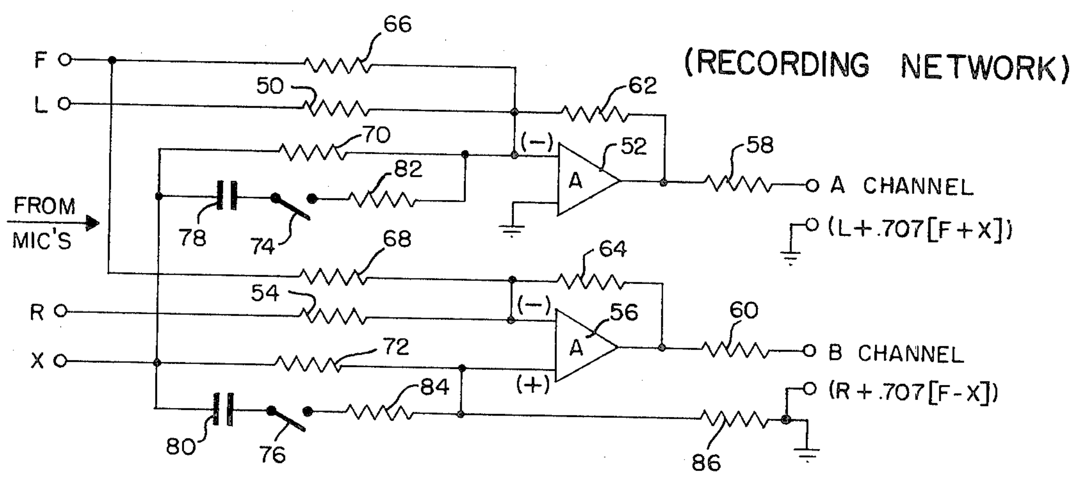
ABSTRACT: A stereophonic sound system is disclosed utilizing a two-channel transmission path yet capable of locating virtual sound sources at any point on a circle around a listener. The two-channel transmission path may consist of conventional stereophonic channels such as records, tapes, broadcasting channels, etc. The recording or transmitting means (as the case may be) of the invention provides two audio signals which may comprise preselected combinations of four (for example) directional inputs. One channel may include the first input plus a signal proportional to the sum of the second and fourth inputs, while the second channel consists of the third input plus a signal proportional to the difference between the second and fourth inputs.

The sound reproducing means couples these signals and various combinations thereof to four (for example) loudspeakers which may be arranged on the circumference of a circle around the listener. The first speaker may be responsive to the signal on one channel, the next adjacent speaker is responsive to the sum of the signals on the two channels, the third successive speaker is responsive to the second channel, and the last speaker is responsive to the difference between the signals on the two channels. Means are disclosed for controlling the gain in the signal paths of the voltages coupled to the various speakers relative to the other speakers to increase the audio separation between adjacent speakers and thus enhance the directional effect.



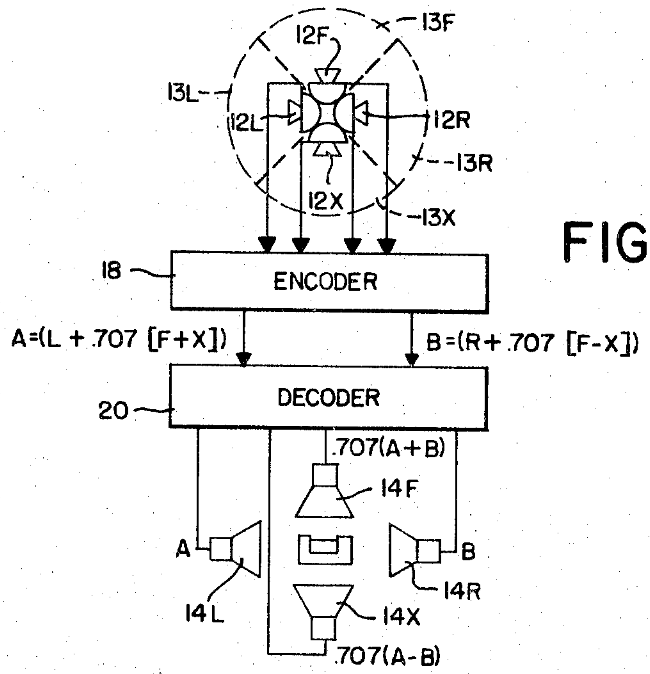
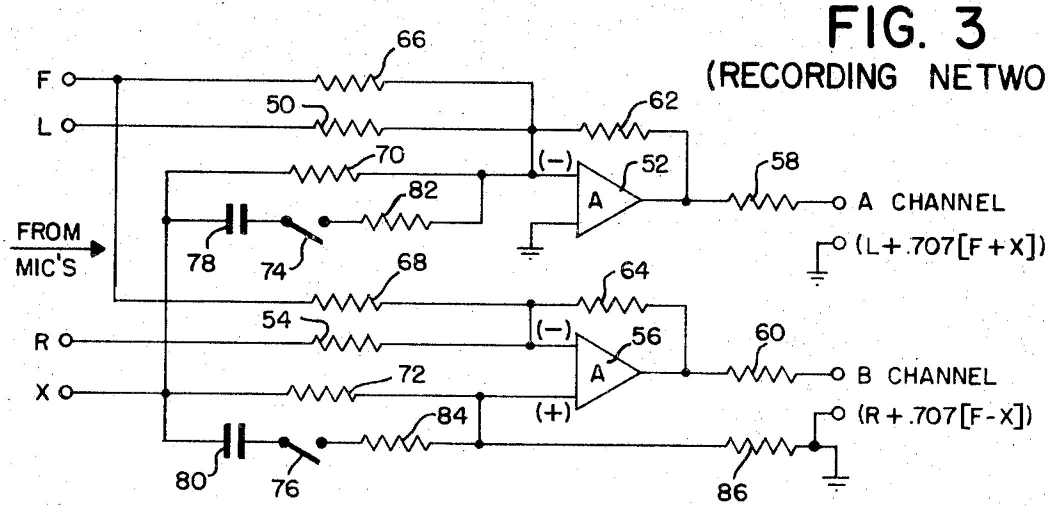


FIG. 1

FIG. 3
(RECORDING NETWORK)



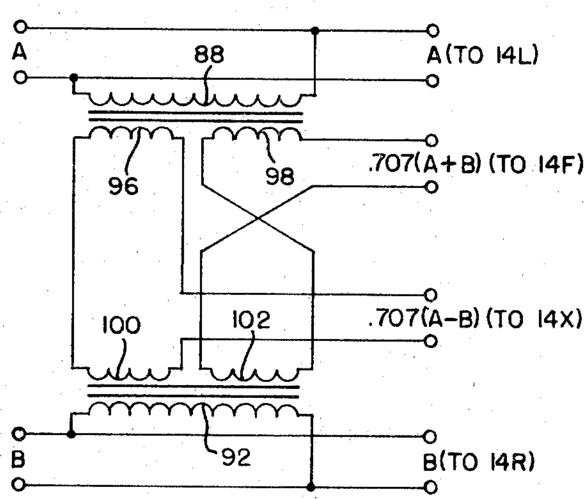


FIG. 4
(PLAYBACK NETWORK)

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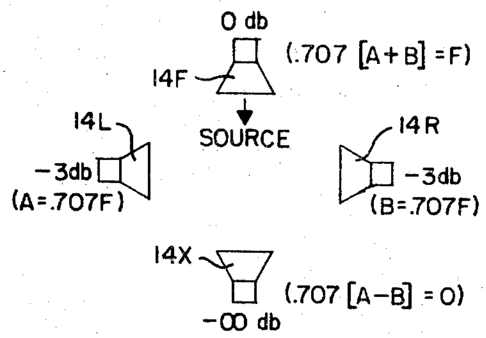
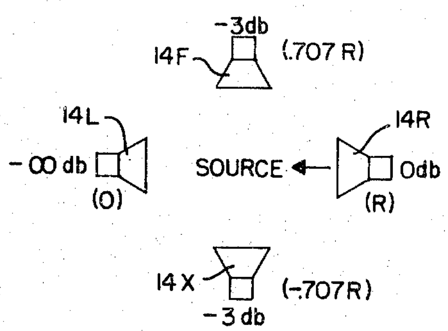


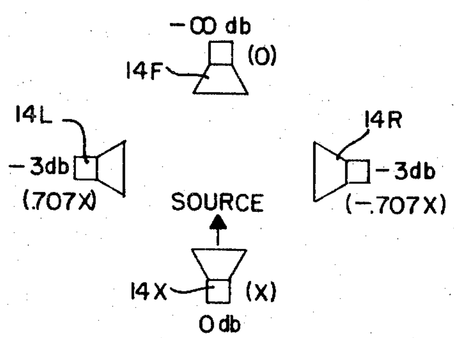
SOURCE IN FRONT
F=1, L=0, X=0, R=0
A=.707F
B=.707F

FIG. 2A



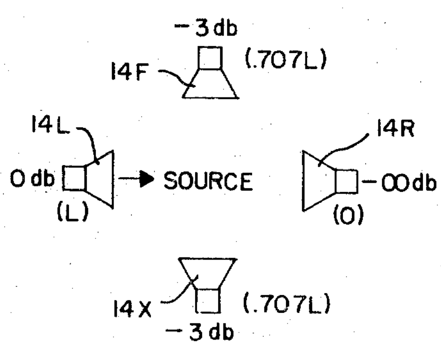
SOURCE AT RIGHT
R=1, F=0, L=0, X=0
A=0
B=R

FIG. 2B



SOURCE AT REAR
X=1, R=0, F=0, L=0
A=.707X
B=−.707X

FIG. 2C



SOURCE AT LEFT
L=1, X=0, R=0, F=0
A=L
B=0

FIG. 2D

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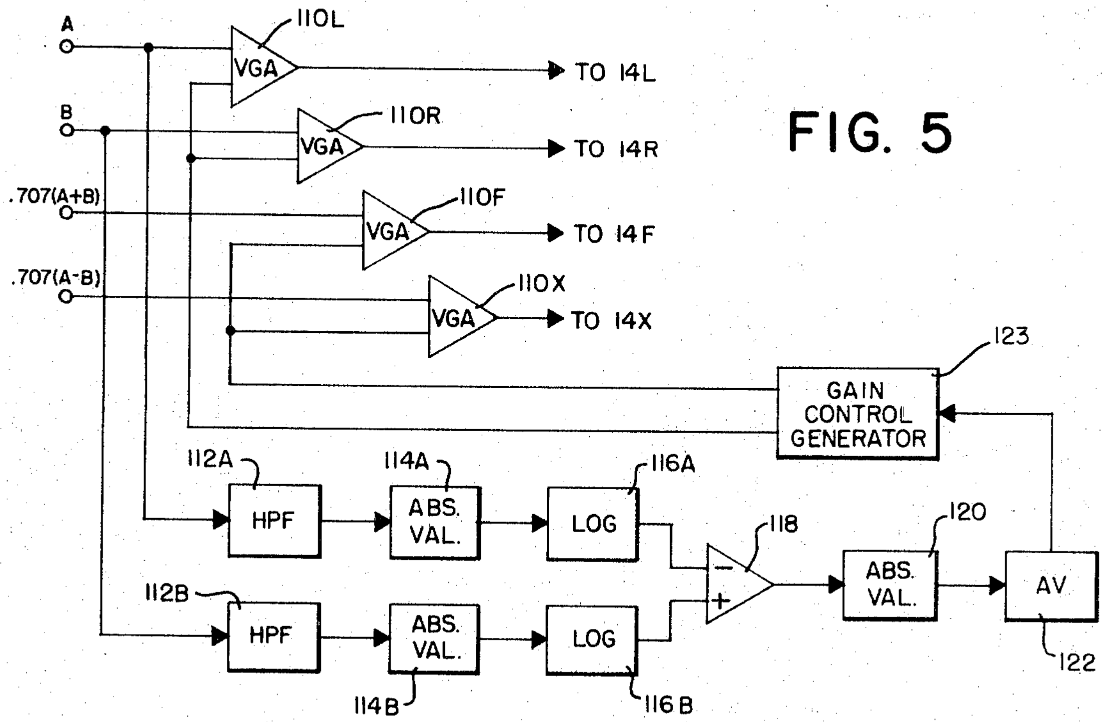

