United States Patent [19]

Bauer

[11] **3,821,471**

[45] **June 28, 1974**

[54] **APPARATUS FOR REPRODUCING QUADRAPHONIC SOUND**

[75] Inventor: **Benjamin B. Bauer,** Stamford, Conn.

[73] Assignee: **CBS Inc.,** New York, N.Y.

[22] Filed: **Mar. 15, 1971**

[21] Appl. No.: **124,135**

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

[51] **Int. Cl.** **H04n 5/00**

[58] **Field of Search** . 179/15 BT, 16, 6 P, 100.1 TP, 179/100.4 ST

[56] **References Cited**

UNITED STATES PATENTS

3,632,886	1/1972	Scheiber	179/15 BT
3,708,631	1/1973	Bauer	179/15 BT
3,746,792	7/1973	Scheiber	179/1 GQ

OTHER PUBLICATIONS

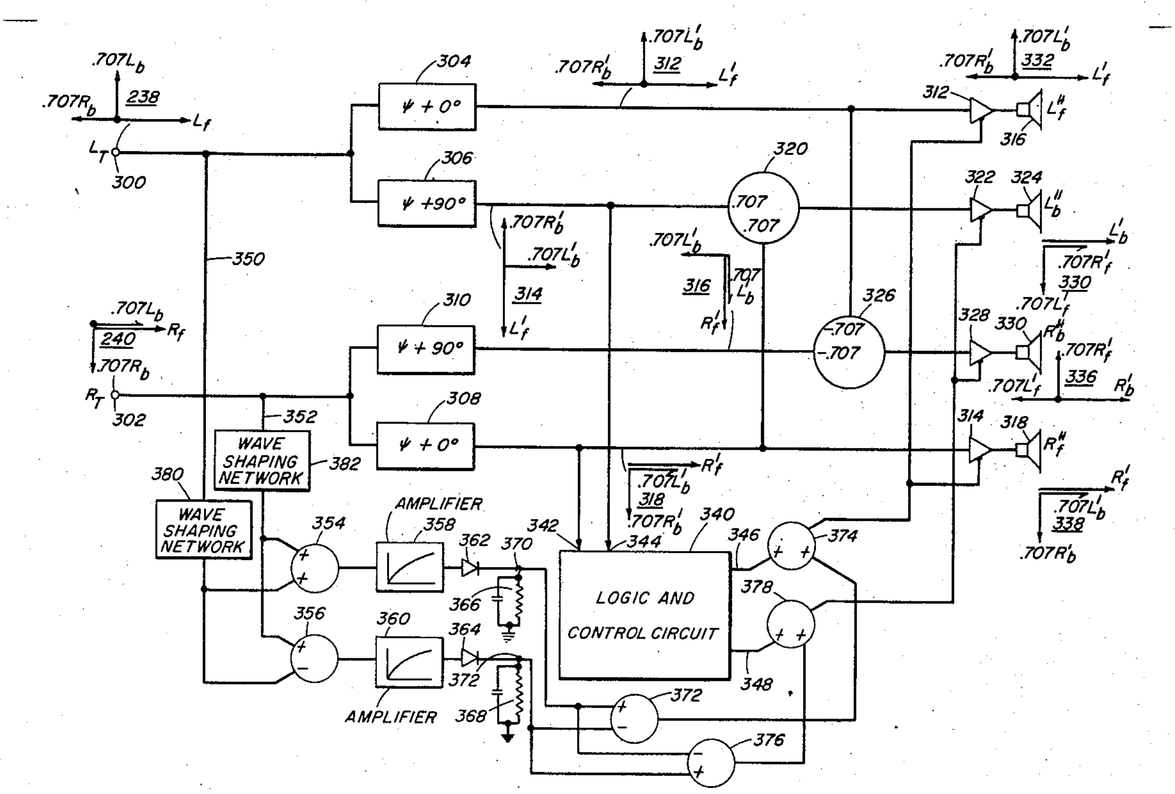
"Four Channels and Compatibility" by Scheiber, Audio Engineering Society Preprint, Oct. 12–15, 1970.

Primary Examiner—Kathleen H. Claffy
Assistant Examiner—Thomas D'Amico
Attorney, Agent, or Firm—Spencer E. Olson

[57] **ABSTRACT**

Apparatus for decoding four separate channels of information transduced from a medium having only two channels and presenting them on four loudspeakers to give the listener the illusion of sound coming from a corresponding number of separate sources. The realism is enhanced by a decoding system which accepts the two outputs from the medium, which may be a disc record, decodes them into four independent channels each carrying as predominant components the information contained in the originally recorded sound signals, and which includes a logic and control circuit and a "front-back" logic for controlling the gains of amplifiers associated with the four loudspeakers. The logic and control circuit utilizes a wave-matching technique for improving the separation of the four independent channels, and the "front-back" logic, which may be used independently of the "wave-matching" logic, improves the separation between generally "front" and generally "back" signals, thereby to enhance the realism of the quadraphonic reproduction.

