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(19)

(54) SOUND REPRODUCTION SYSTEMS

(71) We, NATIONAL RESEARCH DEVELOPMENT CORPORATION, a British Corporation established by Statute, of Kingsgate House, 66—74, Victoria Street, London, SW1, do hereby declare the invention, for which we pray that a patent may be granted to us, and the method by which it is to be performed, to be particularly described in and by the following statement:—

This invention relates to sound reproduction systems and more particularly to sound reproduction systems which enable the listener to distinguish sound from sources extending over 360° of azimuth. Cetrain aspects of the invention are concerned with the provision of a sound reproduction system of this type which in addition enables the listener to distinguish sounds from sources at different heights.

United Kingdom Specification No. 1,369,813 discloses a sound reproduction system which enables the listener to hear sound from sources extending over 360° of azimuth and which employs only two independent transmission channels. In the system described in this specification, one channel carrier so-called omni-directional signal components which contain sound from all horizontal directions with equal gain. The other channel carries so-called azimuth or phasor signal components containing sounds with unity gain from all horizontal directions but with a phase shift relative to the corresponding omni-directional signal component which is related to, and is preferably equal to, the azimuth angle of arrival measured from a suitable reference direction. The phasor signal may be resolved into two components with a phase difference of 90°. When these signal components are applied to four loudspeakers located at the corners of a square, one signal component constitutes a difference signal indicating the difference in signal strength between the signals for a first adjacent pair of loudspeakers and the signals for a second adjacent pair comprising the other two loudspeakers. The other component constitutes a second difference signal indicative of the difference in signal strength between the signals for a third adjacent pair of loudspeakers comprising one loudspeaker from each of the first and second adjacent pairs and the signals for a fourth adjacent pair comprising the other loudspeaker from each of the first and second adjacent pairs.

One object of the present invention is to improve the results obtained with a four loudspeaker array when the four loudspeakers are not uniformly spaced around the centre of a listening area.

The expression "the required uncompensated difference in signal strength", as used hereinafter, means the difference in signal strengths which would be required to reproduce the sound signals encoded in the input signals with the correct sound localisation if the loudspeakers were uniformly spaced around the centre of the listening area.

According to the invention, there is provided a decoder for a sound reproduction system having four loudspeakers surrounding a listening area each located on one of the diagonals of a non-square rectangle between the point of intersection of said diagonals and a respective corner of said rectangle, said decoder comprising input means for receiving at least two input signals comprising complex linear combinations of pressure signal components and velocity signal components representative of the pressure and of velocity at a listening position and signal processing means for producing first and second difference signal components from said velocity signal components, said first difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for a first adjacent pair of said loudspeakers and the sum of the signals for a second adjacent pair comprising the other two loudspeakers and said second difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for a third adjacent pair of loudspeakers comprising one loudspeaker from each of said first and second adjacent pairs and the sum of the signals for a fourth adjacent pair of loudspeakers comprising the other loudspeaker from each of said first and second adjacent pairs of loudspeakers, said decoder further comprising layout control means for applying first and second gains to said first and second difference signal components, the ratio between the first and second gains being substantially equal to the ratio between the sine of half the angle between the diagonals on which said first pair of loudspeakers are located and the sine of half the angle between the diagonals on which said third pair of loudspeakers are located, and output means responsive to said layout controls means and said pressure signal components for producing a responsive output signal for each loudspeaker.

In some embodiments of the invention, the first and second signal components may exist as separate signals, the means for receiving the input signals being arranged to produce an omni-directional signal and two difference signals for supply to the means for producing the respective output signals for the loudspeakers. When the invention is applied to the system described in the above-mentioned specification, the difference signal components are combined in the phasor signal and the relative variation of gains between them is achieved by varying the relative phase shift between the omni-directional and phasor signal components for each loudspeaker so that they exceed the amount of the phase difference for a corresponding loudspeaker in a square array by the same amount as the angular position of such loudspeaker relative to a reference direction is less than that of the corresponding loudspeaker in a square array and *vice versa.*

The invention is also applicable to other decoders in which the first and second difference signal components do not exist as discrete signals and further to systems in which the layout control means is operative on signals where the difference signal components do not exist as discrete signals even though such discrete signals are available elsewhere in the decoder.

The invention may further provide a decoder for a sound reproduction system having eight loudspeakers surrounding a listening area each located on one of the diagonals of a non-cubic cuboid between the point of intersection of said diagonals and a respective corner of said cuboid, said decoder comprising input means for receiving at least three input signals comprising complex linear combinations of pressure and velocity signal components and signal processing means for producing first, second and third difference signal components from said velocity signal components, said first difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a first face of the cuboid and the sum of the signals for the other four loudspeakers, said second difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a second face of the cuboid perpendicular to said first face and the sum of the signals for the other four loudspeakers and said third difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a third face of the cuboid perpendicular to both of said first and second faces and the sum of the signals for the other four loudspeakers, said decoder further comprising layout control means for applying first, second and third gains to said first, second and third difference signal components, the ratio between the first, second and third gains being inversely proportional to the ratio between the distances separating said first, second and third faces of the cuboid from their respective opposite faces, and output means response to said layout control means and said pressure signal components for producing a responsive output signal for each loudspeaker.

The pressure signal components may be omnidirectional signal components i.e. signal components in which sounds from all directions are captured with a constant gain. The velocity signal components may be phasor signal components, i.e. signal components in which sounds from all directions in a plane (e.g. all horizontal directions) are captured with a complex gain proportional to $\cos\theta \pm j \sin\theta$ for a sound arriving at an angle θ to a reference direction, the sign $\pm$ being the same for all sounds.

Some embodiments of the invention provide different treatment for frequencies above and below a particular frequency band. The need for this is fully discussed in M.A. Gerzon "Critères Psychoacoustiques Relatifs à la Réalisation des Systèmes Matriciels et Discrets en Tetraphonie" published in the 1974 Paris International

Festival du son "Journées d'Etudes", Editions Radio, Paris and in M. A. Gerzon "Surround-Sound Psychoacoustics" Wireless World, December, 1974 pages 483 to 486. Briefly, for frequencies appreciably less than the frequency for which the distance between the human ears is less than half a wavelength of sound in air (about 700 Hz), the head offers no obstacle to sound waves so that the amplitude of sound reaching the two ears is virtually identical. Consequently, the only information available at these low frequencies for sound localisation is the phase difference between the sounds received at the two ears. At higher frequencies, the phase relationship is no longer of primary importance in sound localisation; what matters is the directional behaviour of the energy field around the listener. There is a transitional band, referred to above as the predetermined frequency band, between these two conditions.

The transitional frequency may be within the range 100 Hz to 1000 Hz. Transitional frequencies at the lower end of the range give an increased listening area. A preferred value is about 320 Hz.

Embodiments of the invention will now be described by way of example with reference to the accompanying drawings, in which:—

Figure 1 is a schematic diagram of a sound reproduction system illustrating the disposition of the loudspeakers round a listening position and their connection to a decoder,

Figure 2 is a block diagram of a known decoder suitable for use in the system shown in Figure 1,

Figure 3 is a block diagram of a decoder for use in a sound reproduction system providing height information and employing eight loudspeakers,

Figure 4 is a schematic diagram illustrating the disposition of loudspeakers for use with the decoder shown in Figure 3,

Figure 5 shows a decoder in accordance with the invention including a layout control unit,

Figure 6 is a circuit diagram of a layout control unit for use in the decoder shown in Figure 5,

Figure 7 is a schematic diagram, similar to Figure 4, illustrating the layout of an eight loudspeaker cuboid array,

Figure 8 is a schematic diagram of a decoder in accordance with the invention for use with loudspeaker array shown in Figure 7,

Figure 9 is a block diagram of a frequency dependent decoder,

Figure 10 is a circuit diagram of a decoder of the type shown in Figure 9,

Figure 11 is a block diagram illustrating a decoder in accordance with the invention for use with discrete four channel signals,

Figure 12 is a block diagram of an alternative WXY circuit for use with the decoder of Figure 11, and

Figure 13 is a further alternative WXY circuit for use with the decoder of Figure 11.

It should be understood that, in the following description, where reference is made to a set of phase shifting circuits applying different phase shifts to different parallel channels, the phase shift specified in each case is a relative phase shift and a uniform additional phase shift may be applied to all channels if desired. Similarly, where it is specified that particular gains are applied to parallel channels, these gains are relative gains and a common additional overall gain may be applied to all channels if desired.

Before describing embodiments of the invention, it will be convenient to describe the basic form of a type of a decoder for use with rectangular loudspeaker layouts and the corresponding type for use with cuboid loudspeaker layouts. These two types of decoder are hereinafter referred to as WXY decoders and WXYZ decoders respectively. The invention may be applied to any decoder of these types.

Referring to Figure 1, a listening location centred on the point 10 is surrounded by four loudspeakers 11, 12, 13 and 14 which are arranged in a rectangular array. The loudspeakers 11 and 12 each subtend an equal angle θ at the point 10 relative to a reference direction indicated by an arrow 15. A loudspeaker 13 is disposed opposite the loudspeaker 11 and the loudspeaker 14 disposed opposite the loudspeaker 12. Thus, assuming that the reference direction is the forward direction, the loudspeaker 11 is disposed at the left front position, loudspeaker 12 at the right front position, the loudspeaker 13 at the right back position and the loudspeaker 14 at the left back position. All four loudspeakers 11 to 14 are connected to receive respective output signals LF, RF, RB and LB from the decoder 16 which has two input terminals 17 and 18, the received omni-directional signal W_1 being connected to the terminal 17 and the phasor signal P_1 to the terminal 18.

Figure 2 shows a known WXY decoder suitable for use as the decoder 16 when the angle $\theta=45°$. The decoder takes the form of a WXY circuit 20 and an amplitude matrix 22. The WXY circuit 20 produces an omni-directional output signal W, a front-back difference output signal X and a left-right difference output signal Y. These signals are then applied to the amplitude matrix 22 which produces the required output signals LB, LF, RF and RB.

The nature of the WXY circuit depends on the form of the input signals. If, as shown, the input signals comprise an omni-directional signal W_1 and a phasor signal P_1 of the same magnitude as the omni-directional signal but with a phase difference equal to minus the azimuth angle, the outputs of the WXY circuit 20 are related to its inputs as follows:

$$W = W_1$$

$$X = \frac{1}{\sqrt{2}} P_1$$

$$Y = \frac{1}{\sqrt{2}} jP_1$$

The amplitude matrix 22 fulfills the function of the following group of equations:—

$$LB=\tfrac{1}{2}(-X+W+Y)$$

$$LF=\tfrac{1}{2}(X+W+Y)$$

$$RF=\tfrac{1}{2}(X+W-Y)$$

$$RB=\tfrac{1}{2}(-X+W-Y)$$

In fact this decoder is the same as the decoder shown in Figure 5 of the above-mentioned United Kingdom Specification No. 1,369,813, the 90° phase shift circuits serving as the active part of the WXY circuit 20 and the adders and phase inverters serving as the amplitude matrix 22.

Any decoder which produces the four output signals LB, LF, RF and RB is the equivalent of a WXY circuit and an amplitude matrix, and thus constitutes a WXY decoder, provided that

$$\tfrac{1}{2}(-LB+LF-RF+RB)=0$$

The WXY circuit 20 may have more than two inputs.

A WXYZ decoder may be used in systems providing height information and employing eight loudspeakers disposed at respective corners of a cube. Referring to Figure 3, three input signals are applied to a WXYZ circuit 24 which produces output signals W, X and Y having the same significance as the corresponding signals of Figure 2 and an up-down difference signal Z. The signals W, X, Y and Z are applied to a type II amplitude matrix 26 which produces eight loudspeaker signals LBU, LFU, RFU, RBU, LBD, LFD, RFD and RBD, the signals being fed to loudspeakers located at the correspondingly referenced points in Figure 4. The construction of the WXYZ circuit 24 depends on the nature of the input signals. The output signals from the type II matrix 26 are related to the input signals as follows:—

$$LBU=\tfrac{1}{2}(-X+W+Y+Z)$$

$$LFU=\tfrac{1}{2}(X+W+Y+Z)$$

$$RFU=\tfrac{1}{2}(X+W-Y+Z)$$

$$RBU=\tfrac{1}{2}(-X+W-Y+Z)$$

$$LBD=\tfrac{1}{2}(-X+W+Y-Z)$$

$$LFD=\tfrac{1}{2}(X+W+Y-Z)$$

$$RFD=\tfrac{1}{2}(X+Y-Y-Z)$$

$$RBD=\tfrac{1}{2}(-X+W-Y-Z)$$

As for the two-dimensional case, any decoder is the equivalent of a WXYZ circuit and an amplitude matrix, and thus constitutes a WXYZ decoder, if the following equations are satisfied:—

$$(LBU+LBD)-(LFU+LFD)+(RFU+RFD)-(RBU+RBD)=0$$

$$(LBD+RBD)-(LFD+RFD)+(LFU+RFU)-(LBU+RBU)=0$$

$$(LBD+LFD)-(LBU+LFU)+(RBU+RFU)-(RBD+RFD)=0$$

$$(LBU-LBD)-(LFU-LFD)+(RFU-RFD)-(RBU-RBD)=0$$

Reverting to the loudspeaker arrangement and WXY decoder shown in Figures 1 and 2, in accordance with the invention, a layout control unit is provided to adjust the gains of the X and Y signals relative to the W signal to compensate for the non-square layout obtained when $\theta \neq 45°$. For example, when $\theta < 45°$ the gain for the front minus back signal has to be reduced for the increased front-back separation of loudspeakers and similarly, the gain of the left minus right signal Y has to be increased to compensate for the decreased side to side loudspeaker separation.

Referring to Figure 5, a layout control unit 28 is connected between the WXY circuit 20 and the type I amplitude matrix 22. The layout control unit 28 comprises gain adjustment devices 29 and 30 arranged to apply gain

$$\sqrt{2}\sin\theta$$

to the X signal and gain

$$\sqrt{2}\cos\theta$$

to the Y signal respectively to provide inputs W′, X′ and Y′ to the amplitude matrix 22.

One form of layout control unit 28 as shown in Figure 6. The gain control units 29 and 30 comprise respective inverting amplifiers 32 and 34, each of which has a feedback resistor R, an input resistor S and an output resistor T. The outputs X′ and Y′ of the gain control units 29 and 30 are also interconnected by a potentiometer U. The resistance R may have any convenient value and the resistance U may have any convenient value such that

$$U < \sqrt{2}L$$

where L is the input impedance of the amplitude matrix 22 for all input signals. Then, if

$$T = \frac{UL}{\sqrt{2}\,L - U}$$

and

$$S = \frac{\sqrt{2}\,L - U}{(2+\sqrt{2})L}$$

the gains for the X and Y signals are a good approximation to

$$\sqrt{2}\sin\theta$$

and

$$\sqrt{2}\cos\theta$$

respectively when θ is in the range 0° to 90°. In practice, it is preferable to keep θ within the range of about 25° to 65° since, outside this range, the angle subtended at the listening position by two of the pairs of adjacent loudspeakers become inconveniently large. This range may be limited by connecting fixed resistors in series with the potentiometer U and reducing the resistance of the potentiometer so that the overall resistance remains the same.

The W input signal to the layout control unit 28 is also connected to the W' output thereof by an inverting amplifier 35 having feedback and input resistors of equal value R, thus matching the phase inversion introduced to the X and Y signals by the variable gain circuits.

It should be appreciated that changing the relative amplitudes of the X and Y signals has exactly the same effect as changing the phase of the phasor signal P_1 relative to the omni-directional signal W_1.

The above gains of

$$\sqrt{2}\sin\theta$$

in the X signal path and

$$\sqrt{2}\cos\theta$$

in the Y signal path are first order approximation to ideal gains. Better approximations are obtained if the gains are of the form

$$\sqrt{2}\,k\sin\theta \quad\text{and}\quad \sqrt{2}\,k\cos\theta$$

respectively. At frequencies below about 500 Hz, the preferred form of k is given by

$$k=\frac{1}{\sin 2\theta}=\frac{1}{2\sin\theta\cos\theta}$$

which is approximately equal to 1 where θ equals 45°. At higher frequencies, the preferred value is k=1. If, as described above, these gains are not frequency dependent, the choice of k=1, as described above, is satisfactory at all frequencies.

Similar techniques may be used in conjunction with a WXYZ decoder for an eight loudspeaker cuboid array. In order to provide a decoder for the array shown in Figure 7, the decoder shown in Figure 3 is modified as shown in Figure 8 by inserting a layout control unit 36 comprising gain adjustment devices 38, 40 and 42 for the X, Y and Z channels respectively, between the WXYZ circuit 24 and the type II amplitude matrix 26. The approximate optimal gains for frequencies above and below 500 Hz are shown in the Table I.

Table I

channel	high frequency gain	low frequency gain
X	$\frac{\sqrt{3}\,ac}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\frac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,b}$
Y	$\frac{\sqrt{3}\,bc}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\frac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,a}$
Z	$\frac{\sqrt{3}\,ab}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\frac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,c}$

As for the rectangular decoder, if the gains are to be frequency independent, the value shown for high frequencies may be used. These values are equivalent to the values shown in Table II.

Table II

channel	gain
X	$\sqrt{3}\ \sin\theta$
Y	$\sqrt{\frac{3}{2}}\cos\theta\ .\ \sqrt{2}\ \sin\phi$
Z	$\sqrt{\frac{3}{2}}\cos\theta\ .\ \sqrt{2}\cos\phi$

where

$$b:a:c=\frac{1}{\sin\theta}:\ \frac{1}{\cos\theta\sin\phi}:\ \frac{1}{\cos\theta\cos\phi}$$

The gain adjustment devices 38, 40 and 42 may be implemented in a similar manner to the gain adjustment devices 29 and 30 of Figure 6, the gain adjustment devices 40 and 42 each comprising two stages in cascade, one with gain

$$\sqrt{\frac{3}{2}}\cos\theta$$

and the other with gain

$$\sqrt{2}\ \sin\phi$$

for the device 40 and

$$\sqrt{2}\ \cos\phi$$

for the device 42.

The three input signals to the WXYZ circuit 24 of Figure 8 may consist of linear combinations of the signals W_4, Y_4 and V_4 where W_4 is an omni-directional signal that picks up all sound directions with identical gain, Y_4 is a signal resulting from picking up a sound with gain $\sqrt{3}$ y and V_4 is a signal resulting from picking up a sound with directional gain $\sqrt{3}$ (x−qjz), where q is a real constant, and (x, y, z) are the sound directions. Then the outputs of the WXYZ circuit 24 are related to its inputs as follows:—

$$W=W_4$$

$$X=fV_4$$

$$Y=fY_4$$

$$Z=fjd^{-1}V_4$$

where f is a real constant. Ideally at low frequencies f=1; ideally at mid-high frequencies,

$$f=\frac{1}{\sqrt{3(1+q^{-2})}}$$

It is clear that by interchanging axes, other encoding systems may be obtained. For example, one might consider the signals with directional gains 1, x−jqy, z or 1, x, y−jqz. The corresponding decoders are obtained by exchanging the signal paths accordingly.

The decoders described above do not make special provision for the different mechanisms by which the human ears localise sounds above and below about 700 Hz. Decoders which do take into account these differences employ frequency dependent matrices approximating to an "ideal" low frequency design at low freqeuncies and an "ideal" high frequency design at high frequencies. There is also a transition region of frequencies in which the decoder matrix has an intermediate form. Theoretically, the centre of this transition region should be about 700 Hz. It has been found that, in practice satisfactory results can be obtained if the centre of this transition region is within the range of 100 Hz to 1000 Hz but that good listening conditions away from the centre of the listening area are best obtained if the centre of this region is below 700 Hz and a value of 320 Hz has been found to be particularly suitable.

It has been found that there are four localisation criteria. Two of these criteria relate to voltage gain and are dominant at low frequencies. The other two criteria relate to the energy gain to which the signal is subject and are dominant at high frequencies. The symbols LB_V, LF_V, RF_V and RB_V represent the complex voltage gains that a monophonic sound in some direction is subjected to when passed through the entire system, i.e., the original encoder and the decoder to feed the four loudspeakers shown in Figure 1. Then, for a sound for which the desired apparent azimuth angle is ϕ, the more important low frequency condition, known as the Makita condition, is that the quantities x and y given by

$$x = \mathrm{Re}\left(\frac{LF_V + RF_V - LB_V - RB_V}{LF_V + RF_V + LB_V + RB_V}\right)$$

$$y = \mathrm{Re}\left(\frac{LF_V + LB_V - RF_V - RB_V}{LF_V + RF_V + LB_V + RB_V}\right)$$

must be expresssible in the form

$$x \cos \theta = r \cos \phi$$

$$y \sin \theta = r \sin \theta$$

where r is a positive number. The symbol "Re" means "the real part of". If this condition is satisfied, the correct apparent direction of the sound is obtained at low frequencies. However, unless a second condition, known as the velocity condition is also satisfied, the apparent direction of the sound tends to be unstable when the listener moves his head. The velocity condition is

$$(x \cos \theta)^2 + (y \sin \theta)^2 = 1$$

At higher frequencies, above the transition frequency, the most important condition is the so-called energy vector condition that the quantities x_E and y_E given by

$$x_E = \frac{|LF_V|^2 + |RF_V|^2 - |LB_V|^2 - |RB_V|^2}{|LF_V|^2 + |RF_V|^2 + |LB_V|^2 - |RB_V|^2}$$

$$y_E = \frac{|LF_V|^2 + |LB_V|^2 - |RF_V|^2 - |RB_V|^2}{|LF_V|^2 + |RF_V|^2 + |LB_V|^2 + |RB_V|^2}$$

must be expressible in the form

$$x_E \cos\theta = r_E \cos\phi$$

$$y_E \sin\theta = r_E \sin\phi$$

where r_E is a positive number. This determines correct localisation but, if the apparent direction of sound at higher frequencies is to be stable when the listener moves his head, it is in addition necessary, in accordance with the energy magnitude condition for the quantity

$$(x_E \cos\theta)^2 + (y_E \sin\theta)^2$$

to be as large as possible for all directions. In practice, it may be necessary to sacrifice the magnitude of this quantity for some directions in order to improve it in others. The quantity can, of course, never exceed 1.

The Makita condition and the energy vector condition, which determine the basic sound directions at low and high frequencies respectively, are the most important. Since it is not clear precisely which of these theories is more important in the region of frequencies around the transition frequencies, it is important that both conditions are satisfied in this region. It can be shown mathematically that any WXY decoder or WXYZ decoder which satisfies either the Makita condition or the energy vector condition automatically satisfies both conditions. Thus, a WXY decoder or a WXYZ decoder designed to satisfy, for example, the Makita condition at all frequencies will give correct sound localisation at all frequencies. This applies to the decoders described above. In order to improve the stability of apparent sound direction as a listener's head moves, it is necessary to satisfy the velocity condition at lower frequencies and the energy magnitude condition at higher frequencies. This involves the use of frequency dependent decoders.

Figure 9 shows a decoder similar to that shown in Figure 5 but modified to provide the required frequency dependence. Two identical shelf filters 44 and 46, of type I are connected in the X and Y signal paths respectively. A shelf filter 48 of type II is connected in the W signal path. The filters 44, 46 and 48 are filters with substantially identical phase responses and each having one gain at low frequencies, below a transition frequency, another gain at high frequencies above such transition frequency, and which smoothly make the transition from low frequency gain to the high frequency gain across a frequency band around the transition frequency. When, as shown, the input to the decoder takes the form of an omni-directional signal W_1 and a phasor signal P_1, the relative gains of all the shelf filters, 44, 46 and 48 are 1 at frequencies above the transition frequency band in order to give optimum high frequency reproduction according to the energy magnitude condition. At frequencies below the transition frequency band, the gains of the shelf filters I relative to that of the shelf filter II are

$$\frac{2}{\sin 2\theta}$$

which is approximately equal to 2 when θ is in the range 30° to 60°. Consequently, it is satisfactory if the type I shelf filters have twice the gain of the type II shelf filter at frequencies below the transition frequency band.

A particular decoder circuit of this type is illustrated in Figure 10. In order to reduce the number of components required, the shelf filters and layout control are located before a modified WXY circuit 50. This means that a single type I shelf filter 52 is connected in the phasor signal path in place of the two type I shelf filters 44 and 46 in the X and Y signal paths respectively. The layout control unit 28 provides two phasor inputs to the WXY circuit 50 which comprises two 0° phase shift circuits 54 and 56 and one 90° phase shift circuit 58.

The shelf filter 48 is required to have a complex frequency response given by:—

$$\frac{\sqrt{a_1 b_1}\left(\sqrt{\dfrac{a_1}{b_1}} + j\omega T_1\right)}{1 + j\sqrt{\dfrac{a_1}{b_1}}\,(\omega T_1)} \times \frac{1 - j\omega T_2}{1 + j\omega T_2}$$

where a_1 is the low frequency gain and b_1 is the high frequency gain. This filter consists of an amplifier 60 connected to a capacitance resistance network comprising resistances R_1, R_2 and R_3 and capacitance C_1. In turn, this is connected to a parallel circuit having amplifier 62 and capacitor C_2 in one branch and amplifier 64 and resistance R_4 in the other branch. For a transition frequency of 200 Hz, the variables in the expression for frequency response and the circuit components have the values indicated in Table III.

TABLE III

a_1	0.6325
b_1	1
T_1	946.3 μsecs.
T_2	838.8 μsecs.
gain of 60	1.2649
gain of 62	–1
gain of 64	1
R_1	0.1325 R_O
R_2	0.3675 R_O
R_3	0.5 R_O
R_O C_1	3237 μsecs.
R_4 C_2	T_2

The values of R_0 and R_4 are chosen arbitrarily according to design convenience.

The shelf filter 52 for the phasor signal P has the following complex frequency response:—

$$\frac{\sqrt{a_3 b_3}\left(\sqrt{\dfrac{a_3}{b_3}} - j\omega T_3\right)}{1 + j\sqrt{\dfrac{a_3}{b_3}}\,\omega T_3}$$

where a_3 is the low frequency gain and b_3 is the high frequency gain. This filter consists of two parallel paths, one consisting of an amplifier 66 and a resistor R_5 and the other consisting of an amplifier 68 and a capacitor C_3. The values of the various circuit components are shown in Table IV.

TABLE IV

a_3	$2a_1$
b_3	b_1
T_3	669.2 μsecs.
gain of 54	1.2649
gain of 56	−1
R_5 R_3	752.6 μsecs.

The value of the resistance R_5 is chosen arbitrarily according to design convenience.

The layout control unit 28 consists of an amplifier 70 of gain 1.707, two fixed resistances 72 and 74 in series with outputs to the two phase shift circuits 56 and 58 in the WXY circuit 50 and a chain formed by fixed resistances 76 and 78 and a potentiometer 80 connected in parallel with the two outputs of the network. The moving contact of the potentiometer 80 is connected to earth. The two resistances 76 and 78 in series with the potentiometer each have resistance values equal to half that of the potentiometer 80. The two series resistances 72 and 74 each have resistance value equal to 1.414 times the resistance of the potentiometer 80. The amplifier 60 ensures that the sum of the energies at the two ouputs of the layout control unit 28 is effectively equal to the energy at the input thereof.