FIG. 5

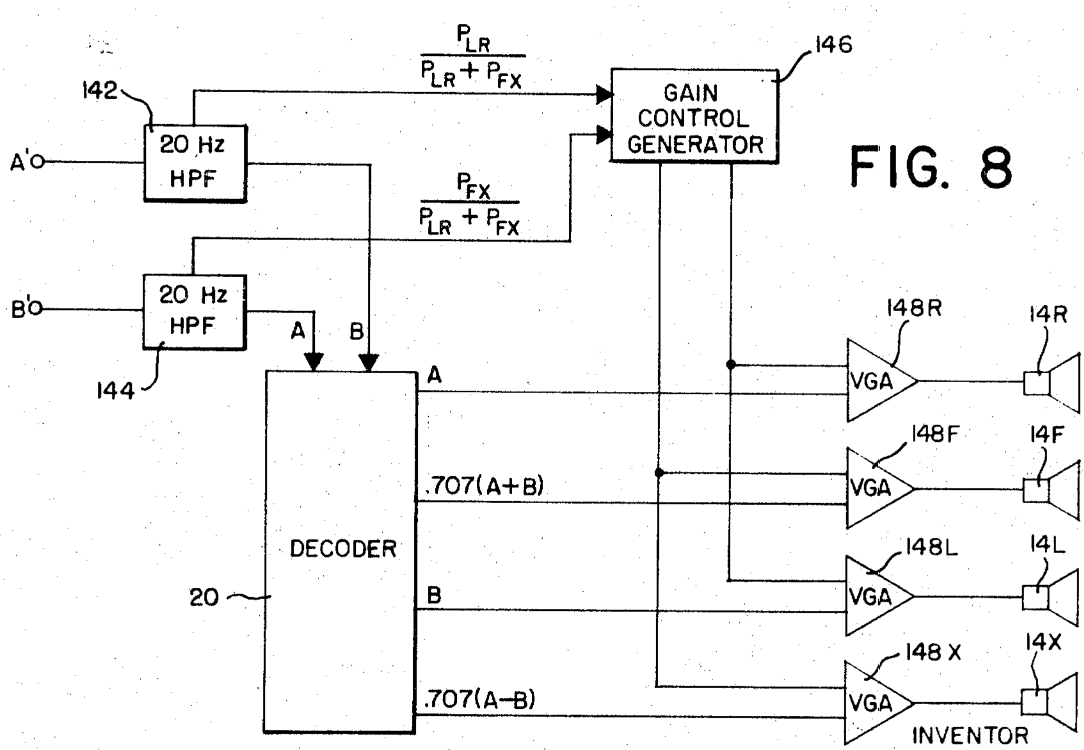

FIG. 8

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FIG. 10

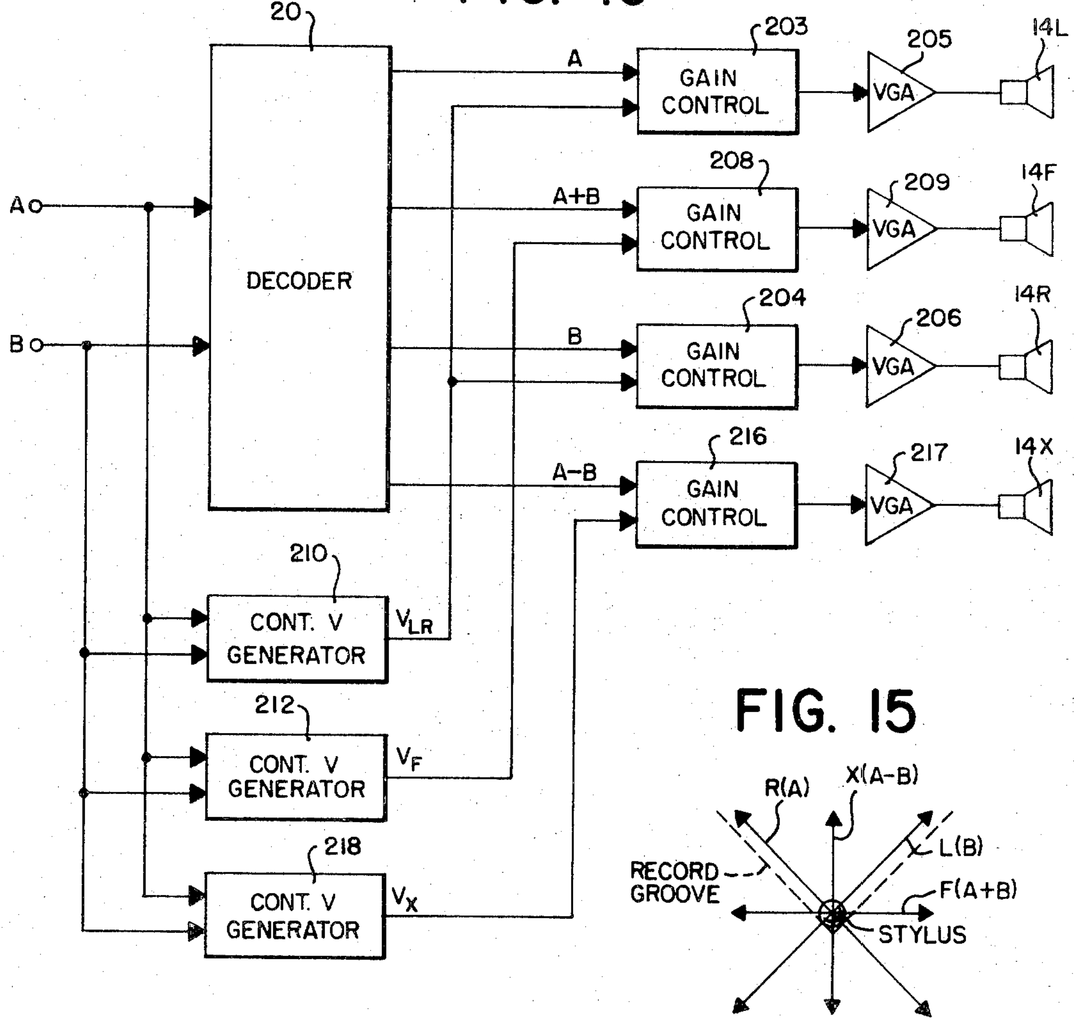
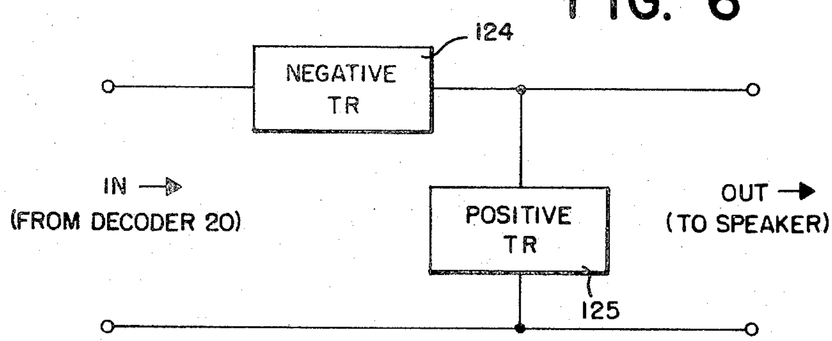


FIG. 15

FIG. 6



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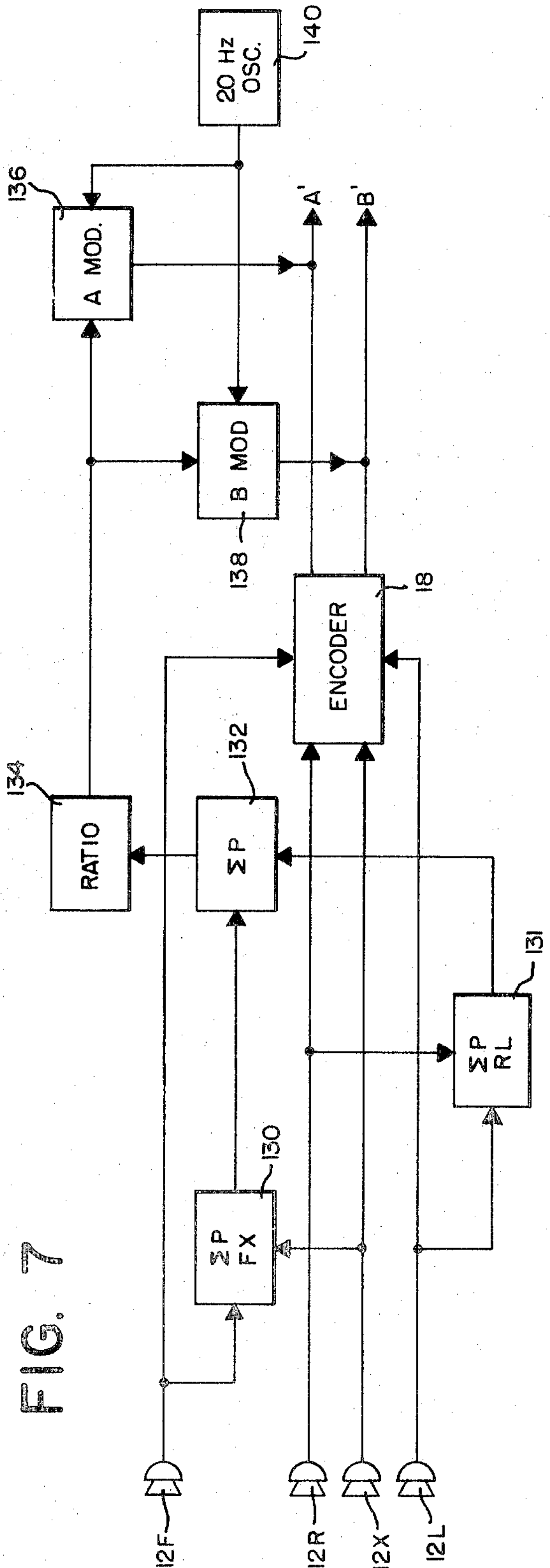

FIG. 7

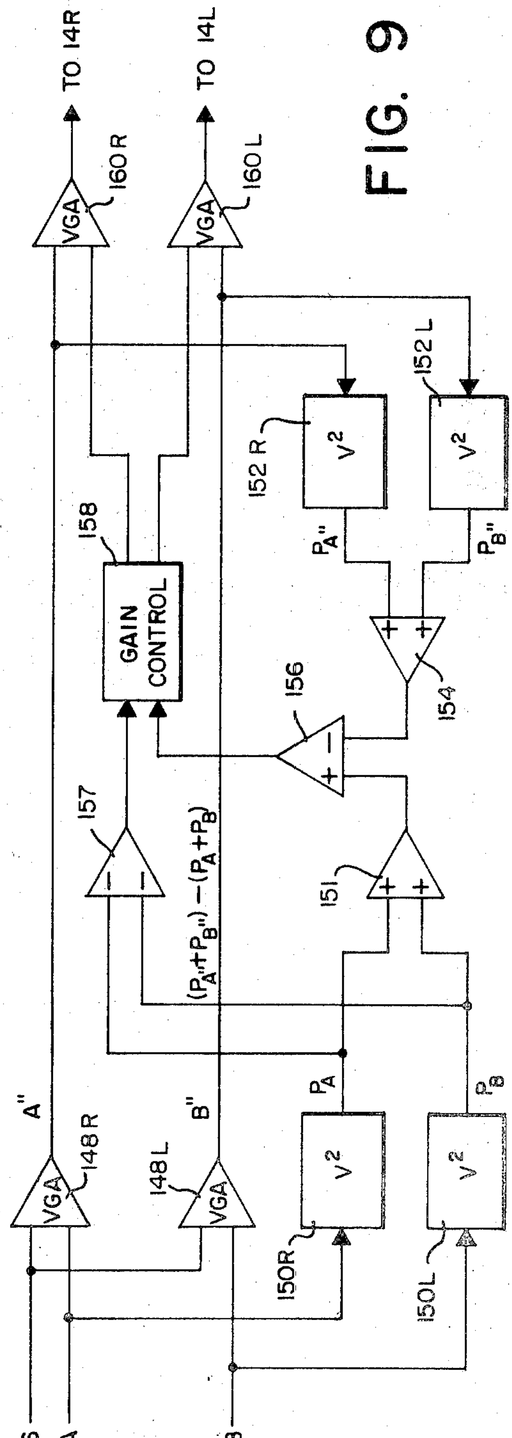

FIG. 9

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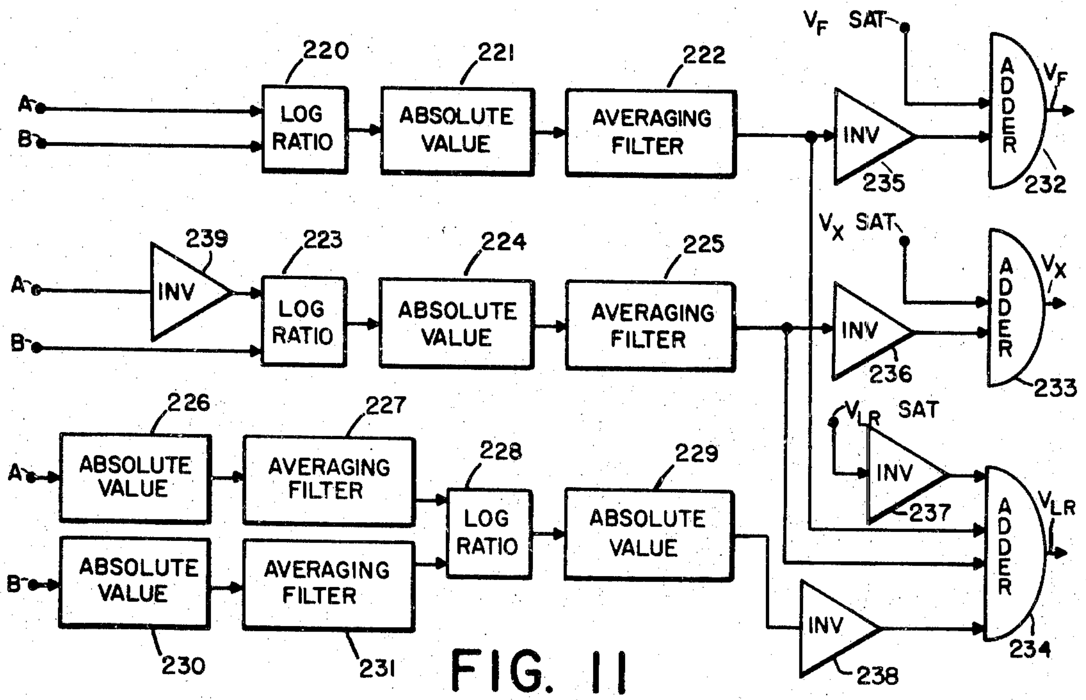