12 Claims, 14 Drawing Figures



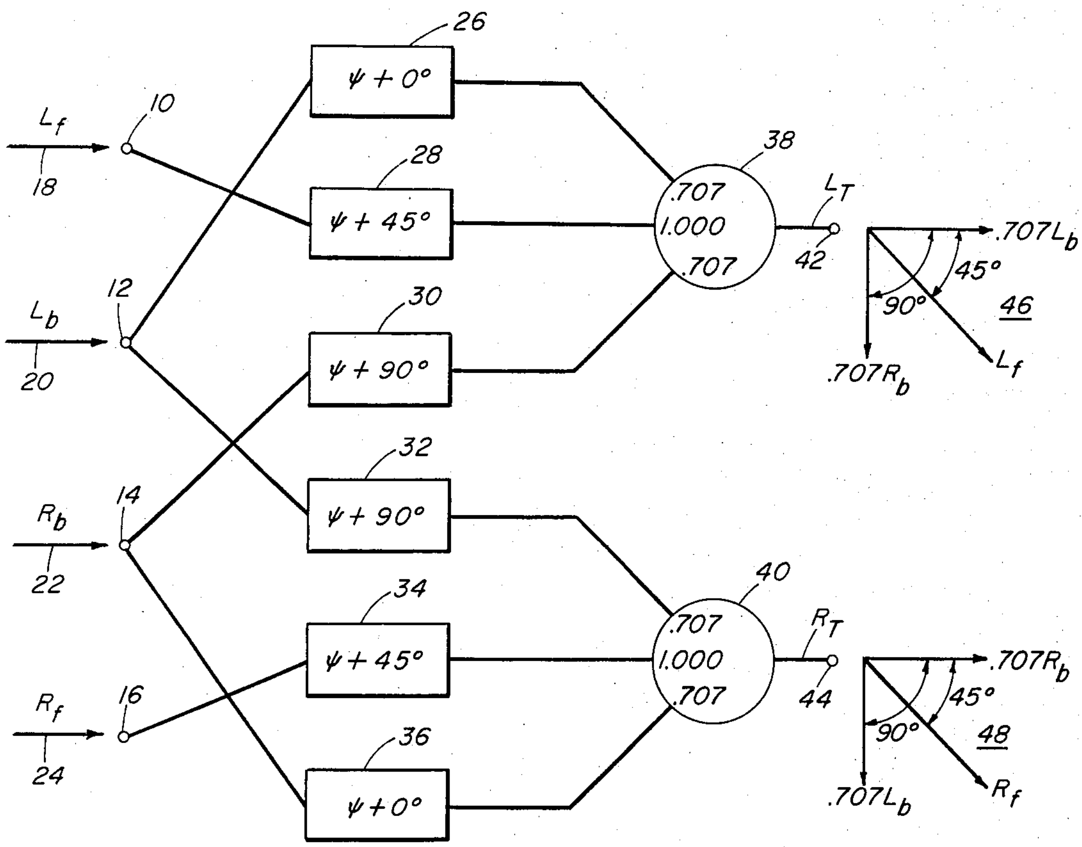


FIG. 1

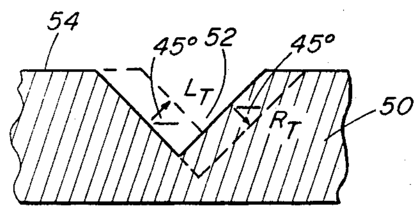


FIG. 2

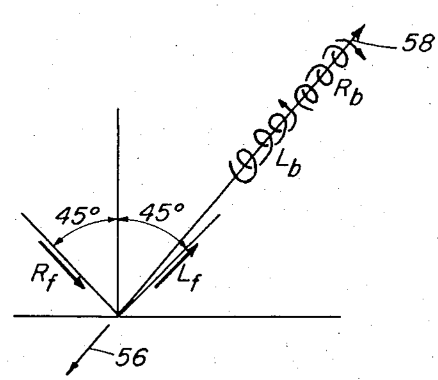


FIG. 3

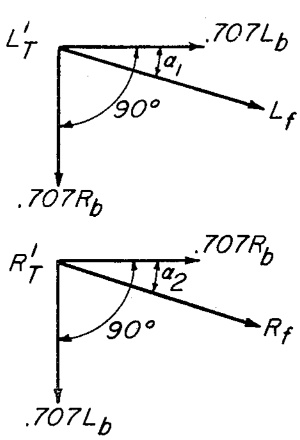


FIG. 4

INVENTOR.
BENJAMIN B. BAUER
BY
Spencer E. Olson
ATTORNEY

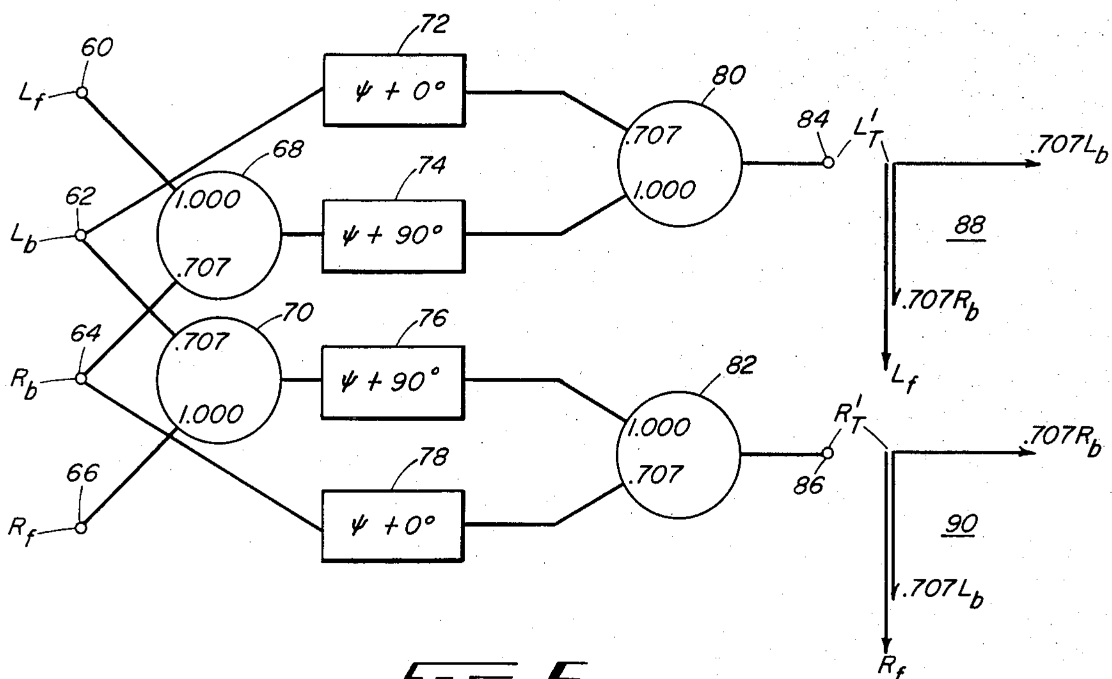


FIG. 5

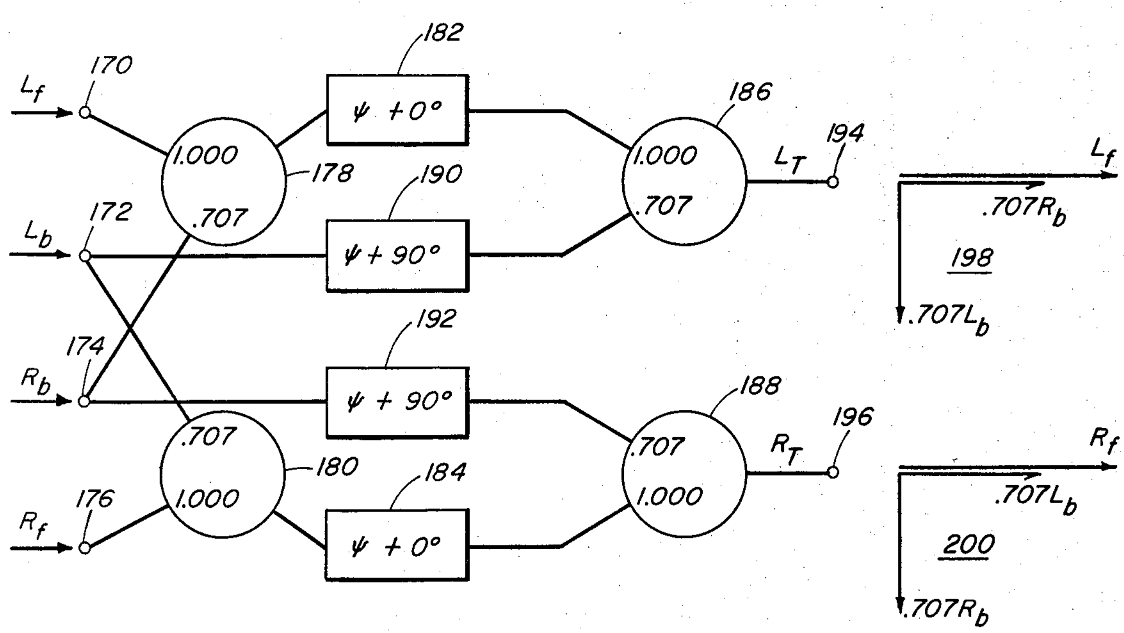


FIG. 7

INVENTOR.
BENJAMIN B. BAUER
BY Spencer E. Olson
ATTORNEY

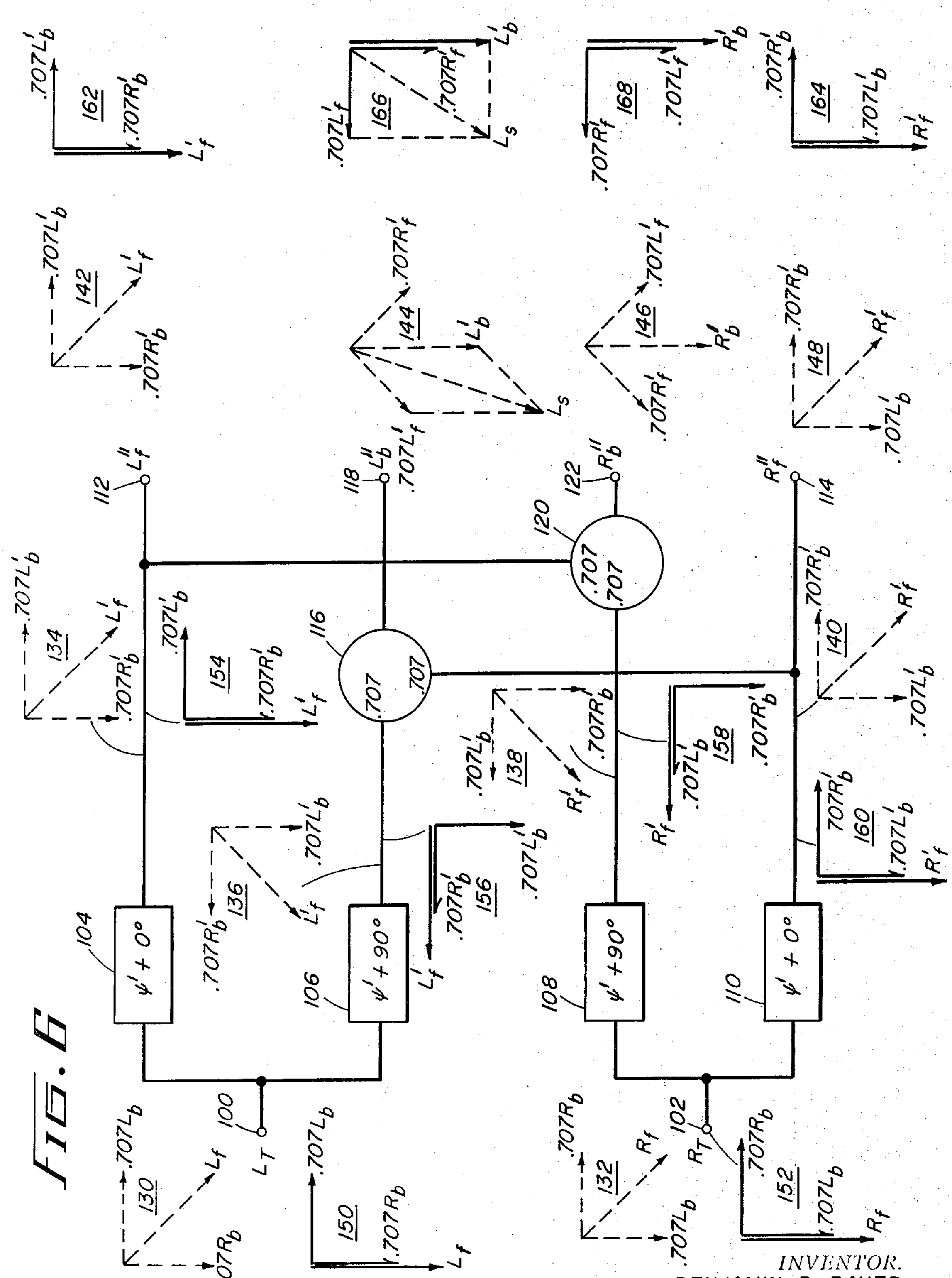


FIG. 6

INVENTOR.
BENJAMIN B. BAUER

BY Spencer E. Olson

ATTORNEY

PATENTED JUN 28 1974

3,821,471

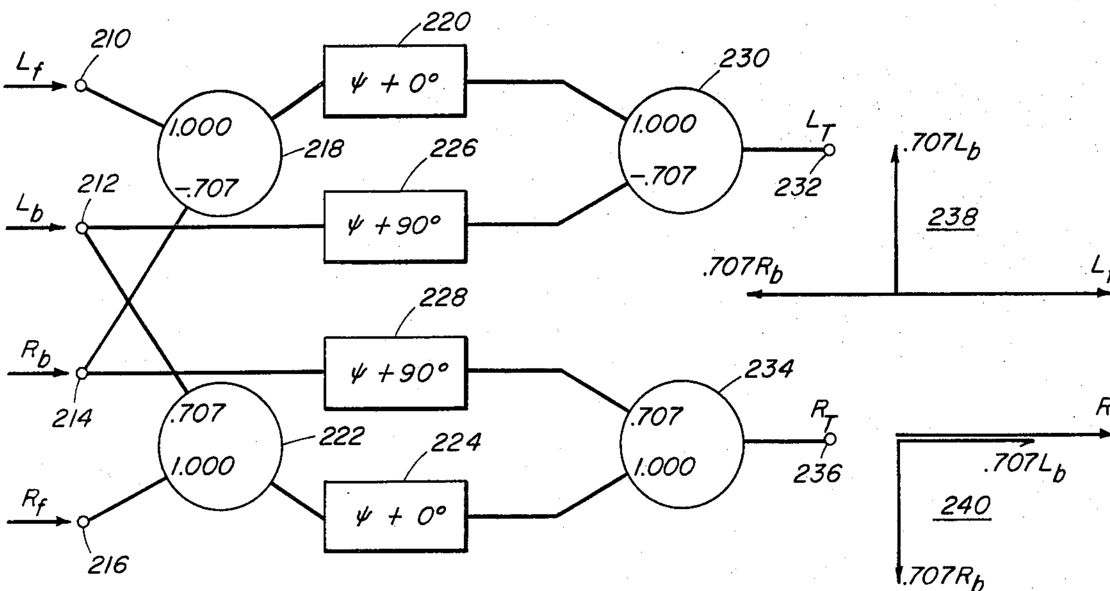


FIG. 8

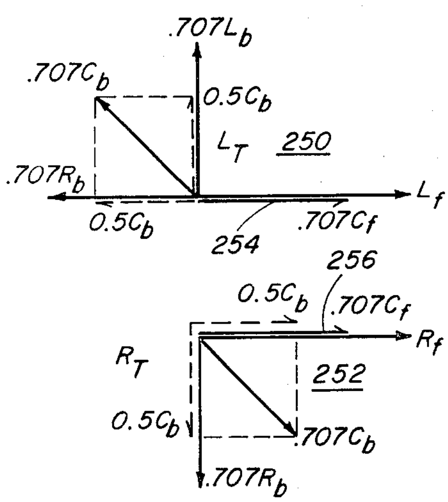


FIG. 10

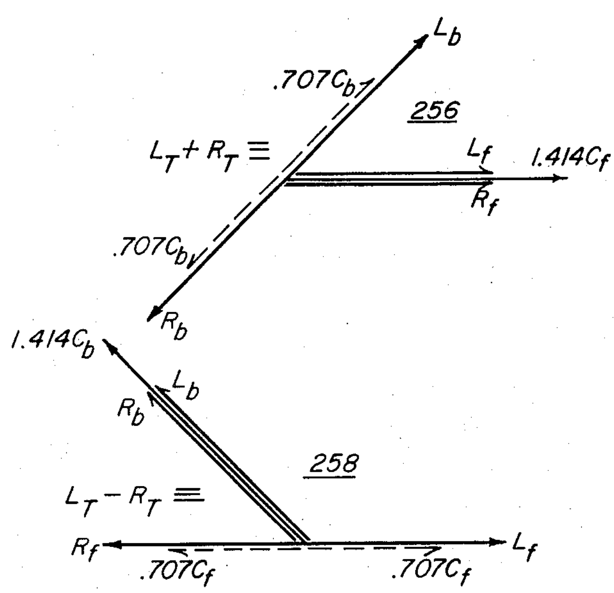


FIG. 11

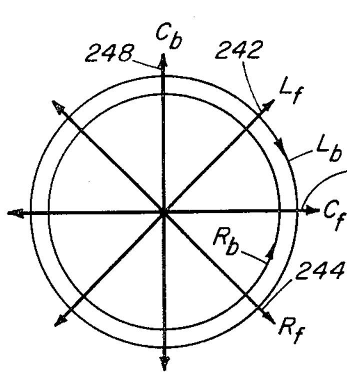


FIG. 9

INVENTOR.
BENJAMIN B. BAUER

BY Spencer E. Olson
ATTORNEY

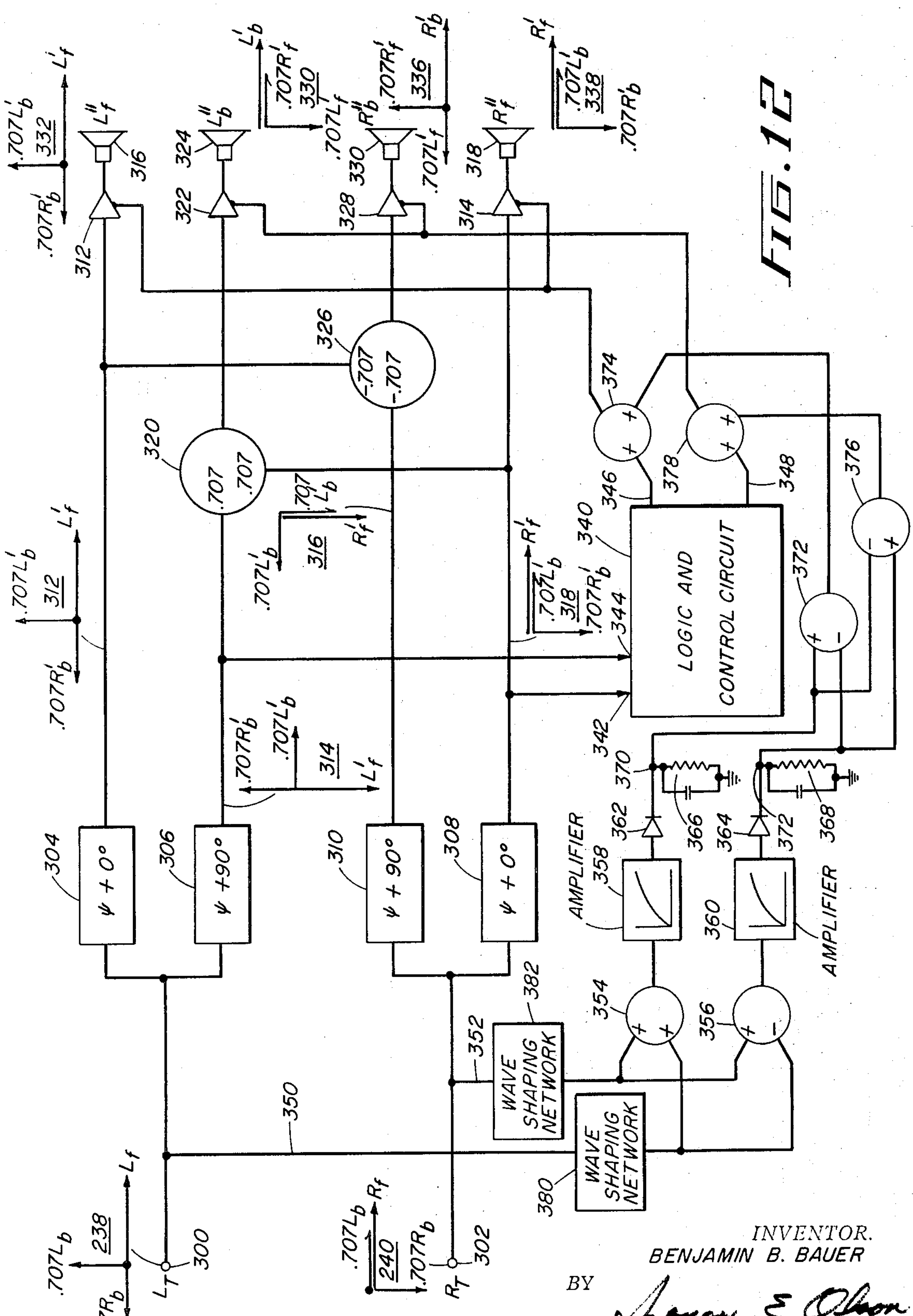


FIG. 12

INVENTOR.
BENJAMIN B. BAUER

BY Spencer E. Olson

ATTORNEY

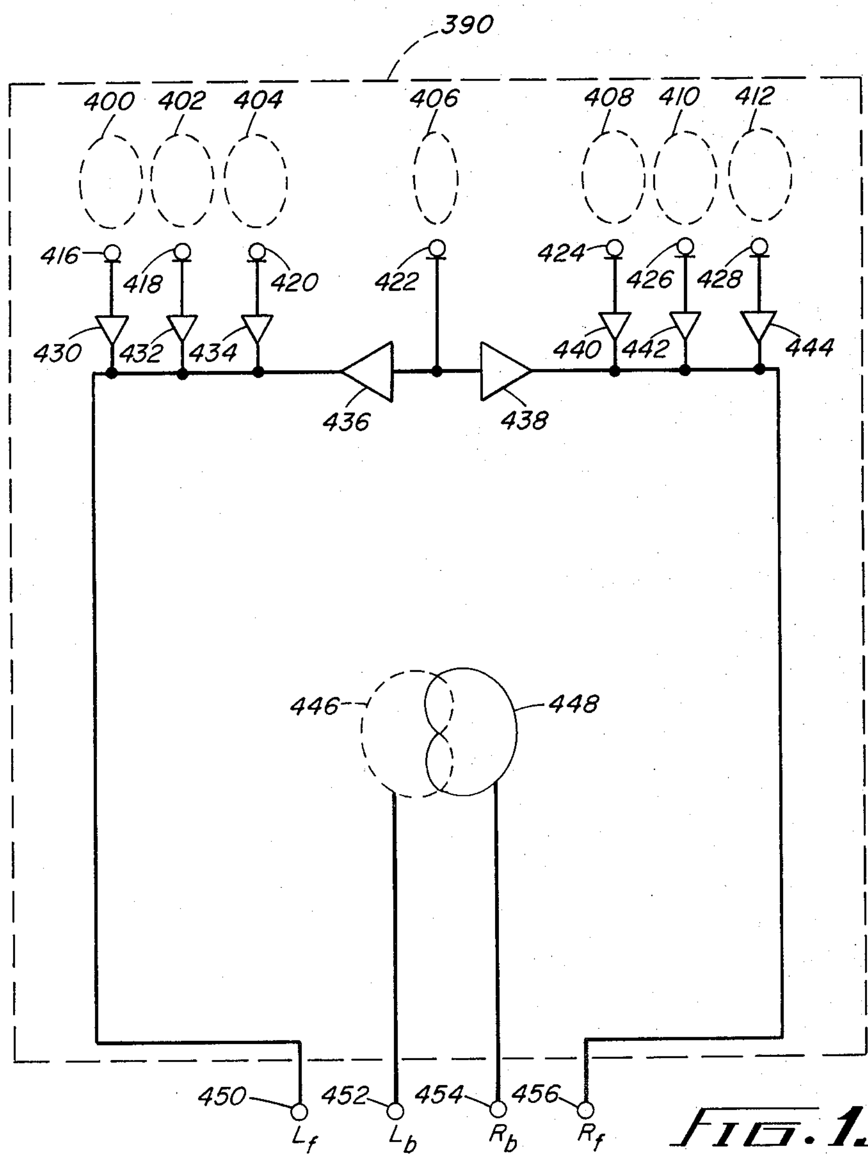


FIG. 13