The circuit shown in Figure 10 also includes a high pass filter 82 in the input path for the signals P_1. The high pass filter 82 consists of a capacitor 84 and a potentiometer 86. The purpose of this filter is to compensate for the effect at the listening position due to the distance between the loudspeakers and a central listener. The effect of a finite lowspeaker distance is to produce a bass boost and phase shift in the low frequency components of the velocity of the sound field at the listener and this, in turn, can degrade the image quality and may in some circumstances cause errors in the location of sound images at both frequencies.

In use, the setting of the potentiometer 86 is adjusted so that the time constant of the filter is equal to the time taken for sound to travel from any of the loudspeakers 11 to 14 to the centre point 10 of the listening area (Figure 1). The potentiometer 86 preferably has an associated scale calibrated in distance to facilitate this setting.

It should be noted that, as illustrated in Figure 1, the loudspeakers 11 to 14 are preferably equidistant from the centre point 10. If it is necessary for the distances of the various loudspeakers from the centre point 10 to differ from one another, the amplitude gains of the signals for the more distant loudspeakers are increased until a subjectively satisfactory result is obtained.

Similar compensation for the different localisation mechanisms used by the human ear at low and high frequencies may be applied to WXYZ decoders, respective type I shelf filters being connected in the X, Y and Z channels and a type II shelf filter in the W channel. Where the input signal is a four channel signal consisting of four linear combinations of an omni-directional signal and three signals resulting from pickup sound from an arrival direction given by direction cosines (x, y, z) with respective direction gains $\sqrt{3}$ x, $\sqrt{3}$ y and $\sqrt{3}$ z, the low and high frequency gains of these shelf filters are as follows:—

Filter	Low frequency gain	High frequency gain
I	1	$\sqrt{2/3}$
II	1	$\sqrt{2}$

Figure 11 illustrates a decoder in accordance with the invention for use with so-called "discrete" or "pairwise mixed" four channel signals. Such four channel signals assign sounds to a horizontal direction between the azimuths of two adjacent loudspeakers of a square layout by feeding them to both channels corresponding to adjacent speakers with the same phase but differing intensities thus, there are four input channels LB_1, LF_1, RF_1 and RB_1. For an azumith ϕ from the front direction, the gains of the signals in the four input channels are shown in Table V.

TABLE V

	$-45° \leqslant \phi \leqslant 45°$	$45° \leqslant \phi \leqslant 135°$	$135° \leqslant \phi \leqslant 225°$	$-135° \leqslant \phi \leqslant -45°$
LB_1	0	$\cos (135°-\phi)$	$\sin (225°-\phi)$	0
LF_1	$\cos (56°-\phi)$	$\sin (135°-\phi)$	0	0
RF_1	$\sin (45°-\phi)$	0	0	$\cos (-45°-\phi)$
RB_1	0	0	$\cos (225°-\phi)$	$\sin (-45°-\phi)$

Such an encoding specification is in common use. It may be decoded using a WXY decoder as shown in Figure 11. The WXY circuit 88 thereof comprises a type III amplitude matrix 90 in the form

$$X_2=\tfrac{1}{2}(-LB_1+LF_1+RF_1-RB_1)$$

$$Y_1=\tfrac{1}{2}(LB_1+LF_1-RF_1-RB_1)$$

$$W_2=\tfrac{1}{2}(LB_1+LF_1+RF_1+RB_1)$$

$$F=\tfrac{1}{2}(-LB+LF_1-RF_1+RB_1)$$

The difference outputs X_2 and Y_2 of the amplitude matrix 90 are connected via respective. 0° phase shift circuits 92 and 94 to provide the X and Y outputs. The pressure signal output W_2 is connected via a 0° phase shift circuit 96 and the diagonal difference output F via a 90° phase shift circuit 98 to a proportional adder 100 which applies gain 0.707 to the W_2 input, gain 0.455 to the jF input and then sums these two signals to provide the W output. The X and Y signals are applied to type I shelf filters 102 and 104 similar to the shelf filter 52 shown in Figure 12 but having unity gain at low frequencies and $\sqrt{\tfrac{3}{4}}$ gain at high frequencies. The W signal is applied to a type II shelf filter 106 which is similar to the shelf filter 48 of Figure 10 but having unity gain at low frequencies and $\sqrt{3/2}$ gain at high frequencies. The outputs of the shelf filters 102 and 104 are connected to variable high pass filters 108 and 110 which are identical with the high pass filter 82 of Figure 10 and have the control of their potentiometers ganged. These filters 108 and 110 provide compensation for loudspeaker proximity as described with reference to Figure 10. The outputs of the filters 108 and 110 are then connected to a layout control unit 112. The layout control unit 112 comprises a pair of input amplifiers 114 and 116, each having gain 2.414 and having their ouputs connected to the outputs of the layout control unit by equal resistors 118 and 120. A resistance chain, consisting of resistor 122, potentioemter 124 and resistor 126 is connected between the ouputs of the distance control unit. The relationship between the resistance values of the potentiometer 124 and the various resistors is as stated in Table VI where S may have any convenient value.

TABLE VI

Component	Resistance
118	0.707 S
120	0.707 S
122	0.25 S
124	0.50 S
126	0.25 S

The use of the resistors 122 and 126 in series with the potentiometer 112 limits the range of adjustment of the layout control so that over which satisfactory results can be achieved as described above with reference to Figure 6.

The decoder illustrated in Figure 11 may also be used as a four loudspeaker decoder for conventional stereo recordings by connecting the two stereo channels L and R to the inputs LF_1 and RF_1 respectively and grounding the other two units LB_1 and RB_1. Such stereo material is thus treated as four channel pairwise mixed material for which all sounds originate in the quadrant $-45°$ to $+45°$.

A decoder in accordance with the invention may be used to decode signals from the TMX three channel system in which the input system to the decoders consists of three channels as follows:—

$$L=\tfrac{1}{2}(W_3+jP_3)$$

$$R=\tfrac{1}{2}(W_3-jP_3)$$

$$T_T=jP_3^*$$

where P_3^* is a signal whose azimuthal gain is the complex conjugate of that of P_3, as described in D. H. Cooper, T. Shiga and T. Takagi "QMX Carrier Channel Disc" Journal of the Audio Engineering Society, Volume 21, Pages 614 to 624, October, 1973. The WXY circuit 88 of Figure 11 is replaced by a WXY circuit as shown in Figure 12. The L and R input signals are connected to a type IV matrix 110 of the form:—

$$W_3=L+R$$

$$jP_3=L-R$$

The W_3 output of the matrix 130 is connected via a 0° phase shift circuit 132 to form the W output of the WXY circu it. The jP_3 output of the matrix 130 is connected both to a 0° phase shift 134 and to a $-90°$ phase shift circuit 136. Similarly, the T_T input signal from the TMX source is connected both to a $-90°$ phase shift circuit 138 and a $-180°$ circuit 140. The outputs of the two $-90°$ phase shift circuits 136 and 138 are added together, each with gain 0.707 in a proportional adder 142, the output of which forms the X output of the WXY circuit. Similarly, the outputs of the 0° phase shift 134 and the $-180°$ phase shift 140 are added together, both with gains 0.707 in a proportional adder 144, the output of which forms the Y output of the WXY circuit.

A decoder in accordance with the invention can also be used for the QMX system as described in D. H. Cooper, T. Shiga and T. Takagi, "QMX Carrier Channel Disc". The QMX disc system incorporates TMX signals in which the T_T signal is of restricted band width and is therefore not available above about 6 kHz. In a decoder for this system, the WXY circuit shown in Figure 12 is replaced by a WXY circuit as shown in Figure 13. It will be seen that this circuit differs from the circuit of Figure 12 in that the W and jP outputs of the type IV matrix 130 are passed through an all-pass filter 146 and a type III shelf filter 148 and the T_T input is passed through a low pass filter 150 with a cut-off frequency of about 2kHz. The all-pass filter 146, the shelf filter 148 and the low pass filter 150 all have substantially the same phase response and all have unity gain at well below 2 kHz. The shelf filter 148 has gain $\sqrt{2}$ at high frequencies and a transition frequency equal to the -6 dB frequency of the low pass filter 150.

The low pass filter 150 comprises two identical resistor-capacitor low pass filters in cascade, the all-pass filter 146 is a resistor-capacitor all-pass filter of the same time constant as the low pass filter 150 and the shelf filter 148 is a resistor-capacitor shelf filter followed by a phase-compensating all-pass filter of a similar design to those used for the type II shelf filter 48 in Figure 10.

In the case of two-input WXY circuits, the input signals need not be the actual omni-direction input signal W_1 and the phasor input signal P_1. Any non-singular linear combination thereof may be used with a suitably modified WXY circuit. The signals Q and R which are related to the signals W and P as follows:—

$$Q=\alpha W_1+\beta P_1$$
$$R=\beta^* W_1+\alpha^* P_1$$

where α an β are complex numbers and α^* and β^* their respective complex conjugates, may be used instead of the signals W_1 and P_1. This is because any such signals have equal amplitude but differing phase.

A decoder in accordance with the invention may also be used to decode inputs which may be regarded as consisting of two signals W_4 and P_4. W_4 is an omni-directional signal with unit gain in all directions and P_4 is a signal with gain

$$m \cos \phi - j \sin \phi$$

where ϕ is the azimuth angle from the front and m is real. When $m=1$, the signal P_4 is, of course a phasor signal. Inputs in the form of signals W_4 and P_4 can be decoded by a WXY circuit in accordance with the following equations:—

$$W=W_4$$

$$X=\frac{1}{m\sqrt{2}}P_4$$

$$Y=\frac{1}{\sqrt{2}}jP_4$$

The encoding systems known as "BBC matrix G" and "BBC matrix H", described in British Broadcasting Corporation Research Department, Engineering Division Report BBC RD 1974—29, "The Subjective Performance of Various Quadraphonic Matrix Systems" November, 1974, produce signals L and R corresponding to the stereo "left" and "right" signals. It can be shown that the signals L and R may be regarded as linear combinations of the signals W_4 and P_4 as follows:—

$$W_4=\gamma L+\gamma^* R$$

$$P_4=\delta L+\delta^* R$$

where γ and δ are non-zero complex numbers of modulus 1 and γ^* and δ^* are their complex conjugates. The signals W_4 and P_4 can then be decoded by the above-described WXY circuit with m approximately equal to 0.68.

In all the embodiments of the invention described above, the signals W′, X′ and Y′, or W′, X′, Y′ and Z′ have been produced as discrete signals and applied to either a type I or a type II amplitude matrix respectively. It should be understood that the invention is also applicable to systems in which these signals do not have a separate discrete existance but take the form of linear combinations of one another, the output signals to the loudspeakers being produced directly from such linear combinations.

Where it is possible to interchange the positions of circuits or to combine circuits without changing the overall function, such modifications are within the scope of the invention. For example, if two successive circuits can be expressed mathematically as respective matrices, then they be replaced by a single circuit which can be represented mathematically by the product of the two matrices.

It should also be understood that, at any point in the systems described, additional amplifiers may be inserted to provide such overall gain as is considered necessary or desirable by one skilled in that art. In particular, the outputs to the various loudspeakers will usually be connected to their respective loudspeakers via power amplifiers.

In all embodiments of the invention, there may be additional direct signal paths between the WXY circuit or the WXYZ circuit and the amplitude matrix providing the loudspeaker signals. For example, in the Figure 9 embodiment, a fourth signal path F may be added directly connecting the WXY circuit 20 to the amplitude matrix 22 which is then arranged to produce output signals as follows:—

$$LB=\tfrac{1}{2}(-X'+W'+Y'-F)$$

$$LF=\tfrac{1}{2}(X'+W'+Y'+F)$$

$$RF=\tfrac{1}{2}(X'+W'-Y'-F)$$

$$RB=\tfrac{1}{2}(-X'+W'-Y'+F)$$

which is as before if the F signal is 0. The addition of the F signal path will not affect the overall directional effect of the decoder provided that F is $\pm$ 90° out of phase with respect to X' and Y' for all directions.

A decoder in which the ratio of the gains applied to pressure and velocity signal components changes with frequency, as described above, is claimed in our copending Application No. 2685/77 (Serial No. 1 494 752).

WHAT WE CLAIM IS:—

1. A decoder for a sound reproduction system having four loudspeakers surrounding a listening area each located on one of the diagonals of a non-square rectangle between the point of intersection of said diagonals and a respective corner of said rectangle, said decoder comprising input means for receiving at least two input signals comprising complex linear combinations of pressure signal components and velocity signal components representative of the pressure and of velocity at a listening position and signal processing means for producing first and second difference signal components from said velocity signal components, said first difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for a first adjacent pair of said loudspeakers and the sum of the signals for a second adjacent pair comprising the other two loudspeakers and said second difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for a third adjacent pair of loudspeakers comprising one loudspeaker from each of said first and second adjacent pairs and the sum of the signals for a fourth adjacent pair of loudspeakers comprising the other loudspeaker from each of said first and second adjacent pairs of loudspeakers, said decoder further comprising layout control means for applying first and second gains to said first and second difference signal components, the ratio between the first and second gains being substantially equal to the ratio between the sine of half the angle between the diagonals on which said first pair of loudspeakers are located and the sine of half the angle between the diagonals on which said third pair of loudspeakers are located, and output means responsive to said layout control means and said pressure signal components for producing a responsive output signal for each loudspeaker.

2. A decoder as claimed in claim 1, in which said output means comprises an amplitude matrix.

3. A decoder as claimed in claim 1 or 2, in which said layout control means comprises means for producing a signal at a first output consisting of said first difference signal components, means for producing a signal at a second output consisting of said second difference signal components, and a resistance having an earthed intermediate tapping connected between said first and said second outputs, whereby the ratio of the resistance between the intermediate tapping and the first ouput to the resistance between the intermediate tapping and the second output determines the ratio between the first and second gains.

4. A decoder as claimed in claim 1, 2 or 3, in which the input means comprises means for producing from said input signals, a pressure signal, a first difference signal and a second difference signal.

5. A decoder as claimed in claim 4, in which the input means comprises an amplitude matrix responsive to four-channel pairwise mixed input signals to produce the pressure signal, the first and second difference signals and a diagonal difference signal and means for applying a 90° phase shift to said diagonal difference signal and adding said phase shifted diagonal difference signal to said pressure signal.

6. A decoder as claimed in claim 4, in which the input means comprises an amplitude matrix responsive to first and second input signals which comprise the sum and the difference of an omni-directional signal component and a phasor signal component respectively, said amplitude matrix being arranged to produce an omni-directional output and a phasor output, said input means also having a third input for receiving a signal comprising the complex conjugate of the phasor signal component, means for subtracting the third input signal from the phasor output of the matrix to form said first difference signal and phase shift means for applying respective 90° phase shifts to the phasor output of the matrix and the third input signal and means for adding said phase shifted signals to form said second difference signal.

7. A decoder as claimed in claim 6, in which the third input is connected to its phase shift means via a low pass filter and the phasor output of the matrix is connected to its phase shift means and the subtraction means via a shelf filter having transition frequency substantially equal to the cut off frequency of the low pass filter and a higher gain above the transition frequency than below the transition frequency.

8. A decoder as claimed in claim 1, 2 or 3, in which the input means is arranged to supply a phasor signal to the layout control means, said layout control means being arranged to apply a first gain to the phasor signal to produce a first output comprising said first difference signal components and to apply a second gain to said phasor signal to produce a second output and means for applying a 90° phase shift to said second output to produce a signal comprising said second difference signal components.

9. A decoder for a sound reproduction system having eight loudspeakers surrounding a listening area each located on one of the diagonals of a non-cubic cuboid between the point of intersection of said diagonals and a respective corner of said cuboid, said decoder comprising input means for receiving at least three input signals comprising complex linear combinations of pressure and velocity signal components and signal processing means for producing first, second and third difference signal components from said velocity signal components, said first difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a first face of the cuboid and the sum of the signals for the other four loudspeakers, said second difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a second face of the cuboid perpendicular to said first face and the sum of the signals for the other four loudspeakers and said third difference signal components being dependent on the required uncompensated difference in signal strength between the sum of the signals for the four loudspeakers adjacent to the corners of a third face of the cuboid perpendicular to both of said first and second faces and the sum of the signals for the other four loudspeakers, said decoder further comprising layout control means for applying first, second and third gains to said first, second and third difference signal components, the ratio between the first, second and third gains being inversely proportional to the ratio between the distances separating said first, second and third faces of the cuboid from their respective opposite faces, and output means responsive to said layout control means and said pressure signal components for producing a responsive output signal for each loudspeaker.

10. A decoder as claimed in claim 9, in which the input means comprises means for producing from said input signals, a pressure signal, a first difference signal, a second difference signal and a third difference signal.

11. A decoder as claimed in any preceding claim, in which said velocity signal components are passed through high pass filter means, the time constant of said filter being substantially equal to the time of travel of sound from the loudspeakers to the centre of the listening area.

12. A decoder for a sound reproduction system as claimed in claim 1, substantially as hereinbefore described with reference to Figure 5 and 6, Figures 7 and 8, Figure 11, Figure 12 or Figure 13 of the accompanying drawings.

13. A decoder for a sound reproduction system as claimed in claim 1, substantially as hereinbefore described with reference to Figure 10 of the accompanying drawings.

14. A sound reproduction system including a decoder as claimed in any preceding claim.

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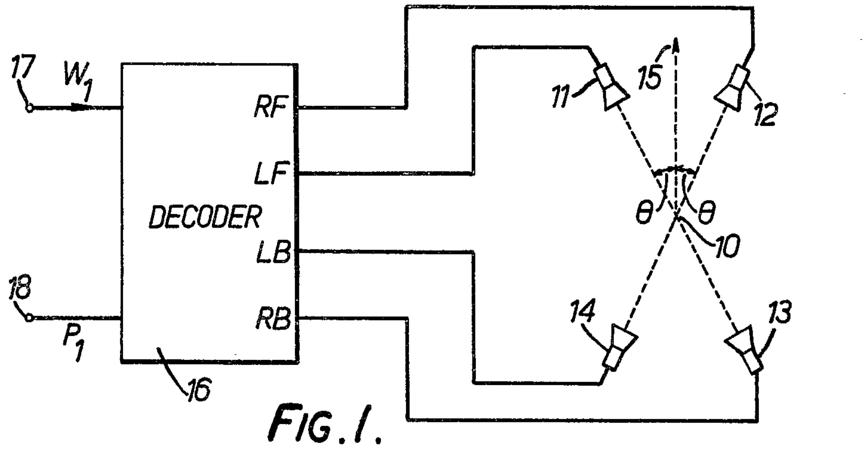


FIG. 1.

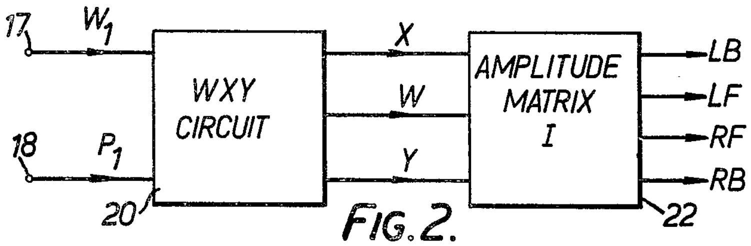


FIG. 2.

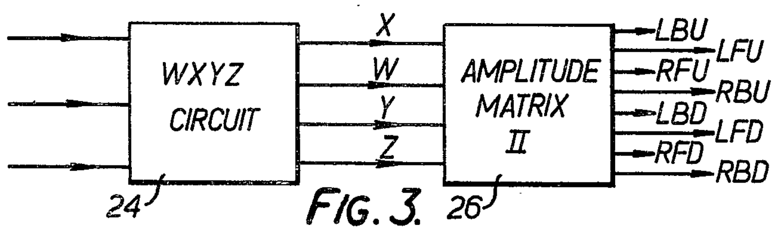


FIG. 3.

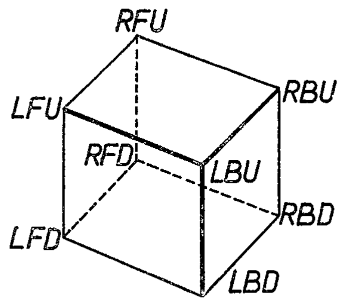


FIG. 4.

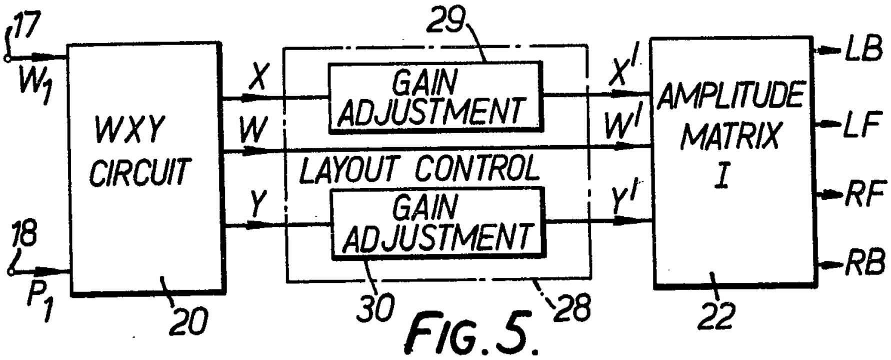


FIG. 5.

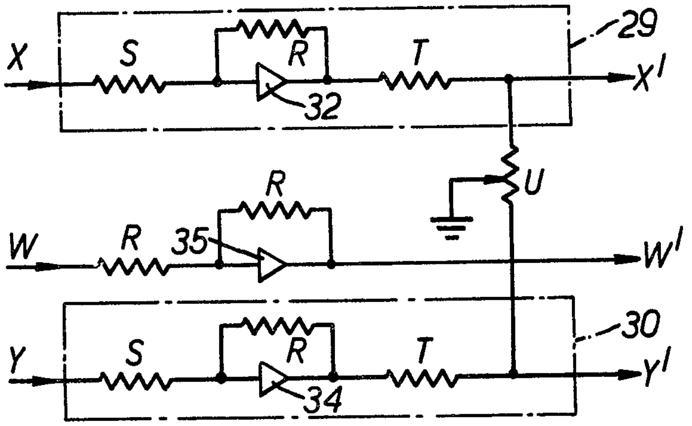


FIG. 6.

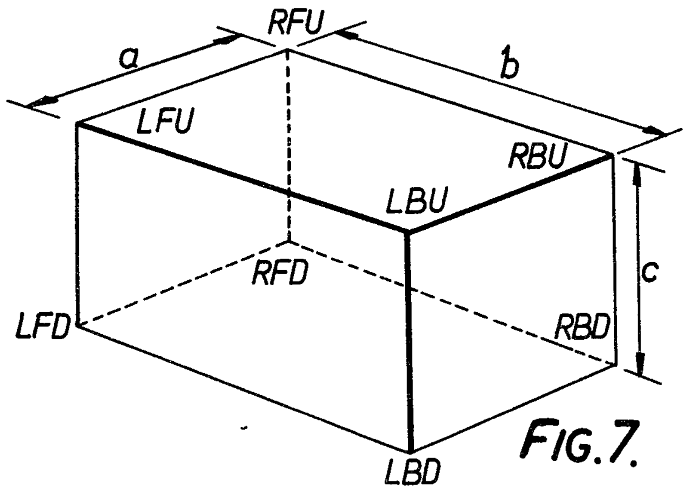


FIG. 7.

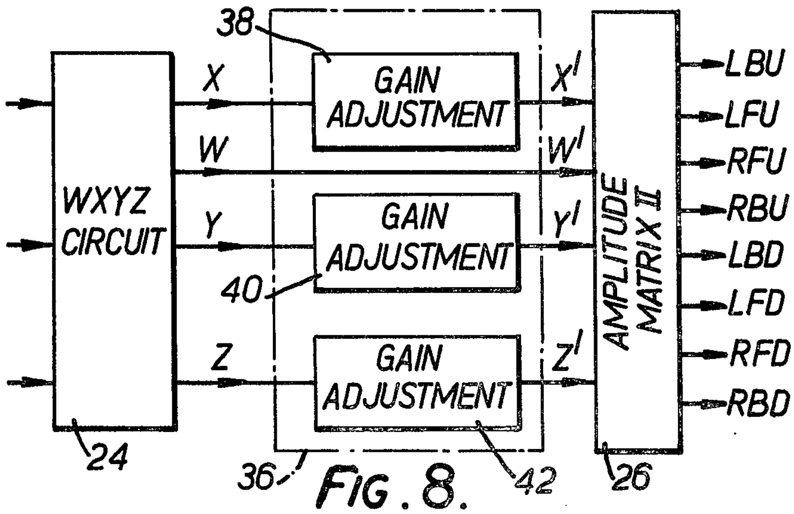

38
WXYZ CIRCUIT
X
W
Y
Z
GAIN ADJUSTMENT
GAIN ADJUSTMENT
GAIN ADJUSTMENT
40
X'
W'
Y'
Z'
AMPLITUDE MATRIX II
LBU
LFU
RFU
RBU
LBD
LFD
RFD
RBD
24
36
42
26

FIG. 8.