FIG. 11

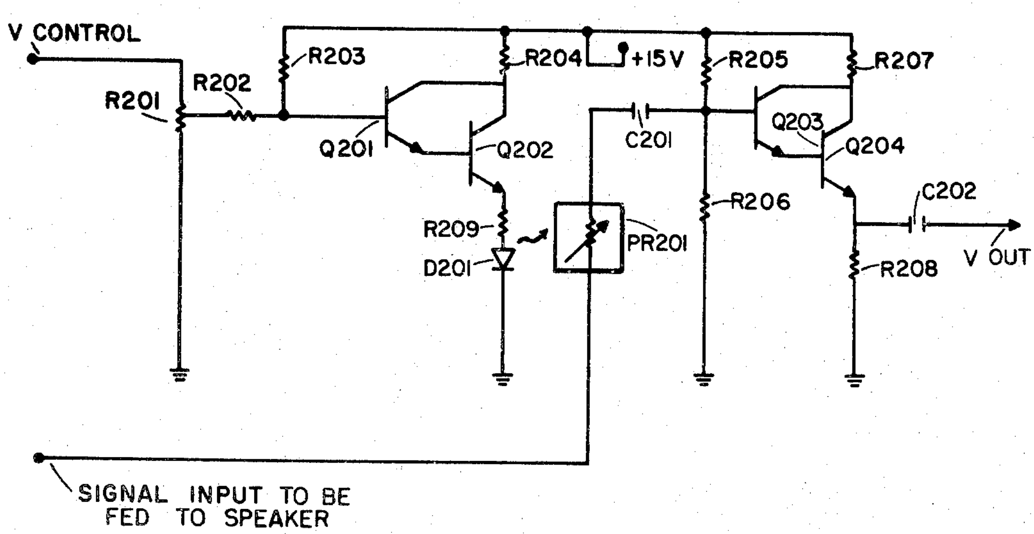

FIG. 12

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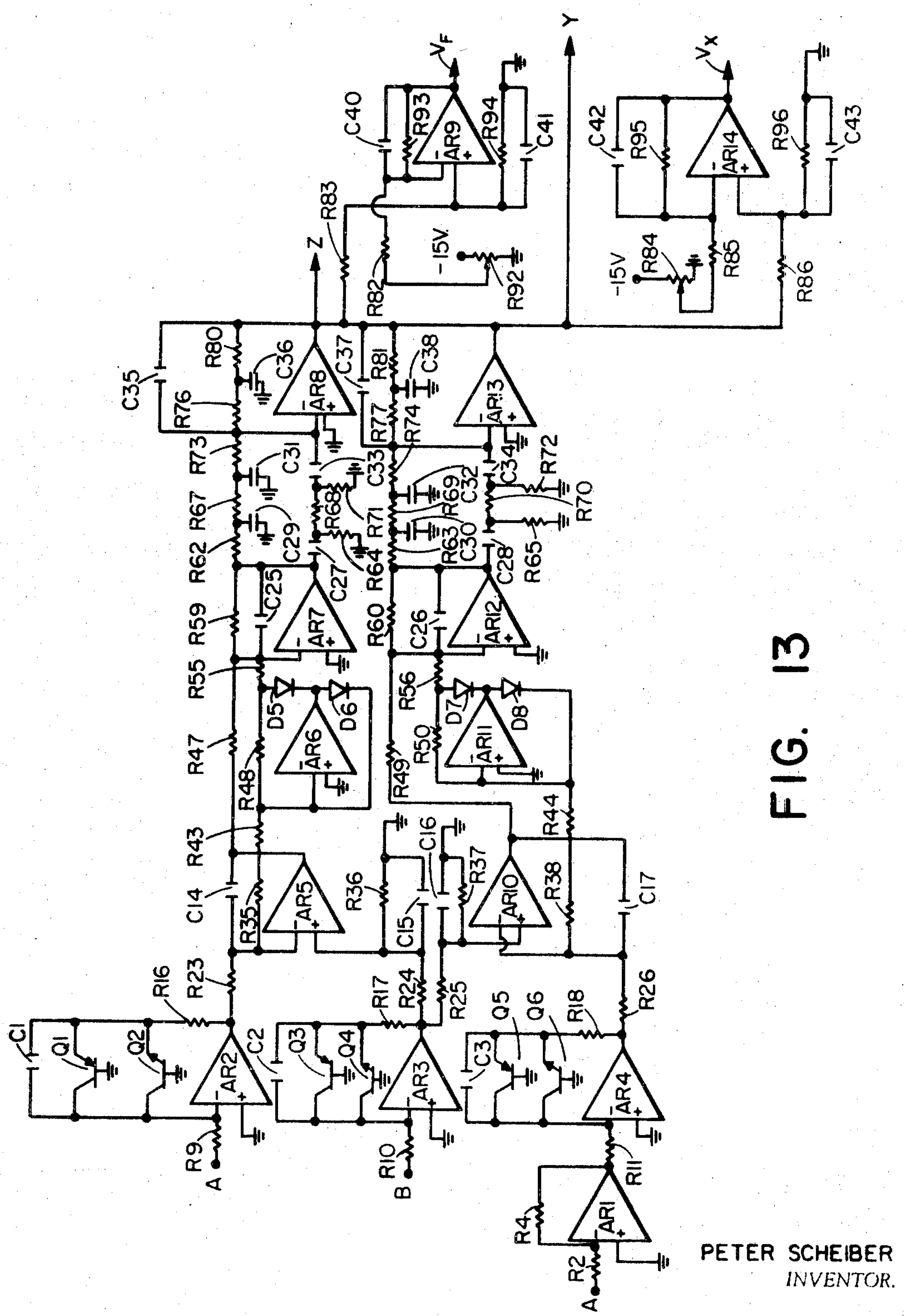

FIG. 13

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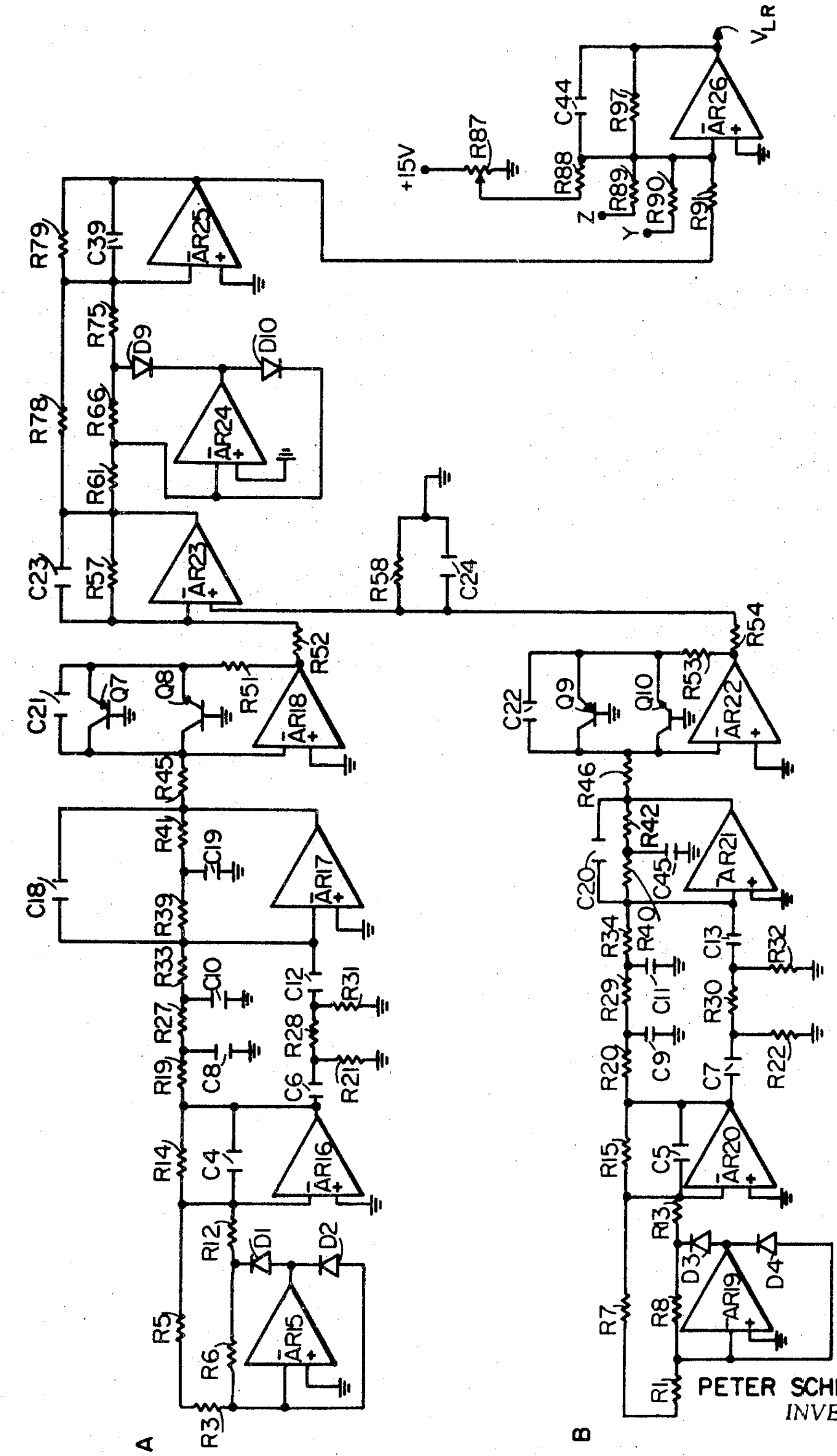

FIG. 14

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QUADRASONIC SOUND SYSTEM

This is a continuation-in-part of U.S. Pat. Application, Ser. No. 853,822 filed on Aug. 28, 1969 in the name of Peter Scheiber and entitled STEREOPHONIC RECORDING AND TRANSMISSION SYSTEM and U.S. Pat. application, Ser. No. 697,103, filed on Jan. 11, 1968 in the name of Peter Scheiber and entitled STEREOPHONIC SOUND SYSTEM.

The present invention relates to audio systems and, in particular, to a stereophonic sound system which is capable of locating virtual sound sources at any point on a full circle around a listener.

PRIOR ART

Commercial stereophonic systems usually include two separate sound channels. This helps to recreate actual listening conditions by effectively increasing the region from which sound sources can emanate. Where two loudspeakers are used, a virtual sound source may be located at either of the two speakers or any point in between. In this respect, a two-channel or binaural system is an improvement over a monaural system which utilizes a single speaker and is therefore only capable of locating a sound source at that speaker.

It has been proposed to use a third speaker positioned between a pair of outer speakers with the third speaker being fed by a simple additive combination of the signals on the two stereo channels. To further improve the audio effect, it is known to record a third channel of audio information on the two stereophonic channels with equal amplitude and same polarity so that when the signals in the two stereo channels are added at the reproducing end, a third channel signal is formed which can locate a source directly at the third speaker. However, even this third channel cannot locate a sound source behind the listener (assuming proper placement of the third channel speaker) because the relatively equal signals appearing in the two existing channels produce a sound "image" between them resulting in a conflict as to the direction of the source.