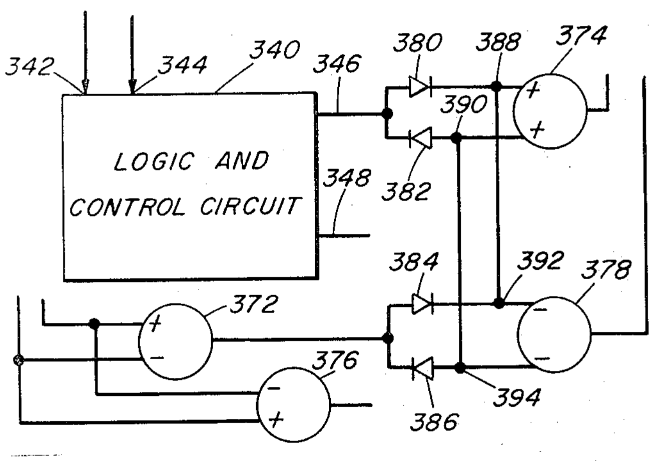


FIG. 12A

INVENTOR.
BENJAMIN B. BAUER

BY Spencer E. Olson

ATTORNEY

APPARATUS FOR REPRODUCING QUADRAPHONIC SOUND

CROSS-REFERENCE TO OTHER APPLICATIONS

This invention is related to the subject matter of the following co-pending applications, all of which are assigned to the assignee of the present invention: Ser. No. 40,510, filed May 26, 1970, now abandoned in favor of continuation-in-part application Ser. No. 164,675 filed July 21, 1971; Ser. No. 44,196, filed June 8, 1970, now Pat. No. 3,708,631; Ser. No. 44,224, filed June 8, 1970, now abandoned in favor of continuation application Ser. No. 251,544 filed Apr. 21, 1972, now also abandoned and a portion of the subject matter thereof incorporated in continuation-in-part application Ser. No. 328,874 filed Feb. 10, 1973; Ser. No. 81,858, filed Oct. 19, 1970, now abandoned in favor of continuation application Ser. No. 251,636 filed May 8, 1972; Ser. No. 112,168 filed Feb. 3, 1971; and Ser. No. 118,271, filed Feb. 24, 1971.