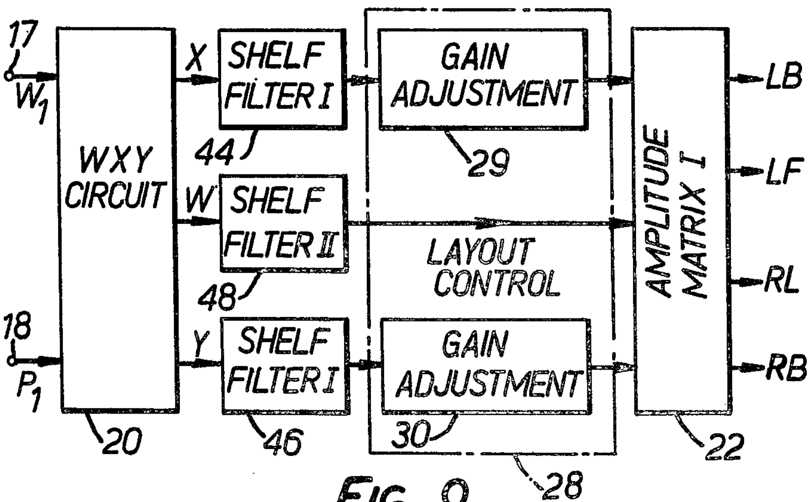
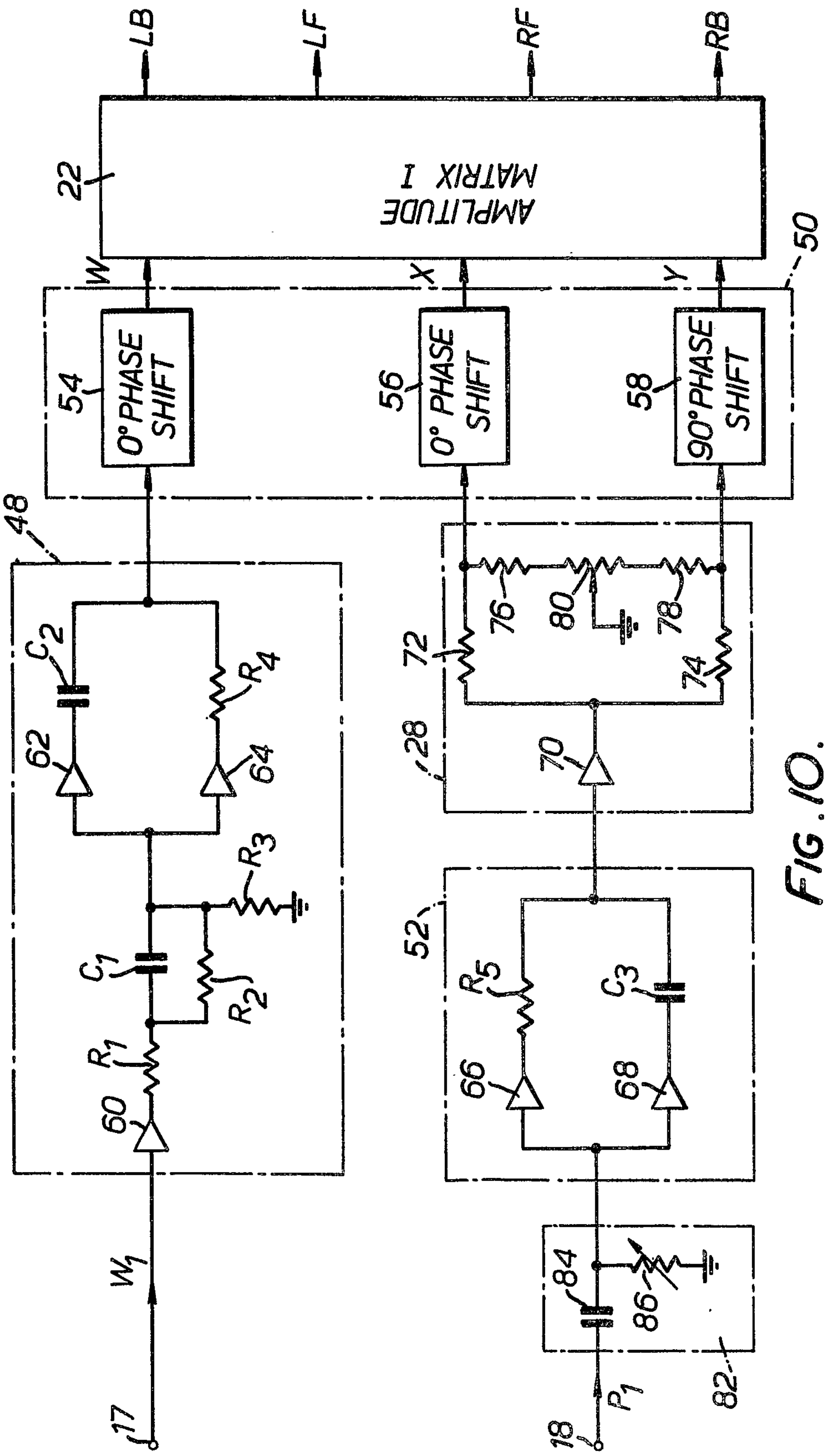

17
W_1
18
P_1
WXY CIRCUIT
X
W
Y
SHELF FILTER I
SHELF FILTER II
SHELF FILTER I
44
48
46
GAIN ADJUSTMENT
GAIN ADJUSTMENT
LAYOUT CONTROL
29
30
AMPLITUDE MATRIX I
LB
LF
RL
RB
20
22
28

FIG. 9.


LB
LF
RF
RB
AMPLITUDE MATRIX I
22
W
X
Y
50
0° PHASE SHIFT
54
0° PHASE SHIFT
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90° PHASE SHIFT
58
48
C2
R4
62
64
R3
C1
R2
R1
60
W1
17
72
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52
R5
C3
66
68
84
86
82
P1
18

FIG. 10.

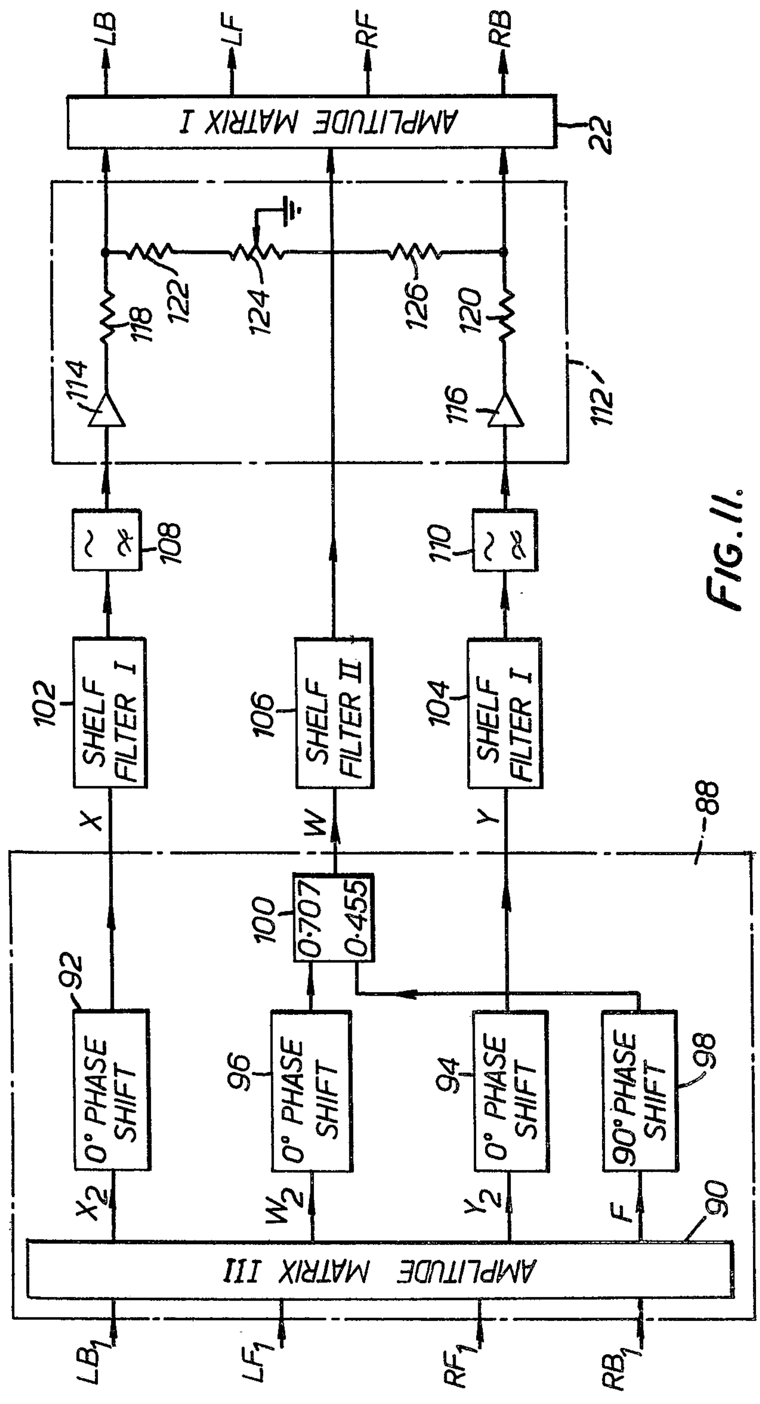
LB
LF
RF
RB
AMPLITUDE MATRIX I
22
118
122
124
126
120
114
116
112
108
110
SHELF FILTER I
SHELF FILTER II
SHELF FILTER I
102
106
104
X
W
Y
88
100
0·707
0·455
92
0° PHASE SHIFT
96
0° PHASE SHIFT
94
0° PHASE SHIFT
90° PHASE SHIFT
98
X2
W2
Y2
F
90
AMPLITUDE MATRIX III
LB1
LF1
RF1
RB1

FIG. 11.

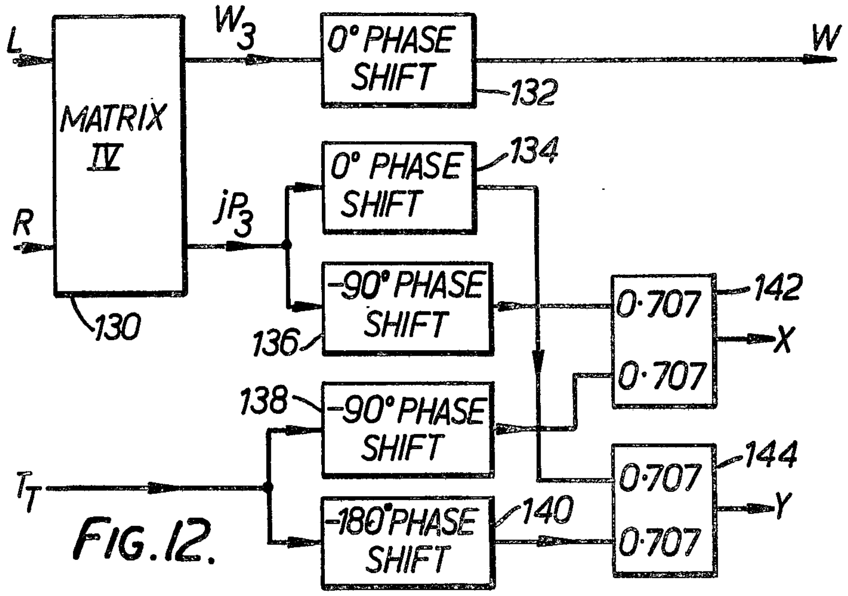
L
R
MATRIX IV
130
W_3
jP_3
0° PHASE SHIFT
132
W
0° PHASE SHIFT
134
-90° PHASE SHIFT
136
0·707
0·707
142
X
T_T
138
-90° PHASE SHIFT
-180° PHASE SHIFT
140
0·707
0·707
144
Y

FIG. 12.

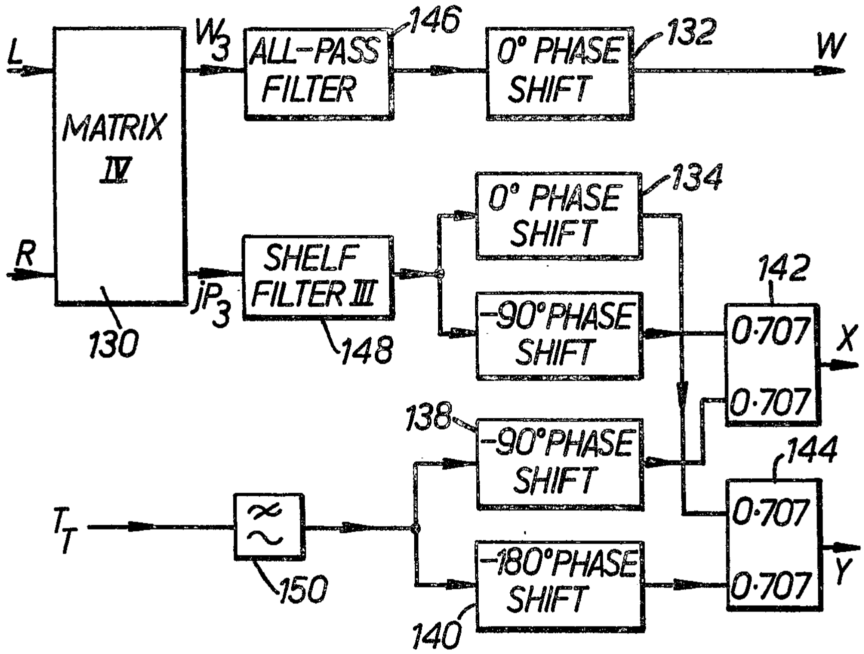
L
R
MATRIX IV
130
W_3
jP_3
ALL-PASS FILTER
146
0° PHASE SHIFT
132
W
SHELF FILTER III
148
0° PHASE SHIFT
134
-90° PHASE SHIFT
142
0·707
0·707
X
138
-90° PHASE SHIFT
144
0·707
0·707
Y
T_T
150
-180° PHASE SHIFT
140

FIG. 13.

PATENT SPECIFICATION

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(23) Complete Specification filed 10 March 1975
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(51) INT CL² H04R 5/04
(52) Index at acceptance H4R 16A
(72) Inventor MICHAEL ANTHONY GERZON

1 494 752

(19)

(54) SOUND REPRODUCTION SYSTEMS

(71) We, NATIONAL RESEARCH DEVELOPMENT CORPORATION, a British Corporation established by Statute, of Kingsgate House, 66—74, Victoria Street, London, S.W.1., do hereby declare the invention, for which we pray that a patent may be granted to us, and the method by which it is to be performed, to be particularly described in and by the following statement:—

This invention relates to sound reproduction systems and more particularly to sound reproduction systems which enable the listener to distinguish sound from sources extending over 360° of azimuth.

United Kingdom Specification No. 1,369,813 discloses a sound reproduction system which enables the listener to hear sound from sources extending over 360° of azimuth and which employs only two independent transmission channels. In the system described in this specification, one channel carries so-called omni-directional signal components which contain sound from all horizontal directions with equal gain. The other channel carries so-called azimuth or phasor signal components containing sounds with unity gain from all horizontal directions but with a phase shift relative to the corresponding omni-directional signal component which is related to, and is preferably equal to, the azimuth angle of arrival measured from a suitable reference direction. The phasor signal may be resolved into two components with a phase difference of 90°. When these signal components are applied to four loudspeakers located at the corners of a square, one signal component constitutes a difference signal indicating the difference in signal strength between the signals for a first adjacent pair of loudspeakers and the signals for a second adjacent pair comprising the other two loudspeakers. The other component constitutes a second difference signal indicative of the difference in signal strength between the signals for a third adjacent pair of loudspeakers comprising one loudspeaker from each of the first and second adjacent pairs and the signals for a fourth adjacent pair.

According to the invention, there is provided a decoder, for a sound reproduction system, comprising output means for providing output signals for at least three loudspeakers surrounding a listening position, input means for receiving at least two input signals comprising pressure signal components representative of the sum of the desired output signals and velocity signal components representative of the velocity of the sound field to be produced at said listening position and gain adjustment means between the input means and the output means and arranged to apply frequency dependent relative gains to said pressure and velocity signal components such that the gain applied to pressure signal components of frequencies above a predetermined frequency band divided by the gain applied to velocity signal components of frequency above said predetermined frequency band is greater than the gain applied to pressure signal components of frequency below said predetermined frequency band divided by the gain applied to velocity signal components of frequencies below said predetermined frequency band.

The input means may be arranged to provide a discrete signal containing only pressure signal components and one or more discrete signals containing only velocity signal components. In this case the gain adjustment means comprises a respective shelf filter, having a first characteristic, respective to each signal containing velocity signal components and a shelf filter, having a second characteristic, respective to the pressure signal.

The pressure signal components may be omni-directional signal components, i.e.

a signal component in which sounds from all directions are captured with a constant gain. The velocity signal components may be phasor signal components, i.e. signal components in which sounds from all directions in a plane (e.g. all horizontal directions) are captured with a complex gain proportional to $\cos\theta \pm j \sin\theta$ for a sound arriving at an angle θ a reference direction, the sign $\pm$ being the same for all sounds.

For four-loudspeaker rectangular arrangements, the velocity signal preferably has a gain of about twice that of the pressure signal for frequencies substantially below said predetermined frequency band.

The need for different treatment for frequencies above and below a particular frequency band is fully discussed in M. A. Gerzon "Criteres Psychoacoustiques Relatifs a la Realisaton des Systemes Matriciels et Discrets en Tetraphonie" published in the 1974 Paris International Festival du Son "Journees d'Etudes", Editions Radio, Paris and in M. A. Gerzon "Surround-sound psychoacoustics" Wireless World, December, 1974 pages 483 to 486. Briefly, for frequencies appreciably less than the frequency for which the distance between the human ears is less than half a wavelength of sound in air (about 700 Hz), the head offers no obstacle to sound waves so that the amplitude of sound reaching the two ears is virtually identical. Consequently, the only information available at these low frequencies for sound localisation is the phase difference between the sounds received at the two ears. At higher frequencies, the phase relationship is no longer of primary importance in sound localisation; what matters is the directional behaviour of the energy field around the listener. There is a transitional band, referred to above as the predetermined frequency band, between these two conditions.

The transitional frequency may be within the range 100 Hz to 1000 Hz. Transitional frequencies at the lower end of the range give an increased listening area. A preferred value is about 320 Hz.

Embodiments of the invention will now be described by way of example with reference to the accompanying drawings, in which:—

Figure 1 is a schematic diagram of a sound reproduction system illustrating the disposition of the loudspeakers round a listening position and their connection to a decoder,

Figure 2 is a block diagram of a known decoder suitable for use in the system shown in Figure 1,

Figure 3 is a block diagram of a decoder for use in a sound reproduction system providing height information and employing eight loudspeakers,

Figure 4 is a schematic diagram illustrating the disposition of loudspeakers for use with the decoder shown in Figure 3,

Figure 5 shows a decoder including a layout control unit,

Figure 6 is a circuit diagram of a layout control unit for use in the decoder shown in Figure 5,

Figure 7 is a schematic diagram, similar to Figure 4, illustrating the layout of an eight loudspeaker cuboid array,

Figure 8 is a schematic diagram of a decoder in accordance with the invention for use with the loudspeaker array shown in Figure 7,

Figure 9 is a block diagram of a frequency dependent decoder in accordance with the invention,

Figure 10 is a circuit diagram of a decoder of the type shown in Figure 9,

Figure 11 is a block diagram illustrating a decoder in accordance with the invention for use with discrete four channel signals,

Figure 12 is a block diagram of an alternative WXY circuit for use with the decoder of Figure 11, and

Figure 13 is a further alternative WXY circuit for use with the decoder of Figure 11.

It should be understood that, in the following description, where reference is made to a set of phase shifting circuits applying different phase shifts to different parallel channels, the phase shift specified in each case is a relative phase shift and a uniform additional phase shift may be applied to all channels if desired. Similarly, where it is specified that particular gains are applied to parallel channels, these gains are relative gains and a common additional overall gain may be applied to all channels if desired.

Before describing embodiments of the invention, it will be convenient to describe the basic form of a type of a decoder suitable for use with rectangular loudspeaker layouts and the corresponding type for use with cuboid loudspeaker layouts. These two types of decoder are hereinafter referred to as WXY decoders and WXYZ decoders respectively. The invention may be applied to any decoder of these types.

Referring to Figure 1, a listening location centred on the point 10 is surrounded by four loudspeakers 11, 12, 13 and 14 which are arranged in a rectangular array. The loudspeakers 11 and 12 each subtend an aqual angle θ at the point 10 relative to a reference direction indicated by an arrow 15. A loudspeaker 13 is disposed opposite the loudspeaker 11 and the loudspeaker 14 disposed opposite the loudspeaker 12. Thus, assuming that the reference direction is the forward direction, the loudspeaker 11 is disposed at the left front position, loudspeaker 12 at the right front position, the loudspeaker 13 at the right back position and the loudspeaker 14 at the left back position. All four loudspeakers 11 to 14 are connected to receive respective output signals LP, RF, RB and LB from the decoder 16 which has two input terminals 17 and 18, the received omni-directional signal W_1 being connected to the terminal 17 and the phasor signal P_1 to the terminal 18.

Figure 2 shows a known WXY decoder suitable for use as the decoder 16 when the angle $\theta=45°$. The decoder takes the form of a WXY circuit 20 and an amplitude matrix 22. The WXY circuit 20 produces an omni-directional output signal W, a front-back difference output signal X and a left-right difference output signal Y. These signals are then applied to the amplitude matrix 22 which produces the required output signals LB, LF, RF and RB.

The nature of the WXY circuit depends on the form of the input signals. If, as shown, the input signals comprise an omni-directional signal W_1 and a phasor signal P_1 of the same magnitude as the omni-directional signal but with a phase difference equal to minus the azimuth angle, the outputs of the WXY circuit 20 are related to its inputs as follows:—

$$W=W_1$$
$$X=\tfrac{1}{2}P_1$$
$$Y=\tfrac{1}{2}jP_1$$

The amplitude matrix 22 fulfills the function of the following group of equations:—

$$LB=\tfrac{1}{2}(-X+W+Y)$$
$$LF=\tfrac{1}{2}(X+W+Y)$$
$$RF=\tfrac{1}{2}(X+W-Y)$$
$$RB=\tfrac{1}{2}(-X+W-Y)$$

In fact this decoder is the same as the decoder shown in Figure 5 of the above-mentioned United Kingdom Specification No. 1,369,813, the 90° phase shift circuits serving as the active part of the WXY circuit 20 and the adders and phase inverters serving as the amplitude matrix 22.

Any decoder which produces the four output signals LB, LF, RF and RB is the equivalent of a WXY circuit and an amplitude matrix, and thus constitutes a WXY decoder, provided that

$$\tfrac{1}{2}(-LB+LF-RF+RB)=0$$

The WXY circuit 20 may have more than two inputs.

A WXYZ decoder may be used in systems providing height information and employing eight loudspeakers disposed at respective corners of a cube. Referring to Figure 3, three input signals are applied to a WXYZ circuit 24 which produces output signals W, X and Y having the same significance as the corresponding signals of Figure 2 and an up-down difference signal Z. The signals W, X, Y and Z are applied to a type II amplitude matrix 26 which produces eight loudspeaker signals LBU, LFU, RFU, RBU, LBD, LFD, RFD and RBD, the signals being fed to loudspeakers located at the correspondingly referenced points in Figure 4. The construction of the WXYZ circuit 24 depends on the nature of the input signals. The output signals from the type II matrix 26 are related to the input signals as follows:—

$$LBU=\tfrac{1}{2}(-X+W+Y+Z)$$
$$LFU=\tfrac{1}{2}(X+W+Y+Z)$$
$$RFU=\tfrac{1}{2}(X+W-Y+Z)$$
$$RBU=\tfrac{1}{2}(-X+W-Y+Z)$$
$$LBD=\tfrac{1}{2}(-X+W+Y-Z)$$
$$LFD=\tfrac{1}{2}(X+W+Y-Z)$$
$$RFD=\tfrac{1}{2}(X+W-Y-Z)$$
$$RBD=\tfrac{1}{2}(-X+W-Y-Z)$$

As for the two-dimensional case, any decoder is the equivalent of a WXYZ circuit and an amplitude matrix, and thus constitutes a WXYZ decoder, if the following equations are satisfied:—

$$(LBU+LBD)-(LFU+LFD)+(RFU+RFD)-(RBU+RBD)=0$$
$$(LBD+RBD)-(LFD+RFD)+(LFU+RFU)-(LBU+RBU)=0$$
$$(LBD+LFD)-(LBU+LFU)+(RBU+FRU)-(RBD+RFD)=0$$
$$(LBU-LBD)-(LFU-LFD)+(RFU-RFD)-(RBU-RBD)=0$$

Reverting to the loudspeaker arrangement and WXY decoder shown in Figures 1 and 2, in accordance with the invention, a layout control unit is provided to adjust the gains of the X and Y signals relative to the W signal to compensate for the non-square layout obtained when $\theta \neq 45°$. For example, when $\theta < 45°$ the gain for the front minus back signal has to be reduced for the increased front-back separation of loudspeakers and similarly, the gain of the left minus right signal Y has to be increased to compensate for the decreased side to side loudspeaker separation.

Referring to Figure 5, a layout control unit 28 is connected between the WXY circuit 20 and the type I amplitude matrix 22. The layout control unit 28 comprises gain adjustment devices 29 and 30 arranged to apply gain

$$\sqrt{2}\sin\theta$$

to the X signal and gain

$$\sqrt{2}\cos\theta$$

to the Y signal respectively to provide inputs W′, X′ and Y′ to the amplitude matrix 22.

One form of layout control unit 28 is shown in Figure 6. The gain control units 29 and 30 comprise respective inverting amplifiers 32 and 34, each of which has a feedback resistor R, an input resistor S and an output resistor T. The outputs X′ and Y′ of the gain control units 29 and 30 are also interconnected by a potentiometer U. The resistance R may have any convenient value and the resistance U may have any convenient value such that

$$U < \sqrt{2}L$$

where L is the input impedance of the amplitude matrix 22 for all input signals. Then, if

$$T = \frac{UL}{\sqrt{2}L - U}$$

$$S = \frac{\sqrt{2}L - U}{(2+\sqrt{2})L}$$

the gains for the X and Y signals are a good approximation to

$$\sqrt{2}\sin\theta$$

and

$$\sqrt{2}\cos\theta$$

respectively when θ is in the range 0° to 90°. In practice, it is preferable to keep θ within the range of about 25° to 65° since, outside this range, the angle subtended at the listening position by two of the pairs of adjacent loudspeakers become inconveniently large. This range may be limited by connecting fixed resistors in series with the potentiometer U and reducing the resistance of the potentiometer so that the overall resistance remains the same.

The W input signal to the layout control unit 28 is also connected to the W′ output thereof by an inverting amplifier 35 having feedback and input resistors of equal value R, thus matching the phase inversion introduced to the X and Y signals by the variable gain circuits.

It should be appreciated that changing the relative amplitudes of the X and Y signals has exactly the same effect as changing the phase of the phasor signal P_1 relative to the omni-directional signal W_1.

The above gains of

$$\sqrt{2}\sin\theta$$

in the X signal path and

$$\sqrt{2}\cos\theta$$

in the Y signal path are first order approximations to ideal gains. Better approximations are obtained if the gains are of the form

$$\sqrt{2}\,k\sin\theta \text{ and } \sqrt{2}\,k\cos\theta$$

respectively. At frequencies below about 500 Hz, the preferred form of k is given by

$$k=\frac{1}{\sin 2\theta}=\frac{1}{2\sin\theta\cos\theta}$$

which is approximately equal to 1 where θ equals 45°. At higher frequencies, the preferred value is k=1. If, as described above, these gains are not frequency dependent, the choice of k=1, as described above, is satisfactory at all frequencies.

Similar techniques may be used in conjunction with a WXYZ decoder for an eight loudspeaker cuboid array. In order to provide a decoder for the array shown in Figure 7, the decoder shown in Figure 3 is modified as shown in Figure 8 by inserting a layout control unit 36 comprising gain adjustment devices 38, 40 and 42 for the X, Y and Z channels respectively, between the WXYZ circuit 24 and the type II amplitude matrix 26. The approximate optimal gains for frequencies above and below 500 Hz are shown in the Table I.

Table I

channel	high frequency gain	low frequency gain
X	$\dfrac{\sqrt{3}\,ac}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\dfrac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,b}$
Y	$\dfrac{\sqrt{3}\,bc}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\dfrac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,a}$
Z	$\dfrac{\sqrt{3}\,ab}{\sqrt{a^2b^2+b^2c^2+c^2a^2}}$	$\dfrac{\sqrt{a^2+b^2+c^2}}{\sqrt{3}\,c}$

As for the rectangular decoder, if the gains are to be frequency independent, the values shown for high frequencies may be used. These values are equivalent to the values shown in Table II.