The logical extension of a binaural system would include third and fourth channels with speakers positioned in front of and behind the listener and such systems (known as quadrasonic systems) are already commercially available in (at least) four channel tape systems. Where a four-channel system is used, it is possible to locate a source of sound at any point on a full circle around the listener. This ability to locate sound sources behind a listener has a significant audio effect and compares with binaural systems in much the same way a binaural system compares with a monaural system.

Where a full four-channel system is used, there are no special problems involved in feeding audio signals to the respective speakers to create the desired effect. For example, four microphones may be used at the pickup end, corresponding to four different directions, with the output of each microphone coupled by a separate transmission path (e.g., a tape channel) to suitably placed speakers on the circumference of a circle and corresponding to the four microphones. However, as indicated previously, virtually all stereophonic sound systems presently in use include only two signal transmission paths. For example, conventional stereophonic records include grooves wherein the surfaces are inclined at 45 degrees with respect to the record face, each surface serving as the equivalent of a separate "transmission path." FM multiplex broadcasting techniques approved by the U.S. Federal Communications Commission and currently in widespread use, provide for the transmission of only two separate signals and, obviously, most multiplex receivers presently used are capable of receiving only these two separate channels. Most tape recording equipment in use today records and reproduces only two channels and could not readily be modified to record and/or play back four separate audio signals, as presently required for full-circle sound.

BRIEF DESCRIPTION OF THE INVENTION

The present invention provides a stereophonic system which is capable of locating a sound source at any place on a full circle around a listener wherein the information is transmitted (or recorded) on only two separate transmission paths. The invention may be used with three speakers, but, preferably, four separate speakers are used.

In accordance with the invention, a signal is transmitted on one or both channels with a selected relative polarity or phase difference (and, preferably, gain also) depending upon the desired direction of the source represented by that signal. For example, the system may employ four separate microphones each of which is adapted to respond to sound sources from a preselected quadrant of the circle around the microphones. The signals from two oppositely directed microphones are connected directly to the respective stereo channels. Additive and subtractive combinations of the other two microphones are also coupled to these two channels so that when the signals are replayed (or received) with suitable combining, the input to any microphone will be effectively "located" or replayed at a corresponding speaker. The invention is fully compatible with existing binaural systems, and produces two outputs (at the recording or transmitting end) which can be replayed over existing binaural equipment.

At the receiver or playback end, the four speakers are fed successively with (1) the signal on the first channel, (2) the sum of the signals on the two channels, (3) the signal on the second channel, and (4) the difference of the signals on the two channels. As explained in detail in the following specification, this permits location of a virtual sound source at any point on a full circle around the listener.

The invention also contemplates the use of special gain control techniques both of a basic and more sophisticated form for the purpose of increasing the signal separation between any speaker and the two adjacent speakers. Such techniques may serve to increase the directional effect and thus enhance the audio effect.

In the following specification and claims, the term "transmission path" is used to include both recording and actual transmission (e.g., FM multiplex) inasmuch as the invention is not dependent upon the actual physical or electrical nature of the two separate signals. Reference to locating sound on a circle around a listener means the ability to locate virtual sound sources in front of, behind, or to the sides of a listener but is not intended to limit or define the precise placement of the speakers. The benefits of the invention can best be provided where at least one of the speakers is located behind the listener and the speakers are arranged in fact on the circumference of a circle, with the listener located at the center. Different arrangements of the speakers may vary the directional effect achieved by this invention yet remain within its scope.

The invention is described in detail with reference to the attached drawings, wherein:

FIG. 1 is a block diagram of a system incorporating the principles of the invention;

FIGS. 2A, B, C and D are explanatory diagrams used to illustrate the operation of the system of FIG. 1;

FIG. 3 is a circuit diagram of a preferred embodiment of an encoder used with the invention;

FIG. 4 is a circuit diagram of a preferred embodiment of a decoder used with the invention;

FIG. 5 is a block diagram of a gain control circuit intended for use with the system of FIG. 1;

FIG. 6 is a block diagram of a second gain control circuit intended for use with the system of FIG. 1;

FIGS. 7 and 8 comprise block diagrams of the transmitting and receiving ends of a further gain control circuit;

FIG. 9 is a block diagram of still a further embodiment of a gain control circuit;

FIG. 10 is a block diagram of yet a further gain control embodiment which can provide the desired results apart from the basic system of FIG. 1;

FIG. 11 is a detailed block diagram of the control voltage generator of FIG. 10;

FIG. 12 is a schematic diagram of the gain control circuit of FIG. 10; and

FIGS. 13 and 14 are schematic diagrams of the control voltage generator of FIG. 11; and

FIG. 15 is a diagrammatic illustration of a conventional 45°×45 record groove for explanatory purposes.

The preferred embodiment of the invention is described below for use with four separate speakers arranged on the circumference of a circle around a listening position, so that they are capable of locating a virtual sound source at any point around the listener. For purposes of this description, two speakers will be considered to be arranged directly in front of and behind the listener, with the other two speakers arranged to the left and right. In explaining the operation of the system, the letters F and X are used with numerals to indicate parts of the system corresponding to the front and rear speakers, respectively. The letters L and R are used with numerals to represent parts corresponding to the left and right speakers, respectively. The two transmission paths on which the audio signals are conveyed are represented by the letters A and B, it again being emphasized that such transmission paths may (and frequently will) comprise recording media.

Referring now to FIG. 1, four microphones **12L**, **12F**, **12R** and **12X** are diagrammatically shown as being arranged to receive acoustical energy arriving from any direction around the microphones. For purposes of explanation, each of the microphones **12** may be considered as a directional device capable of receiving sound from any direction within a 90° arc so that four microphones can be arranged to cover a full circle. The receptivity pattern of each of the microphones **12** may be represented by the dashed lines **13F,R,X,L**. In practice, of course, the receptivity patterns of the respective microphones may overlap each other (which is required to locate sound sources between adjacent speakers) or include regions of relatively weak reception, but consideration of this simplified situation is helpful in appreciating the principles by which the invention can effectively transmit four audio signals on only two transmission paths.

In FIG. 1, the four loudspeakers are shown at **14F**, **14R**, **14X** and **14L**. These loudspeakers are shown around a listener's position (represented by a chair **16**). If the electrical outputs of the microphones **12F**, **12R**, **12X** and **12L** were conveyed (by any conventional technique) on four channels directly to the respective speakers **14F**, **14R**, **14X** and **14L**, the speakers would reproduce the directional audio information received by the microphones. In this way, it would be possible to locate a virtual sound source in any quadrant of a circle around the listening position **16**. This ability to locate sounds behind and to the respective sides of a listener is the principal way in which a quadrasonic system differs from conventional binaural stereophonic systems.

As mentioned above, because of practical considerations it is not feasible in all cases to convey the audio signals directly from the microphones **12** to the respective loudspeakers **14**. Because of the great number of binaural stereophonic systems presently in use, it is highly desirable to convey these four distinct audio signals to the four loudspeakers using only the two conventional binaural channels or transmission paths indicated as A and B in FIG. 1.

For this purpose, and pursuant to the invention, an encoder **18** at the recording or transmitting station combines the signals from the four microphones into two separate audio signals which can be conveyed by the separate transmission paths A and B. For example, if a stereophonic record were being made, the encoder **18** would be used to generate the two signals which would be recorded in the standard 45°×45° record groove. In the case of a live broadcast or other direct transmission system, the encoder **18** would be used to generate the A and B signals which would ultimately be broadcast or otherwise transmitted.

The encoder serves to feed each of the four input signals to one or both of the two channels A and B with an amplitude and polarity which determine the relative amplitude of that signal at the various speakers and thus the effective location of the sound source relative to the listener. In this sense, directional information is encoded. In the preferred embodiment the audio signal in transmission path A includes the sum of the output from microphone **12L** and the outputs from microphones **12F** and **12X**, with the amplitude of the F and X signals being attenuated (decreased in amplitude) by a preselected constant, for example, 0.707. The audio signal applied to the transmission path B comprises the output from microphone **12R** plus a signal equal to the difference between the front and rear signals, also preferably attenuated by the same preselected constant, i.e., 0.707.

At the playback or receiving station, a decoder **20** receives the A and B signals and couples these signals in preselected combinations to the loudspeakers **14F**, **14R**, **14X** and **14L**. As indicated in FIG. 1, the signal fed to loudspeaker **14L** is the A signal alone. The sum of the A and B signals is fed to the front speaker **14F** after being attenuated by a factor 0.707. The B signal alone is fed to the right loudspeaker **14R**, and a signal representative of the difference of the audio signals on A and B is fed to the rear speaker **14X** after being attenuated by a factor of 0.707. As explained below, this combination of the four signals (preferably with the indicated gain control) permits location of a virtual sound source at any of the four loudspeakers **14** (or any point on a circle defined by the speakers).

The theory of operation of the system illustrated in FIG. 1 is now explained with reference to the diagrams of FIGS. 2A, B, C and D. In the following, the letters F, R, X and L alone are used to represent the respective voltage outputs from microphone **12F**, R, X and L.

Consider the relative amplitudes of the respective microphone outputs F, R, X and L on the transmission paths A and B to be as indicated in FIG. 1. Assume a sound source directly in front of microphone **12F** so that audio signal F has a relative output of one, while the L, X and R signals are zero. With the gain criteria established in FIG. 1, the signal on transmission path A will be equal to 0.707F. The signal on channel B also will be equal to 0.707F. Accordingly, the signal (0.707 [A+B) coupled to speaker **14F** will be equal to F; the signal coupled to speaker **14L**(A) will be equal to 0.707F; the signal coupled to speaker **14R**(B) will be equal to 0.707F; and the signal (0.707[A−B]) coupled to speaker **14X** will be equal to zero. This means that the audio power outputs of the left and right speakers **14L** and **14R** (proportional to $[0.707F]^2$) will be three decibels down from that produced by the front speaker **14F** (proportional to F^2), while no sound energy at all will be emitted from speaker **12X**. As a result, a listener located at position **16** will consider the apparent location of the sound source to be the speaker **12F**.

FIG. 2B is a similar diagram for the case where the sound source is directly to the right so that only microphone **12R** produces an electrical output. In this case, the signal on transmission path B is equal to R, while there is no signal on transmission path A. Consequently, the signal at speaker **14R** is equal to R, the signals coupled to the adjacent front and rear speakers **12F** and **12X** have a relative amplitude of 0.707R (so that the audio output is down 3 db.) and no sound is produced at speaker **14L**.

When the sound source is in the rear so that only microphone **12X** picks up a signal, as illustrated schematically in FIG. 2C, the signals in both paths A and B will be equal to 0.707X. However, the polarity of the two signals will be opposite. Thus, the A signal may be considered equal to 0.707X and the B signal equal to −0.707X. As a result, the A+B signal fed to speaker **14F** will be equal to zero and the A−B signal at the rear speaker **12X** (attenuated by 0.707) will be equal to X. The signals fed to the adjacent left and right speakers **14L** and **14R** will be 0.707X (i.e., audio output down 3 db.) and of opposite polarity and the virtual sound source will in this instance appear to be the rear speaker **14X**.

By the same logic, where the source is coupled only to the left microphone 12L as illustrated in FIG. 2D, the sound source will be located at the left loudspeaker 14L, with the sound from the adjacent front and rear speakers 14F and 14X being down 3db. (and of opposite polarity), and no signal at the right speaker 14R.

The principles, as explained with reference to FIGS. 1, and 2A, B, C and D, are applicable regardless of where the actual sound source is located. By a similar analysis, it can be shown that a sound source can be located at any point on the circumference of a circle around the listener by effectively creating a sound "null" at the location opposite the desired source location. The fact that the audio output of two speakers is reduced for sounds which should appear to be coming from the speaker between those two speakers also aids in creating this directional effect.

Obviously, the speakers can be positioned in any desired way (for example, two front speakers and two rear speakers) to provide the same or similar effects. Moreover, it is unnecessary that the microphones be directional or that they be arranged to respond to sounds coming from the full circumference of a circle around the microphones. Indeed, it is expected that in many cases, the microphones will be arranged with respect to a musical group so that each will respond to a different instrument or section to create an artificial effect during playback by causing the music to "surround" the listener. These and other techniques are standard with respect to four-channel stereophonic systems and are not a part of this invention. The invention provides a way to recreate virtual sound sources at any location with respect to a listener wherein the audio signals are transferred on only two separate transmission paths. As such, the invention may be used to simulate actual listening conditions or to create artificial effects.

Any number of microphones can be used, including one, to achieve the desired effect. For example, if the output of a single microphone were coupled directly to channel A or channel B, an audio output would appear at loudspeaker 14L or 14R, respectively. If a signal of equal amplitude and polarity from a single microphone were coupled to the A and B channels, the sound source would be located at the loudspeaker 14F. If audio signals of equal amplitude but opposite polarity were applied to the A and B channels, the sound source would be located at the rear speaker 14X. Creating a continuous transition between these conditions would give the effect of a moving sound source.