BACKGROUND OF THE INVENTION

This invention relates to apparatus for recording and reproducing four separate channels of information on a medium having only two independent tracks, and more particularly to improved methods and apparatus for reproducing such information and presenting it on four loudspeakers to give the listener the illusion of sound coming from a corresponding number of separate sources. More particularly, this invention is concerned with an improved decoder for four channel sound recorded on a two-track medium in accordance with the method described in the aforementioned co-pending application Ser. No. 328,874.

The recording method disclosed in application Ser. No. 328,874 is based on an encoding function which results from passing four signals associated with the four channels of sound (which, for convenience, are identified as L_f, L_b, R_f and R_b, for "left-front," "left-back," "right-front" and "right-back," respectively) through six all-pass phase-shifting networks and thereafter combining them in appropriate proportions to produce two composite signals, L_T and R_T. The encoder of the aforementioned application, illustrated in FIG. **1**, has four input terminals **10, 12, 14** and **16,** to which are respectively applied signals four channels of a quadruphonic program, L_f, L_b, R_b and R_f, which are represented by phasors **18, 20, 22** and **24**, respectively. It will be remembered from elementary considerations that a phasor generally represents a sinusoidal wave of a particular frequency, with the length of the phasor arrow portraying its amplitude and its direction representing the phase angle. In the interest of clarity of description to follow, the four phasors are shown as all being of equal amplitude and direction.

In the encoding process the positions of the individual phasors are modified by a plurality of all-pass phase-shifting networks **26, 28, 30, 32, 34** and **36** which have the capability of transmitting all of the frequencies within the range of interest (typically, the full audio range from 20 to 20,000Hz.) without change in amplitude but with a change of relative phase, which includes a reference phase-shift ψ, which is a function of frequency, and a differential phase-shift which may be any desired angle, typically 45° or 90°. The all-pass networks **26** and **36** (hereinafter called ψ-networks) each introduce the reference phase-shift ψ; networks **28** and **34** provide a phase-shift of $\psi + 45°$, and networks **30** and **32** provide a phase-shift of $\psi + 90°$. Networks having these properties being well known in the electrical engineering art, they will not here be described in detail. As indicated, the L_f signal is applied to network **28,** the L_b signal is applied to both of networks **26** and **32,** signal R_b is applied to both of networks **30** and **36,** and signal R_f is applied to network **34.**

The outputs of the six ψ-networks are recombined into two groups by a pair of adding junctions **38** and **40.** The junction **38** is operative to add a unity measure of the signal L_f after passage through network **28,** 0.707 of the signal L_b after passage through network **26,** and 0.707 of the signal R_b after it passes through network **30** to produce at output terminal **42** a composite signal, L_T. Similarly, the adding junction **40** is operative to add 0.707 of signal L_b, after passage through network **32,** a unity measure of signal R_f after passage through network **34,** and 0.707 of signal R_b after passage through network **36,** to produce at output terminal **44** a second composite signal, R_T. The signals L_T and R_T, representing the total left and right channel signals, respectively, may be broadcast through an FM-multiplex transmitter and received with an FM-stereo receiver for subsequent decoding, or they may be recorded on a two-track medium, such as a stereophonic phonograph record or two-track tape, for subsequent replay and decoding, all as explained in co-pending application Ser. No. 328,874.

It will be observed from examination of phasor groups **46** and **48** portraying composite signals L_T and R_T, respectively, that the encoder positions the phasors L_f, L_b, R_b and R_f at specific relative phase angles and amplitude relationships with respect to each other, which, as explained in the aforementioned co-pending application, results in an excellent conventional stereophonic record, and at the same time admits of a subsequent decoding operation to produce with a high degree of realism four separate sound channels.

As additional background for an understanding of the present invention, FIG. **2** illustrates how the signals encoded with the system of FIG. **1** are recorded on an ordinary stereophonic record **50** having a groove **52** in the surface **54** thereof cut by a recording stylus for subsequent replay in a well-known manner. By well-known convention, the left channel of a stereophonic record causes modulation of the inner groove wall, that is, the groove wall nearest to the center of the record, with the motion at 45° to the surface as portrayed by the arrow L_T, and the right channel causes modulation of the other groove wall, that is, the wall furthest from the center, causing it to move at 45° to the surface, as indicated by the arrow R_T.

Referring now to FIG. **3,** in which these modulations are portrayed as motions of the stylus (either recording or playing back the groove) the arrow **56** indicates the direction of groove motion as the record turns, and the oppositely directed arrow **58** consequently represents the relative direction of stylus motion with respect to the groove. It is seen that the signal L_f causes the stylus to move at +45° and the signal R_f causes stylus motion at −45°, these motions being identical with those found in a conventional stereophonic record. Considering now the motion associated with the signal L_b, which lags in the right channel with respect to the left channel by 90°, it is seen that such motion is in the form of a counter-clockwise helix. Similarly, the modulation R_b, which lags in the left channel by 90° relative to the right channel, appears as a clockwise helix. An advantage, therefore, of the encoding technique illustrated in FIG. 1 is that the right-front and left-front channels retain a theoretically infinite separation as with a conventional stereophonic record, while the left-back channel and the right-back channel appear more or less centered and somewhat dispersed between the two loudspeakers during playback. This is a very satisfactory way of displaying four-channel quadruphonic information in a two-track stereo system. A record made in accordance with the principles described above is campatible, that is capable of fully satisfactory performance on all conventional stereophonic phonographs, with the added capability of admitting to a decoding operation to convert its signals back into a four-channel program.

It will be observed that in phasor diagrams **46** and **48** of FIG. **1** the dominant signals L_f and R_f are both at 45° relative to the subdominant L_b and R_b components present in both; as a result of this phase relationship, the decoders described in the aforementioned co-pending application Ser. No. 118,271 produce output signals L_f and R_f which are in phase, and signals L_b and R_b which are also in phase with each other but displaced in phase relative to L_f and R_f. While this does not present a serious flaw in the performance of the system, it is preferable that all of the four phasors L_f, L_b, R_b and R_f, after decoding, be in phase. Accordingly, it is one object of the present invention to provide a system for encoding four signals into a pair of composite signals which, upon being decoded, results in four dominant signals which are all in phase with each other. Another object of the invention is to provide a system for encoding four sound channels into two composite signals which avoids exaggeration of output signal intensity upon decoding when equal signals are applied to the side terminals (e.g., left-front and left-back) of the encoder. Another object of the invention is to provide a system including an encoder for encoding four sound channels into two composite signals and a decoder for use therewith capable of resolving ambiguities in reproduction caused by the appearance of equal signals on the two "front" channels, or on the two "back" channels. Still another object is to provide an encoder having the foregoing features and advantages while at the same time simplifying and reducing the cost of circuitry for accomplishing them.

SUMMARY OF THE INVENTION

Briefly, the object of encoding four input signals into a pair of composite signals for recording on a two-track medium which, upon being decoded, results in four dominant signals which are in-phase with each other, is attained by an arrangement of phase-shift networks and adding circuits which produces the following relationships of the components of the respective composite signals. Designating (for convenience) the four input signals as L_f, L_b, R_b and R_f (for left-front, left-back, right-back and right-front), they are combined in such a manner that both the left and right composite signals contain subdominant signals L_b and R_b in quadrature relationship, and respectively contain dominant L_f and R_f components which are in phase with each other and also in phase with their associated R_b and L_b components, respectively. In one embodiment, the L_b component leads the R_b component in the left composite signal and lags the R_b component in the right composite signal, and in a second embodiment the L_b component lags the R_b component in the left composite signal and leads the R_b component in the right composite signal. The composite signals in both cases, upon decoding, produce four predominant signals which are all in phase, and also have characteristics which avoids exaggeration of output signal intensity when equal signals are applied to the side terminals (e.g., left-front and left-back) of the encoder.

In another embodiment, the encoder comprises an arrangement of phase-shifting networks and summing circuits which again causes subdominant L_b and R_b components to appear in quadrature relationship in both the left and right composite signals, with the L_b component leading the R_b component in the left composite signal and lagging the R_b component in the right composite signal, and the dominant L_f and R_f components to be in phase with each other, but in this case, the L_f component is 180° out of phase with its associated subdominant R_b component while the R_f component is in-phase with its associated subdominant L_b component. This phase relationship of the signal components in the left and right composite signals makes it possible to resolve ambiguities in situations when the sound signals are "panned"; that is, inserted into adjacent channels in an in-phase relationship.

Another important aspect of the invention resides in decoding apparatus which includes a matrix for decoding the composite signals to recover the four separate signals for presentation on four separate loudspeaker systems, particularly in the addition of further logic to the decoder control logic described in the aforementioned Bauer application Ser. No. 118,271 which is responsive to the composite signals produced by the last-described encoder to distinguish between front and back signals and to promptly and automatically adjust the gain of the front and back loudspeakers to enhance the realism of four-channel reproduction.

BRIEF DESCRIPTION OF THE DRAWINGS

Other objects, features and advantages of the invention, and a better understanding of its construction and operation, will be evident from the following detailed description, taken in conjunction with the accompanying drawings, in which:

FIG. **1** is a schematic diagram of the encoder described and claimed in co-pending application Ser. No. 44,224, to which reference has been made in discussing the background of the invention;

FIGS. **2** and **3**, to which reference has already been made, illustrate the operation of the encoding apparatus of FIG. **1**;

FIG. **4** is a pair of phasor diagrams useful in explaining the principles of the present invention;

FIG. **5** is a schematic diagram of encoding apparatus embodying the invention;

FIG. **6** is a schematic diagram of decoder apparatus described in co-pending application Ser. No. 118,271, useful in explaining the efficacy of the present encoding technique;

FIG. **7** is a schematic diagram of an alternative form of encoding apparatus embodying the invention;

FIG. **8** is a schematic diagram of still another alternative form of encoding apparatus embodying the invention;

FIG. 9 is a diagram illustrating the motion of the cutting or playback stylus of stereophonic recording or reproducing apparatus in response to signals encoded in accordance with the invention;

FIGS. 10 and 11 are phasor diagrams useful in explaining the operation and advantages of the encoder of FIG. 8;

FIG. 12 is a schematic diagram of a system for decoding signals encoded with the encoder of FIG. 8;

FIG. 12A is a schematic diagram of a modification of the system of FIG. 12; and

FIG. 13 is a schematic diagram illustrating how four original sound channels may be produced for recording and reproduction in accordance with the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

The encoders to be described herein are also described in the aforementioned application Ser. No. 328,874, and the disclosure thereof is incorporated herein by reference; however, the description to follow, particularly in view of the illustration of this previous encoder in FIG. 1, is believed to be sufficiently complete to enable one skilled in the art to understand its operation without recourse to the co-pending application.