Table II

channel	gain
X	$\sqrt{3}\sin\theta$
Y	$\sqrt{\frac{3}{2}}\cos\theta\,.\,\sqrt{2}\sin\phi$
Z	$\sqrt{\frac{3}{2}}\cos\theta\,.\,\sqrt{2}\cos\phi$

where

$$b:a:c=\frac{1}{\sin\theta}:\frac{1}{\cos\theta\sin\phi}:\frac{1}{\cos\theta\cos\phi}$$

The gain adjustment devices 38, 40 and 42 may be implemented in a similar manner to the gain adjustment devices 29 and 30 of Figure 6, the gain adjustment devices 40 and 42 each comprising two stages in cascade, one with gain

$$\sqrt{\frac{3}{2}}\cos\theta$$

and the other with gain

$$\sqrt{2}\sin\phi$$

for the device 40 and

$$\sqrt{2}\cos\phi$$

for the device 42.

The three input signals to the WXYZ circuit 24 of Figure 8 may consist of linear combinations of the signals W_4, Y_4 and V_4 where W_4 is an omni-directional signal that picks up all sound directions with identical gain, Y_4 is a signal resulting from picking up a sound with gain $\sqrt{3}$ y and V_4 is a signal resulting from picking up a sound with directional gain $\sqrt{3}(x-qj\cancel{z})$, where q is a real constant, and $(x, y, \cancel{z})$ are the sound directions. Then the outputs of the WXYZ circuit 24 are related to its inputs as follows:—

$$W=W_4$$
$$X=fV_4$$
$$Y=fY_4$$
$$Z=fjq^{-1}V_4$$

where f is a real constant. Ideally at low frequencies $f=1$; ideally at mid-high frequencies,

$$f=\frac{1}{\sqrt{3(1+q^{-2})}}$$

It is clear that by interchanging axes, other encoding systems may be obtained. For example, one might consider the signals with directional gains 1, $x-jqy$, $\cancel{z}$ or 1, x, $y-jq\cancel{z}$. The corresponding decoders are obtained by exchanging the signal paths accordingly.

The decoders described above do not make special provision for the different mechanisms by which the human ears localise sounds above and below about 700 Hz. Decoders which do take into account these differences employ frequency dependent matrices approximating to an "ideal" low frequency design at low frequencies and an "ideal" high frequency design at high frequencies. There is also a transition region of frequencies in which the decoder matrix has an intermediate form. Theoretically, the centre of this transition region should be about 700 Hz. It has been found that, in practice satisfactory results can be obtained if the centre of this transition region is within the range of 100 Hz to 1000 Hz but that good listening conditions away from the centre of the listening area are best obtained if the centre of this region is below 700 Hz and a value of 320Hz has been found to be particularly suitable.

It has been found that there are four localisation criteria. Two of these criteria relate to voltage gain and are dominant at low frequencies. The other two criteria relate to the energy gain to which the signal is subject and are dominant at high frequencies. The symbols LB_V, LF_V, RF_V, and RB_V represent the complex voltage gains that a monophonic sound in some direction is subjected to when passed through the entire system, i.e., the original encoder and the decoder to feed the four loudspeakers shown in Figure 1. Then, for a sound for which the desired apparent azimuth angle is ϕ, the more important low frequency condition, known as the Makita condition, is that the quantities x and y are given by

$$x = \mathrm{Re}\left(\frac{LF_V + RF_V - LB_V - RB_V}{LF_V + RF_V + LB_V + RB_V}\right)$$

$$y = \mathrm{Re}\left(\frac{LF_V + LB_V - RF_V - RB_V}{LF_V + RF_V + LB_V + RB_V}\right)$$

must be expressible in the form

$$x \cos\theta = r \cos\phi$$
$$y \sin\theta = r \sin\phi$$

where r is a positive number. The symbol "Re" means "the real part of". If this condition is satisfied, the correct apparent direction of the sound is obtained at low frequencies. However, unless a second condition, known as the velocity condition is also satisfied, the apparent direction of the sound tends to be unstable when the listener moves his head. The velocity condition is

$$(x \cos\theta)^2 + (y \sin\theta)^2 = 1$$

At higher frequencies, above the transition frequency, the most important condition is the so-called energy vector condition that the quantities x_E and y_E given by

$$x_E = \frac{|LF_V|^2 + |RF_V|^2 - |LB_V|^2 - |RB_V|^2}{|LF_V|^2 + |RF_V|^2 + |LB_V|^2 - |RB_V|^2}$$

$$y_E = \frac{|LF_V|^2 + |LB_V|^2 - |RF_V|^2 - |RB_V|^2}{|LF_V|^2 + |RF_V|^2 + |LB_V|^2 + |RB_V|^2}$$

must be expressible in the form

$$x_E \cos\theta = r_E \cos\phi$$
$$y_E \sin\theta = r_E \sin\phi$$

where r_E is a positive number. This determines current localisation but, if the apparent direction of sound at higher frequencies is to be stable when the listener moves his head, it is in addition necessary, in accordance with the energy magnitude condition for the quantity

$$(x_E \cos \theta)^2 + (y_E \sin \theta)^2$$

to be as large as possible for all directions. In practice, it may be necessary to sacrifice the magnitude of this quantity for some directions in order to improve it in others. The quantity can, of course, never exceed 1.

The Makita condition and the energy vector condition, which determine the basic sound directions at low and high frequencies respectively, are the most important. Since it is not clear precisely which of these theories is more important in the region of frequencies around the transition frequencies, it is important that both conditions are satisfied in this region. It can be shown mathemathically that any WXY decoder or WXYZ decoder which satisfies either the Makita condition or the energy vector condition automatically satisfies both conditions. Thus, a WXY decoder or a WXYZ decoder designed to satisfy, for example, the Makita condition at all frequencies will give correct sound localisation at all frequencies. This applies to the decoders described above. In order to improve the stability of apparent sound direction as a listener's head moves, it is necessary to satisfy the velocity condition at lower frequencies and the energy magnitude condition at higher frequencies. This involves the use of frequency dependent decoders.

Figure 9 shows a decoder similar to that shown in Figure 5 but modified to provide the required frequency dependence. Two identical shelf filters 44 and 46, of type I are connected in the X and Y signal paths respectively. A shelf filter 48 of type II is connected in the W signal path. The shelf filters 44, 46 and 48 are filters with substantially identical phase responses and each having one gain at low frequencies, below a transition frequency, another gain at high frequencies above such transition frequency and which smoothly make the transition from low frequency gain to the high frequency gain across a frequency band around the transition frequency. When, as shown, the input to the decoder takes the form of an omni-directional signal W_1 and a phasor signal P_1, the relative gains of all the shelf filters, 44, 46 and 48 are 1 at frequencies above the transition frequency band in order to give optimum high frequency reproduction according to the energy magnitude condition. At frequencies below the transition frequency band, the gains of the shelf filters I relative to that of the shelf filter II are

$$\frac{2}{\sin 2\theta}$$

which is approximately equal to 2 when θ is in the range 30° to 60°. Consequently, it is satisfactory if the type I shelf filters have twice the gain of the type II shelf filter at frequencies below the transition frequency band.

A particular decoder circuit of this type is illustrated in Figure 10. In order to reduce the number of components required, the shelf filters and layout control are located before a modified WXY circuit 50. This means that a single type I shelf filter 52 is connected in the phasor signal path in place of the two type I shelf filters 44 and 46 in the X and Y signal paths respectively. The layout control unit 28 provides two phasor inputs to the WXY circuit 50 which comprises two 0° phase shift circuits 54 and 56 and one 90° phase shift circuit 58.

The shelf filter 48 is required to have a complex frequency response given by:—

$$\frac{\sqrt{a_1 b_1}\left(\sqrt{\frac{a_1}{b_1}} + j\omega T_1\right)}{1 + j\sqrt{\frac{a_1}{b_1}}\,(\omega T_1)} \times \frac{1 - j\omega T_2}{1 + j\omega T_2}$$

where a_1 is the low frequency gain and b_1 is the high frequency gain. This filter consists of an amplifier 60 connected to a capacitance resistance network comprising resistances R_1, R_2 and R_3 and capacitance C_1. In turn, this is connected to a parallel circuit having amplifier 62 and capacitor C_2 in one branch and amplifier 64 and resistance R_4 in the other branch. For a transition frequency of 200 Hz, the variables in the expression for frequency response and the circuit components have the values indicated in Table III.

TABLE III

a_1	0.6325
b_1	1
T_1	946.3 μsecs.
T_2	838.8 μsecs.
gain of 60	1.2649
gain of 62	−1
gain of 64	1
R_1	0.1325 R_0
R_2	0.3675 R_0
R_3	0.5 R_0
R_0 C_1	3237 μsecs.
R_4 C_2	T_2

The values of R_0 and R_4 are chosen arbitrarily according to design convenience.

The shelf filter 52 for the phasor signal P has the following complex frequency response:—

$$\frac{\sqrt{a_3 b_3}\left(\sqrt{\dfrac{a_3}{b_3}} - j\omega T_3\right)}{1 + j\sqrt{\dfrac{a_3}{b_3}}\,\omega T_3}$$

where a_3 is the low frequency gain and b_3 is the high frequency gain. This filter consists of two parallel paths, one consisting of an amplifier 66 and a resistor R_5 and the other consisting of an amplifier 68 and a capacitor C_3. The values of the various circuit components are shown in Table IV.

TABLE IV

a_3	$2a_1$
b_3	b_1
T_3	669.2 μsecs.
gain of 54	1.2649
gain of 56	−1
R_5 C_3	752.6 μsecs.

The value of the resistance R_5 is chosen arbitrarily according to design convenience.

The layout control unit 28 consists of an amplifier 70 of gain 1.707, two fixed resistances 72 and 74 in series with the outputs to the two phase shift circuits 56 and 58 in the WXY circuit 50 and a chain formed by fixed resistances 76 and 78 and a potentiometer 80 connected in parallel with the two outputs of the network. The moving contact of the potentiometer 80 is connected to earth. The two resistances 76 and 78 in series with the potentiometer each have resistance values equal to half that of the potentiometer 80. The two series resistances 72 and 74 each have resistance value equal to 1.414 times the resistance of the potentiometer 80. The amplifier 60 ensures that the sum of the energies at the two outputs of the layout control unit 28 is effectively equal to the energy at the input thereof.

The circuit shown in Figure 10 also includes a high pass filter 82 in the input path for the signals P_1. The high pass filter 82 consists of a capacitor 84 and a potentiometer 86. The purpose of this filter is to compensate for the effect at the listening position due to the distance between the loudspeakers and a central listener. The effect of a finite loudspeaker distance is to produce a bass boost and phase shift in the low frequency components of the velocity of the sound field at the listener and this, in turn, can degrade the image quality and may in some circumstances cause errors in the location of sound images at both frequencies.

In use, the setting of the potentiometer 86 is adjusted so that the time constant of the filter is equal to the time taken for sound to travel from any of the loudspeakers 11 to 14 to the centre point 10 of the listening area (Figure 1). The potentiometer 86 preferably has an associated scale calibrated in distance to facilitate this setting.

It should be noted that, as illustrated in Figure 1, the loudspeakers 11 to 14 are preferably equidistant from the centre point 10. If it is necessary for the distances of the various loudspeakers from the centre point 10 to differ from one another, the amplitude gains of the signals for the more distant loudspeakers are increased until a subjectively satisfactory result is obtained.

Similar compensation for the different localisation mechanisms used by the human ear at low and high frequencies may be applied to WXYZ decoders, respective type I shelf filters being connected in the X, Y and Z channels and a type II shelf filter in the W channel. Where the input signal is a four channel signal consisting of four linear combinations of an omni-directional signal and three signals resulting from picking up sound from an arrival direction given by direction cosines (x, y, z) with respective directional gains $\sqrt{3}x$, $\sqrt{3}y$ and $\sqrt{3}z$, the low and high frequency gains of these shelf filters are as follows:—

Filter	Low frequency gain	High frequency gain
I	1	$\sqrt{2/3}$
II	1	$\sqrt{2}$

Figure 11 illustrates a decoder in accordance with the invention for use with so-called "discrete" or "pairwise mixed" four channel signals. Such four channel signals assign sounds to a horizontal direction between the azimuths of two adjacent loudspeakers of a square layout by feeding them to both channels corresponding to adjacent speakers with the same phase but differing intensities thus, there are four input channels LB_1, LF_1, RF_1 and RB_1. For an azimuth ϕ from the front direction, the gains of the signals in the four input channels are shown in Table V.

TABLE V

	$-45° \leqslant \phi \leqslant 45°$	$45° \leqslant \phi \leqslant 135°$	$135° \leqslant \phi \leqslant 225°$	$-135° \leqslant \phi \leqslant -45°$
LB_1	0	$\cos(135°-\phi)$	$\sin(225°-\phi)$	0
LF_1	$\cos(45°-\phi)$	$\sin(135°-\phi)$	0	0
RF_1	$\sin(45°-\phi)$	0	0	$\cos(-45°-\phi)$
RB_1	0	0	$\cos(225°-\phi)$	$\sin(-45°-\phi)$

Such an encoding specification is in common use. It may be decoded using a WXY decoder as shown in Figure 11. The WXY circuit 88 thereof comprises a type III amplitude matrix 90 in the form

$$X_2=\tfrac{1}{2}(-LB_1+LF_1+RF_1-RB_1)$$
$$Y_2=\tfrac{1}{2}(LB_1+LF_1-RF_1-RB_1)$$
$$W_2=\tfrac{1}{2}(LB_1+LF_1+RF_1+RB_1)$$
$$F=\tfrac{1}{2}(-LB_1+LF_1-RF_1+RB_1)$$

The difference outputs X_2 and Y_2 of the amplitude matrix 90 are connected via respective 0° phase shift circuits 92 and 94 to provide the X and Y outputs. The pressure signal output W_2 is connected via a 0° phase shift circuit 96 and the diagonal difference output F via a 90° phase shift circuit 98 to a proportional adder 100 which applies gain 0.707 to the W_2 input, gain 0.455 to the jF input and then sums these two signals to provide the W output. The X and Y signals are applied to type I shelf filters 102 and 104 similar to the shelf filter 52 shown in Figure 12 but having unity gain at low frequencies and $\sqrt{\tfrac{3}{4}}$ gain at high frequencies. The W signal is applied to a type II shelf filter 106 which is similar to the shelf filter 48 of Figure 10 but having unity gain at low frequencies and $\sqrt{3/2}$ gain at high frequencies. The outputs of the shelf filters 102 and 104 are connected to variable high pass filters 108 and 110 which are identical with the high pass filter 82 of Figure 10 and have the control of their potentiometers ganged. These filters 108 and 110 provide compensation for loudspeaker proximity as described with reference to Figure 10. The outputs of the filters 108 and 110 are then connected to a layout control unit 112. The layout control unit 112 comprises a pair of input amplifiers 114 and 116, each having gain 2.414 and having their outputs connected to the outputs of the layout control unit by equal resistors 118 and 120. A resistance chain, consisting of resistor 122, potentiometer 124 and resistor 126 is connected between the outputs of the distance control unit. The relationship between the resistance values of the potentiometer 124 and the various resistors is as stated in Table VI where S may have any convenient value.

TABLE VI

Component	Resistance
118	0.707 S
120	0.707 S
122	0.25 S
124	0.50 S
126	0.25 S

The use of the resistors 122 and 126 in series with the potentiometer 112 limits the range of adjustment of the layout control to that over which satisfactory results can be achieved as described above with reference to Figure 6.

The decoder illustrated in Figure 11 may also be used as a four loudspeaker decoder for conventional stereo recordings by connecting the two stereo channels L and R to the inputs LF_1 and RF_1 respectively and grounding the other two inputs LB_1 and RB_1. Such stereo material is thus treated as four channel pairwise mixed material for which all sounds originate in the quadrant $-45°$ to $+45°$.

A decoder in accordance with the invention may be used to decode signals from the TMX three channel system in which the input system to the decoders consists of three channels as follows:—

$$L=\tfrac{1}{2}(W_3+jP_3)$$
$$R=\tfrac{1}{2}(W_3-jP_3)$$
$$T_T=jP_3{}^*$$

where $P_3{}^*$ is a signal whose azimuthal gain is the complex conjugate of that of P_3, as described in D. H. Cooper, T. Shiga and T. Takagi "QMX Carrier Channel Disc" Journal of the Audio Engineering Society, Volume 21, Pages 614 to 624, October, 1973. The WXY circuit 8 of Figure 11 is replaced by a WXY circuit as shown in Figure 12. The L and R input signals are connected to a type IV matrix 110 of the form:—

$$W_3=L+R$$
$$jP_3=L-R$$

The W_3 output of the matrix 130 is connected via a 0° phase shift circuit 132 to form the W output of the WXY circuit. The jP_3 output of the matrix 130 is connected both to a 0° phase shift 134 and to a $-90°$ phase shift circuit 136. Similarly, the T_T input signal from the TMX source is connected both to a $-90°$ phase shift circuit 138 and a $-180°$ circuit 140. The outputs of the two $-90°$ phase shift circuits 136 and 138 are added together, each with gain 0.707 in a proportional adder 142, the output of which forms the X output of the WXY circuit. Similarly, the outputs of the 0° phase shift 134 and the $-180°$ phase shift 140 are added together, both with gains 0.707 in a proportional adder 144, the output of which forms the Y output of the WXY circuit.

A decoder in accordance with the invention can also be used for the QMX system as described in D. H. Cooper, T. Shiga and T. Takagi, "QMX Carrier Disc". The QMX disc system incorporates TMX signals in which the T_T signal is of restricted band width and is therefore not available above about 6 kHz. In a decoder for this system, the WXY circuit shown in Figure 12 is replaced by a WXY circuit as shown in Figure 13. It will be seen that this circuit differs from the circuit of Figure 12 in that the W and jP outputs of the type IV matrix 130 are passed through an all-pass filter 146 and a type III shelf filter 148 and the T_T input is passed through a low pass filter 150 with a cut-off frequency of about 2 kHz. The all-pass filter 146, the shelf filter 148 and the low pass filter 150 all have substantially the same phase response and all have unity gain at well below 2 kHz. The shelf filter

148 has gain $\sqrt{2}$ at high frequencies and a transition frequency equal to the -6 dB frequency of the low pass filter 150.

The low pass filter 150 comprises two identical resistor-capacitor low pass filters in cascade, the all-pass filter 146 is a resistor-capacitor all-pass filter of the same time constant as the low pass filter 150 and the shelf filter 148 is a resistor-capacitor shelf filter followed by a phase-compensating all-pass filter of a similar design to those used for the type II shelf filter 48 in Figure 10.

In the case of two-input WXY circuits, the input signals need not be the actual omni-directional input signal W_1 and the phasor input signal P_1. Any non-singular linear combination thereof may be used with a suitably modified WXY circuit. The signals Q and R which are related to the signals W and P as follows:—

$$Q=\alpha W_1+\beta P_1$$
$$R=\beta^* W_1+\alpha^* P_1$$

where α and β are complex numbers and α^* and β^* their respective complex conjugates, may be used instead of the signals W_1 and P_1. This is because any such signals have equal amplitude but differing phase.

A decoder in accordance with the invention may also be used to decode inputs which may be regarded as consisting of two signals W_4 and P_4. W_4 is an omni-directional signal with unit gain in all directions and P_4 is a signal with gain

$$m \cos \phi - j \sin \phi$$

where ϕ is the azimuth angle from the front and m is real. When $m=1$, the signal P_4 is, of course, a phasor signal. Inputs in the form of signals W_4 and P_4 can be decoded by a WXY circuit in accordance with the following equations:—

$$W=W_4$$

$$X=\frac{1}{m\sqrt{2}}P_4$$

$$Y=\frac{1}{\sqrt{2}}jP_4$$

The encoding systems known as "BBC matrix G" and BBC matrix H", described in British Broadcasting Corporation Research Department, Engineering Division Report BBC RD 1974—29, "The subjective Performance of Various Quadrophonic Matrix Systems" November, 1974, produce signals L and R corresponding to the stereo "left" and "right" signals. It can be shown that the signals L and R may be regarded as linear combinations of the signals W_4 and P_4 as follows:—

$$W_4=\gamma L+\gamma^* R$$
$$P_4=\delta L+\delta^* R$$

where γ and δ are non-zero complex numbers of modulus 1 and γ^* and δ^* are their complex conjugates. The signals W_4 and P_4 can then be decoded by the above-described WXY circuit with m approximately equal to 0.68.

In all the embodiments of the invention described above, the signals W′, X′ and Y′ or W′, X′, Y′ and Z′ have been produced as discrete signals and applied to either a type I or a type II amplitude matrix respectively. It should be understood that the invention is also applicable to systems in which these signals do not have a separate discrete existance but take the form of linear combinations of one another, the output signals to the loudspeakers being produced directly from such linear combinations.

Where it is possible to interchange the positions of circuits or to combine circuits without changing the overall function, such modifications are within the scope of the invention. For example, if two successive circuits can be expressed mathematically as respective matrices, then they can be replaced by a single circuit which can be represented mathematically by the product of the two matrices.

It should also be understood that, at any point in the systems described, additional amplifiers may be inserted to provide such overall gain as is considered necessary or desirable by one skilled in that art. In particular, the outputs to the various loudspeakers will usually be connected to their respective loudspeakers via power amplifiers.

In all embodiments of the invention, there may be additional direct signal paths between the WXY circuit or the WXYZ circuit and the amplitude matrix providing the loudspeaker signals. For example, in the Figure 9 embodiment, a fourth signal path F may be added directly connecting the WXY circuit 20 to the amplitude matrix 22 which is then arranged to produce output signals as follows:—

$$LB=\tfrac{1}{2}(-X'+W'+Y'-F)$$
$$LF=\tfrac{1}{2}(X'+W'+Y'+F)$$
$$FR=\tfrac{1}{2}(X'+W'-Y'-F)$$
$$RB=\tfrac{1}{2}(-X'+W'-Y'+F)$$

which is as before if the F signal is 0. The addition of the F signal path will not affect the overall directional effect of the decoder provided that F is $\pm 90°$ out of phase with respect to X′ and Y′ for all directions.

A decoder having layout control as described above is claimed in our copending Application No. 13292/74 (Serial No. 1494751).

WHAT WE CLAIM IS:—

1. A decoder, for a sound reproduction system, comprising output means for providing output signals for at least three loudspeakers surrounding a listening position, input means for receiving at least two input signals comprising pressure signal components representative of the sum of the desired output signals and velocity signal components representative of the velocity of the sound field to be produced at said listening position and gain adjustment means between the input means and the output means and arranged to apply frequency dependent relative gains to said pressure and velocity signal components such that the gain applied to pressure signal components of frequencies above a predetermined frequency band divided by the gain applied to velocity signal components of frequency above said predetermined frequency band is greater than the gain applied to pressure signal components of frequency below said predetermined frequency band divided by the gain applied to velocity signal components of frequencies below said predetermined frequency band.

2. A decoder as claimed in claim 1, in which said input means is arranged to provide a discrete signal containing only pressure signal components and a discrete signal containing only velocity signal components and said gain adjustment means comprises a shelf filter having a first characteristic responsive to the velocity signal and a shelf filter having a second characteristic responsive to the pressure signal.

3. A decoder as claimed in claim 1, in which said input means is arranged to provide a discrete signal containing only pressure signal components and a plurality of discrete signals containing only velocity signal components and said gain adjustment means comprises a respective shelf filter, having a first characteristic, responsive to each velocity signal and a shelf filter, having a second characteristic, responsive to the pressure signal.

4. A decoder as claimed in claim 1, 2 or 3, in which the pressure signal components are omni-directional signal components and the velocity signal components are phasor signal components.

5. A decoder as claimed in any preceding claim, wherein the gain adjustment means comprises a filter having a frequency response of

$$\frac{\sqrt{a_1 b_1}\left(\sqrt{\dfrac{a_1}{b_1}} + j\omega T_1\right)}{1 + j\sqrt{\dfrac{a_1}{b_1}}\,(\omega T_1)} \times \frac{1 - j\omega T_2}{1 + j\omega T_2}$$

for the pressure signal components, where a_1 is the low frequency gain and b_1 is the high frequency gain, and a filter having a frequency response of

$$\frac{\sqrt{a_3 b_3}\left(\sqrt{\frac{a_3}{b_3}} - j\omega T_3\right)}{1 + j\sqrt{\frac{a_3}{b_3}}\,\omega T_3}$$

for the velocity signal components, where a_3 is the low frequency gain and b_3 is the high frequency gain, and T_1, T_2 and T_3 are time constants chosen in accordance with the required frequency of transition from low frequency gain to high frequency gain and so that the phase response of said filters are substantially identical at all frequencies.

6. A decoder as claimed in any preceding claim, in which said velocity signal components are passed through high pass filter means, the time constant of which is chosen to be substantially equal to the time of travel of sound from the loudspeakers to the centre of the listening area.

7. A decoder as claimed in claim 1, substantially as hereinbefore described with reference to the accompanying drawings.

8. A sound reproduction system including a decoder as claimed in any preceding claims.

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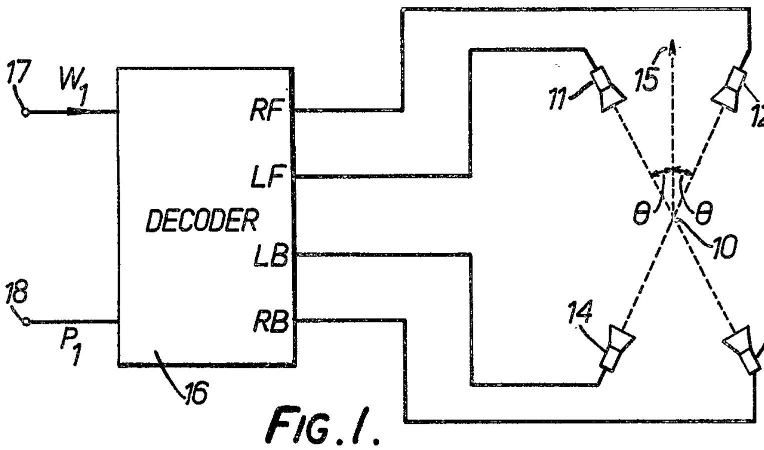


FIG. 1.

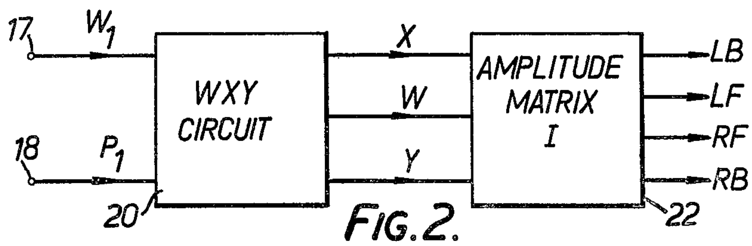


FIG. 2.

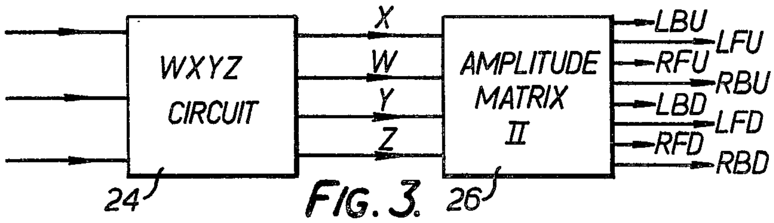


FIG. 3.