The principles of the invention may also be used in a three-speaker system. Thus, consider two speakers 14L and 14R located in front (left and right) of the listener and a third speaker 14X directly behind the listener position. The signal on channel A may be equal to L −0.5R +0.5X and the signal on channel B equal to R −0.5L +0.5X. The A and B signals would be coupled directly to the left and right speakers, respectively, with the sum of the A and B signals being coupled to the rear speaker X so that partial cancellation of the R and L signals would occur at the rear speaker thus effectively locating the X channel audio source at the rear speaker.

The foregoing principles may be applied to combine only two signals (e.g., L +0.707X and R −0.707X) on each of the transmission paths A and B so that when the amplitudes of the respective signals are properly combined with suitable consideration for polarity (e.g., A, B and 0.707[A−B]) a desired microphone output will predominate at a corresponding speaker. In this case, the rear signal X will appear in all three speakers with opposite polarity at the left and right speakers 14L and 14R. This will not prevent the desired directional effect since the combined outputs of two speakers fed by out of phase voltages results in a virtual sound source without a definitive location. Hence, in the case of an X signal alone, the source can be located at the rear speaker 14X.

FIG. 3 illustrates the specific details of the encoding circuit. The input signal from microphone 12L is applied through a first resistor 50 to the negative input terminal of a first operational amplifier 52. Similarly, the input signal from microphone 12R is applied through a second resistor 54 to the negative input terminal of a second operational amplifier 56. The input signal from microphone 12F is applied equally to the negative input terminals of both operational amplifiers 52 and 56 through resistors 66 and 68 respectively.

The input signal from the microphone 12X is applied through a resistor 70 to the negative input terminal of operational amplifier 52 and through a resistor 72 to the positive input terminal of operational amplifier 56. The effect of this arrangement is that the input signal from microphone 12X is applied to coded channels A and B with opposite polarity and equal amplitude.

Coded channel A is obtained through a small isolating resistor 58 at the output of operational amplifier 52 and coded channel B is obtained through a similar resistor 60 at the output of operational amplifier 56. Both operational amplifiers 52 and 56 are provided with resistive feedback loops connecting their output terminal and their negative input terminal. These feedback paths include resistors 62 and 64 respectively.

The operational amplifiers used in the encoder are well known devices and are represented in a standard way. Each includes negative and positive input terminals and a feedback resistor as indicated. The signal levels on the negative inputs are independent of the resistors in the other negative inputs and each appears at the output with an amplitude equal to its input level times the ratio of its feedback resistor (e.g. 62) to the input resistor (e.g. 70) (assuming input channel source impedance is low). All negative inputs are combined at the output with the same polarity. The signal on the positive input is subtracted from the signals on the negative inputs at the amplifier output, but is dependent to an extent on the resistors in the negative input channels. Hence, a compensating resistor 86 is provided in the positive input to provide the precise amplitude relationship desired.

The relative amplitudes of the four input signals from microphones 12F, R, L, and X are adjusted by selection of the values of the resistors in the circuit. Thus, the level at which the signal from microphone 12X is applied to coded channel A is determined by the ratio of the resistance of resistor 62 to the resistance of resistor 70. The level at which the signal from microphone 12X is applied to coded channel B is determined by relationships in the resistances of resistors 68, 54, 72, 86 and 64.

In a preferred embodiment of the invention, as explained above, the amplitude level at which the signals from microphones 12F and 12X are applied to coded channels A and B is reduced (relative to the gain of the L and R signals) by a factor of approximately 0.707.

Additionally, it may be desirable to weight the frequency response of the signals from microphone 12X for reasons to be described hereinafter. This may be accomplished by closing switches 74 and 76, thereby bringing into the circuit, in parallel with resistors 70 and 72, the series combinations of capacitor 78 with resistor 82, and capacitor 80 with resistor 84, respectively. The positive input terminal of operational amplifier 56 is connected through an appropriate resistor 86 to ground, and the positive input terminal of operational amplifier 52 is grounded. The series combination of capacitor 80 and resistor 84 and the series combination of capacitor 78 and resistor 82 can be selected to have the effect of emphasizing the high frequency content of the signals in the rear channel.

In a preferred embodiment of the invention, the values of capacitors 74 and 76 and the resistance of resistors 82 and 84 are selected so that the components of the signal in the rear channel having a frequency over approximately 3,000 Hz. are emphasized approximately 6 db. over those components below 3,000 Hz. The preemphasis introduced by the closing of switches 74 and 76 will be reversed at the reproduction end of the system at a point after the signal to be applied to speaker 14X has been derived from the coded signals A and B. This use of preemphasis and deemphasis serves to minimize transmission from the left and right channels into the rear output channel.

FIG. 4 shows a simple decoding circuit for use in conjunction with the system of FIG. 1. The signal in coded channel A is applied directly to speaker 14L and the signal in coded channel B is applied directly to speaker 14R. The signal for speaker 14F is derived by taking the sum of the Signals in coded channels A and B, added in phase and with a selected relative decrease in amplitude. The signal coupled to speaker 14X is derived from the difference of the signals in coded signal channels A and B at equal amplitudes with a selected relative decrease in amplitude.

Specifically, in the circuit of FIG. 4, the signal in coded channel A is applied at the primary winding 88 of a first transformer and the coded signal in coded channel B is applied at the primary winding 92 of a second transformer. The signal for speaker 14L is taken across primary winding 88 while speaker 14R is connected directly across primary winding 92. Both transformers are provided with first and second secondary windings; the first transformer having secondary windings 96 and 98 and the second transformer having secondary windings 100 and 102. The signal for speaker 14F is taken across secondary windings 96 and 100 connected together in series addition as shown. The signal for speaker 14X is taken across secondary windings 98 and 102 which are connected together in series opposition so that it equals the difference between the signals in channels A and B. The turns ratio between the primary and secondary windings is selected so that the voltage at each of the four secondaries will be 0.707 times the voltage at the primary.

If preemphasis is employed in the rear or X channel of the encoder, deemphasis should be employed in the corresponding X channel of the decoder. This may be accomplished by employing a simple circuit functionally similar to that introduced by the closing of switches 74 and 76 in FIG. 2, but having the opposite effect. Any of such circuits, well known in the art, may be used.

The invention as described in FIGS. 1 through 4 is a relatively simple device which, by virtue of the preferred selection of amplitude ratios, provides three-decibel separation between adjacent speakers and complete separation between opposite speakers. It is obvious that numerous circuits other than those illustrated in FIGS. 3 and 4 can be used to provide the functions of the encoder and decoder. Operational amplifiers or transformers or combinations thereof may be used with either or both devices to provide the required functions. It is possible for the decoder to be located preceding or following the power amplifiers. In the latter case, only two power amplifiers are required for all four channels which means that existing stereo systems can be adapted merely by adding the decoder and two speakers at the receiving or playback station.

The three-decibel separation between adjacent speakers provided by the preferred embodiment of the invention provides the desired result, i.e., location of a virtual sound source at any place on a circle around a listener. However, to further enhance the effect, it may be desired to increase the separation between adjacent speakers beyond 3db. This can be accomplished in a number of ways, some of which are of particular utility with the invention and, as such, may be considered to be an improvement over the basic system of FIG. 1. Four such improvements are described below with reference to FIGS. 5, 6, 7, 8 and 9.

FIG. 5 is a block diagram of a simplified form of a gain control circuit which varies the gain of any pair of "diagonal" channels (e.g., left—right) with respect to the other diagonal channels. Hereinafter, by diagonal channels is meant the combined left and right channels or the combined front and rear channels as defined with reference to FIG. 1.

Since, in accordance with the invention, an input from any given microphone will appear in three adjacent speaker channels with maximum gain in the speaker channel corresponding to the microphone, the directional effect can be enhanced by decreasing the gain of the two speakers on either side of the desired speaker. In the system of FIG. 1 these two speakers must always be either the front and rear speakers 14F and 14X, or the left and right speakers 14L and 14R, i.e., the speakers of a diagonal pair of channels. It can also be shown that where the absolute value of the A signal is equal to the absolute value of the B signal (i.e., the waveforms are identical except for possibly opposite polarity), the sound source should either be located at the front or rear speaker. Thus, when this condition exists it is desirable that the gain of the A+B (front) and A−B (rear) channels be maximum relative to the gain of the A (left) and B (right) channels. Similarly, when either the A or B channel signal is zero or the waveforms in A and B are unrelated, the sound source should be located at the left or right speaker, in which case the gain of the A+B and A−B channels should be minimum relative to the gain of the A and B channels.

The foregoing shows that the separation between the output channels of FIG. 1 (and thus the directional characteristics of the audio output) can be improved by simultaneously varying the gain in each pair of two diagonal channels alternatively to controlling the gain in each of the individual channels. It is also necessary that the gain in one pair of diagonal channels be accompanied by an appropriate decrease in the gain of the other diagonal channels. Otherwise, an increase in gain (for example) to enhance the directional characteristic of a signal would result in a volume change of the audio output as a function of direction. However, by simultaneously decreasing the gain in one pair of diagonal channels while increasing the gain in the other diagonal channels it is possible to maintain the total power at the speakers constant, and separation between adjacent speakers can be improved without changing the total volume of the system. These functions are performed by the systems illustrated in FIG. 5.

In FIG. 5, the decoder output is shown at the left. The four signals A, B, 0.707(A+B) and 0.707(A−B) are coupled to respective variable gain amplifiers 110L, R, F, and X, which provide the signals for driving the four speakers as indicated. The A and B channels are also coupled directly through high-pass filters 112A and 112B to absolute value circuits 114A and 114B. The absolute value circuits 114 may comprise full-wave rectifiers the outputs of which are of the same polarity. These signals, representing the absolute value of the A and B signals, are fed to respective logarithmic amplifiers 116A and 116B, which are well-known devices, providing output voltages equal to the logarithm of the applied input voltage. The outputs of these amplifiers 116A and 116B are coupled to the negative and positive inputs, respectively, of an operational amplifier 118 which subtracts the two signals providing an output equal to $\log |A| - \log |B|$ (i.e., $\log |A/B|$). This signal is then fed through another absolute value circuit 120 and an averaging network 122 (an integrating circuit) to a gain control generator 123 which controls the gain of the two pairs of variable gain amplifiers 110L, 110R and 110F, 110X as a function of the output of amplifier 118.

As indicated above, where the absolute values of the A and B signals are equal, the gain of amplifiers 110F and 110X should be maximum and the gain of amplifiers 110L and 110R a minimum. When this condition exists, the output from the operational amplifier 118 will be equal to zero and the gain of amplifiers 110F and 110X should be a maximum (e.g., unity) while the gain of amplifiers 110L and 110R is a minimum (e.g., zero). At the other extreme, where either the B or A signal is equal to zero or the waveforms in A and B are unrelated, the output of the amplifier 118 will be a maximum (theoretically infinite but limited in practice to a definite value, for example, 9 volts). This maximum voltage causes the gain control generator 123 to provide output voltages which maximize the gain of amplifiers 110L and 110R and minimize the gain of amplifiers 110F and 110X.

For conditions between those described above, the gains of the respective pairs of amplifiers 110 will be appropriately controlled by generator 123. Mathematically, it can be shown that the curve of the required gain as a function of log A-log B (the voltage output of 118) approximates a square root curve to yield constant total acoustical power output, with the gains in the respective diagonal channels being equal (for example, to 0.707) when the ratio of A to B (or B to A) is about 2.4. The following equations may be used to determine the gain control voltages from

$$V_{LR}=K\sqrt{\int_{t-T}^{t}\log\left|\frac{A}{B}\right|dt},\quad V_{FX}=K\sqrt{1-\int_{t-T}^{t}\log\left|\frac{A}{B}\right|dt}$$

where T is the time constant of the averaging circuit 122, and K is a constant.

The purpose of the high-pass filters 112A and 112B is to prevent the passage of low-frequency signals which might otherwise appear on the inputs to the variable gain amplifiers 110 and possibly modulate the amplifier inputs. The averaging circuit 122 should respond to changes in the output of the amplifier 118 quickly enough so that the ear does not notice the delay, but not so fast as to pass the actual wave form to the amplifiers 110. As an example, a 20 millisecond charging rate has been found satisfactory for practical purposes.

To stabilize the action of the gain control circuits of the decoder (of FIG. 5) it may be desirable to mix slightly the A and B signals at the encoder output (or the left and right inputs to the encoder) to minimize excursions of the A/B signal ratio. Conversely, to prevent the log of this ratio from going to zero, constant phase differences may be introduced between the respective front (rear) signals applied to the A and B channels at the encoder. This has the effect of restricting gain control action to a relatively narrow range so that the gain control action will not be audible at the speakers. Such mixing can be done in proportions which will accomplish the desired result without materially degrading the audio characteristics. Where extreme channel separation is required, this technique would not be used.