The present invention is based on the discovery, illustrated in FIG. 4, that while it is important in the encoding process to maintain a 90° relationship between the phasors $0.707L_b$ and $0.707R_b$, the relative positions of the phasors L_f and R_f may be arbitrarily chosen insofar as decoding is concerned. In other words, any decoder designed to decode the composite signals L_T and R_T in FIG. 1 (such as the decoder described and claimed in co-pending application Ser. No. 118,271 will also satisfactorily decode the signals L_T' and R_T' shown in FIG. 4, regardless of the size of the angles α_1 and α_2 between phasors L_f and $0.707L_b$, and between phasors R_f and $0.707R_b$, respectively. Since the α's (45° in FIG. 1) are established by ψ-networks 28 and 34, by suitable design of these networks it is possible to place phasors L_f and R_f at any desired position with respect to the other two phasors in the group. An especially beneficial relationship is established by making both the angles $\alpha = 90°$ so that in the phasor group portraying L_T', the phasor L_f coincides with phasor $0.707R_b$, and in the other phasor group, the phasor R_f coincides with phasor $0.707L_b$. This relationship is readily obtained by modifying ψ-networks 28 and 34 so that instead of providing a phase-shift $\psi = 45°$ they provide a phase-shift $\psi + 90°$. Moreover, as one of the benefits of this invention, the above relationship permits the encoding function to be performed with only four ψ-networks, instead of the six required in the encoder of FIG. 1.

Referring now to FIG. 5, the encoder in accordance with the invention has four input terminals **60, 62, 64** and **66** to which input signals L_f, L_b, R_b and R_f originating with the quadruphonic program, and represented by phasors corresponding to the same signals depicted in FIG. 1, are respectively applied. Rather than being applied directly to a ψ-network as in the system of FIG. 1, input terminals **60** and **64** are connected to a summing junction **68** which is operative to add a unity measure of signal L_f to .707 of signal component R_b. Similarly, terminals **62** and **66** are connected to a second summing junction **70** which is operative to add a unity measure of signal R_f to 0.707 of signal L_b. Terminal **62** is also connected to the input of ψ-network **72** which introduces a relative phase-shift of ψ to the signals L_b, and terminal **64** is connected to the input of a second $(\psi + 0°)$ network **78**. The outputs of summing junctions **68** and **70** are respectively applied to the input terminals of ψ-networks **74** and **76**, both of which introduce a relative phase-shift of $\psi + 90°$.

The full output of ψ-network **74** is added in summing junction **80** to 0.707 of the output of network **72**, and similarly, the full output of network **76** is added at summing junction **82** to 0.707 of the output of network **78**. As a consequence of this phase-shifting and combining of signals, there appears at output terminal **84** a composite signal L_T' depicted by phasor group **88**, and at output terminal **86** a composite signal R_T', depicted by phasor group **90**. It will be observed that there is a one-to-one correspondence between phasor groups **88** and **90** and the corresponding phasor groups in FIG. 4 if the angle α in the latter group is set at 90°, which is in accordance with the teaching of this invention.

That the encoder of FIG. 5 is compatible with decoders intended for use with the signals encoded in accordance with the system of FIG. 1 is demonstrated by the comparative analysis presented in FIG. 6, as applied to the decoder described in co-pending application Ser. No. 118,271. This decoder includes a pair of input terminals **100** and **102** to which composite signals L_T and R_T are respectively applied. The signal applied to terminal **100** is applied in parallel to and phase-shifted by a pair of ψ-networks **104** and **106**, and the composite R_T signal applied to input terminal **102** is applied in parallel to ψ-networks **108** and **110**. These ψ-networks are of the type previously described, the networks **104** and **110** introducing a phase-shift of $(\psi' + 0°)$ and networks **106** and **108** introducing a phase-shift of $(\psi' + 90°)$. It will be noted that the reference angle is designated ψ' instead of ψ, as used in the encoder; this is to call attention to the fact that the reference phase-shift in the decoder need not be the same as in the encoder, provided the same reference phase-shift is used in all four of ψ-networks **104, 106, 106** and **110**. The outputs of networks **104** and **110** are applied directly to the left-front output terminal **112** and to the right-front output terminal **114**, respectively. Equal portions of the outputs of networks **106** and **110** are summed in a summing junction **116**, the output of which is applied to the left-back output terminal **118**, and equal portions of the outputs of networks **104** and **108** are summed in a second summing network **120**, the output of which is applied to the right-back output terminal **122**.

A comparison will now be made of the performance of the decoder of FIG. 6 in response to signals encoded in accordance with co-pending application Ser. No. 328,874, and to composite signals encoded in accordance with the present invention. Phasor groups corresponding to signals encoded in accordance with application Ser. No. 328,874 are shown in dotted lines, and the phasor groups encoded by the encoder of FIG. 5 are shown in solid lines. Phasor groups **130** and **132** portray the two input signals L_T and R_T which, upon being shifted in phase by the all-pass networks **104, 106, 108** and **110** appear as new phasor groups **134, 136, 138** and **140**. The phasors in these latter four groups are labeled with a prime to differentiate them from the corresponding phasors prior to introduction of the relative phase-shifts. The signal represented by phasor group **134** appears at output terminal **112** as phasor group **142** and contains a dominant component L_f' together with the smaller components $0.707L_b'$ and $0.707R_b'$. The phasor groups **136** and **140** after summing in junction **116** result in a signal at output terminal **118** represented by phasor group **144** containing a dominant phasor L_b' and subsidiary phasors $0.707L_f'$ and $0.707R_f'$. The sum of phasors **134** and **138** appearing at the output of summing junction **120** (output terminal **122**) is a composite signal represented by phasor group **146** having a dominant phasor R_b' accompanied by subsidiary signals $0.707R_f'$ and $0.707L_f'$. Finally, the phasor group **140** appears at output terminal **114** as phasor group **148**, and contains a dominant signal R_f' together with subsidiary signals $0.707R_b'$ and $0.707L_b'$. Thus, the decoded signals appearing at output terminals **112, 118, 122** and **114** each contains its appropriate dominant signal together with signals from two other channels diminished by the factor 0.707. It will be noted that the two principal front channel phasors, namely L_f' in group **142** and R_f' in group **148** are in phase, and that the two principal back channel vectors, L_b' in phasor group **144** and R_b' in group **146**, are also in phase with each other, but not in phase with the L_f' and R_f' phasors. While this phase relationship does not represent a major flaw in the performance of the system, it has been found preferable that the four predominant phasors all be in phase.

It will now be demonstrated that this favorable phase relationship is achieved when signals encoded with the encoder of FIG. **5** are decoded in the decoder of FIG. **6.** It will be remembered from the description of FIG. **5** that the encoded signals L_T and R_T are as portrayed by phasor groups **150** and **152,** the former after being acted upon by ψ-networks **104** and **106** appearing as phasor groups **154** and **156**, respectively, and the R_T signal after passage through ψ-networks **108** and **110** appearing as phasor groups **158** and **160**, respectively. Phasor groups **154** appears at output terminals **112** as phasor group **162,** and phasor group **160** appears at output terminal **114** as phasor group **164.** These output signals contain predominant signals L_f' and R_f', respectively. Phasor group **154** appears at output terminal **112** as phasor group **162,** and phasor group **160** appears at output terminal **114** as phasor group **164.** These output signals contain predominant signals L_f' and R_f', respectively, which are in phase with each other, and each includes subsidiary signals $0.707R_b'$ and $0.707L_b'$.

The phasor groups **156** and **160** upon being summed in summing junction **116** produces at output terminal **118** the composite signal portrayed by phasor group **166,** and the sum of the signals represented by phasor groups **154** and **158** appearing at the output terminal **122** of summing junction **120** is as portrayed by phasor group **168.** It will be noted that phasor groups **166** and **168** contain predominant phasors L_b' and R_b', respectively, which are in phase with each other, and also in phase with the predominant phasors in groups **162** and **164,** and each accompanied by two subsidiary signals $0.707R_f'$ and $0.707L_f'$. Comparison of phasor groups **162** with **142, 166** with **144, 168** with **146,** and **164** with **148** reveals that they contain the same respective subsidiary signals in the same magnitude and in the same intergroup phase relationships. Therefore, the respective signals will be capable of properly activating the enhancing logic and control circuits described in co-pending applications Ser. Nos. 328,874 and 118,271.

Another advantage of the present encoding technique will be seen from a comparison of phasor groups **144** and **166,** for example, in each of which there is shown in dotted line a signal L_s which results from applying equal signals to the left-front and left-back terminals of the encoder. Because of the angular relationship between phasors $0.707L_f'$ and $0.707L_b'$ in phasor group **144** as compared to the guadrature relationship between the corresponding phasors in group **166,** the resulting phasor L_s in group **144** is of greater magnitude than the corresponding phase in group **166.** The lack of exaggeration of the signal L_s is of significant advantage, and by symmetry, it will be recognized that exaggeration of the R_s signal which would result from application of equal signals to terminals **64** and **66** of the encoder of FIG. **5** is likewise avoided.

In summary, the improved encoder of FIG. **5** offers three significant advantages over the encoder of FIG. **1**: **1**) it provides encoding with four, instead of six ψ-networks, with an attendant reduction in the cost of the encoder; **2**) it produces encoded signals which, upon decoding, cause the predominant signals to all be in phase; and **3**) it avoids exaggeration of output signal intensity from the decoder when equal signals are applied to the side terminals of the encoder.

FIG. **7** illustrates a modification of the encoder of FIG. **5,** differing therefrom in the manner in which the four input signals are added and phase-shifted. In this case, the full L_f signal applied at terminal **170** is added to 0.707 of the R_b signal applied to input terminal **174** in a summing junction **178,** and the full R_f signal applied at input terminal **176,** is added in summing junction **180** to 0.707 of the L_b signal applied at input terminal **172.** The sum signals from summing junctions **178** and **180** pass through respective ψ-networks **182** and **184** and are added in respective summing junctions **186** and **188** to 0.707 of signals L_b and R_b, respectively, after being shifted in phase by $\psi + 90°$ in respective ψ-networks **190** and **192.** The L_T and R_T signals appearing at output terminals **194** and **196,** represented by phasor groups **198** and **200,** respectively, are similar to the corresponding phasor groups **88** and **90** in FIG. **5** except that in group **198** the $0.707R_b$ phasor leads the $0.707L_b$ phasor, wherein in group **88** the L_b phasor leads the R_b phasor; the positions of the L_b and R_b phasors in groups **200** and **90** are similarly interchanged. While this decoder provides perfectly consistent signals, it has the slight disadvantage stemming from the fact that the $0.707R_b$ phasor in group **198** leading the corresponding signal in phasor group **200** tends to cause this right-back signal to appear to lean slightly toward the left-front channel when the record is replayed stereophonically over two loudspeakers. By symmetry, the left-back signal likewise will tend to lean slightly to the right when the record is replayed stereophonically. Thus, while the alternative encoder of FIG. **7** produces acceptable composite signals for reproduction over four loudspeakers, it is inferior to the encoder of FIG. **5** if the record carrying the encoded signals is to be played over a two-channel stereophonic playback apparatus.