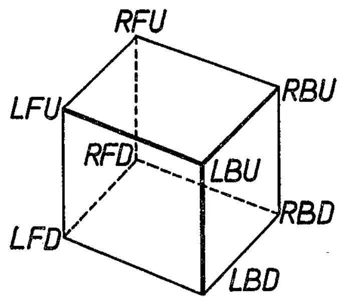


FIG. 4.

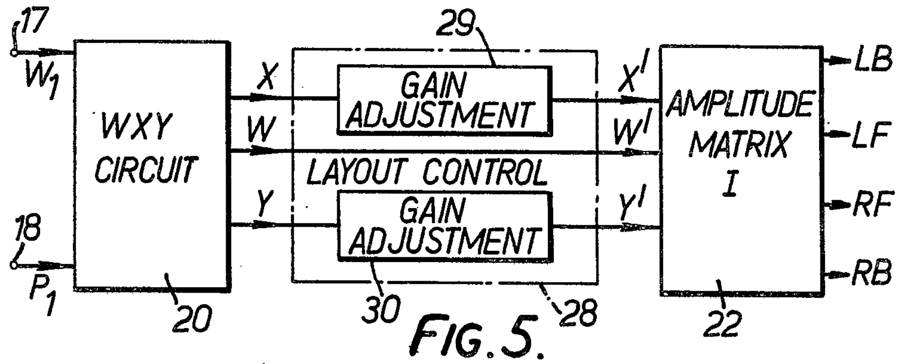
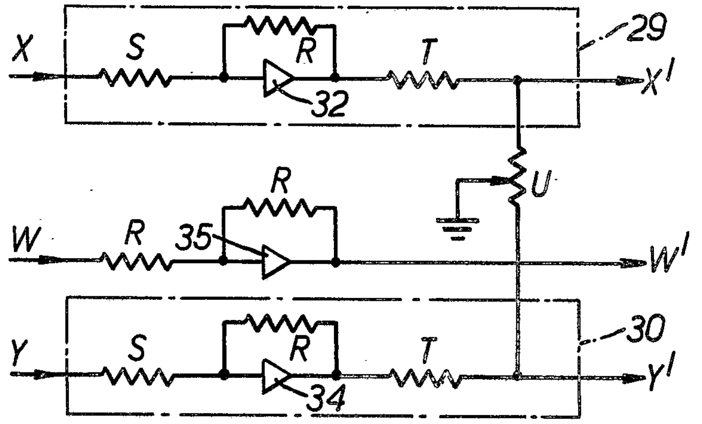


FIG. 5.



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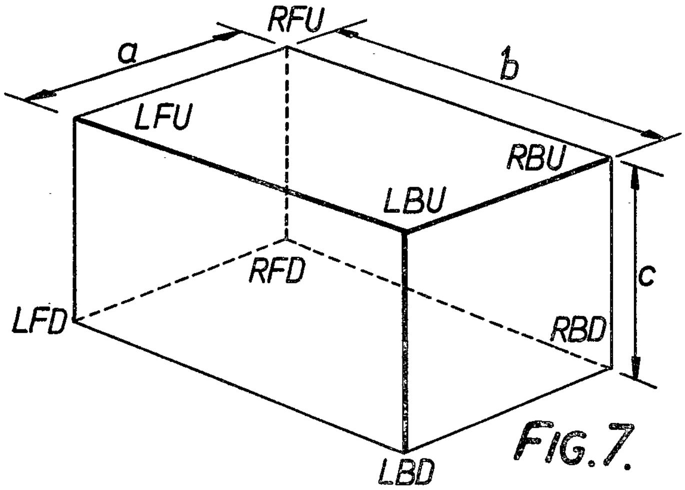

FIG. 7.

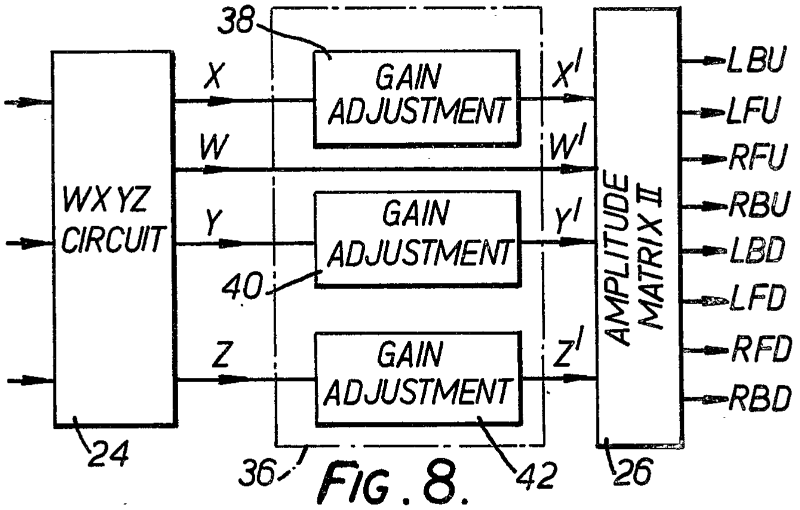
38
X
W
Y
Z
WXYZ CIRCUIT
GAIN ADJUSTMENT
GAIN ADJUSTMENT
GAIN ADJUSTMENT
X'
W'
Y'
Z'
40
AMPLITUDE MATRIX II
LBU
LFU
RFU
RBU
LBD
LFD
RFD
RBD
24
36
42
26

FIG. 8.

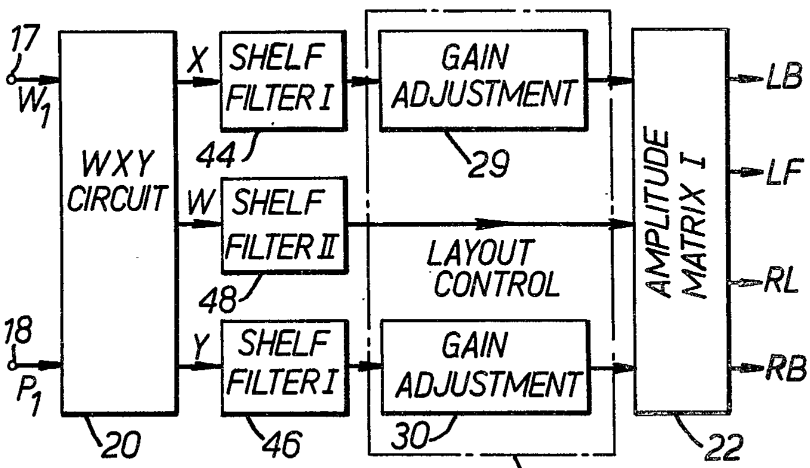
17
W1
18
P1
WXY CIRCUIT
X
W
Y
SHELF FILTER I
SHELF FILTER II
SHELF FILTER I
44
48
GAIN ADJUSTMENT
29
LAYOUT CONTROL
GAIN ADJUSTMENT
30
AMPLITUDE MATRIX I
LB
LF
RL
RB
20
46
28
22

FIG. 9.

1 494 752 COMPLETE SPECIFICATION
6 SHEETS This drawing is a reproduction of the Original on a reduced scale.
SHEET 4

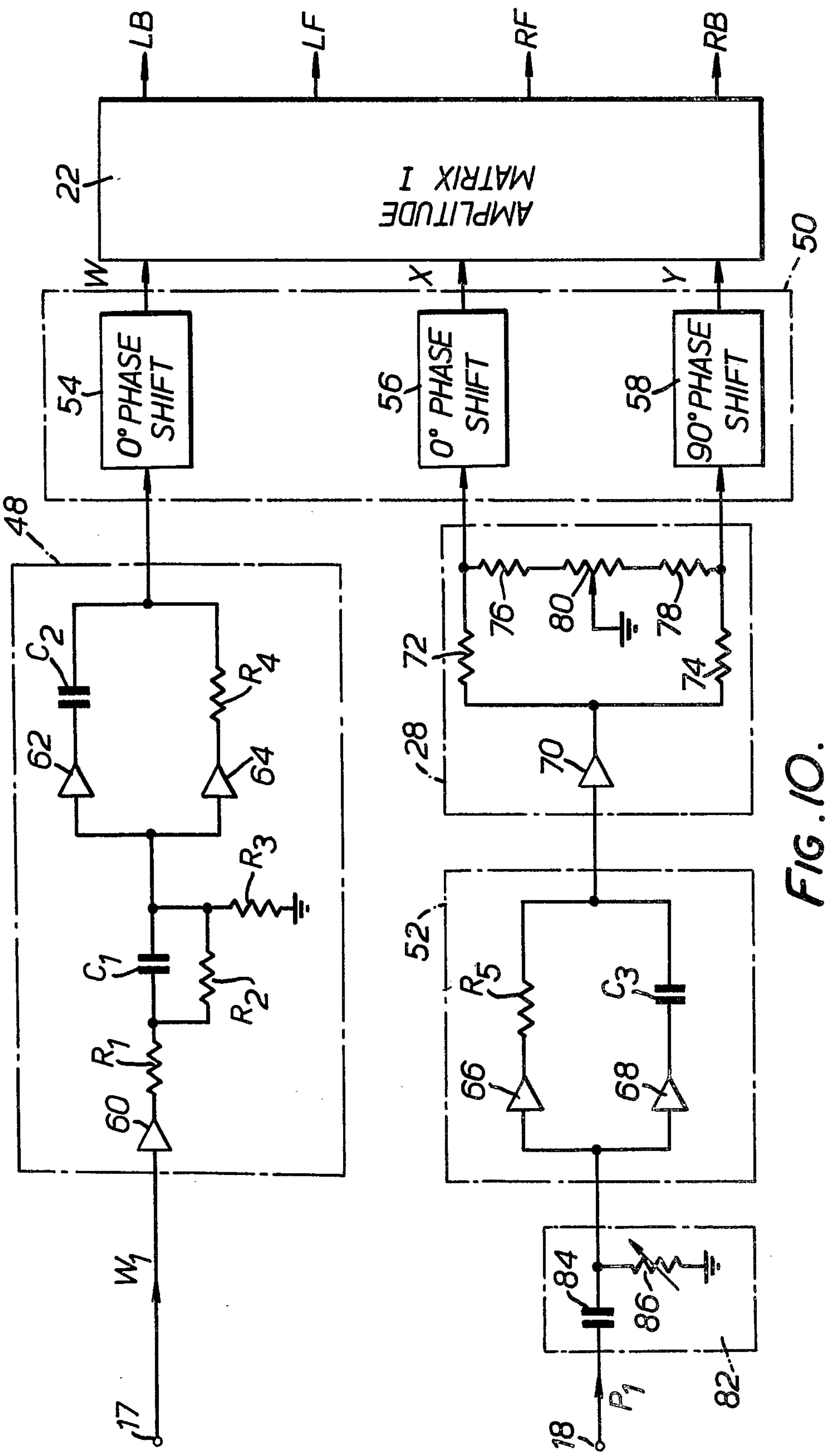

FIG. 10.

1 494 752 COMPLETE SPECIFICATION

6 SHEETS *This drawing is a reproduction of the Original on a reduced scale.*

SHEET 5

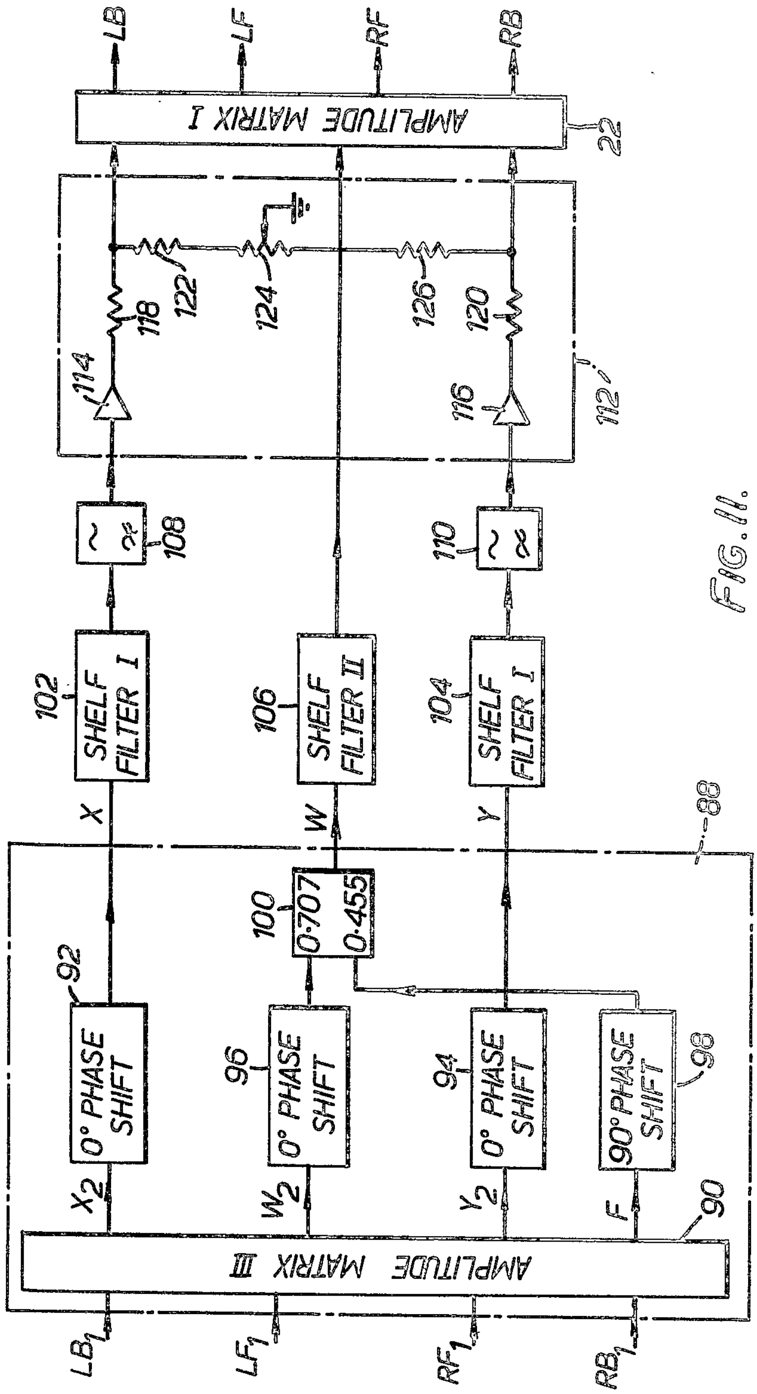

FIG. 11.

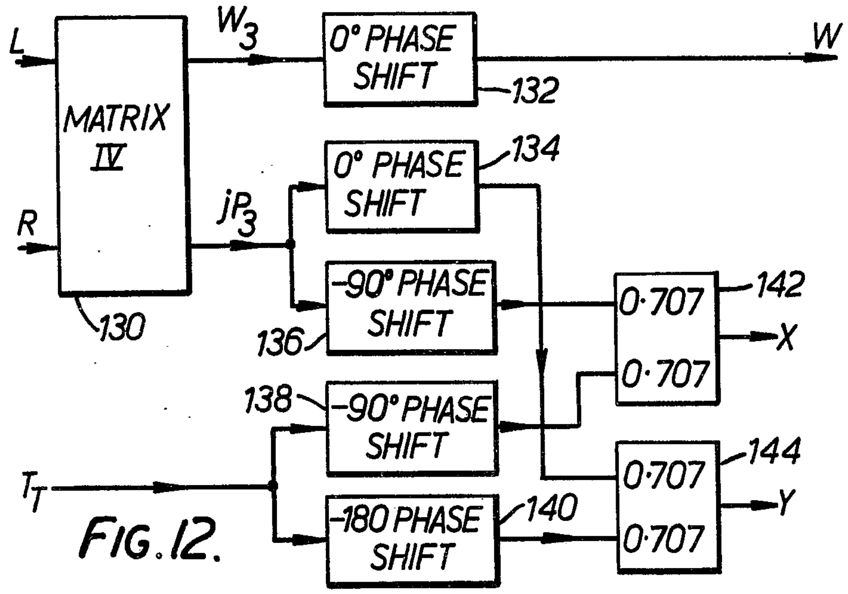
L
R
MATRIX IV
130
W3
jP3
0° PHASE SHIFT
132
W
0° PHASE SHIFT
134
-90° PHASE SHIFT
136
138
-90° PHASE SHIFT
TT
-180 PHASE SHIFT
140
0·707
0·707
142
X
0·707
0·707
144
Y

FIG. 12.

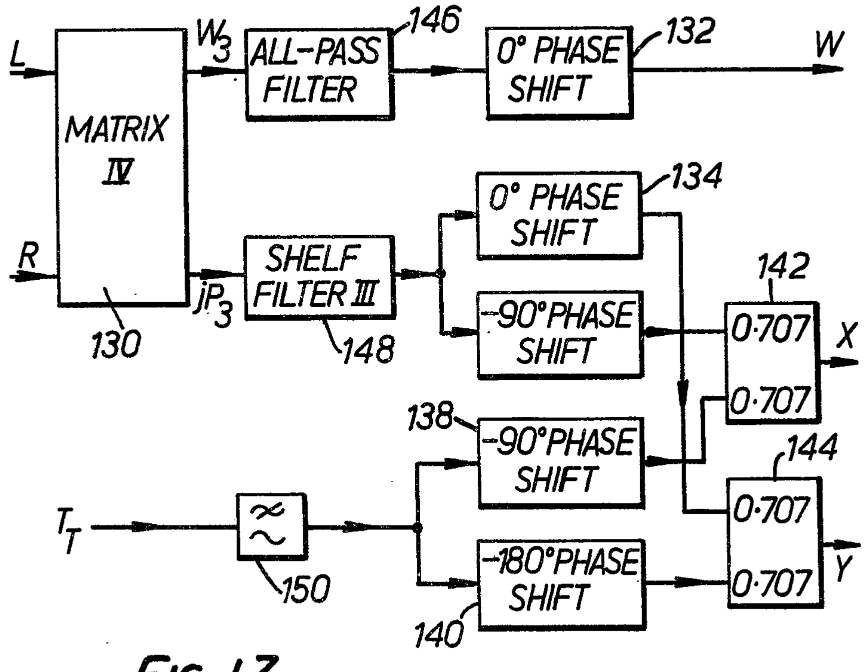
L
R
MATRIX IV
130
W3
ALL-PASS FILTER
146
0° PHASE SHIFT
132
W
jP3
SHELF FILTER III
148
0° PHASE SHIFT
134
-90° PHASE SHIFT
142
0·707
0·707
X
138
-90° PHASE SHIFT
144
0·707
0·707
Y
TT
150
-180° PHASE SHIFT
140

FIG. 13.

PATENT SPECIFICATION (11) 1 550 627

1 550 627

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(52) Index at Acceptance
H4R SDX
(72) Inventor: MICHAEL ANTHONY GERZON

(54) SOUND REPRODUCTION SYSTEMS

(71) We, National Research Development Corporation, a British Corporation established by Statute, of Kingsgate House, 66–74, Victoria Street, London, S.W.1, do hereby declare the invention, for which we pray that a patent may be granted to us, and the method by which it is to be performed, to be particularly described in and by the following statement:–

This invention relates to sound reproduction systems and more particularly to sound reproduction systems which enable the listener to distinguish sounds from sources extending over 360°C of azimuth.

United Kingdom Specifications Nos. 1,369,813 and 1,494,751 are concerned with sound reproduction systems which enable the listener to distinguish sounds from sources extending over 360° of azimuth and which employ only two independent transmission channels. In one of these systems, one channel carries so-called omnidirectional signal components which contain sounds from all horizontal directions with equal gain. The other channel carries so-called azimuth or phasor signal components containing sounds with unity gain from all horizontal directions but with a phase shift relative to the corresponding omnidirectional signal component which is related to, and is preferably equal to the azimuth angle of arrival measured from a suitable reference direction. In other systems, the signals of the two channels comprise linear combinations of the omnidirectional and phasor signals.

The phasor signal P may be resolved into components X and Y with a phase difference of 90°. For a sound at azimuth ϕ from the forward direction, the localisation is determined by

$$\cos \phi : \sin \phi = \mathrm{Re}\,\frac{X}{W} : \mathrm{Re}\,\frac{Y}{W}$$

where W is the omnidirectional signal and Re means 'the real part of'. Thus the imaginary parts of X/W and Y/W do not contribute substantially to the sound localization. Instead they cause the sound signals to have an unpleasant quality commonly called 'phasiness' which manifests itself in broad images that are hard to localize and sound very unnatural. It has been found that for a particular azimuth, the larger the ratio of the imaginary part of Y/W to the real part of Y/W, the worse the phasiness for signals from that particular azimuth.

An omnidirectional signal is a particular one of a class of signals which represent the acoustic pressure signal available at a listening position. Similarly a phasor signal is a particular one of a class of signals which represent the acoustic velocity signals available at the same listening position. It should be understood that in the present specification the signal W may be any signal representing said acoustic pressure signal and the signals X and Y may be any signals representing orthogonal components of said acoustic velocity signals.

The present invention is concerned with minimising the phasiness of the psychoacoustically most important signals. In general these are the signals from in front of the listener. However, if at any time, there is a dominant signal from a particular azimuth, it may be preferred to minimise the phasiness for this azimuth and to change the parameters of the decoding matrix as the azimuth of the most important sound alters. The invention is also applicable to decoders for systems which are subject to phasiness and have a higher number of channels than two and to decoders for three-dimensional systems which additionally distinguished between sounds originating at different heights and have a third signal Z, representing a third orthogonal component of the acoustic velocity signals, for this purpose.

According to the invention, there is provided a decoder for a sound reproduction system having at least three loudspeakers surrounding a listening area, the deocder comprising input means for receiving at least two input signals comprising pressure signal components and velocity signal components respectively representative of acoustic pressure and acoustic velocity at a chosen listening position, subtractor means, responsive to the input means, for subtracting from the signal components representative of the acoustic velocity vector in a chosen direction at said listening position a directional bias signal comprising a signal all the components of which differ in phase relation with respect to the pressure signal components by 90° and output means, responsive to the input means and the subtractor means, for producing a respective output signal for each loudspeaker.

This subtraction procedure is hereinafter called 'directional biasing'. In general the chosen direction will be the direction of the dominant or most significant signal. When the chosen direction is the forward direction, the procedure is called 'forward biasing'.

In the circumstances when all significant sound sources are, or a dominant sound source is, located at a particular azimuth at any one instant of time, the invention may provide means for determining such particular azimuth from the input signals and applying a bias signal dependent on such azimuth so as to compensate for phasiness of sources located thereat.

The pressure signal components may be omnidirectional signal components and velocity signal components may be phasor signal components.

Thus, in accordance with an embodiment of the invention, the signals W, X and Y used to produce the output signals for a two channel input signal, in which compensation for phasiness in the forward direction is required are as follows:–

$$W = W_{in}$$

$$X = \frac{1}{\sqrt{2}} P$$

$$Y = \frac{1}{\sqrt{2}} j\,(P - k\,W_{in})$$

$$= \frac{1}{\sqrt{2}} j\,P - \frac{1}{\sqrt{2}} j\,k\,W_{in}$$

where k is a positive constant between 0 and 1, preferably between ⅓ and ½. Subtraction of jkW_{in} from Y does not alter sound localizations in any way but merely alters the phasiness of reducing the imaginary part of Y/W.

It should of course, be understood that decreasing the phasiness at the front has the effect of increasing the phasiness of the back where P is negative. However, phasiness at the rear of the listener is psychoacoustically less important and an overall improvement is obtained.

Embodiments of the invention will now be described by way of example with reference to the accompanying drawings in which:

Figure 1 is a schematic diagram of a sound reproduction system illustrating the disposition of the loudspeakers round a listening position and their connection to a decoder.

Figure 2 is a block diagram of a known decoder suitable for use in the system shown in Figure 1,

Figure 3 is a block diagram of a decoder in accordance with another embodiment of the invention,

Figure 4 is a block diagram of a decoder in accordance with a further embodiment of the invention, and

Figure 5 is a block diagram of part of a decoder in accordance with a third embodiment of the invention.

It should be understood that, in the following description, where reference is made to a set of phase shifting circuits applying different phase shifts to different parallel channels, the phase shift specified in each case is a relative phase shift and a uniform additional phase shift may be applied to all channels is desired. Similarly, where it is specified that particular gains are applied to parallel channels, these gains are relative gains and a common additional overall gain may be applied to all channels if desired.

Before describing embodiments of the invention, it will be convenient to describe the basic form of a type of decoder suitable for use with rectangular loudspeaker laoyouts, hereinafter referred to as a WXY decoder. The invention may be applied to any decoder of this type.

Referring to Figure 1, a listening location centred on the point 10 is surrounded by four loudspeakers 11, 12, 13 and 14 which are arranged in a rectangular array. The loudspeakers 11 and 12 each subtend an equal angle Θ at the point 10 relative to a reference direction indicated by an arrow 15. A loudspeaker 13 is disposed opposite the loudspeaker 12. Thus, assuming that the reference direction is the forward direction, the loudspeaker 11 is disposed at the left front position, loudspeaker 12 at the right front position, the loudspeaker 13 at the right back position and the loudspeaker 14 at the left back position.

All four loudspeakers 11 to 14 are connected to receive respective output signals LF, RF, RB and LB from the deocder 16 which has two input terminals 17 and 18, the received omnidirectional signal W_I being connected to the terminal 17 and the phasir signal P_I to the terminal 18.

Figure 2 shows a known WXY decoder suitable for use as the decoder 16 when the angle $\Theta = 45°$. The decoder takes the form of a WXY circuit 20 and an amplitude matrix 22. The WXY circuit 20 produces an output signal W representing pressure, an output signal X representing front-back velocity and an output signal Y representing left-right velocity. These signals are then applied to the amplitude matrix 22 which produces the required output signals LB, LF, RF and RB.

The amplitude matrix 22 fulfils the function of the following group of equations:–

$LB = ½(-X + W + Y)$
$LF = ½(X + W + Y)$
$RF = ½(X + W - Y)$
$RB = ½(-X + W - Y)$

Any decoder which produces the four output signals LB, LF RF and RB is the equivalent of a WXY circuit and an amplitude matrix, and thus constitutes a WXY decoder, provided that

$½(-LB + LF - RF + RB) = 0$

The WXY circuit 20 may have more than two inputs. In fact this decoder is the same as the decoder shown in Figure 5 of the above-mentioned United Kingdom Specification No. 1,369,813 the 90° phase shift circuits serving as the active part of the WXY circuit 20 and the adders and phase inverters serving as the amplitude matrix 22.