As noted previously, there are many different ways of controlling the gain in the respective channels to provide the desired directional enhancement at the speakers. The embodiment illustrated and described with reference to FIG. 5 is a relatively inexpensive way of providing the desired gain control of the respective pairs of speakers. An equivalent circuit could compare the logs of A and B and the logs of A and −B, take the absolute value of these two signals, average them and use the combined value as a control signal for gain control. This equivalent circuit would accomplish the same results, but would probably require more components and thus be more costly.

Of course, it is not necessary that the gain of diagonal channels be concurrently controlled to improve the directional characteristics of the audio output. Various well-known means may be provided in each of the output channels (A, B, A+B and A−B) to increase or decrease the gain of the signal, depending upon its relationship to a preselected level. For example, FIG. 6 illustrates a circuit which may be placed in each of the four output channels to control the gain of these channels independently. It includes a series connected device 124 having a high negative thermal resistance coefficient and a device 125 having a high positive thermal resistance coefficient across the channel. If the signal from the decoder increases above a value which may be nominally selected to provide a gain of 1, the resultant temperature increase of devices 124 and 125 will cause a simultaneous decrease in resistance of device 124 and increase in resistance of device 125. This impedance change will cause an increase in gain at the channel output. Similarly, reduced input signals will be attenuated by increasingly greater amounts as the signal decreases from the preselected nominal value. In this way, those channels which carry the predominant signal will be provided with an increase in gain relative to the other channels. This will enhance the desired signals while attenuating the adjacent channel signals thus improving the separation between adjacent channels.

By way of example, device 134 may comprise a doped silicon or other semiconductor element and device 125 may be a common incandescent lamp. If desired, a capacitor may be placed in series with the parallel device 125 so that gain control is essentially provided only for high-frequency components. Numerous other well-known devices also may be used in place of the devices 124 and 125 or both of them.

FIGS. 7 and 8 illustrate a further embodiment of a gain control system employing the basic principles of the invention wherein subsonic control tones are impressed upon the A and B channels for the purpose of controlling the gain of the two pairs of diagonal channels from the decoder 20. In describing the operation of FIGS. 7 and 8, the microphones 12, speakers 14, encoder 18 and decoder 20 perform the same function as the corresponding parts of FIG. 1 and therefore are not described further. Ultrasonic, or other, tones can also be used.

To facilitate an understanding of this embodiment it is convenient to refer to power ratios rather than voltage ratios as previously. The power which can be derived from a given signal is directly proportional to the square of the voltage level of that signal.

The signal recording means is illustrated in FIG. 7. The outputs of the front and rear microphones 12F and 12X are sensed and coupled to a power-adding circuit 130, while a similar power-adding circuit 131 sums the power outputs from microphones 12L and 12R. These two power-adding circuits are devices which produce output voltages directly proportional to the total power which can be derived from the applied input voltages. Their output voltages are then summed in an adding circuit 132, the output of which is thus proportional to the total power in the four input channels.

The output of summing circuit 130 is also coupled to a multiplier 132 which doubles this signal, thus providing a voltage output equal to twice the power in the front and rear input channels. The ratio of these two quantities (twice the power in the front-rear channels to total power in all four channels) is then computed by a ratio circuit 134 which, in turn, causes respective A and B modulators 136 and 138 to modulate a 20-cycle (or other subsonic) tone from oscillator 140. Ratio circuit 134 may be any of a number of well-known circuits and, for example, may produce a direct output voltage having an amplitude proportional to the ratio of the applied input voltages and a polarity indicative of whether the measured ratio is greater or less than one.

Modulators 136 and 138 may be adapted to amplitude-modulate the 20-cycle tone from oscillator 140 with respect to a preselected level, depending upon the magnitude and polarity of the applied control voltage from the ratio circuit 134. When the A modulator 136 provides a tone of increased amplitude, the B modulator 138 should be providing a tone of proportionately decreased amplitude. These modulated tones are then added to the A and B outputs of encoder 18 to provide the signals which are to be conveyed by the two-channel transmission path and which, in this particular embodiment, are indicated as A′ and B′.

The receiving end of the system is illustrated in FIG. 8. Two high pass filters 142 and 144 are used to separate the audio control tones from the A and B audio signals on the A′ and B′ channels. These A and B signals from the filters 142 and 144 are coupled to decoder 20 to provide the four output channels described above with respect to FIG. 1.

The control tones from the filters 142 and 144 are coupled to a gain control generator 146 which controls the gain of variable gain amplifiers 148R, F, L, and X to increase the gain of one pair of diagonal channels while appropriately decreasing the gain of the other pair of diagonal channels. From the preceding discussion of FIG. 7, it follows that the amplitudes of the control tones will each be equal to twice the desired power in their corresponding diagonal input channels divided by the total power in the system. Each of these signals varies from a value of zero to two and their sum should always equal one. Accordingly, since the desired power ratios (i.e., the power ratios at the microphones) are directly represented in the control tone signals it is a simple matter for gain control generator 146 to utilize these known ratios to control the gain of amplifiers 148R, L and 148X, F to recreate the same ratios at the outputs of the amplifiers **148**. This will necessarily enhance the desired signals while deemphasizing these signals which are not in their corresponding channels. The total power will also not be varied due to directionality changes. Generator **146** also serves as a normalizer to maintain the total gain of the four channels such that the sum of the power in the respective channels is maintained equal to a constant. This prevents unwanted changes in the amplitude of the control tone from affecting the volume of the outputs from the respective loudspeakers.

The embodiment of FIGS. **7** and **8** is capable of providing substantial separation between any two adjacent channels. However, in its basic form, this embodiment is not capable of recreating the exact power ratio at the output speakers that existed at the input from the microphones. Obviously, this would be desirable since if it could be accomplished it would recreate the power relationship existing at the microphones. As explained with reference to FIG. **9**, a further modification is possible which enables the exact recreation of the power ratio which existed on the input channels.

After encoding and decoding some signal "spreading" occurs at the speaker due to each input appearing at half power in each of the outputs adjacent to the desired output (as well as at full power in the desired output). That is, the outputs will contain spurious signals in addition to wanted signals.

The following consequences follow:

1. The spurious signal power is equal to the wanted signal power.
2. The sum of the powers of the signals in one pair of diagonal channels is equal to that of the other pair of diagonal channels.
3. The spurious powers in the two outputs of one pair of diagonal channels are equal.

One step in bringing output powers to a level corresponding to the respective input powers is to compute the necessary change in the ratio of the total power in the right-left diagonal channels to the power in the front-rear diagonal channels from equality (as exists in the decoder output) to arrive at the proper ratio. The diagonal pairs input power relation is available from the subaudible frequency code signals or approximately from the analyzer circuits described herein.

Simply increasing the power gain for one diagonal pair while decreasing it for the other would adequately adjust the power relation between the two pairs of diagonal channels. It would not assure the proper relation within a diagonal pair (e.g., relation of P_A to P_B).

Note however that if one considers that half of the total power (A, B, A+B, A−B) is spurious power, and if the adjustment of power in diagonal pairs of outputs is such as to halve the total power, then the reduction (in absolute terms) in power in each output of a diagonal pair should be equal. This is true because the origin of spurious power is such that it is fed equally into each output of a diagonal pair. Putting this another way, one should maintain the power difference between A and B (and between (A−B) and (A+B)) while reducing overall power by an amount (one-half) corresponding to the level of spurious signal power.

In order to restore the decoder output powers (P_A, P_B, P_{A+B}, P_{A-B}) to relative values corresponding to encoder inputs the adjusted powers Q_A, Q_B, Q_{A+B}, Q_{A-B} should be as follows:

$$Q_A=\tfrac{1}{2}([1+C_{LR}]P_A-[1-C_{LR}]P_B)$$
$$Q_B=\tfrac{1}{2}([1+C_{LR}]P_B-[1-C_{LR}]P_A)$$
$$Q_{A+B}=\tfrac{1}{2}([1+C_{FR}]P_{A+B}-[1-C_{FR}]P_{A-B})$$
$$Q_{A-B}=\tfrac{1}{2}([1+C_{FR}]P_{A-B}-[1-C_{FR}]P_{A+B})$$

C_{LR} and C_{FX} are the code signal levels representing the part of total input power in the left-right and front-rear pairs of diagonal channels, respectively. $C_{LR}+C_{FX}=1$.

The gain G_A for the A channel amplifier to produce the power Q_A is obviously

$$G_A=(Q_A/P_A)=\tfrac{1}{2}([1+C_{LR}]-[1-C_{LR}](P_B/P_A)$$

Electronic analog computation systems implementing the above formulas may be used to set each channel gain precisely at the value to restore the output power to a relative value corresponding to the corresponding input power. In practice this need only be approximated to achieve excellent results. Various types of such approximations are utilized in the systems herein disclosed.

Obviously, the gains of all channels may be multiplied by any common factor without disturbing the relative power levels.

There are many circuits which can be used to achieve this objective. One embodiment is illustrated in block diagram form in FIG. **9**, which shows only that portion of the circuit which would be used for the right and left speakers **14R** and **14L**. This particular embodiment incorporates portions of the circuit of FIG. **8** and such portions are numbered accordingly. The control circuit for the front and rear channels (A+B and A−B) will be the same as that illustrated in FIG. **9**, and is not illustrated merely for purposes of convenience.

It is recalled that the outputs of the amplifiers **148** comprise two pairs of signals (left-right and front-rear) which are directly proportional in power to the power signals provided by the input microphone pairs **12F**, **12X** and **12R**, **12L**. The left and right channel signals are represented in FIG. **9** by the letters A″ and B″. The inputs to the amplifiers **148R** and **148L** are the A and B signals prior to any modification. In FIG. **9**, signals proportional to the power in and out of the respective amplifiers are summed and a difference signal generated to control the gain of two additional variable gain amplifiers which in turn feed the speakers.

Suitable voltage squaring circuits **150R** and **150L** are shown responsive to the A and B signals for producing direct voltages which are proportional to the power to be derived from the signals. These outputs are fed to a summing circuit **151** which produces a signal proportional to the sum of these power signals, i.e., P_A+P_B.

Similar voltage squaring circuits **152R** and **152L** produce signals proportional to the power to be derived from the outputs of the amplifiers **148R** and **148L** represented as P_A'' and P_B'', and these latter signals are summed in a second summing circuit **154**.

The difference between these two power sums is determined by a difference circuit **156** which produces a signal proportional to this power difference as indicated on the drawing.

As in the preceding circuits, amplifiers **151**, **154** and **156** may comprise conventional operational amplifiers. In practice, the functions of the three illustrated amplifiers may be accomplished by a single operational amplifier in an obvious way.

A further operational amplifier **156** compares the magnitude of the P_A and P_B signals and determines which of the two channels should be increased and which decreased. In each case, the change in gain is dependent upon the output of the amplifier **156** but this output does not indicate which of the two signals should be increased and which decreased. However, it can be shown that when P_A is greater than P_B the signal level and the A channel should be increased while the B signal is decreased, and vice versa. Accordingly, a gain control circuit **158** is responsive to the outputs of amplifiers **156** and **157** and produces two signals which are coupled to two additional variable gain amplifiers **160R** and **160L** in the A and B channels, respectively. The gain control circuit **158** will increase the gain of one of amplifiers **160R** while equivalently decreasing the gain of the other amplifier **160**, depending upon the polarity of the output of amplifier **157** and the amplitude of the output of amplifier **156**. The gain control voltages are such as to cause the respective outputs of amplifiers **160R** and **160L**, which are fed to the speakers **14R** and **14L**, to have the same power ratio as existed at the outputs of the microphones **12R** and **12L**.

As noted previously, the present invention is compatible, and has utility, with all present commercial binaural stereophonic systems. All that is required is that the two encoded channels derived pursuant to the principles of FIG. **1** be recorded or transmitted, as the case may be, on the two available channels. In the case of stereophonic records where the two channels are recorded in respective grooves disposed at 45 degrees with respect to the horizontal, the desired output channels may be derived from suitable resolution of the recorded signals in these channels alone. For example, the left and right channels (i.e., A and B) would each be derived from components of groove modulation (i.e., stylus motion) at 45° from the surface of the record; the front or A+B channel would be derived from components parallel to the record surface (actually equal to 0.707A+0.707B); and the rear or A−B channel would be derived from components perpendicular to the record surface. These components are shown in FIG. 15 as indicated thereon.

Where a stereo record has been recorded in accordance with the invention, the record itself will differ from prior art records in that the A+B signal can contain frequencies which are not present in the A−B signal. In a prior art binaural stereo record, the A+B and A−B channels necessarily will contain the same frequencies. Because of the special form of encoding provided by the invention, where there is complete separation between the front and rear channels, as explained previously, the rear signals will be cancelled in the front channel (A+B) and the front frequencies will be cancelled in the rear channel (A−B).