Referring again to FIG. **4,** although the values of angles α_1 and α_2 of 90° are preferred, it may be desirable in some cases to chose other values, for example, 0° or

180°, or α_1 and α_2 may be different, as will now be obvious to those skilled in the art.

By seemingly slight modifications of the encoder of FIG. **7**, and of the decoder shown in FIG. **6** (which is fully described and claimed in the aforementioned co-pending application, Ser. No. 118,271, the performance of the overall system can be significantly improved, particularly in its ability to resolve ambiguities in cases when the sound signals are panned; that is, inserted into adjacent channels in an in-phase relationship. The cause of such ambiguities will become apparent from analysis of the decoded signals delivered by the decoder of FIG. **6** which, it will be remembered, contain predominant signals L_f, L_b, R_f and R_b, respectively, together with two "contaminating" signals from other channels. Actually, the contaminating signals are not noticed when all four predominant signals are simultaneously present, as when four different performers produce four parts of a musical selection in all four channels, since there is sufficient mixing of sound in the room or listening area that the presence of the contaminating signals in the individual channels is inconsequential. They are noticable, however, when sound is present in only a single channel, or in at most two channels, because in these instances, when the sound should be coming from a single loudspeaker or from two loudspeakers, it is instead heard from all four, which is readily noticable and sometimes objectionable. This situation is improved, and the realism of four channel reproduction enhanced by the logic and control systems described in the aforementioned co-pending applications Ser. No. 328,874 and 118,271 which recognize the presence of sounds in individual channels and generate signals for controlling the gain of gain control amplifiers in the individual loudspeaker circuits in response to the instantaneous presence of the predominant signals. Thus, if a signal appears only in the left-front channel, for example, (and which, because of the protocol of the decoder, also appears at reduced level in both of the back channels) the logic functions to enhance the gain of the front loudspeaker amplifiers and to turn down the gain of the back loudspeaker amplifiers thereby to cause the sound to appear to originate at the left-front loudspeaker only. The logic and control circuitry operates similarly with respect to the other three loudspeakers with the consequence that when artists are performing in concert in all four channels the gain of the respective amplifiers are increased and decreased to instantaneously enhance the channel or channels in which signals are predominant at a particularly instant to give a highly realistic replication of the original four channel program.

The above-described methods of encoding four signals into two and decoding them back into four works very well in the majority of circumstances, one exception, however, being when the sounds are panned, that is, inserted into adjacent channels in an in-phase relationship, and then only in two specific instances: namely, when the sound is panned precisely in the middle of the two front channels by application of equal signals to the L_f and R_f encoder terminals, or when it is panned precisely in the middle of the two back channels L_b and R_b. It can be shown that in these two circumstances the modulation produced on the stereophonic disc is the same when the two front channels are panned as it is when the two back channels are panned. Consequently, the logic and control system used with the decoder is unable to distinguish whether such panned sound signal belongs to the front channels or to the back channels, resulting in an ambiguity. In accordance with another aspect of the present invention this ambiguity is resolved by modification of the encoder and the decoder, thereby to provide a significant improvement in performance of the system.

The modified encoder, illustrated in FIG. **8**, has four input terminals **210**, **212**, **214** and **216** to which the four signals L_f, L_b, R_b and R_f, depicted as in-phase signals of equal amplitude, are respectively applied. The total L_f signal is added in a summing junction **218** to −0.707 of the R_b signal, the output of the summing junction being applied to a phase-shifting network **220** which introduces a reference phase-shift ψ, which is a function of frequency. The full R_f signal at terminal **216** is added in summing network **222** to 0.707 of the L_b signal appearing at input terminal **212**, and the output is passed through the ψ-network **224**, which also provides the reference phase-shift ψ. The L_b and R_b signals are also applied to respective ψ-networks **226** and **228**, each of which provides a phase-shift of $\psi + 90°$. The full signal appearing at the output of network **220** is added in a summing circuit **230** to −0.707 of the signal appearing at the output of network **226** to produce at its output terminal **232** a composite signal designated L_T. Similarly, the full signal from network **224** is added in summing junction **234** to 0.707 of the signal from network **228**, the latter in this case being in the positive sense. The signal appearing at the output terminal **236** is the composite signal designated R_T. As in the case of the other encoders, the signals L_T and R_T may be transmitted by FM multiplex radio, or they may be recorded on any two-channel medium such as a two-track tape or stereophonic record for later reproduction.

The significance of the modifications to the encoder of FIG. **7** to provide the encoder of FIG. **8** (namely, the reversal of the phase of the 0.707 terminals of the two summing circuits in the upper half of the diagram) will be appreciated from an analysis of the phasor relationship of the L_T and R_T composite signals portrayed as phasor groups **238** and **240**, respectively. It will be observed that phasor group **238** consists of the signal L_f (which although shown in the same phase relationship as the input signal L_f has a ψ-as-a-function-of-frequency angle difference between them), a signal $0.707R_b$ in a negative sense with respect to its corresponding input phasor, and a $0.707L_b$ signal which lags phasor $0.707R_b$ by 90° because of the action of network **226**. Phasor group **240** consists of the original signal R_f in the same relative phase position as its corresponding input signal, a signal $0.707L_b$ in phase with the R_f signal, and a **.707**R_b signal lagging the $0.707L_b$ signal by 90° due to the action of ψ-network **228**. As has been pointed out hereinabove, in the interest of providing better realism of image placement when the record is played in conventional stereophonic mode over two loudspeakers, it is preferable to arrange the phasor $0.707L_b$ in phasor group **240** to lag behind the similarly numbered phasor in phasor group **238**, and conversely, to arrange the phasor $0.707R_b$ in phasor group **238** to lag behind the corresponding phasor in group **240**. Thus, the connections shown in FIG. **8** are preferred, but it is possible to interchange the phase positions of the phasors and still obtain the principal benefits of the invention.

With reference again to FIG. 2, when the encoded signals represented by phasor groups **238** and **240** are applied to a stereophonic photograph record, the left groove is modulated during the cutting process in the direction of the arrow L_T (which is at 45° to the surface of the record) under the influence of the left-channel signal L_T, while the groove is modulated in the direction of the arrow R_T under the influence of the right-channel signal R_T.

Referring now to FIG. **10**, the effect of the panning is to divide the signal (as by means of two coupled attenuators) between two channel inputs. At the midpoint of the panning operation, the signal becomes precisely divided between the front channels L_f and R_f, or between the back channels L_b or R_b; this condition will now be examined. The phasor groups **238** and **240** from FIG. **8** are repeated here as phasor groups **250** and **252**, respectively, and the panned "center" signals have been added. The front center signal, C_f, is placed in the proportion $0.707C_f$ and in-phase in the phasor groups **250** and **252**, appearing as phasors **254** and **256**. Since these phasors are equal and in-phase they cause the arrows L_T and R_T in FIG. **2** to combine to produce a horizontal motion; accordingly, the center front signal C_f appears as a horizontal arrow **246** in FIG. **9**. It is seen, thus far, that the phasor group in FIG. **9** depicts the left-front channel phasor L_f, the center channel phasor C_f, and the right-front channel phasor R_f in a relationship which those skilled in the art will recognize as portraying the modulation of a conventional stereophonic record.

Reverting now to FIG. **10**, it will be noted that the center-back channel C_b is divided in the proportion 0.707 in the left back and right back channels, and since these two phasors appear as a 0.707 fraction, the corresponding fraction of the C_b signal is 0.5 in phase with the $0.707L_b$ phasor and **0.5** in phase with the $0.707R_b$ phasors in both phasor groups. With this convention in mind, it is seen that the two phasors in each group add to the larger phasors $0.707C_b$ in each of the phasor groups **250** and **252**; however, it should also be observed that the phasor $0.707C_b$ in phasor group **250** is out-of-phase with the corresponding phasor in group **252**. This is an important quality of the encoder of FIG. **8** because now the centerback signal C_b is of an entirely different character than the center front signal C_f. It will be recognized that the signal C_b having an out-of-phase relationship in the two channels will result in a vertical modulation of the groove **52** in FIG. **2**, which is depicted by the arrow **248** in FIG. **9**. It will be realized that any signal recorded in this manner cannot be reproduced by a monophonic phonograph pick-up, nor by the monophonic section of an FM multiplex transmitting station; consequently, when using the encoder of FIG. **8** the "center-back" location should preferably be used for occasional sounds such as reverberation, motion during planning, etc., and not for the placement of an important artist since he would not be heard when the signal is broadcast over AM radio or over monophonic FM radio. Such signals would, however, be fully audible with stereophonic or quadruphonic modes of reproduction, and all other locations of the artist would be transmitted satisfactorily.

Another significant feature of the encoder is illustrated by the phasor groups **256** and **258** in FIG. **11**, the former depicting the situation which results when the phasor groups **250** and **252** of FIG. **10** are added and the latter depicting the situation when the composite signal R_T (phasor group **252**) is subtracted from L_T (phasor group **250**). It will be noted that when L_T and R_T are added the phasors L_f, L_b, R_b and R_f all have an intensity equal to unity, whereas the front center signal C_f is augmented by a factor 1.414, which is exactly what happens when a stereophonic record is played over a monophonic player. The back center signal C_b, is cancelled, however, because of the aforementioned out-of-phase relationship. When the phasor groups are subtracted, the phasors L_f, L_b, R_b and R_f again all appear with unity amplitude, but this time the center back signal, C_b, is augmented by the factor 1.414 while the center front signal, C_f, is cancelled. The relationship portrayed by phasor groups **256** and **258** are extremely important since they indicate that if only a center front signal is present, i.e., no center back signal, the phasor group **256** will be greater than group **258**, and, conversely, if there is only a center back signal but no center front signal, the phasor group **258** will be the larger. This interesting property is used to advantage to enhance the operation of the decoder to be utilized with the encoder of FIG. **8**, which will now be described.