The nature of the WXY circuit depends on the form of the input signals. If, as shown, the input signals comprise an omnidirectional signal W_I and a phasor signal P_I of the same magnitude as the omnidirectional signal but with a phase difference equal to minus the azimuth angle, the outputs of the WXY circuit 20 are related to its inputs as follows:–

$$W = W_I$$
$$X = \frac{1}{\sqrt{2}} P_I$$
$$Y = \frac{1}{\sqrt{2}} jP_I$$

Figure 3 shows a decoder similar to that of Figure 2 but forward biased in accordance with the invention. The forward biased decoder comprises a WXY circuit 24 which is similar to the WXY circuit 20 except that it has an additional jW output. The X and W outputs are connected directly to the amplitude matrix 22 as before. The jW output is connected via a variable gain amplifier 26 to a subtraction circuit 28 where it is subtracted from the Y output of the WXY circuit 24. The output Y of the subtraction circuit 28 is connected to the amplitude matrix 22. The gain of the amplifier 26 is set to k, i.e. a positive value between 0 and 1 as stated above. Conveniently, in the case when the WXY circuit 20 received two input signals comprising omnidirectional and phasor signal components k may be in the range from ⅓ to ½.

A similar modification may be made to any of the WXY decoders described in United Kingdom Specification No. 1,494,751. The subtraction of the jW signal from the Y signal may be carried out at any convenient point between the WXY circuit and the amplitude matrix. Conveniently, this subtraction is carried out on the output signals from the WXY circuit but other arrangements are possible. For example, as shown in Figure 4 of the present specification, the output of the WXY circuit 24 may be connected to respective shelf filters 30 to 33, the shelf filters 31 for the W signal being a type I shelf filter and the shelf filters 30 and 32 will be X and Y signals being type II shelf filters as described in the above mentioned co-pending application. The shelf filter 33 for the jW signal is a type III shelf filter which has a matched phase response identical to thoe of the types I and II shelf filters. This enables the constant k to be frequency dependent so that the degree of residual phasiness can be controlled according to the sensitivity of the human ear to phasiness at each frequency. However a design simplification or economy or apparatus may be achieved by making the type III shelf filter the same as the type I shelf filter in which case the function of these two filters can be performed by a single filter operating on the W signal, and a 90°

phase shift circuit used to produce the jW signal from the output of this filter. The signals are then applied to a lay-out control stage 34 and a distance control stage 38 substantially as described in the above-mentioned co-pending application.

The subtraction of the jW signal may also be performed after the lay-out control stage 34 and/or the distance control stage 38 although this will mean that the resulting compensation for phasiness will vary with these adjustments.

The application of the invention is not limited to decoders having omnidirectional and phasor inputs but can also be applied to more general classes of signals encoded on two channels. For example it may be applied to an encoding method such that one linear combination A of the two channels replaces the omnidirectional signal and another linear combination B is ($\cos\phi - j\,q\sin\phi$) times that of the linear combination A, where ϕ is chosen suitably for each encoded sound position and q is the real non-zero constant. ϕ may be equal to the intended azimuth angle during the encoding process or may be some function of that angle. In the following decoding equations, ϕ is treated as the angle from which the sound will be heard after decoding.

The decoder for such signals will have the following equations;

$W = A$

$X = \alpha B$

$Y = \alpha j\, q^{-1} (B - kA)$

where α is a constant which may be frequency-dependent and k is a positive constant less than 1. The subtraction of kA from the signal Y is the process of forward biasing in accordance with the invention so as to minimise 90° phase shifted components of Y for sounds for which ϕ is near zero. The value of α will ideally be about $\sqrt{2}$ at frequenices substantially below 350 Hz and around $1/\sqrt{2}$ at substantially higher frequencies.

The effect of the forward bias term in the above expression for Y is not only to reduce the phasiness of sounds towards the front but also to increase the gain of sounds from the back and to reduce that of sounds from the front. This may help to compensate for any relative excessive gains at the front in the signals A and B during encoding. There are several systems in which such excessive front gain exists.

For example, the invention may be applied to two channel signals where the signals in the two channels are linear combinations of C and D (possibly involving phase shifts) where C has gain ($1 + \mu\cos\phi + \mu\, j\sin\phi$) and D has gain ($\mu + \cos\phi - j\sin\phi$) where μ is a non-zero constant. Both signals have the equal gains for each azimuth ϕ and the signal D lags the signal C by a phase angle ϕ, just as for an omnidirectional/phasor encoding, but C does not have constant gain with angle, its actual energy gain being ($1 + \mu^2 + 2\mu\cos\phi$) at azimuth ϕ. Where μ is positive, this gain is higher at the front than at the back and these signals may be decoded by treating C as an omnidirectional signal and D as a phasor signal and using the forward biasing to help to restore equality to the gains during reproduction as well as giving lower phasiness for sounds from the front.

The invention may also be applied to three channel systems of the type in which the third channel is of poorer quality than the other two channels. For example, on a three-channel record, the two high quality channels may be base band channels and the third channel recorded using a subcarrier.

In one three-channel system the three transmitted signals are W_{in}, P and P* where P* is the signal whose directional gain is the complex conjugate of that of P. The respective gains of the three signals at azimuth ϕ are 1, ($\cos\phi - j\sin\phi$) and ($\cos\phi + j\sin f$). An 'ideal' WXY circuit for these three channels, without forward biasing, is given by

$W = W_{in}$

$X = \beta\,(\tfrac{1}{2}P + \tfrac{1}{2}P^*)$

$Y = \beta\,(\tfrac{1}{2}jP - \tfrac{1}{2}jP^*)$

where β is a real constant which may be frequency-dependent. This decoder does not suffer from phasiness but gives equal significance to the signals P and P*. In order to reduce the significance of the supposedly low quality signal P*, the following type of decoder has been proposed;

$W = W_{in}$

$X = \beta[tP + (1-t)P^*]$

$Y = \beta[tjP - (1-t)jP^*]$

where t is a positive number between ½ and 1. If $t = ½$ the resulting decoder is the full three-channel decoder described above and where $t = 1$ the resulting decoder is a two-channel decoder. t can vary with frequency if desired. This system is subject to phasiness and in order to reduce phasiness for front images, it may be forward biased as follows:

$W = W_{in}$

$X = \beta[tP + (1-t)P^*]$

$Y = \beta\,[tjP - (1-t)jP^* - k(2t-1)jW_{in}$

Although the undesirable side effects of increase in the gain at the back relative to that at the front also occurs, the magnitude of this effect is less than that for a two-channel decoder.

In a full three-channel system, there are signals other than jW that have 90° phase shift relative to W for all azimuths. Any real linear combination of jW, j(P + P*) and (P – P*) has the required 90° phase shift. Consequently, a three-channel decoder can be forward biased without affecting its basic image localization by adding any real linear combination of these three signals to X and Y in the basic decoder equation. Such bias need not necessarily be in the forward direction (in which case it is not forward bias) and may be used to alter the gain of the decoder in some directions relative to others.

With some encloded signals, all significant sound sources or a dominant sound source may be located at a particular azimuth at any one instant of time. In these circumstances, it may be desirable to apply a bias signal to reduce the imaginary components of the velocity signal components signal for this particular azimuth.

More specifically, a decoder matrix for this purpose may have the following decoding equations:

$W = W_{in}$

$X = \alpha(P + ju\ W_{in})$

$Y = \alpha(jP + j\ v\ W_{in})$

where α is a real constant which may be frequency dependent and u and v are real numbers, representing gains, which vary according to the deduced distribution of sounds in the encoded signals.

If it is deduced that all the sounds in the encoded signals are at azimuth f then the ideal values of u and v are

$u \cong \sin\phi$

$v \cong -\cos\phi$

in order to cancel out the 90° phase shifted components of X and Y. If the general tendency of sounds is to be twoards azimuth ϕ, but with a certainity of $r < 1$. (where r may be related to the spread of sound sources away from azimuth ϕ), then putting:

$u \cong r\sin\phi$

$v \cong -r\cos\phi$

gives acceptable results. Inaccuracies in the estimates for ϕ and r do not affect the subjective results very critically because azimuths near ϕ are also decoded with relatively low phasiness.

Several methods estimating ϕ and r are known and one technique will be described by way of example. Figure 5 illustrates a WXY circuit incorporating variable bias in accordance with the invention for decoding the signals W_{in} and jP.

The W_{in} signal is applied to a 0° phase shift circuit 50 for producing the signal W and to a 90° phase shift circuit 52 for producing the signal jW_{in} Similarly, the phasor signal jP is applied to a –90° phase shift circuit 54 and 0° phase shift circuit 56. The outputs of the phase shift circuits 54 and 56 are connected via respective adders 58 and 60 to the X and Y outputs of the WXY circuit, the adders 58 and 60 being used to apply the required biasing as will now be described.

It can be shown that for practical purposes $\cos\phi$ and $\sin\phi$ can be considered as given by

$$-2r\cos\phi = \frac{En(W_{in} - P) - En(W_{in} + P)}{En(W_{in})}$$

and

$$2r\sin\phi = \frac{En(W_{in} + jP) - En(W_{in} - jP)}{En(W_{in})}$$

where En(S) means the envelope of a wave form S.

In the circuit shown in Figure 5, the omnidirectional signal W_{in} is applied to an envelope detector 58′ to produce the signal $En(W_{in})$ which is the denominator of both the above expressions. The signal $En(W_{in} + P)$ produced by an envelope detector 60′ responsive to an adder 62 and the signal $En(W_{in} - P)$ is produced by an envelope detector 64 which is responsive to a subtraction circuit 66. The outputs of the envelope detectors 60′ and 64 are applied to a subtraction circuit 68 to produce the numerator of the expression for cos ϕ and this is divided by the output of the envelope detector 58′ in a divider 70. The output of the divider 70 is multiplied by jW_{in} in a multiplier 72 to obtain the required biasing signal for the Y output. This biasing signal is then applied via a variable gain amplifier 74 to the adder 60.

The biasing signal for the X output is obtained in a similar manner. The signal $En(W_{in} +$

jP) is produced by an envelope detector 76 which is responsive to an adder 78. The signal En (W_{in} –JP) is produced by an envelope detector 80 which is responsive to a subtraction circuit 82. The outputs of the envelope detectors 76 and 80 are applied to a subtraction circuit 84, the output of which is divided by the output of the enevelope detector 58′ in a divider 86. The output of the divider 68 is multiplied by the output of the phase shift circuit 52 in a multiplier 88 and the resulting biasing signal is applied to the adder 58 via an amplifier 90.

Thus the biasing signals applied to the X and Y outputs of the circuit shown in Figure 5 are dependent on the azimuth of the dominant sound represented by the coded signals W_{in} and P and the magnitude of the biasing signals depends on the amplitude of the dominant signal as compared with the amplitude of signals from other directions. If sounds of equal intensity come from directions of widely differing azimuth so that there is no dominant signal, the inputs to the subtraction circuits 68 and 84 will be equal so that their outputs are zero.

A simplified variable bias decoder may be obtained by applying a variable bias signal only to the Y output of the WXY circuit and not to the X output, i.e. by putting *u* equal to zero. This will 'enhance' directional resolution to the front and/or the back but not at the sides.

Directional biasing may also be applied to non-rectangular loudspeaker layouts. For example, in a regular polygonal array, the signal fed to each loudspeaker may be:

$k_1 W + k_2 (X' + k_3 jW) \cos \Theta + k_2 (Y' + k_4 jW) \sin \Theta$ where X' and Y' are the velocity signal outputs of the WXY circuit and k_1 and k_2 are both greater than zero and where Θ is the azimuth of the loudspeaker to which the signal is fed. The terms $k_3 jW$ and $k_4 jW$ are the directional bias terms. k_1 k_2 k_3 and k_4 may be frequency dependent and/or may be dependent on the supposed instantaneous direction of the dominant signals but otherwise they are real constants. The circuitry required to implement sich polygonal decoders differs from that illustrated in Figures 2 to 5 only in that the output amplitude matrix 22 is replaced by an amplitude matrix having *n* outputs S_i (corresponding to loudspeakers at azimuths $\Theta_1, \ldots \Theta_n$ spaced apart by 360°/n) given by

$$S_i = k_1 W + k_2 X \cos \Theta_i + k_3 Y \sin \Theta_i$$

When directional biasing is applied to three-dimensional systems, biasing may be applied to the Z component of the velocity signal as well as or instead of the X and/or Y components.

What we claim is:

1. A decoder for a sound reproduction system having at least three loudspeakers surrounding a listening area, the decoder comprising input means for receiving at least two input signals comprising pressure signal components and velocity signal components respectively representative of acoustic pressure and acoustic velocity at a chosen listening position, subtractor means, responsive to the input means, for subtracting from the signal components representative of the acoustic velocity vector in a chosen direction at said listening position a directional bias signal comprising a signal all the components of which differ in phase relation with respect to the pressure signal components by 90° and output means, responsive to the input means and the subtractor means, for producing a respective output signal for each loudspeaker.

2. A decoder according to claim 1, wherein the directional bias signal is a fraction of the pressure signal phase shifted by 90°.

3. A decoder according to claim 1 or 2, wherein the fraction of the pressure signal is in the range of one third to one half thereof.

4. A decoder according to claim 1, 2, or 3, wherein the pressure signal components are omnidirectional signal components and the velocity signal components are phasor signal components.

5. A decoder according to any preceding claim, wherein the input means is adapted to receive three input signals and derive therefrom a pressure signal and two velocity signals which, for all sounds, are either in phase or 180° out of phase with the pressure signal, the signal subtracted from the velocity signal being a real linear combination of the pressure signal phase shifted by 90° and the velocity signals phase shifted by 90°.

6. A decoder according to claim 5, wherein the two velocity signals are respectively the sum of a phasor signal and its complex conjugate and the difference between the phasor signal phase shifted by 90° and its complex conjugate.

7. A decoder according to any preceding claim, including means responsive to the input signals for determining the azimuth angle of the most significant sound source and means for applying a directional bias signal dependent on said azimuth angle.

8. A decoder according to claim 7. including means for producing first and second mutually orthogonal components of the velocity signal of azimuths 0° and 90° respectively and means for applying a first directional bias signal to said first component and a second directional bias signal to said second components.

9. A decoder according to claim 8, wherein the magnitude of the first directional bias signal is proportional to minus the cosine of said azimuth angle and the magnitude of the second directional bias signal is proportional to the sine of said azimuth angle.

10. A decoder according to claim 9, wherein the bias signal applied to the first of said mutually orthogonal components is proportional to the difference between the envelope of the difference between the pressure and velocity signals and the envelope of the sum of the pressure and velocity signals divided by the envelope of the pressure signal and the bias signal applied to the second of said mutually orthogonal components is the difference between the envelope of the sum of the pressure signal and the omnidirectional signal phase shifted by 90° and the envelope of the difference between the pressure signal and the omnidirectional signal phase shifted by 90° divided by the envelope of the pressure signal.

11. A sound reproduction system incorporating a decoder according to any preceding claim.

12. A decoder for a sound reproduction system substantially as hereinbefore described with reference to Figure 3, Figure 4 or Figure 5 of the accompanying drawings.

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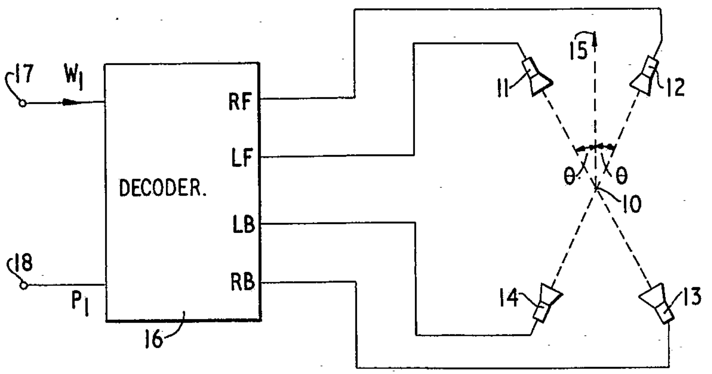
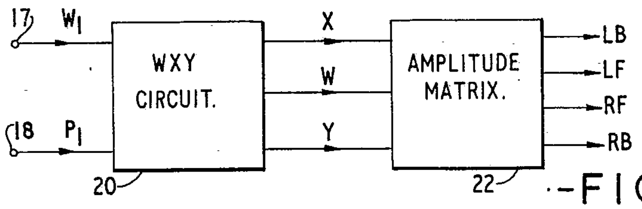
1550627 COMPLETE SPECIFICATION

2 SHEETS *This drawing is a reproduction of the Original on a reduced scale*

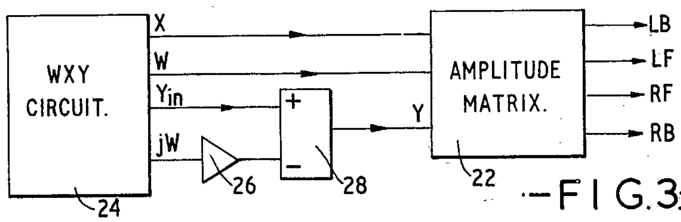
Sheet 1



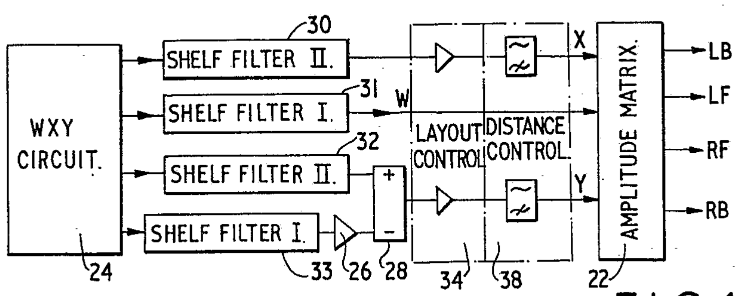
-FIG.1-



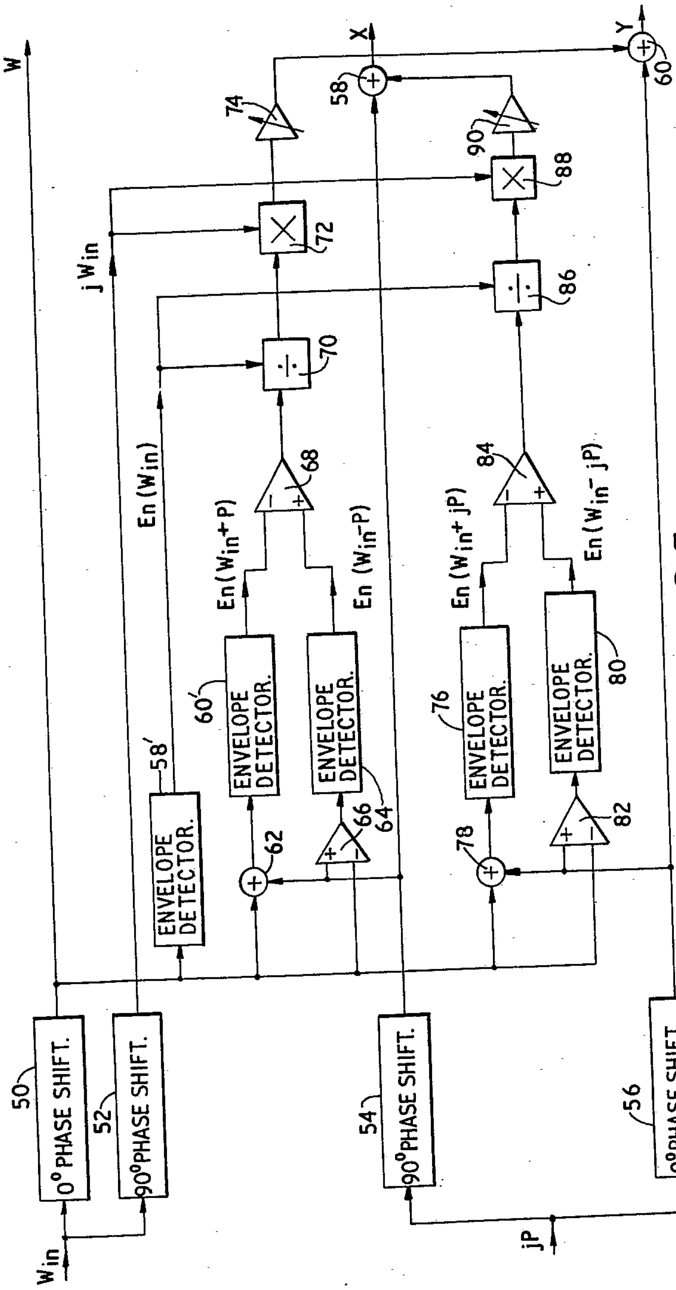
-FIG.2-



-FIG.3-



-FIG.4-


W
jWin
En (Win)
En (Win+P)
En (Win-P)
En (Win+jP)
En (Win-jP)
X
Y
ENVELOPE DETECTOR.
ENVELOPE DETECTOR.
ENVELOPE DETECTOR.
ENVELOPE DETECTOR.
ENVELOPE DETECTOR.
0° PHASE SHIFT.
90° PHASE SHIFT.
90° PHASE SHIFT.
0° PHASE SHIFT.
Win
jP
50
52
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58'
60
60'
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64
66
68
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86
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Michael Anthony Gerzon
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(54) **Sound reproduction systems**

(57) A decoder is provided for feeding an irregular array of *m* (being three or more) pairs of diametrically opposite loudspeakers, each loudspeaker being disposed at an equal distance *r* from a common reference point. The decoder incorporates a WXY circuit 10 for producing output signals W, X, Y and −jW from the input signals, and shelf filters 12, 14, 16 and 22 and high-pass filters 18, 20 and 24 for producing output signals W′, X′, Y′ and −jW″ᵢ. In addition the decoder includes an amplitude matrix circuit 26 which is such that the sum of the signals S^+_i and S^-_i fed to the loudspeakers of each pair is the same for all pairs of loudspeakers, and

$$S^+_i = W' + \alpha_i X' + \beta_i Y' - \delta_i jW''_i$$
$$S^-_i = W' - \alpha_i X' - \beta_i Y' + \delta_i jW''_i$$

where $KM = \frac{k}{\sqrt{2}} mr I$,

M being the $2 \times m$ matrix

$$\begin{pmatrix} \alpha_1 & \beta_1 \\ \vdots & \\ \alpha_m & \beta_m \end{pmatrix}$$

K being the $m \times 2$ matrix

$$\begin{pmatrix} x_1 & \cdots & x_m \\ y_1 & & y_m \end{pmatrix}$$

I being the 2 x 2 identity matrix, and *k* being a positive real constant which may be frequency dependent.

A decoder is also provided for feeding a three-dimensional loudspeaker layout.

Thus the outputs of the loudspeakers may be adapted to irregular positioning of the loudspeakers which may be dictated by room geometry.

FIG.1.

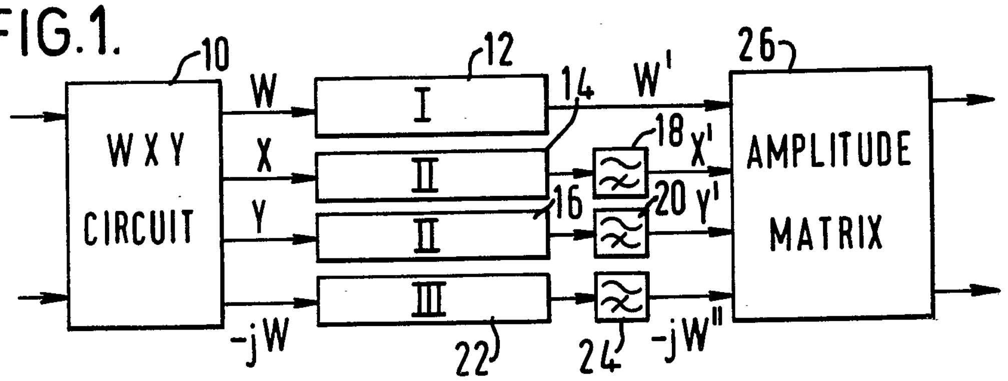

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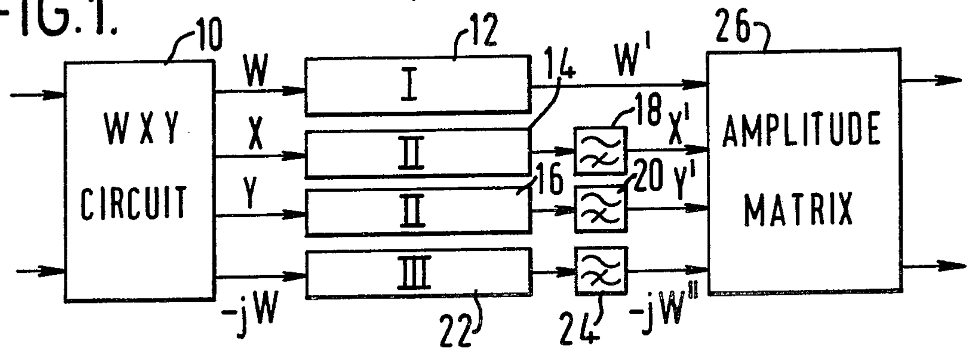
FIG.1.



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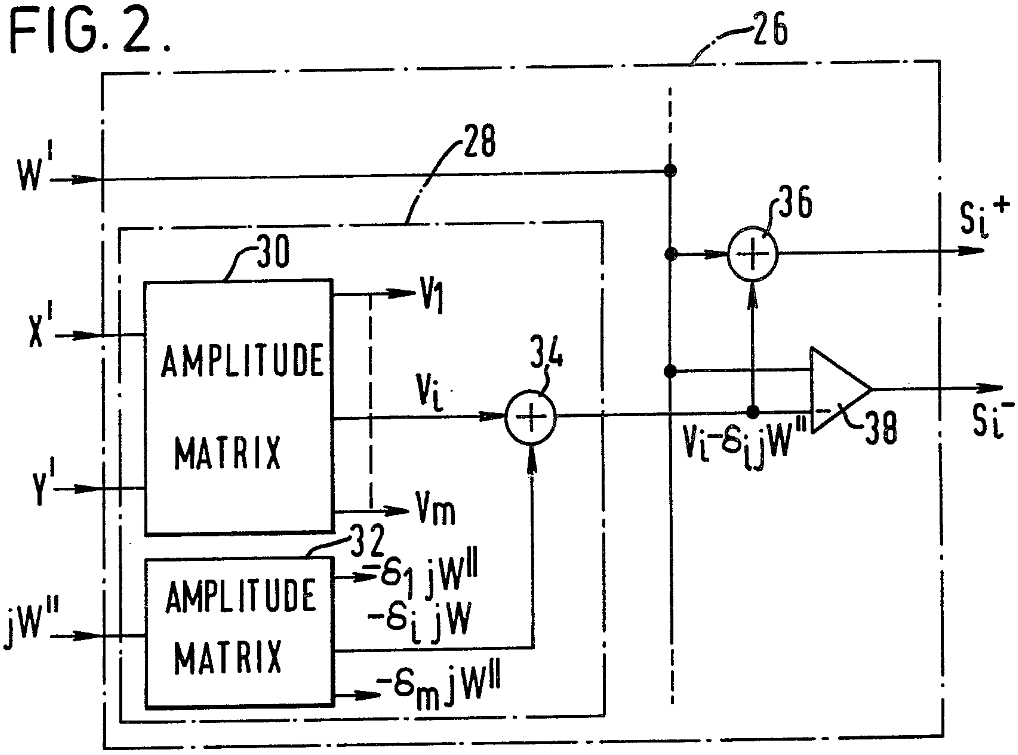

FIG.3.