No effort has been made in describing the basic invention and preferred embodiments of various improvements upon the basic invention to consider every possible method and apparatus for practicing the invention. The following is a description of an early form of the circuit intended to be used as a decoder. In the equations below the term "log" is used to designate logarithmic functions which have a polarity as determined by the polarity of the antilog.

In this embodiment of the invention, the gain associated with each speaker is determined by a combination of a gain control element serially connected in the respective channel, and a gain control voltage generator whose output is coupled to the gain control element. The audio signal in each playback channel (A, B, A+B, or A−B) passes through the respective gain control element. Then, according to the output signal of the control voltage generator, the signal in the gain control element is either enhanced or attenuated. When the output signal of the control voltage generator is at a maximum, the output of the gain control element is at a maximum, and vice versa.

The gain control elements **203** and **204** are thus controlled by an output voltage V_{LR} produced by the control voltage generator **210**. The gain control element **208** is controlled by an output voltage V_F produced by the control voltage generator **212**, and the gain control element **216** is controlled by an output voltage V_X produced by the control voltage generator **218**. If desired, separate control voltage generators may be respectively coupled to gain control elements **203** and **204** rather than just one (generator **210**).

The expressions for each of the control voltages V_{LR}, V_F, and V_X are dictated by design considerations of the various control voltage generators which produce these expressions, as well as by the specific phase, waveform, and level cues present in the original signals A and B which are to activate the respective speakers.

For example, the desired acoustical reproduction requires that the gain associated with the speakers of the left and right channels increases as the ratio of the intensity levels of the signals A and B diverges from unity, or their waveforms become increasingly dissimilar. To achieve this result, the control voltage V_{LR} applied to the gain control elements **203** and **204** may be represented by one of various expressions below. In the following three equations A & B are absolute values.

$$V_{LR}=k\left|(A-B/A+B)\right|;\quad V_{LR}=k\left|\log\ (A-B/A+B)\right|;\quad V_{LR}=k\left|\log\ (a/b)\right|.$$

In accordance with this embodiment, however, the control voltage generator **210** comprises analog circuitry which furnishes a control voltage

$$V_{LR}=\left|\log(A/B)\right|+\left|\log-(A/B)\right|-k\left|\log(\mathrm{env}A/\mathrm{env}B)\right|-V_{SAT},$$

where the term "env" indicates the instantaneous value of the intensity of the envelope voltage, independent of phase and polarity, obtained by full wave rectification and smoothing of the particular signal A or B. This control voltage V_{LR} then increases as the loudness level associated with either the A or B signals becomes stronger with respect to the other, or their waveforms become increasingly dissimilar. A constant voltage V_{SAT} assures a positive value for the control voltage.

The gain in the front channel, on the other hand, is to increase as the ratio of the intensity levels of each of the signals A and B approaches unity (L=0), and as their waveforms become similar and in phase. Accordingly, the expression for the control voltage produced by the control voltage generator **212** may be

$$V_F=V_{SAT}-\left|\log\ (A/B)\right|.$$

The gain associated with the rear channel is to be activated increasingly as the intensity level ratio of the signals approaches unity (L=0), and as their waveforms become similar and of opposite polarity. Accordingly, the control voltage generator **218** may provide an output expression

$$V_X=V_{SAT}-\left|\log\ (-A/B)\right|.$$

The construction of the control voltage generators **210**, **212**, and **218** is based upon analog circuits providing these control voltages V_{LR}, V_F, and V_X, respectively. For example, in order to sense loudness level differences, circuit means for obtaining the log ratio of the intensity levels of the signals A and B are required. These circuits may be incorporated into a single structure as depicted in FIGS. **11**, **12** and **13**. Accordingly, the signals A and B are fed into the log ratio unit **220**, the output of log ratio unit **220** being coupled to the absolute value unit **221**, its output being coupled to the averaging filter unit **222**. The output of the averaging filter is then directed through an inverter **235**, whose output along with the voltage V_{FSAT} is applied to the input of the adder **232**. The output voltage $V_F=V_{SAT}-\left|\log\ (A/B)\right|$ is then applied to one input terminal of the gain control element **208** in the front channel.

Similarly, the control voltage V_X is provided by applying the signal voltage B and the inverted signal voltage A (the output of the inverter **239**) to the log ratio means **223**. The output of the means **223** is coupled to the absolute value means **224**, which output is coupled to the averaging filter **225**. The output signal of the filter **225** is applied to the input of the inverter **236**, the resulting inverted output being applied along with the constant voltage V_{XSTAT} to the adder **233**. The output voltage $V_X=V_{SAT}-\left|\log\ (-A/B)\right|$ is then applied to one input terminal of the gain control element **216** in the rear channel X.

The control voltage generator **210** comprises the absolute value circuit means **226** and **230**, averaging filters **227** and **231**, log ratio means **228**, absolute value means **229**, inverters **237** and **238**, and adder **234**. The output of the filters **222** and **225** are respectively coupled along with the outputs of the inverters **238** and **237** to the input of the adder **234**. The output voltage $V_{LR}=\left|\log\ (A/B)\right|+\left|\log-(A/B)\right|-k\left|\log\ (\mathrm{env}A/\mathrm{env}B)\right|-V_{SAT}$ is then applied to an input terminal of the gain control elements **203** and **204**.

The specific circuitry constituting the various blocks of FIG. **11** is illustrated in FIGS. **13** and **14**. Accordingly, the log ratio means **220** and **223** comprise operational amplifiers AR_2, AR_3, AR_4, AR_5, and AR_{10}, transistors Q_1–Q_6, capacitors C_1–C_3 and C_{14}–C_{17}, and resistors R_9–R_{11}, R_{16}–R_{18}, R_{23}–R_{26}, and R_{35}–R_{38}. The absolute value means **221** and **224** comprise operational amplifiers AR_6, AR_7, AR_{11}, and AR_{12}, diodes D_5–D_{8a}, resistors R_{43}, R_{44}, R_{47}–R_{50}, R_{55}, R_{56}, R_{59}, and R_{60}, and capacitors C_{25} and C_{26}.

The averaging filters **222** and **225** comprise operational amplifiers AR_8 and AR_{13}, capacitors C_{27}–C_{38}, and resistors R_{62}–R_{65}, R_{67}–R_{74}, R_{76}, and R_{77}. The inverters **235** and **236** shown in block form in FIG. **11** are unnecessary when negative voltages are utilized for V_{FSAT} and V_{XSAT} applied through resistors R_{82} and R_{85} to the negative terminals of the amplifiers AR_5 and AR_{14}, as illustrated in FIG. **13**.

The inverter 239 comprises operational amplifier AR_1, and resistors R_2 and R. The adder 232 comprises operational amplifier AR_9, resistors R_{93}, and R_{94}, and capacitors C_{40} and C_{41}. The adder 233 comprises operational amplifier AR_{14}, capacitors C_{42} and C_{43}, and resistors R_{95} and R_{96}.

As illustrated by FIG. 14, the absolute value means 226 comprises operational amplifiers AR_{15} and AR_{16}, resistors R_3, R_5, R_6, R_{12}, and R_{14}, diodes D_1 and D_2, and capacitor C_4. Absolute value means 230 comprises AR_{19} and AR_{20}, resistors R_1, R_7, R_8, R_{13}, and R_{15}, diodes D_3 and D_4, and capacitor C_5.

Averaging filter 227 comprises operational amplifier AR_{17}, resistors R_{19}, R_{27}, R_{33}, R_{39}, R_{41}, R_{28}, R_{21}, and R_{31}, and capacitors C_6, C_8, C_{10}, C_{12}, C_{18}, and C_{19}. Averaging filter 231 comprises operational amplifier AR_{21}, resistors R_{20}, R_{29}, R_{34}, R_{40}, R_{42}, R_{22}, R_{30}, and R_{32}, and capacitors C_7, C_9, C_{11}, C_{13}, C_{20}, and C_{45}.

Log ratio means 228 comprises operational amplifier AR_{18}, AR_{22}, AR_{23}, transistors Q_7–Q_{10}, resistors R_{45} and R_{46}, R_{51}–R_{54}, R_{57}, and R_{58}, and capacitors C_{21}–C_{24}. Absolute value means 229 comprises operational amplifiers AR_{24} and AR_{25}, resistors R_{61}, R_{66}, R_{75}, R_{78}, and R_{79}, and diodes D_9 and D_{10}. Adder 234 comprises operational amplifier AR_{26}, resistors R_{88}–R_{91}, R_{97}, and capacitor C_{44}.

It is to be noted from FIG. 13 that the outputs of the operational amplifiers AR_8 and AR_{13} are actually negative, and are applied to the positive terminals of the adder amplifier AR_9 and AR_{14}. This fact, plus the fact that negative supply voltages (−15 volts) are utilized for V_{SAT}, enables the omission of the inverters 235 and 236. In like manner, the use of a positive supply voltage for F_{SAT} (FIG. 9) applied to the negative input of amplifier AR_{26}, and the application of the positive voltage of the output of the amplifier AR_{25} obviates the need for inverters 237 and 238. Negative outputs at the Z and Y terminals shown in FIG. 13 are then applied to the negative input terminal of amplifier AR_{26} as shown in FIG. 14.

Neither the expressions for V_{LR}, V_F, or V_X nor the specific circuitry illustrated in FIGS. 11, 13 and 14 for providing these functions is critical to the present invention. Accordingly various other circuitry providing the desired result may be utilized to control the gain of the respective speakers. Even if the control voltage generators are employed in conjunction with the gain control elements to provide this gain control function, a different spatial arrangement of the speakers, a different number of channels, or mere economy may dictate entirely different circuitry for the generators.

As a specific feature of the present invention, each of the gain control elements 203, 204, 208, and 216 comprise circuitry as illustrated in FIG. 12. The heart of the circuitry includes a semiconductor diode light source D_{201} and a photoresponsive resistor PR_{201}. The light emitting diode D_{201} may be of gallium arsenide, for example.

As the current through the diode D_{201} increases, an increasing amount of light is thereby emitted which reaches the photoresistor PR_{201}. This results in a corresponding decrease in the resistance of the photoresistor, and current flowing through the photoresistor PR_{201} is thereby enhanced. Any decrease in current through the light emitting diode then similarly results in a decrease of current flowing through the resistor PR_{201}.

Accordingly, current representing the audio signal, for example, the left channel, passes through the photoresistor PR_{201} and thereon through the emitter follower amplifier arrangement including Q_{203} and Q_{204}. The control voltage, V_{LR} for example, is applied to the other input terminal. As V_{LR} increases in response to the various level, waveform, and/or phase cues, as described above, this causes the corresponding enhancement of the signal current through the photoresistor, and consequently the enhancement of the output voltage V_{OUT}. This increased voltage V_{OUT} thus increases the gain associated with the loudspeaker in that channel, in this instance loudspeaker 14L. As the control voltage V_{LR} decreases, the opposite effect occurs. In like manner the gain of the other transducers may be controlled.

Although not illustrated in the drawing, it may be desirable to add a series resistor and capacitor (RC circuit) in parallel with the photoresistor PR_{201} so that both gain and frequency response may be simultaneously controlled. Thus, the frequency response may be narrowed for low levels of current, and turntable rumble and 60 cycle noise may be eliminated at these low levels.