The decoder, illustrated in FIG. **12**, is in many respects similar to the decoder of FIG. **6**. The signals L_T and R_T represented by phasor groups **238** and **240**, respectively, are applied to respective input terminals **300** and **302**, from whence they are applied in parallel to respective pairs of ψ-networks **304**, **306**, **308** and **310**. In this manner, each of the signals L_T and R_T passes without relative phase-shift through networks **304** and **308**, respectively, and also pass with a relative phase-shift of 90° through networks **306** and **310**. In the phasor groups **312**, **314**, **316** and **318** portraying the output signals from networks **304**, **306**, **310** and **308**, respectively, the individual phasors, essentially identical with the corresponding phasors in groups **238** and **240**, are differentiated with a prime to indicate that they have been subjected to the action of a ψ-network and thus differ from the input phasors by an angle which is a ψ-function of frequency, in addition to the differential angle introduced by the networks.

The outputs of networks **304** and **308** are applied directly to the input terminals of respective gain control amplifiers **312** and **314**, the outputs of which are applied to respective loudspeakers **316** and **318**. The signals applied to loudspeakers **316** and **318** contain predominant original signals L_f' and R_f', respectively, the two subdominant "contaminating" signals $0.707L_b'$ and $0.707R_b'$. Equal proportions, namely, 0.707, of the outputs of networks **306** and **308** are summed at a summing junction **320** to produce a composite signal consisting of a predominant signal L_b', which is applied to a gain control amplifier **322** and thence to loudspeaker **324**. Equal negative portions, namely, **−0.707**, of the outputs of networks **304** and **310** are summed at a second summing network **326** to produce for application to a fourth gain control amplifier **328** a composite signal composed of a dominant signal, R_b', together with $0.707R_f'$ and $0.707L_f'$; after amplification, this composite signal is applied to loudspeaker **330**. It will be observed from the phasor groups **332**, **334**, **336** and **338** which respectively portray the composite signals appearing at loudspeakers **316**, **324**, **330** and **318**, that the predominant phasors at all four loudspeakers are in-phase.

As has been explained previously, the "contaminating" signals in each of the composite signals have little effect on the listening quality of the decoder as long as all of the predominant signals are simultaneously present because there is sufficient confusion and mixing in the air of sounds from different sources that the precise location of each sound is not easy to determine. However, if the nature of the original program signals are such as to be best presented over one or two of the loudspeakers, it becomes desirable to enhance or sharpen the realism of reproduction. This is accomplished with a logic and control circuit **340** which preferably is of the type disclosed in detail in co-pending application Ser. No. 118,271, the teaching of which is hereby incorporated by reference. The logic described in this co-pending application is characterized as "wave-matching" logic which makes an instantaneous comparison of the wave shape of the signals and makes a decision to either increase the gain of amplifiers **312** and **314** which feed the two front loudspeakers **316** and **318** and diminish the gain of the amplifiers which supply the back loudspeakers, or, conversely, whether to increase the gain of the back amplifiers **322** and **328** and reduce the gain of amplifiers **312** and **314.** The information on which the logic circuit bases its decisions is derived from the outputs of ψ-networks **308** and **306** and is applied to input terminals **342** and **344**, respectively. The logic and control circuit **340** is operative to produce output signals at its output terminals **346** and **348** as follows: If a single channel signal L_f, or two uncorrelated signals L_f and R_f are present in the input, the logic **340** produces output signals which are operative to increase the gain of amplifiers **312** and **314** and to decrease the gain of amplifiers **322** and **328.** If, on the other hand, the principal signals are L_b alone, or uncorrelated signals L_b and R_b, the logic is operative to produce signals which increase the gain of amplifiers **322** and **328** and to diminish the gain of amplifiers **312** and **314.** However, as has been explained above, if the signals L_f and R_f are equal and in-phase, or, conversely, if the signals L_b and R_b are equal, in-phase signals, the logic described in the aforementioned co-pending application Ser. No. 118,271 lacks the information necessary to determine which set of amplifiers is to be turned on and which is to be turned off, resulting in a spread of the center signals to all four loudspeakers. It should be emphasized that this does not produce an unpleasant reproduction, but for the sake of greater realism in more sophisticated decoder equipment it is desirable that the gain of the front and back loudspeakers be promptly and automatically adjusted in response to the center front and center back signals. This desired action is made possible by the encoding technique described in connection with FIG. **8,** and from the recognition that differentiation between back and front center signals, illustrated in FIG. **11,** as achievable when the signals are encoded in this way.

The decoder of FIG. **12** is given this additional capability by the provision of logic in addition to that provided by logic circuit **340.** The input signals L_T and R_T at input terminals **300** and **302** are applied via conductors **350** and **352**, respectively, to a summing junction **354** and to a subtracting junction **356.** The sum signal appearing at the output of summing junction **354** being the sum of L_T and R_T is as depicted by phasor group **256** in FIG. **11,** and the output of subtracting junction **356** is a composite signal having the properties portrayed by phasor group **258.** It will be evident from reconsideration of FIG. **11** that if the front center signal predominates the output of summing circuit **354** will exceed the output of subtraction circuit **356,** and, conversely, if the back center signal predominates, the output of the subtracting junction will exceed the output of the summing junction. In order to obtain the relative ratio between these two quantities, the outputs of junctions **354** and **356** are amplified in respective quasi-logarithmic amplifiers **358** and **360** and then rectified by respective rectifiers **362** and **364,** which are preferably full-wave rectifiers, and integrated with leaky integrators **366** and **368,** respectively. The difference of the of outputs of the integrators, appearing at terminals **370** and **372,** respectively, is proportional to the relative magnitudes of the sum and difference signals produced by junctions **354** and **356.** Inasmuch as the output at terminal **370** arises from the sum of L_T and R_T while the output at terminal **372** arises from the difference of these signals, the former will be greater than the latter with the presence of a front-center signal. To make the decision that a front-center signal is present, a subtracting junction **372** is provided which subtracts the signal at terminal **372** from the signal appearing at terminal **370.** The output of subtracting junction **372** is applied to a summing junction **374** where it is added to the output signal at terminal **346** of logic and control circuit **340,** the output of junction **374** being applied in parallel to the gain control electrodes of amplifiers **312** and **314** (which supply the front loudspeakers **316** and **318**) to augment their gain. A second subtracting junction **376** subtracts the signal appearing at terminal **370** from the signal appearing at terminal **372,** which produces a negative output when a center-front signal is present; this is combined in a summing junction **378** with the output at terminal **348** of logic circuit **340,** the sum signal serving to diminish the gain of amplifiers **322** and **328** which supply the rear loudspeakers **324** and **330,** respectively. Conversely, if a center-back signal is present, the just-described logic would be operative to partially or completely turn off the front loudspeakers and to augment the gain of the rear loudspeakers.

It will be observed that with the logic of FIG. **12** the control action is derived from the sum of a pair of control signals. It may be advantageous, and it is possible by modification of the system of FIG. **12,** to instead control the action of the gain control amplifiers by the stronger of the control signals rather than their sum. To this end, the circuit of FIG. **12** is modified as shown in FIG. **12A** (which illustrates only the portion of the FIG. **12** circuit affected by the modification) so as to combine the control signals from the "wave-matching" logic **340** and from the just-described "center front-center back" logic in an OR circuit, rather than summing them. Since the output signal at terminal **348** of the logic and control circuit **340** is simply the inverse of the signal at terminal **346,** only the latter need by used; similarly, the output signal from summing junction **376** being the inverse of the output signal from junction **372,** only the latter need be used.

The OR circuit takes the form of two pairs of rectifiers **380** and **382,** and **384** and **386,** so connected that one of the rectifiers in each pair is forward conducting and the other is backward connecting. The signal appearing at terminal **346** is applied to rectifiers **380** and **382** and the output signal from summing junction **372** is applied to rectifiers **384** and **386.** By reason of a connection therebetween the outputs of rectifiers **380** and **384** are added at points **388** and **392,** and will be the greater of positive voltages at terminal **346** or at the output of junction **372.** At the same time, the outputs of rectifiers **382** and **386** are combined at points **390** and **394** and will be the greater of negative voltages at terminal **346** or junction **372.** The voltages appearing at points **388** and **390** are added at summing junction **374**; consequently, the output of summing junction **374** will be the greater of the outputs of either the wave-matching logic or the "center front-center back" control logic, and not the sum as in FIG. **12.** The negative voltage outputs at points **392** and **394** are added at summing junction **378,** thereby producing the inverse of the output of junction **374.** The outputs of the junctions **374** and **378** are applied to the pairs of gain control amplifiers **312** and **314** and **322** and **328,** respectively, as shown in FIG. **12.**

It will be recognized from the foregoing description that the basic decoder of FIG. **12** is the same as that described in FIG. **6** with the single exception that the adding junction **326** inverts the phase of the R_b'' signal; alternatively, this phase reversal action may be obtained in the amplifier **328,** or in its associated loudspeaker.

If desired, the composite input signals L_T and R_T may be shaped by frequency-dependent wave-shaping networks **380** and **382,** respectively, before application to the logic circuitry so as to limit the action of the logic to voice signals, for example, or other instrumental sound, to make the logic less susceptible to erroneous decisions with certain high power, low frequency instruments, such as the bass drum.

There are obvious precautions which should be taken to obtain best results from the encoder of FIG. **8** and the decoder of FIG. **12.** For example, it is not advisable to record an important signal, such as a soloist, in the center-back because the signal will disappear or be greatly attenuated when played over a monophonic phonograph. However, it is perfectly acceptable to record auxiliary signals, such as the sound of a vehicle moving around the room, or the sound of a symphony orchestra in a reverberant concert hall, as diagrammatically illustrated in FIG. **13.** In this example the musical groups, arranged in the front of a concert hall represented by the dashed-line enclosure **390,** are designated by the dotted ellipses **400–412,** the sounds produced by these groups being picked up by microphones **416–428,** respectively. The microphone circuits are isolated from each other by a plurality of buffer amplifiers **430–444,** connected as shown. Microphones **416, 418** and **420** are connected together and to the left front terminal **450,** and microphones **424, 426** and **428** are likewise connected together and to the right front terminal **456.** The output of the center microphone **422,** which may be reserved for the soloist, is fed in parallel to equal-gain amplifiers **436** and **438** so as to feed equal parts of the signal to the left front and right front terminals, thus constituting the center-front channel of the record.