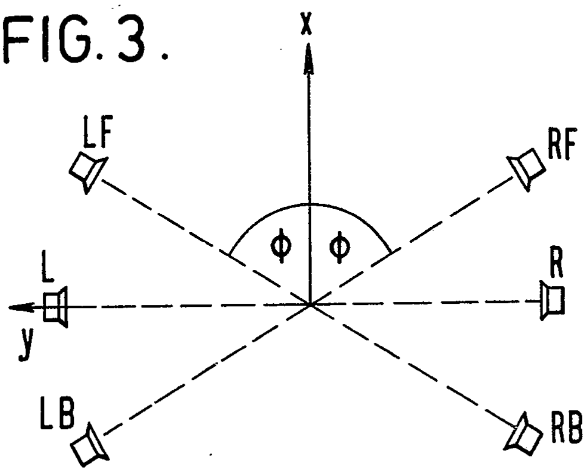

FIG. 4.

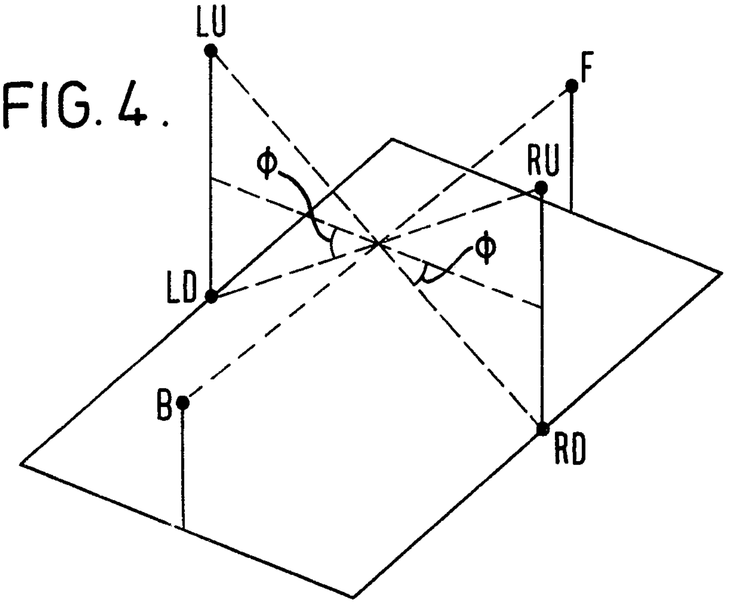
LU
F
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FIG. 5.

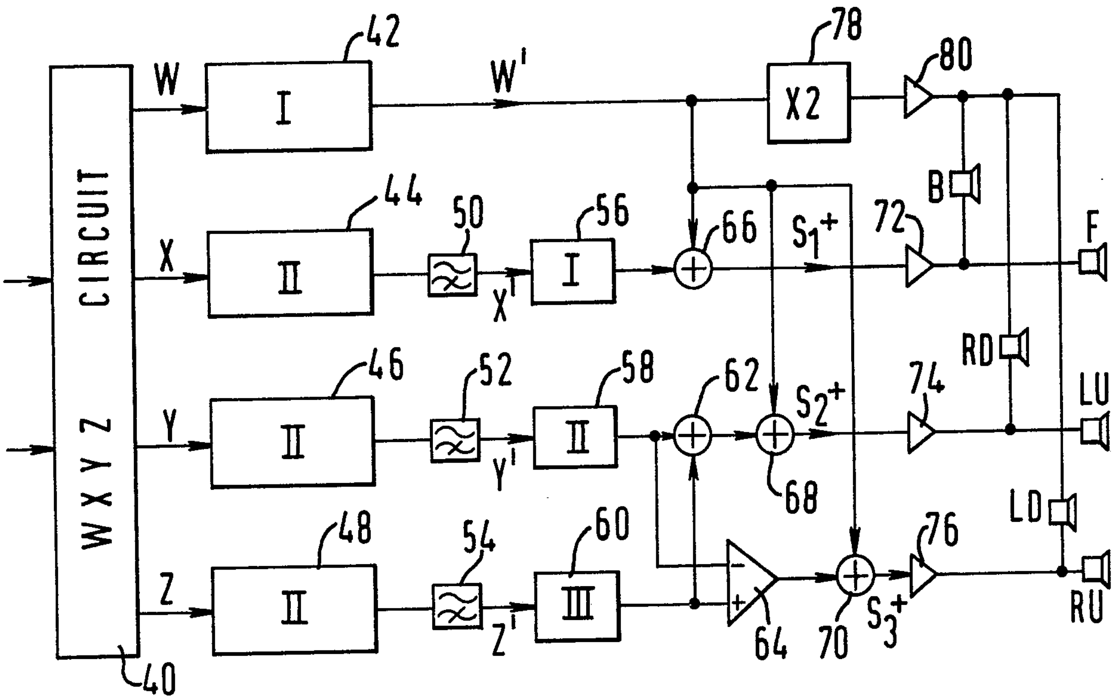
WXYZ CIRCUIT
40
W
X
Y
Z
I
42
II
44
II
46
II
48
W'
50
52
54
X'
Y'
Z'
I
56
II
58
III
60
62
64
66
68
70
S_1^+
S_2^+
S_3^+
X2
78
80
72
74
76
B
F
RD
LU
LD
RU

FIG. 6.

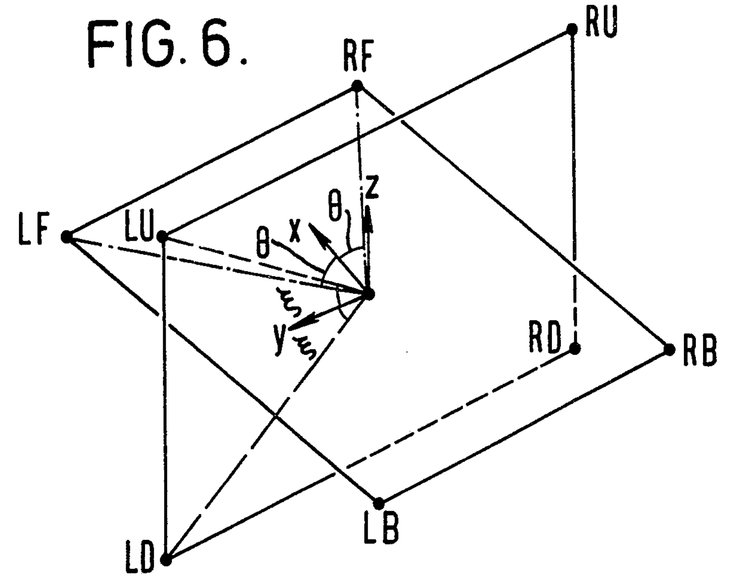
RU
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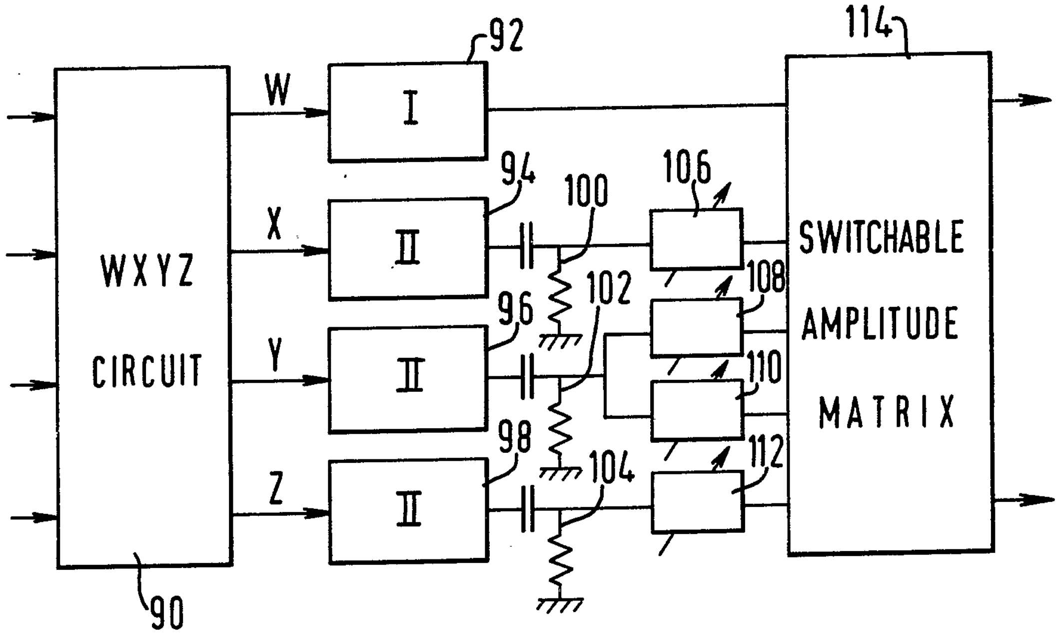
FIG. 7.


92
114
W
I
94
100
106
X
II
SWITCHABLE
WXYZ
108
102
96
AMPLITUDE
Y
II
CIRCUIT
110
MATRIX
98
104
112
Z
II
90

SPECIFICATION

Sound reproduction systems

This invention relates to sound reproduction systems and more particularly to sound reproduction systems which enable a listener to distinguish sound from source extending over 360° of azimuth. Such systems are hereinafter called surround sound systems. The invention is also applicable to surround sound systems which, in addition, enable the listener to distinguish sound from sources at difference heights.

Surround sound systems for loudspeaker arrays in which the loudspeakers are disposed at the corners of a geometrically regular polygon, or, in the case of with-height surround sound systems, the corners of a regular solid, are already known. Such systems are also known for loudspeaker arrays where the loudspeakers are disposed at corners of a rectangle or rectangular cuboid. The present invention is concerned with the provision of a decoder for use in a surround sound system where the loudspeakers are disposed at other locations. Such loudspeaker arrays will hereinafter be referred to as irregular loudspeaker arrays and it should be understood that this term excludes rectangular and rectangular cuboid arrays in spite of the fact that these are, in strict mathematical terms, not regular shapes.

It has already been proposed in U.K. Patent Specification No. 1,411,994 to feed each loudspeaker of an irregular array with a signal having an effective directional pick-up characteristic for encoded sounds which points in the direction of that loudspeaker. However, the results achieved with irregular arrays are not psychoacoustically correct.

Two important theories of sound localisation are the "Makita" theory and the "energy vector" theory. The "Makita" theory is applicable to frequencies less than 700 Hz and has some applicability up to about 1500 Hz. According to this theory, for a loudspeaker array with *n* loudspeakers all placed at the same distance *r* from a central reference point at positions indicated by respective rectangular cartesian co-ordinates (x_i, y_i, z_i) where $i = 1, 2, \ldots, n$, the localisation of the sound fed to these loudspeakers, where g_i is the complex gain of the sound emerging from the ith loudspeaker, is given by:

$$w = \sum_{i=1}^{n} g_i \qquad \ldots.(1)$$

$$x_o = \mathrm{Re}\left\{\left(\sum_{i=1}^{n} g_i x_i\right)/rw\right\} \qquad \ldots.(2)$$

$$y_o = \mathrm{Re}\left\{\left(\sum_{i=1}^{n} g_i y_i\right)/rw\right\} \qquad \ldots.(3)$$

$$z_o = \mathrm{Re}\left\{\left(\sum_{i=1}^{n} g_i z_i\right)/rw\right\} \qquad \ldots.(4)$$

where "Re" means "the real part of" and (x_o, y_o, z_o) is a vector pointing to the apparent localisation of the sound with respect to the origin of the cartesian co-ordinates.

For frequencies in the range from approximately 700 Hz to 5kHz, the "energy vector" theory of localisation is appropriate, the apparent sound direction being the direction of the vector sum of a set of vectors, one pointing to each loudspeaker with a respective length equal to the energy gain of the sound at that loudspeaker. Then, with a loudspeaker array as described above, the energy vector localisation is the direction of the vector (x_E, y_E, z_E) given by:

$$x_E = \left(\sum_{i=1}^{n} |g_i|^2 x_i\right)/\left(\sum_{i=1}^{n} |g_i|^2 r\right) \qquad \ldots.(5)$$

$$y_E = \left(\sum_{i=1}^{n} |g_i|^2 y_i\right)/\left(\sum_{i=1}^{n} |g_i|^2 r\right) \qquad \ldots.(6)$$

$$z_E = \left(\sum_{i=1}^{n} |g_i|^2 z_i\right)/\left(\sum_{i=1}^{n} |g_i|^2 r\right) \ldots.(7)$$

The present invention is concerned with the provision of a decoder for an irregular loudspeaker layout which satisfies both the "Makita" and the "energy vector" theories.

According to the invention, there is provided a decoder for feeding an irregular array (as hereinbefore defined) of *m* (being three or more) pairs of diametrically opposite loudspeakers, each loudspeaker being disposed substantially at an equal distance *r* from the common reference point, comprising an amplitude matrix circuit so arranged that, in operation, the sum of the signals S_i^+ and S_i^- fed to the loudspeakers of each pair is the same for all pairs of loudspeakers, and such that, if the ith pair of loudspeakers has cartesian coordinates (x_i, y_i, z_i) and $(-x_i, -y_i, -z_i)$ with respect to rectangular cartesian axes *x*, *y* and *z* at the reference point,

$$S_i^+ = W' + \alpha_i X' + \beta_i Y' + \gamma_i Z' - \delta_i jW_i''$$

$$S_i^- = W' - \alpha_i X' - \beta_i Y' - \gamma_i Z' + \delta_i jW_i''$$

where W′ is a signal representative of the acoustical pressure at the reference point and is independent of *i*,

X′, Y′ and Z′ are signals representative of the components of a desired acoustical velocity along the *x*, *y* and *z* axes and are independent of *i*,

jW_i'' is any signal bearing a 90° phase relationship to W′ for all encoded sound directions, and

$\alpha_i, \beta_i, \gamma_i$, and δ_i are real gain coefficients such that α_i, β_i, and γ_i substantially satisfy the following matrix equation:

$$KM = \frac{k \quad mrI}{\sqrt{2}}$$

where K is the $m \times 3$ matrix:

$$\begin{pmatrix} x_1 & x_2 & & x_m \\ y_1 & y_2 & \cdots & y_m \\ z_1 & z_2 & & z_m \end{pmatrix}$$

M is the $3 \times m$ matrix of coefficients:

$$\begin{pmatrix} \alpha_1 & \beta_1 & \gamma_1 \\ \alpha_2 & \beta_2 & \gamma_2 \\ & \vdots & \\ \alpha_m & \beta_m & \gamma_m \end{pmatrix}$$

I is the identity matrix:

$$\begin{pmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{pmatrix}$$ for a three-dimensional loudspeaker layout

or $$\begin{pmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 0 \end{pmatrix}$$ for a two-dimensional horizontal loudspeaker layout,

and k is a positive real constant which may be frequency dependent. It can be shown that the condition $S_i^+ + S_i^- = 2W'$ for all i is sufficient to ensure that the Makita and energy vector localisations always coincide.

It should be understood that, although jW_i'' may be the same for all pairs of diametrically opposite loudspeaker, this signal may also differ for different loudspeaker pairs provided that each signal bears a 90°phase relationship to W' for all encoded sound directions.

When the invention is to be applied to a decoder having a "WXY" circuit, as described in U.K. Patent Specification No. 1,494,751, having outputs W, X, Y such that the intended direction of sound localisation is an azimuth ø, measured anticlockwise from the x-axis, where:

$$\cos ø : \sin ø = \mathrm{Re}\,(X/W) : \mathrm{Re}\,(Y/W) \qquad (8)$$

then a decoder in accordance with the invention for feeding an irregular horizontal array of loudspeakers consisting of m diametrically opposite pairs of loudspeakers (where m is 3 or more) produces signals to be fed to the loudspeakers of each pair given by S_i^+ and S_i^-, where $i = 1, 2 \ldots, m$ and

$$S_i^+ = W' + \alpha_i X' + \beta_i Y' - \delta_i jW_i'' \qquad (9)$$
$$S_i^- = W' - \alpha_i X' - \beta_i Y' + \delta_i jW_i'' \qquad (10)$$

where α_i, β_i and δ_i are real gains, arranged such that the apparatus sound localisation according to Makita's theory is substantially equal to the azimuth ø. It will be understood that, in equation (8), the convention has been used of letting the symbols W, X and Y representing signals also denote the complex gains of these signals for a given single encoded sound direction.

If desired, the gains of the signals W, X and Y may be altered provided that the gains in the X and Y channels are identical and the phase responses in all three channels are identical. Gains applied may be frequency-dependent. A fourth signal path is provided for conveying a signal proportional to $-jW_i''$ which is used to apply directional biasing as described in U.K. Patent Specification No. 1,550,627, the biasing signals applied to the loudspeakers of each pair being of equal magnitude but opposite polarity.

Embodiments of the invention will now be described by way of example with reference to the accompanying drawings, in which:

Figure 1 is a block schematic diagram of a decoder for a horizontal surround sound decoder in accordance with the invention,

Figure 2 is a block schematic diagram of part of an amplitude matrix for the decoder shown in Figure 1,

Figure 3 shows an irregular hexagonal loudspeaker array suitable for use with the decoder shown in Figure 1,

Figure 4 shows an irregular octahedral loudspeaker array,

Figure 5 is a block schematic diagram of a decoder in accordance with the invention for use with the loudspeaker array shown in Figure 4,

Figure 6 is an irregular three dimensional array of eight loudspeakers, and

Figure 7 is a block schematic diagram of a decoder in accordance with the invention for use with a loudspeaker array as shown in Figure 4 or 6.

Referring to Figure 1, a decoder for a horizontal surround sound system has a WXY circuit 10 arranged to receive coded input signals and produce output signals W, X and Y. In addition, the circuit 10 produces a second output W phase-shifted by 90° to give the signal −jW. The signal W is applied to a type I shelf filter 12 to produce the signal W'. The signals X and Y are applied to respective type II shelf filters 14 and 16 and respective high-pass filters 18 and 20 to produce the signals X' and Y', and the signal −jW is applied to a type III shelf filter 22 and a high-pass filter 24 to produce the signal −jW". The shelf filters 12, 14 and 16 have substantially identical phase responses, and are used to achieve a different ratio of velocity to pressure information at the reference listening position at low frequencies, for example less than 400 Hz, and at high frequency, for example greater than 700 Hz. The high-pass filters 18, 20 and 24 are used to compensate for curvature of the sound field due to finite loudspeaker distance and optimally have their −3dB points at a frequency (53/r) Hz where r is the distance of the loudspeakers from the reference point in metres. The signal −jW" is used to apply directional biasing. The nature and functions of the various filters 12 to 24 is more fully described in U.K. Patent Specifications Nos. 1,494,751, 1,494,752 and 1,550,627.

The signals W', X', Y', −jW" are applied to an amplitude matrix 26. Referring to Figure 2, the matrix 26 comprises a $3 \times m$ amplitude matrix 28, to which the signals X', Y' and −jW" are applied and which produces m outputs, $V_1 - \delta_1\, jW''$ to $V_m - \delta_m\, jW''$, one for each pair of loudspeakers. The matrix 28 comprises a $2 \times m$ amplitude matrix 30, to which the signals X' and Y' are applied and which produces m outputs V_1 to V_m, a $1 \times m$ amplitude matrix 32, to which the signal −jW" is applied and which produces m directional biasing signals $-\delta_1\, jW''$ to $-\delta_m\, jW''$, and m addition circuits 34 for adding $-\delta_i\, jW''$ to V_i to produce a respective signal $V_i - \delta_i\, jW''$ for each pair of loudspeakers, where the real coefficients δ_i are chosen to achieve the desired degree of directional biasing. In addition, the matrix 26 includes an addition circuit 36 and a subtraction circuit 38 for each pair of loudspeakers of which only the circuits for the ith loudspeaker pair are shown in Figure 2. The signal W′ and the output $V_i - \delta_i jW''$ are applied to the addition circuit 36, the output of which comprises the signal:

$$S_i^+ = W' + V_i - \delta_i jW'' \quad \ldots(11)$$

and forms a feed signal for one of the loudspeakers of the *i*th pair. The signal W′ is also applied to the positive input of the subtractor 38 and the signal $V_i - \delta_i jW''$ from the amplitude matrix 34 is applied to the negative input thereof, the output of which is given by:

$$S_i^- = W' - V_i + \delta_i jW'' \quad \ldots(12)$$

and forms the feed signal for the other loudspeaker of the ith pair.

It will be understood that, if no directional biasing is required, then $\delta_i = 0$ for every *i*, and that in that case all parts of the circuit of Figure 1 and Figure 2 concerned with the handling of the signal –jW may be omitted. Also, it will be understood that any amplitude matrix producing outputs identical to those of the circuit 26 falls within the scope of the invention, and that, in particular, it may often be convenient to perform the addition of the bias signal –jW″ prior to the amplitude matrix 28 rather than subsequent to it.

Since the amplitude matrix 28 having matrix coefficients such that:

$$V_i = \alpha_i X' + \beta_i Y' \quad \ldots(13)$$

that is

$$\begin{pmatrix} V_1 \\ \vdots \\ V_m \end{pmatrix} = \begin{pmatrix} \alpha_1 & \beta_1 \\ \vdots & \vdots \\ \alpha_m & \beta_m \end{pmatrix} \begin{pmatrix} X' \\ Y' \end{pmatrix} \quad \ldots(14)$$

is required to satisfy Makita localisation criteria of Equation (8), the coefficients α_i and β_i of the matrix must satisfy the equations:

$$\sum_{i=1}^{m} \alpha_i x_i = \sum_{i=1}^{m} \beta_i y_i = \frac{mr}{\sqrt{2}} \quad \ldots(15)$$

$$\sum_{i=1}^{m} \alpha_i y_i = \sum_{i=1}^{m} \beta_i x_i = 0 \quad \ldots(16)$$

Since $S_i^+ + S_i^- = 2W'$ for such an amplitude matrix, it also follows that the energy vector localisation coincides with the Makita localisation for this amplitude matrix.

If we write the $2 \times m$ matrix of the coefficients:

$$\begin{pmatrix} \alpha_1 & \beta_1 \\ \vdots & \vdots \\ \alpha_m & \beta_m \end{pmatrix}$$

of the matrix 28 as M and the $m \times 2$ matrix

$$\begin{pmatrix} x_1 & \cdots & x_m \\ y_1 & & y_m \end{pmatrix}$$

as K, then the Equations 15 and 16 can be rewritten in matrix form as:

$$KM = \frac{1}{\sqrt{2}} mrI \quad \ldots(17)$$

where I is the 2×2 identity matrix and *r* is the distance of the loudspeakers from the reference listening position as before.

In practice, any positive real multiple *k* of the matrix M satisfying Equation 17 may be used, that is one can multiply all gain α_i, β_i by a fixed positive gain *k*. However, it is preferably to use a multiple *k* of M which also ensures the condition:

$$k^2\{(\mathrm{Re}\,(X'/W'))^2 + (\mathrm{Re}\,(Y'/W'))^2\} = 2 \quad \ldots(18)$$

is satisfied, since, if this condition is met, not only the Makita theory, but also other low frequency localisation theories are satisfied. This last mentioned condition is satisfied, for example, when W′ has unity gain for all sounds, and X′ has gain $\sqrt{2}\cos\Theta$ and Y′ has gain $\sqrt{2}\sin\Theta$ for a sound originating from an azimuth Θ measured anticlockwise from the front direction and when *k* equals 1.

The constant *k* may be implemented by means of gain or shelf filter circuits affecting the signals X′, Y′ and W′ prior to the final output matrix circuitry, and additional changes of gain, phase response and frequency response may be applied to these signals, provided that all the signals are affected equally by these additional changes.

A convenient way of devising a matrix M satisfying the condition:

$$KM = \frac{k}{\sqrt{2}} mrI \quad \ldots(19)$$

, and therefore giving correct localisation, is as follows.

For each pair of loudspeakers:

$$\begin{pmatrix} \alpha_i \\ \beta_i \end{pmatrix} = \frac{k}{\sqrt{2}} mr \left(\sum_{h=1}^{m} \begin{pmatrix} x_h^2 & x_h y_h \\ x_h y_h & y_h^2 \end{pmatrix} \right)^{-1} \begin{pmatrix} x_i \\ y_i \end{pmatrix} \quad \ldots(20)$$

where the power −1 indicates a matrix inverse.

The application of Equation 20 to the irregular hexagonal loudspeaker array shown in Figure 3 will now be described. The array of Figure 3 consists of a due left loudspeaker L, a due right loudspeaker R and four loudspeakers LB, LF, RF and RB placed at respective azimuths 180°−ø, ø, −ø, and −180° + ø measured anticlockwise from due front. Putting due front as the *x* direction, due left as the *y* direction and $S_1^+, S_2^+, S_3^+, S_1^-, S_2^-, S_3^-$ equal to the signals fed to the respective loudspeakers LB, L, LF, RF, R, RB we have:

$(x_1, y_1) = (-r\cos\varnothing, r\sin\varnothing)$ ....(21)

$(x_2, y_2) = (O, r)$ ....(22)

$(x_3, y_3) = (r\cos\varnothing, r\sin\varnothing)$ ....(23)

so that:

$$\begin{pmatrix}\alpha_i\\ \beta_i\end{pmatrix} = \frac{3r^{-1}k}{\sqrt{2}}\begin{pmatrix}2\cos^2\phi & 0\\ 0 & 1+2\sin^2\phi\end{pmatrix}^{-1}\begin{pmatrix}x_i\\ y_i\end{pmatrix}$$

$$= \frac{r^{-1}k}{\sqrt{2}}\begin{pmatrix}\frac{3}{2\cos^2\phi} & 0\\ 0 & \frac{3}{1+2\sin^2\phi}\end{pmatrix}\begin{pmatrix}x_i\\ y_i\end{pmatrix} \quad \ldots(24)$$

From Equation 24:

$$M = \begin{pmatrix}\alpha_1 & \beta_1\\ \alpha_2 & \beta_2\\ \alpha_3 & \beta_3\end{pmatrix} = \frac{k}{\sqrt{2}}\begin{pmatrix}\frac{-3}{2\cos\phi} & \frac{3\sin\phi}{1+2\sin^2\phi}\\ 0 & \frac{3}{1+2\sin^2\phi}\\ \frac{3}{2\cos\phi} & \frac{3\sin\phi}{1+2\sin^2\phi}\end{pmatrix} \quad \ldots(25)$$

and the amplitude matrix 28 in Figures 1 and 2 feeds the following signals to the loudspeakers of Figure 3:

$$S_1^+ = W' - \frac{3k}{2\sqrt{2}\cos\phi}X' + \frac{3k\sin\phi}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(26)$$

$$S_2^+ = W' + \frac{3k}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(27)$$

$$S_3^+ = W' + \frac{3k}{2\sqrt{2}\cos\phi}X' + \frac{3k\sin\phi}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(28)$$

$$S_1^- = W' + \frac{3k}{2\sqrt{2}\cos\phi}X' - \frac{3k\sin\phi}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(29)$$

$$S_2^- = W' - \frac{3k}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(30)$$

$$S_3^- = W' - \frac{3k}{2\sqrt{2}\cos\phi}X' - \frac{3k\sin\phi}{\sqrt{2}(1+2\sin^2\phi)}Y' \quad \ldots(31)$$

The matrix coefficients of the amplitude matrix 30 may have any real value chosen to provide the required directional biasing, if any, or alternatively directional biasing may be achieved by modifying the signals X' and Y' as described in above-mentioned Patent Specification No. 1,550,627.