The circuits described with reference to FIGS. 12, 13 and 14 may be constructed using conventional circuit components having desired values. The circuits may be of conventional discrete components or may utilize integrated circuits of monolithic or hybrid construction. As an example, however, of typical component values, the circuits illustrated in these figures were constructed employing components having the following values:

OPERATIONAL AMPLIFIERS		
AR_1, AR_5–AR_{17}, AR_{19}–AR_{21}, AR_{23}–AR_{26}		P85AU(Philbrick-Nexus)
AR_2–AR_4, AR_{13}, AR_{22}		P25AU
TRANSISTORS		
Q_1, Q_3, Q_5, Q_7, Q_9,		½-PL1/P (Philbrick-Nexus)
Q_2, Q_4, Q_6, Q_8, Q_{10}		½-PL1/N
Q_{201}, Q_{203}		2N3707
Q_{202}, Q_{204}		2N3704
DIODES		
D_1–D_{10}		1N914
D_{201}		TIXL09
RESISTORS		
R_1, R_3, R_5–R_8, R_{14}, R_{15}, R_{43} R_{44}, R_{47}–R_{50}, R_{59}–R_{61}, R_{66} R_{78}, R_{79}		200k
R_9–R_{11}, R_{23}–R_{26}, R_{45}, R_{46} R_{52}, R_{54}		10k
R_{12}, R_{13}, R_{55}, R_{56}, R_{75}, R_{82} R_{83}, R_{85}, R_{86}, R_{88}–R_{91}, R_{93}–R_{97} R_{202}		100k
R_2, R_4		22k
R_{16}–R_{18}, R_{51}, R_{53}		4.7k
R_{19}, R_{20}, R_{33}, R_{34}, R_{62}, R_{63} R_{73}, R_{74}		56k
R_{21}, R_{22}, R_{27}, R_{29}, R_{31}, R_{32} R_{64}, R_{65}, R_{67}, R_{69}, R_{71}, R_{72}		220k
R_{28}, R_{30}, R_{68}, R_{70}		860k
R_{35}–R_{38}, R_{57}, R_{58}, R_{205}, R_{206}		180k
R_{39}–R_{42}, R_{76}, R_{77}, R_{80}, R_{81}		156k
R_{203}		470k
R_{204}, R_{207}	180	
R_{209}		330
R_{208}		1.8k
R_{84}, R_{87}, R_{92}		25k linear pot.
R_{201}		10k linear pot.
CAPACITORS		
C_1–C_3, C_{21}, C_{22}		6.8pf.
C_4, C_5, C_{14}–C_{17}, C_{23}–C_{26} C_{39}–C_{44}		100pf.
C_6, C_7, C_{12}, C_{13}, C_{27}, C_{28}, C_{33}, c_{34}		0.03mf.
C_8–C_{11}, C_{29}–C_{32}		0.12mf.
C_{18}, C_{20}, C_{35}, C_{37}		0.012mf.
C_{19}, C_{36}, C_{38}		0.05mf.
C_{201}		10mf.
C_{202}		120mf.
PHOTORESISTOR		
PR_{201}		CL3AL (Clairex)

In the gain control embodiment of the invention shown in FIG. 5, where the channel gain is controlled as a function of the instantaneous value of the A and B signals, an error can occur if different input signals of approximately equal amplitudes are coupled to the encoder by the front and rear microphones 12F and 12X. In this case, assuming no other input, the A signal will differ from the B signal which, as explained previously, will cause the gain in the left and right output channels to be maximized while the gain in the front and rear channels is minimized. Analogously, where the logs of the envelopes are compared (as in the embodiment of FIGS. 10–14), the appearance of different signals of approximately equal amplitudes on the left and right microphone inputs will cause an error in the output by minimizing the gain of the left and right speakers while maximizing the gain of the front and rear speakers. From a practical viewpoint, this condition in most cases does not affect the utility of the invention particularly where there is suitable control of the audio material being recorded (or transmitted). However, if this error should prove to be a problem, it is conceivable that two systems such as shown in FIGS. 5 and 11 could be combined by deriving both control signals simultaneously and averaging these control signals to provide the final gain control. One way to effectively "combine" the two systems would be to include an averaging circuit (such as 221 and 222) in series after each of the absolute value circuits 116A, 116B of FIG. 2, with an adjustable resistance in shunt around the added components. When the averaging circuit is shorted out (the resistance is set to zero), the instantaneous ratio of A/B is sensed as in FIG. 5. At infinite resistance the circuit senses the ratio of envelope A to envelope B (FIG. 11). At intermediate resistance, combinations of the two are sensed.

The signal source of the present invention having three or more signals encoded in two is also compatible with and will playback through conventional monaural and two channel stereophonic speaker arrangements. Various modifications of the disclosed embodiments may also be employed while still utilizing the principles of the present invention. For example, while the reproduction or playback apparatus has particular applicability when a two channel signal source is used, additional playback channels may also be coupled to a three or more channel signal source, the speakers associated with each channel being activated by cues present in one or all of the three or more source signals.

Additional channels may be coupled to the stereo signal source, their respective gains being determined by other defined relationships in the signals A and B. Thus, a fifth or sixth channel may be added, for example, the gain associated with the left and right channels increasing responsive to dissimilar waveforms and unequal loudness levels of signals A and B; the gain associated with the fifth and sixth channels increasing with similar waveforms and unequal levels. The loudspeakers in these channels may be added at any point around the listener, thereby enabling the location of the virtual sound source at additional points around a 360° circle, or even in another plane perpendicular to that of the four channels.

For purpose of convenience reference has been made herein to polarity differences which may be considered as phase differences of 180°. This is preferred because of the relative ease of "inverting" a signal to obtain a reversal of polarity. The principles of the invention may also be employed with other phase relationships by using known devices which are capable of providing substantially constant phase shifts for all frequencies in the audio band of interest. Instead of polarity or phase relationships, it may be possible to utilize time delays for encoding the respective channels with complementary time delays in the decoder.

Various other modifications may be made to the above described embodiments by one ordinarily skilled in the art without departing from the spirit and scope of the invention.

What is claimed is:

1. For use with a stereophonic sound system wherein a receiver means couples audio data to at least four separate speakers and wherein audio signals are transmitted to said receiver means on only two channels A and B, a recording or transmitting network, comprising

- means for coupling a first signal desired to be coupled to a first speaker to at least one of said channels with a preselected amplitude and polarity,
- means for coupling a second signal desired to be coupled to a second speaker to at least the other of said channels with a preselected amplitude and polarity,
- means for coupling a third signal desired to be coupled to a third speaker to both A and B channels with the same polarity in both channels and preselected amplitudes in said channels, and
- means for coupling a fourth signal desired to be coupled to a fourth speaker to both A and B channels with opposite polarities in the two channels and preselected amplitudes in said channels.

2. A recording or transmitting network according to claim 1, further including means for reducing the amplitude of said third and fourth signals relative to said first and second signals, respectively, in said A and B channels.

3. A recording or transmitting network according to claim 2, wherein said amplitude reducing means reduces the amplitude of said third and fourth signals by substantially the same amount.

4. A recording or transmitting network according to claim 3, wherein said amplitude reducing means reduces the amplitude of said third and fourth signals by a factor of approximately 0.707.

5. A recording or transmitting network according to claim 1, including means for superimposing a control tone on at least one of said channels, said control signal being dependent upon the relative signal strengths of said first, second, third and fourth signals.

6. A recording or transmitting network according to claim 5, including means for superimposing a control tone on each of said A and B channels, each of said control tones being dependent upon the ratio of the total signal power in two of the channels to the total power in all four channels.

7. For use with a stereophonic sound system wherein a receiver means couples audio data to at least four separate speakers and wherein audio signals are transmitted to said receiver means on two channels A and B, a recording or transmitting network, comprising:

- at least four acoustical-electrical transducer means for producing, respectively, at least four audio signals to be transmitted,
- means for coupling a first signal desired to be coupled to a first speaker to at least one of said channels with a preselected amplitude and polarity,
- means for coupling a second signal desired to be coupled to a second speaker to at least the other of said channels with a preselected amplitude and polarity,
- means for coupling a third signal desired to be coupled to a third speaker to both A and B channels with the same polarity in both channels and preselected amplitudes in said channels, and
- means for coupling a fourth signal desired to be coupled to a fourth speaker to both A and B channels with opposite polarities in the two channels and preselected amplitudes in said channels.

8. For use in a sound reproducing system capable of locating virtual sound sources at essentially any point on a circle surrounding a listener position, wherein audio information corresponding to four separate directional inputs is transmitted on a two-channel transmission path and wherein there are provided at least four loudspeakers adapted to be arranged around a listener, the combination comprising:

- means for coupling the signal on at least one of said channels to a first of said speakers,
- means for coupling the signals on said first and second channels to a second speaker with the same polarity and a preselected amplitude relationship,
- means for coupling the signal on at least the second of said channels to a third speaker, and
- means for coupling the signals on said two channels to a fourth speaker with opposite polarities and a preselected amplitude relationship.

9. The combination according to claim 8, wherein said first and third speakers define a diagonal intersecting a diagonal defined by said second and fourth speakers.

10. The combination according to claim 8, wherein the signals fed to said second and fourth speakers are reduced in amplitude relative to the signals coupled to the first and third speakers.

11. For use in a sound reproducing system capable of locating virtual sound sources at essentially any point on a circle surrounding a listener position, wherein audio information corresponding to four separate directional inputs is transmitted on a two-channel transmission path and wherein there are provided at least four loudspeakers adapted to be arranged around a listener, the combination comprising:

means for coupling a signal on at least one of said channels to a first of said speakers,

means for coupling the signal on at least the second of said channels to a third speaker,

means for coupling the signals on said first and second channels to a second speaker with the same polarity and with their amplitudes reduced by a factor of approximately 0.707 relative to the amplitude of the signals coupled to said first and third speakers, and

means for coupling the signals on said two channels to a fourth speaker with opposite polarities and with their amplitudes reduced by a factor of approximately 0.707 relative to the amplitude of the signals coupled to said first and third speakers.

12. The combination according to claim **10**, wherein each of said coupling means includes variable gain amplifier means, said combination further including gain control means for varying the gain of said variable gain amplifier means so as to increase separation between adjacent speakers.

13. The combination according to claim **12**, wherein the gains of the variable gain amplifier means associated with the first and third speakers are maintained equal and the gains of the variable gain amplifier means associated with said second and fourth speakers are maintained equal.

14. The combination according to claim **12**, wherein said gain control means includes means for computing the instantaneous value of the ratio of the signal amplitude in one of said channels to the signal amplitude in the other of said channels.

15. The combination according to claim **12**, wherein said gain control means comprises means for comparing the envelopes of the signals in said channels.

16. The combination according to claim **13**, wherein the gain of said variable gain amplifiers associated with said first and third speakers is minimum and the gain associated with said second and fourth speakers is maximum when said ratio is equal to one, and wherein the gain of the variable gain amplifiers associated with said first and third speakers is a maximum and the gain of the amplifiers associated with said second and fourth speakers a minimum when there is no signal in either one of said channels.

17. The combination according to claim **16**, wherein said gain control means varies the gain of said variable gain amplifiers so as to maintain the total power coupled to said speakers at a substantially constant level.

18. The combination according to claim **12**, wherein said gain control means includes means responsive to a modulated tone on at least one of said channels.

19. The combination according to claim **12**, wherein said gain control means includes means responsive to modulated control tones on each of said channels.

20. The combination according to claim **12**, including further variable gain amplifier means and gain control means, said further gain control means adapted to control the gain of said further variable gain amplifiers and responsive to voltages dependent upon (a) the power differential between the inputs and outputs of the first-named variable gain amplifiers associated with the first and third speakers and (b) the power differential between the inputs and outputs of the first named variable gain amplifiers associated with the second and fourth speakers.

21. A stereophonic sound system wherein audio signals are transmitted on only two channels A and B, in combination, an encoder and a decoder,

said encoder comprising

means for coupling a first signal desired to be coupled to a first speaker to at least the A channel,

means for coupling a second signal desired to be coupled to a second speaker to at least the B channel,

means for coupling a third signal desired to be coupled to a third speaker to both A and B channels with the same polarity in both channels, and

means for coupling a fourth signal desired to be coupled to a fourth speaker to both A and B channels with opposite polarity in both channels,

said decoder comprising

means for coupling the signal on one of said channels to a first of said speakers,

means for coupling the signals on said first and second channels to a second speaker with the same polarity,

means for coupling the signal on the second of said channels to a third speaker, and

means for coupling the signals on said two channels to a fourth speaker with opposite polarities.

22. A method of recording or transmitting at least four directional audio information signals on two channels comprising

forming at least two composite signals each of which includes at least three of said audio information signals in preselected amplitude and phase relationships, at least one of said audio information signals having a different phase in each of said composite signals, said amplitude and phase relationships being such that four combinations of said two composite signals with preselected phase and amplitude relationships will yield four separate signals in which respective ones of said four directional signals are predominant.

23. A method of recording or transmitting at least four directional audio information signals according to claim **22**, wherein at least one of said audio information signals has the same relative phase in each of said composite signals and at least one of said audio information signals has opposite phase in each of said composite signals.

24. A method of recording or transmitting at least four directional audio information signals according to claim **22**, wherein at least some of said phase relationships are equal to a phase difference of 180°.

25. A method of substantially reproducing at least four individual directional audio information signals which are contained in two information channels, wherein the signal in each of said channels comprises a combination of at least three of said directional signals with preselected phase and amplitude relationships, comprising

deriving said four output signals from the signals in said two channels, at least one pair of said output signals being derived by combining the signals in said channels with preselected phase and amplitude relationships wherein different phase relationships are used to derive each signal of said pair of signals, the remaining output signals, if any, being derived from the signals in one of said channels, whereby a different desired one of said four directional signals is predominant in each of said four output signals.

26. A method of substantially reproducing at least four individual directional audio information signals according to claim **25**, further including the step of controlling the gain of at least some of said four output signals depending on a preselected comparison of the signals in said two channels.

27. A method of substantially reproducing at least four individual directional audio information signals according to claim **25**, wherein at least one of said output signals is derived by combining the signals in said two channels with a zero phase shift and at least one other of said four output signals is derived by combining said signals in said two channels with one of said signals in said two channels being shifted in phase by 180°.

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