Reverberation and other space effects may be conveniently picked up by a suitable arrangement of a dual cardioid microphone, exhibiting polar patterns depicted by **446** and **448,** placed near the center of the hall to provide a realistic reverberation delay. The outputs of the two transducers of the microphone are fed to terminals **452** and **454** which correspond to the back channels. This arrangement of microphones produces two correlated signals: the sound produced by the soloist which is applied to the front channels **450** and **456,** and that due to reverberation picked up by microphone sections **446** and **448** and applied directly and in-phase to the terminals **452** and **454.** Since during the performance, the orchestra and the soloist will produce the stronger signals, the output of the summing junction **354** (FIG. **12**) will exceed that of the subtracting junction **356,** resulting in an increase in the gain of the front loudspeakers. However, during prolonged pauses, the sound of the orchestra will decay quickly, while the reverberant sound will continue for some time. During this time the reverberation sound will be the stronger, resulting in a greater output at the subtractor junction **356** than at the summing junction **354,** thereby causing the gain of the front loudspeakers to be attenuated and the rear loudspeakers to be increased in gain to reproduce with realism the space effect of the concert hall. The time constants of the integrators **366** and **368** are preferably adjustable to permit adjustment to give a pleasing performance; attack times of about one to five milliseconds and decay times of the order of **0.4** seconds are typical.

Although the invention has been described in terms of combining signals in specific proportions, namely, 1.00 and 0.707, and in terms of specific relative phase angles, it is to be understood that minor departures from these specified values may be made without affecting system operation. Thus, although it is convenient to describe and claim the invention in specific terms, it is intended that the claims shall encompass such departures from the stated values.

I claim:

1. Signal decoding apparatus for decoding first and second composite signals L_T and R_T respectively containing dominant signals L_f and R_f in phase with each other and each including two subdominant signal components L_b and R_b in quadrature relationship, with said L_b and R_b components in one of said composite signals leading and lagging, respectively, the L_b and R_b components in the other composite signal, and with one of the L_f and R_f signals in phase with its respective associated R_b and L_b component and the other in phase opposition with the said associated component, said apparatus comprising, in combination:

decoder means including first and second input terminals to which said first and second composite signals are respectively applied and at least two all-pass phase-shifting networks connected to respective ones of said input terminals, said phase-shifting networks being operative to shift the phase of one of said composite signals relative to the other by substantially 90° thereby to position the L_b and R_b components in one of the relatively phase-shifted first and second composite signals either in phase coincidence or in phase opposition with corresponding components in the other relatively phase-shifted composite signal,

signal-combining networks for combining said relatively phase-shifted first and second composite signals to derive third and fourth composite signals respectively containing dominant signal components L_b and R_b which are in phase with each other and each including two subdominant signal components L_f and R_f, and means for applying composite signals respectively containing said signal components L_f, R_f, L_b and R_b as predominant components to first, second, third and fourth signal amplitude-modifying means, respectively, for reproduction over four corresponding sound-reproducing devices;

control circuit means operative in response to at least said relatively phase-shifted first and second composite signals to detect whether said relatively phase-shifted first and second composite signals contain substantially equal amplitude signal components in phase coincidence or in phase opposition and to produce a first control signal operative when applied to said first and second signal amplitude-modifying means to enhance the gain thereof when said first and second composite signals do not contain said signal components L_b and R_b either in phase or in phase opposition and to produce a second control signal operative when applied to said third and fourth signal amplitude-modifying means to enhance the gain thereof when said third and fourth composite signals do not contain substantially equal signal components either in phase or in phase opposition,

a logic circuit connected to said first and second input terminals operative to compare the sum and the difference of said first and second composite signals and to produce a third control signal when the sum exceeds the difference and to produce a fourth control signal when the difference exceeds the sum, and

means for combining said first and third control signals and applying the combined control signal to said first and second signal amplitude-modifying means to control the gain thereof, and means for combining said second and fourth control signals and applying the combined control signal to said third and fourth signal amplitude-modifying means to control the gain thereof.

2. Apparatus in accordance with claim **1,** wherein said combining means includes circuit means connected to said control circuit means and to said logic circuit and operative to select the larger of said first and said third control signals, and means for applying the larger of said control signals to said signal amplitude-modifying means.

3. Apparatus in accordance with claim **2** wherein said combining means is an OR circuit.

4. Signal-decoding apparatus for decoding first and second composite signals L_T and R_T respectively containing dominant signals L_f and R_f in phase with each other and each including two subdominant signal components L_b and R_b in quadrature relationship, with the L_b and R_b components in one of said composite signals leading and lagging, respectively, the L_b and R_b components in the other of said composite signals, and with one of the L_f and R_f signals in phase opposition with its respective associated R_b or L_b component and the other in phase with the said associated component, the apparatus including:

phase-shifting means operative to shift the phase of one of said composite signals relative to the other by substantially 90° thereby to position the L_b and R_b components in one of the relatively phase-shifted first and second composite signals either in phase coincidence or in phase opposition with corresponding components in the other;

signal-combining networks for combining said relatively phase-shifted first and second composite signals to derive third and fourth composite signals respectively containing dominant signal components L_b and R_b which are in phase with each other and each including two subdominant components L_f and R_f;

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

a control circuit operative in response to said relatively phase-shifted first and second composite signals to produce a first control signal operative when applied to said first and second gain control means to enhance the gain thereof when said first and second composite signals do not contain substantially equal signal components either in phase or in phase opposition and to produce a second control signal operative when applied to said third and fourth gain control means to enhance the gain thereof when said third and fourth composite signals do not contain substantially equal signal components either in phase or in phase opposition;

a logic circuit connected to receive and operative to compare the sum and the difference of said first and second composite signals and to produce a third control signal when the sum exceeds the difference and to produce a fourth control signal when the difference exceeds the sum; and

means for combining the control signals from said control circuit with the control signals from said logic circuit for producing fifth and sixth control signals for application to the said gain control means to control the gain thereof.

5. Apparatus in accordance with claim **4,** wherein said control circuit is operative to produce first and second control signals, and said logic circuit is operative to produce third or fourth control signals depending upon whether the sum exceeds the difference of the first and second composite signals, or vice versa, and wherein said combining means comprises means for adding the third and fourth control signals to the first and second control signals, respectively.

6. Apparatus in accordance with claim **4,** wherein said combining means includes a circuit connected to said control circuit and to said logic circuit and operative to select the larger of the control signals produced by the control circuit and the logic circuit, and means for applying the larger of the signals to said gain control means.

7. Apparatus for reproducing on four sound-reproducing devices four directional audio information signals respectively designated L_f, R_f, L_b and R_b contained in first and second composite signals respectively containing to the extent they are present dominant L_f and R_f component signals, and each including to the extent they are present sub-dominant L_b and R_b component signals, with said L_b and R_b component signals in one of said composite signals in substantially quadrature relationship with the corresponding component signals in the other of the composite signals, the combination comprising:

decoding circuit means including first and second pairs of all-pass phase-shifting networks connected to receive said first and second composite signals, respectively, a first phase-shifting network of each pair being operative to shift the phase of the applied signal by a predetermined reference angle and a second phase-shifting network of each pair being operative to shift the phase of the applied signal by an angle differing from said reference angle by substantially 90°,

signal-combining networks for combining the relatively phase-shifted first and second composite signals from the first phase-shifting network of one of said pairs and from the second phase-shifting network of the other of said pairs to derive third and fourth composite signals respectively containing dominant signal components L_b and R_b which are in phase with each other and each including two subdominant signal components L_f and R_f,

signal-coupling means connected to receive and operative to couple composite signals respectively containing said signal components L_f, R_f, L_b and R_b as their predominant components to respective ones of first, second, third and fourth sound-reproducing devices, said signal-coupling means including signal amplitude-modifying means for separately adjusting the amplitude of the composite signal applied thereto,

a control circuit operative in response to relatively phase-shifted first and second composite signals from the first phase-shifting network of one of said first and second pairs of phase-shifting networks and from the second phase-shifting network of the other pair to produce a first control signal operative when applied to said first and second signal amplitude-modifying means to enhance the gain thereof when said first and second composite signals do not contain substantially equal signal components either in phase or in phase opposition or a second control signal operative when applied to said third and fourth signal amplitude-modifying means to enhance the gain thereof when said third and fourth composite signals do not contain substantially equal signal components either in phase or in phase opposition,

a logic circuit connected to receive and operative to compare the sum and the difference of the first and second composite signals and to produce at least one of third and fourth control signals depending upon whether the sum exceeds the difference, or vice versa, and

means for combining the first or the second control signals from said control circuit with the third or the fourth control signals, respectively, from said logic circuit for producing fifth and sixth control signals for application to said first and second and to said third and fourth signal amplitude-modifying means, respectively, to control the transmission characteristic thereof.

8. Apparatus in accordance with claim **7**, wherein said last-mentioned means comprises means for adding the third and fourth control signals to the first and second control signals, respectively.

9. Apparatus in accordance with claim **7**, wherein said last-mentioned means includes a circuit connected to said control circuit and to said logic circuit and operative to select the larger of the control signals produced by the control circuit and by the logic circuit, and means for applying the larger of the signals to said signal amplitude-modifying means.

10. Apparatus in accordance with claim **7**, wherein said logic circuit includes first and second summing junctions each connected to receive both of said first and second composite signals and operative to produce sum and difference signals, respectively,

first and second logarithmic amplifiers connected to receive and operative to amplify said sum and difference signals, respectively, and

third and fourth summing junctions each connected to receive both of said amplified sum and difference signals and operative to produce at least one of said third and fourth control signals depending upon whether the sum of the first and second composite signals exceeds the difference of the first and second composite signals, or vice versa.

11. Apparatus for reproducing four individual audio information signals respectively designated L_f, R_f, L_b and R_b contained in first and second composite signals respectively containing to the extent they are present dominant L_f and R_f component signals and each including to the extent they are present sub-dominant L_b and R_b component signals with a phase-shift angle of substantially **90**° between said L_b component signals and between said R_b component signals, the combination comprising:

decoding circuit means connected to receive said first and second composite signals and operative in response thereto to derive third and fourth composite signals respectively containing predominant L_b and R_b component signals and each including sub-dominant L_f and R_f component signals,

first, second, third and fourth signal amplitude-modifying means respectively connected to receive and operative to couple said first, second, third and fourth composite signals to respective ones of first, second, third and fourth sound-reproducing means,

a logic circuit connected to receive and operative to compare the sum and the difference of said first and second composite signals and to produce a first control signal when the sum exceeds the difference and to produce a second control signal when the difference exceeds the sum, and

means for applying said first control signal to said first and second signal amplitude-modifying means, and means for applying said second control signal to said third and fourth signal amplitude-modifying means.

12. Apparatus in accordance with claim **11**, wherein said logic circuit includes

first and second summing junctions to each of which both said first and second composite signals are applied and operative to produce sum and difference signals, respectively,

first and second logarithmic amplifiers connected to receive and operative to amplify said sum and difference signals, respectively, and

means connected to receive said amplified sum and difference signals and operative to produce one or the other of said first and second control signals depending upon whether the amplified sum signal exceeds the amplified difference signal, or vice versa.

* * * * *