It should be understood that, in all these decoders, the signals X and Y from the WXY circuit 10 may be replaced by two independent real linear combinations of X and Y provided that the amplitude matrix 26 derives from these linear combinations the required output signals S_i^+ and S_i^-. Moreover, matrices may be combined or rearranged in the circuitry wherever this is of design or contructional convenience so that a part of the output amplitude matrix function might, for example, be combined with the function of the WXY circuit.

It will be appreciated that the gains α_1, α_2 and α_3, β_1, β_2 and β_3 of the above decoder for a hexagonal loudspeaker layout depend on the angle Ø, and that it will often be desirable to incorporate means for providing a continuous adjustment of the value of Ø in the decoder circuit. To this end the gains $\alpha_1 = -\alpha_3$ (which result in a signal component $\alpha_1X' = -\alpha_3X'$) may be implemented by a first variable gain circuit placed in the X' signal path, the gain β_2 (which results in a signal component β_2Y') may be implemented by a second variable gain circuit placed in the Y' signal path, and the gains $\beta_1 = \beta_3$ (which result in a signal component $\beta_1Y' = \beta_3Y'$) may be implemented by a third variable gain circuit placed in the Y' signal path. Simultaneous adjustment of these three variable gain circuits will then permit the decoder to be adapted to loudspeaker layouts with various different values of Ø.

The invention may also be applied to irregular three-dimensional loudspeaker arrays where the loudspeakers are placed in *m* diametrically opposite pairs at a distance *r* from the reference listening point. In the following discussion it is assumed that the *i*th of *m* pairs of loudspeakers have positions given by the cartesian coordinates (x_i, y_i, z_i) and $(-x_i, -y_i, -z_i)$ and are fed with respective signals S_i^+ and S_i^-. W, X, Y and Z are signals representative respectively of the desired pressure and x-axis, y-axis, and z-axis components of velocity of sound at the reference listening position. Such signals may be subjected to shelf filters having identical phase responses and to RC high-pass filters compensating for loudspeaker distance, analogous to the filters described with reference to Figure 1, provided only that the filtering on each of the X, Y and Z signal paths is identical, producing modified signals W', X', Y', Z'. Then, in accordance with the invention, the Makita and energy vector localisation theories give the same direction of sound provided that:

$S_i^+ + S_i^- = 2W'$ for $i=1,2,\ldots,m$ ....(32)

In addition, it is often desired that this localisation be at the direction of the point (Re (X/W), Re (Y/W), Re (Z/W)), and in that case the signals S_i^+ and S_i^- are given by:

$S_i^+ = W' + \alpha_iX' + \beta_iY' + \gamma_iZ' - \delta_ijW_i''$ ....(33)

$S_i^- = W' - \alpha_iX' - \beta_iY' - \gamma_iZ' + \delta_ijW_i''$ ....(34)

where α_i, β_i, γ_i and δ_i are real coefficients, where jW_i'' is any signal having a 90° phase relation to W' for all sound directions, and where the $3 \times m$ matrix:

$$M = \begin{pmatrix} \alpha_1 & \beta_1 & \gamma_1 \\ \alpha_2 & \beta_2 & \gamma_2 \\ & \vdots & \\ \alpha_m & \beta_m & \gamma_m \end{pmatrix} \quad \ldots(35)$$

satisfies the matrix equation:

$$KM = \frac{1}{\sqrt{2}}\, kmrI \quad \ldots.(36)$$

where k is a positive constant and

$$K = \begin{pmatrix} x_1 & & x_m \\ y_1 & \cdots & y_m \\ z_1 & & z_m \end{pmatrix} \quad \ldots(37)$$

and I is the 3×3 identity matrix and where δ_i are the arbitrary real coefficients of directional biasing signals.

The Equation (36) may alternatively be written as:

$$\sum_{i=1}^{m} \alpha_i x_i = \sum_{i=1}^{m} \beta_i y_i = \sum_{i=1}^{m} \gamma_i z_i = \frac{kmr}{\sqrt{2}}$$

$$\sum_{i=1}^{m} \alpha_i y_i = \sum_{i=1}^{m} \alpha_i z_i = \sum_{i=1}^{m} \beta_i x_i = \sum_{i=1}^{m} \beta_i z_i =$$

$$\sum_{i=1}^{m} \gamma_i x_i = \sum_{i=1}^{m} \gamma_i y_i = 0$$

In particular, the matrix M may be given by the equation:

$$\begin{pmatrix} \alpha_i \\ \beta_i \\ \gamma_i \end{pmatrix} = \frac{1}{\sqrt{2}}\, kmr \left\{ \sum_{h=1}^{m} \begin{pmatrix} x_h^2 & x_h y_h & x_h z_h \\ x_h y_h & y_h^2 & y_h z_h \\ x_h z_h & y_h z_h & z_h^2 \end{pmatrix} \right\}^{-1} \begin{pmatrix} x_i \\ y_i \\ z_i \end{pmatrix} \quad \ldots..(38)$$

A matrix M satisfying Equation 36 yields correct localisation according to all major low frequency localisation theories provided that the constant k is chosen to ensure that:

$$k^2\{(\mathrm{Re}\,(X'/W'))^2 + (\mathrm{Re}\,(Y'/W'))^2 + (\mathrm{Re}(Z'/W'))^2\} = 2 \quad \ldots\ldots(39)$$

for encoded sounds.

The constant k may be implemented by means of gain or shelf filter circuits affecting the signals X', Y', Z' and W' prior to the final output matrix circuitry, and additional changes of gain, phase response and frequency response may be applied to these signals, provided that all the signals are affected equally by these additional changes.

For horizontally encoded sounds, Z = O, in which case the Z signal path may be omitted, and the system reduces to that previously described with reference to Figures 1 and 2, except that the values of α_i and β_i may be somewhat altered in accordance with Equation 36.

Figure 4 indicates an irregular octahedral layout of six loudspeakers F, B, LU, LD, RU and RD placed at a distance r from a reference point and respectively disposed in front, behind, at an angle ø above due left, at an angle ø below due left, at an angle ø above due right, and at an angle ø below due right. The corresponding loudspeaker feed signals S_1^+, S_1^-, S_2^+, S_3^-, S_3^+, S_2^-, are fed to the loudspeakers at $\pm (x_i, y_i, z_i)$ where:

$$(x_1, y_1, z_1) = (r,\ 0\ \ 0)$$
$$(x_2, y_2, z_2) = (0, r\cos ø, r\sin ø)$$
$$(x_3, y_3, z_3) = (0, -r\cos ø, r\sin ø) \quad \ldots\ldots(40)$$

Use of the above-mentioned matrix Formula 38 gives:

$$\begin{matrix} \alpha_1 & \beta_1 & \gamma_1 \\ \alpha_2 & \beta_2 & \gamma_2 \\ \alpha_3 & \beta_3 & \gamma_3 \end{matrix} = \frac{k}{\sqrt{2}} \begin{pmatrix} 3 & 0 & 0 \\ 0 & \frac{3}{2\cos\phi} & \frac{3}{2\sin\phi} \\ 0 & \frac{-3}{2\cos\phi} & \frac{3}{2\sin\phi} \end{pmatrix} \quad \ldots(41)$$

so that the loudspeaker feed signals are:

$$S_1^+ = W' + \frac{3k}{\sqrt{2}} X'$$

$$S_1^- = W' - \frac{3k}{\sqrt{2}} X'$$

$$S_2^+ = W' + \frac{3k}{2\sqrt{2}\cos\phi} Y' + \frac{3k}{2\sqrt{2}\sin\phi} Z'$$

$$S_2^- = W' - \frac{3k}{2\sqrt{2}\cos\phi} Y' - \frac{3k}{2\sqrt{2}\sin\phi} Z'$$

$$S_3^+ = W' - \frac{3k}{2\sqrt{2}\cos\phi} Y' + \frac{3k}{2\sqrt{2}\sin\phi} Z'$$

$$S_3^- = W' + \frac{3k}{2\sqrt{2}\cos\phi} Y' - \frac{3k}{2\sqrt{2}\sin\phi} Z' \quad \ldots(42)$$

Figure 5 illustrates a decoder for use in the case when the signal W has unit gain for sounds encoded from all directions in space, and where X, Y and Z have respective gains $\sqrt{2}\cos\Theta\cos\eta$, $\sqrt{2}\sin\Theta\cos\eta$ and $\sqrt{2}\sin\eta$ for sounds having source azimuth Θ measured anticlockwise from due front and source elevation measured upwards from horizontal such as may occur in the decoders of certain four-channel encoding systems with full-sphere directionality. The signals W, X, Y and Z are produced from the received input signals by a WXYZ circuit 40. The W signal is applied to a type I shelf filter 42 while the X, Y and Z signals are applied to respective type II shelf filters 44, 46 and 48. The function of the shelf filters 42 to 48 is analogous to that of shelf filters 12, 14 and 16 of

Figure 1 and the transition frequency between low and high frequency gains is preferably centred at about 350 Hz, the shelf filters of both types having unity gain at low frequencies while the type I shelf filter has gain $\sqrt{2}$ and the type II shelf filters have gain $\sqrt{2/3}$ at frequencies well above the transition frequencies. The ratio of gains of the type II shelf filters to the type I shelf filter may be considered to implement a part of the factor k referred to in Equations (41) and (42). In this case it will be seen that k is a frequency dependent gain. The X, Y and Z signal paths also include high-pass filters 50, 52 and 54 to compensate for sound field curvature due to finite loudspeaker distance as previously described.

The X, Y and Z signal paths also include amplifiers 56, 58 and 60 applying respective gains I, II and III in order to implement matrix Equation 35. For the loudspeaker layout of Figure 4, these gains are given by:

$$\text{gain I} = \frac{3}{\sqrt{2}}$$

$$\text{gain II} = \frac{3}{2\sqrt{2}\cos\phi}$$

$$\text{gain III} = \frac{3}{2\sqrt{2}\sin\phi}$$

The output signals in the Y and Z channels from the amplifiers 58 and 60 respectively are added by an addition circuit 62 to give the difference signals for the LU and RD pair of loudspeakers, and subtracted by a subtraction circuit 64 to obtain the difference signal for the RU and LD pair of loudspeakers. The output signal in the X channel, from the amplifier 56, itself constitutes the difference signal for the F and B pair of loudspeakers. Each of these difference signals is combined by a respective addition circuit 66, 68 and 70 to give the signals S_1^+, S_2^+, and S_3^+ which are amplified by respective power amplifiers 72, 74 and 76 and fed through the loudspeakers F, LU and RU.

The output signal in the W channel from the shelf filter 42 is also applied to an amplifier 78 having a gain of 2 and thence to a power amplifier 80 having equal gain to that of the power amplifiers 72, 74 and 76. Each of the signals S_1^+, S_2^+, and S_3^+ is subtracted from the output of the amplifier 80 by connecting a respective one of the loudspeakers B, RD and LD between the output of the amplifier 80 and the output of the corresponding one of the amplifiers 72, 74 and 76. Thus only four power amplifiers are needed to feed the six loudspeakers. This so-called loudspeaker matrixing technique forms the subject of U.K. Patent Specification No. 1,548,674 and may also be applied to decoders for feeding horizontal loudspeaker arrays in accordance with the present invention, such as the decoder illustrated in Figures 2 and 3.

It will be appreciated that other spatial orientations of the octahedral layout of Figure 4 may be used, provided that the signals X, Y and Z are matrixed or interchanged to correspond to components of sound velocity along the reorientated spatial axes.

The invention may also be applied to more complex irregular loudspeaker layouts. For example, the invention may be applied to a three dimensional layout of eight loudspeakers LF, RF, LB, RB, LU, LD, RU and RD as shown in Figure 6, placed at the cartesian co-ordinates (x_i, y_i, z_i) and $(-x_i, -y_i, -z_i)$ with respective feed signals S_i^+ and S_i^- of the form given in Equations (33) and (34), where i has the values 1 to 4, and where, for radius r:

$$(x_1, y_1, z_1) = (r\cos\phi, r\sin\phi, o)$$
$$(x_2, y_2, z_2) = (r\cos\phi, -r\sin\phi, o)$$
$$(x_3, y_3, z_3) = (o, r\cos\xi, r\sin\xi)$$
$$(x_4, y_4, z_4) = (O, -r\cos\xi, r\sin\xi) \qquad (44)$$

This corresponds to a loudspeaker layout consisting of a combination of a horizontal array of four loudspeakers with an angle 2ϕ subtended at the centre by the front loudspeaker pair, and a vertical rectangular array of four loudspeakers with an angle 2ξ subtended at the centre by one of the vertical loudspeaker pairs.

Such a loudspeaker layout can be made to satisfy the directional requirements of the Makita and energy vector theories if one applies Equation (38) to the layout. A calculation then shows that:

$$\begin{pmatrix} \alpha_1 & \beta_1 & \gamma_1 \\ \alpha_2 & \beta_2 & \gamma_2 \\ \alpha_3 & \beta_3 & \gamma_3 \\ \alpha_4 & \beta_4 & \gamma_4 \end{pmatrix} = \sqrt{2}\,k \begin{pmatrix} \frac{1}{\cos\phi} & \frac{\sin\phi}{(\sin^2\phi + \cos^2\xi)} & 0 \\ \frac{1}{\cos\phi} & \frac{-\sin\phi}{(\sin^2\phi + \cos^2\xi)} & 0 \\ 0 & \frac{\cos\xi}{(\sin^2\phi + \cos^2\xi)} & \frac{1}{\sin\xi} \\ 0 & \frac{-\cos\xi}{(\sin^2\phi + \cos^2\xi)} & \frac{1}{\sin\xi} \end{pmatrix}$$

so that the loudspeaker feed signals are given by Equations (33) and (34) by using these values of α_i, β_i, γ_i for a suitable positive gain k (which may be chosen to be frequency dependent).

Figure 7 illustrates a decoder for use with a variety of three dimensional loudspeaker layouts in accordance with this invention, including those described above in reference to Figures 4 and 6. This decoder is also suitable for use with a cuboid of loudspeakers as described in U.K. Patent Specifications Nos. 1,494,751 and 1,494,752 and incorporates a WXYZ circuit 90, type I and II shelf filters 92, 94, 96 and 98 and also high-pass filters 100, 102 and 104 to compensate for loudspeaker distance as described in the aforementioned specifications. The decoder also incorporates a switchable amplitude matrix 114. By providing several variable gain amplifiers 106, 108, 110 and 112, and by making the output amplitude matrix coefficients switchable to match the type of loudspeaker layout chosen, a single decoder can be made which is suitable for a number of different loudspeaker layouts. In particular, the variable gain amplifiers permit adjustment of the angles ϕ and ξ describing the exact shape of the loudspeaker layout and thus act as a "layout control".

Any of the decoders described above can be used in conjunction with additional gain and time delay circuitry which serves to modify the output signals from the decoder prior to feeding these to the loudspeakers in order to compensate for loudspeakers at unequal distances from the common reference point, in accordance with the provisions of U.K. Patent Specification No. 1,552,478.

It will also be appreciated that the designation of the x-axis as being "forward", the y-axis as being "leftward" and the z-axis as being "upward" in this specification is purely arbitrary, and that x, y and z axes could equally as well be chosen to be any other set of 3 orthogonal cartesian axes at the common reference point. Thus, for example, by making the x-axis point leftward and the y-axis point forward, the decoders described with reference to Figures 3 to 6 will be suitable for alternative orientations of loudspeaker layouts. Thus, the L loudspeaker of Figure 3 will become a front loudspeaker, the R loudspeaker will become a back loudspeaker, and the left front, left back, right front, right back loudspeaker will become respectively left front, right front, left back and right back loudspeakers. In a similar way, the octahedral layout of Figure 4 will consist of front and back vertical pairs of speakers and one loudspeaker at each side. Finally, the layout of Figure 6 will consist of front and back vertical pairs of loudspeakers and left and right side pairs of loudspeakers.

It will also be appreciated that the amplitude matrix described above may also incorporate any additional overall gain (including phase inversion where appropriate) such as might be considered desirable by one skilled in the art.

CLAIMS

1. A decoder for feeding an irregular array (as hereinbefore defined) of m (being three or more) pairs of diametrically opposite loudspeakers, each loudspeaker being disposed substantially at an equal distance r from a common reference point, comprising an amplitude matrix circuit so arranged that, in operation, the sum of the signals S_i^+ and S_i^- fed to the loudspeakers of each pair is the same for all pairs of loudspeakers, and such that, if the ith pair of loudspeakers has cartesian coordinates (x_i, y_i, z_i) and $(-x_i, -y_i, z_i)$ with respect to rectangular cartesian axes x, y, and z at the reference point,

$$S_i^+ = W' + \alpha_i X' + \beta_i Y' + \gamma_i Z' - \delta_i jW_i''$$

$$S_i^- = W' - \alpha_i X' - \beta_i Y' - \gamma_i Z' + \delta_i jW_i''$$

where W′ is a signal representative of the acoustical pressure at the reference point and is independent of i,

X′, Y′ and Z′ are signals representative of the components of a desired acoustical velocity along the x, y and z axes and are independent of i,

jW_i'' is any signal bearing a 90° phase relationship to W′ for all encoded sound directions, and

$\alpha_i, \beta_i, \gamma_i$, and δ_i are real gain coefficients such that α_i, β_i and γ_i substantially satisfy the following matrix equation:

$$KM = \frac{k \ \ mrI}{\sqrt{2}}$$

where

K is the $m \times 3$ matrix:

$$\begin{pmatrix} x_1 & x_2 & \cdots & x_m \\ y_1 & y_2 & \cdots & y_m \\ z_1 & z_2 & & z_m \end{pmatrix}$$

M is the $3 \times m$ matrix of coefficients:

$$\begin{pmatrix} \alpha_1 & \beta_1 & \gamma_1 \\ \alpha_2 & \beta_2 & \gamma_2 \\ & \vdots & \\ \alpha_m & \beta_m & \gamma_m \end{pmatrix}$$

I is the identity matrix:

$$\begin{pmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{pmatrix}$$ for a three-dimensional loudspeaker layout

or

$$\begin{pmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 0 \end{pmatrix}$$ for a two-dimensional, horizontal loudspeaker, layout,

and k is a positive real constant which may be frequency dependent.

2. A decoder according to claim 1, wherein jW_i'' is the same for all pairs of diametrically opposite loudspeakers.

3. A decoder according to claim 1 or 2, for a two-dimensional loudspeaker layout, wherein the amplitude matrix is so arranged that, if the ith pair of loudspeakers has cartesian coordinates (x_i, y_i) and $(-x_i, -y_i)$ with respect to rectangular cartesian axes x and y at the reference point,

$$S_i^+ = W' + \alpha_i X' + \beta_i Y' - \delta_i jW_i''$$

$$S_i^- = W' - \alpha_i X' - \beta_i Y' + \delta_i jW_i''$$

where α_i, β_i and δ_i are real gain coefficients such that α_i and β_i substantially satisfy the following equations:

$$\sum_{i=1}^{m} \alpha_i x_i = \sum_{i=1}^{m} \beta_i y_i = \frac{kmr}{\sqrt{2}}$$

$$\sum_{i=1}^{m} \alpha_i y_i = \sum_{i=1}^{m} \beta_i x_i = 0$$

4. A dedoder according to claim 3, wherein the gain coefficients α_i and β_i are substantially given by the matrix equations:

$$\begin{pmatrix} \alpha_i \\ \beta_i \end{pmatrix} = \frac{1}{\sqrt{2}} kmr \left(\sum_{h=1}^{m} \begin{pmatrix} x_h^2 & x_h y_h \\ x_h y_h & y_h^2 \end{pmatrix} \right)^{-1} \begin{pmatrix} x_i \\ y_i \end{pmatrix}$$

where the power –1 indicates the matrix inverse.

5. A decoder according to claim 3 or 4, wherein, considering the signal W′ as having unity gain and incorporating encoded sounds from all directions, the signal X′ has gain $\sqrt{2}\cos\Theta$, and the signal Y′ has gain $\sqrt{2}\sin\Theta$ for a sound originating from an azimuth Θ.

6. A decoder according to claim 3, 4 or 5, wherein a first shelf filter circuit is provided for producing the signal W′, and identical second shelf filter circuits are provided for producing the signals X′ and Y′.

7. A decoder according to claim 6, wherein the first and second shelf filter circuits have substantially identical phase responses at all audio frequencies.

8. A decoder according to any one of claims 3 to 7, wherein the constant k at low frequencies is such as to ensure that:

$$k^2\{(\mathrm{Re}\,(X'/W'))^2 + (\mathrm{Re}\,(Y'/W'))^2\} = 2$$

for all horizontal sounds encoded into the signals W′, X′ and Y′, where Re denotes "the real part of".

9. A decoder according to claim 4, for feeding respective signals S_1^+, S_1^-, S_2^+, S_2^-, S_3^+ and S_3^- to an irregular arrangement of six loudspeakers placed at the respective cartesian coordinates:

$(x_1^+, y_1^+) = (-r\cos\varnothing, r\sin\varnothing)$,
$(x_1^-, y_1^-) = (r\cos\varnothing, -r\sin\varnothing)$,
$(x_2^+, y_2^+) = (0, r)$,
$(x_2^-, y_2^-) = (0, -r)$,
$(x_3^+, y_3^+) = (r\cos\varnothing, r\sin\varnothing)$,
$(x_3^-, y_3^-) = (-r\cos\varnothing, -r\sin\varnothing)$,

where $$\alpha_1 = \frac{-3k}{2\sqrt{2}\cos\varnothing},$$

$$\beta_1 = \beta_3 = \frac{(3\sin\varnothing)k}{\sqrt{2}(1+2\sin^2\varnothing)},$$

$$\alpha_2 = 0,$$

$$\beta_2 = \frac{3k}{\sqrt{2}(1+2\sin^2\varnothing)}, \text{ and}$$

$$\alpha_3 = \frac{3k}{2\sqrt{2}\cos\varnothing}.$$

10. A decoder according to any one of claims 3 to 9, wherein the amplitude matrix circuit comprises adjustment means for matching a range of loudspeaker arrangements by adjusting the gains of the signals X′ and Y′ before they are fed into a fixed matrix circuit.

11. A decoder according to claim 10 when appended to claim 9, wherein a first variable gain circuit is provided for multiplying the signal X′ by the gain coefficients α_1 and α_3, a second variable gain circuit is provided for multiplying the signal Y′ by the gain coefficient β_2, and a third variable gain circuit is provided for multiplying the signal Y′ by the gain coefficients β_1 and β_3.

12. A decoder according to claim 1 or 2, for a three-dimensional loudspeaker layout, wherein the gain coefficients α_i, β_i and γ_i substantially satisfy the following equations:

$$\sum_{i=1}^{m}\alpha_i x_i = \sum_{i=1}^{m}\beta_i y_i = \sum_{i=1}^{m}\gamma_i z_i = \frac{kmr}{\sqrt{2}},$$

$$\sum_{i=1}^{m}\alpha_i y_i = \sum_{i=1}^{m}\alpha_i z_i = \sum_{i=1}^{m}\beta_i x_i = \sum_{i=1}^{m}$$

$$\beta_i z_i = \sum_{i=1}^{m}\gamma_i x_i = \sum_{i=1}^{m}\gamma_i y_i = 0.$$

13. A decoder according to claim 12, wherein the gain coefficients α_i, β_i and γ_i are substantially given by the matrix equations:

$$\begin{pmatrix}\alpha_i\\ \beta_i\\ \gamma_i\end{pmatrix} = \frac{1}{\sqrt{2}}\,kmr\left\{\sum_{h=1}^{m}\begin{pmatrix}x_h^2 & x_h y_h & x_h z_h\\ x_h y_h & y_h^2 & y_h z_h\\ x_h z_h & y_h z_h & z_h^2\end{pmatrix}\right\}^{-1}\begin{pmatrix}x_i\\ y_i\\ z_i\end{pmatrix}$$

where the power –1 indicates the matrix inverse.

14. A decoder according to claim 12 or 13, wherein considering the signal W′ as having unity gain and incorporating sounds from all directions, the signals X′, Y′ and Z′ have gain $\sqrt{2}\cos\Theta\cos\eta$, $\sqrt{2}\sin\Theta\cos\eta$ and $\sqrt{2}\sin\eta$ for a sound having a source azimuth Θ measured anticlockwise from the x-axis and a source elevation η measured upward from the xy-plane to the z-axis.

15. A decoder according to claim 12, 13 or 14, wherein a first shelf filter circuit is provided for the signal W′ and identical second shelf filter circuits are provided for the signals X′, Y′ and Z′.

16. A decoder according to claims 15, wherein the first and second filter circuits have substantially identical phase responses at all audio frequencies.

17. A decoder according to any one of claims 12 to 16, wherein the contant k at low frequencies is such as to ensure that:

$$k^2\{(\mathrm{Re}\,(X'/W'))^2 + (\mathrm{Re}\,(Y'/W'))^2 + (\mathrm{Re}\,(Z'/W'))^2\} + 2$$

for all directional sounds encoded into the signals W′, X′, Y′ and Z′.

18. A decoder according to claim 13, for feeding respective signals S_1^+, S_1^-, S_2^+, S_2^-, S_3^+ and S_3^- to an irregular arrangement of six loudspeakers placed at the vertices of an irregular octahedron at a distance r from the origin of the cartesian coordinates.

19. A decoder according to claim 18, wherein the loudspeaker coordinates are respectively:

$(x_1^+, y_1^+, z_1^+) = (r, 0, 0)$,
$(x_1^-, y_1^-, z_1^-) = (-r, 0, 0)$,
$(x_2^+, y_2^+, z_2^+) = (0, r\cos\varnothing, r\sin\varnothing)$,
$(x_2^-, y_2^-, z_2^-) = (0, -r\cos\varnothing, -r\sin\varnothing)$,
$(x_3^+, y_3^+, z_3^+) = (0, -r\cos\varnothing, r\sin\varnothing)$,
$(x_3^-, y_3^-, z_3^-) = (0, r\cos\varnothing, -r\sin\varnothing)$,

where $\beta_1 = \gamma_1 = \alpha_2 = \alpha_3 = 0$,

$$\alpha_1 = \frac{3k}{\sqrt{2}},$$

$$\beta_2 = \frac{3k}{(2\sqrt{2}\cos \emptyset)},$$

$$\beta_3 = \frac{-\ 3k}{(2\sqrt{2}\cos \emptyset)},$$

and

$$\gamma_2 = \gamma_3 = \frac{3k}{(2\sqrt{2}\sin \emptyset)}.$$

20. A decoder according to any one of claims 12 to 19, wherein the amplitude matrix circuit comprises adjustment means for matching a range of loudspeaker arrangements by adjusting the gains of the signals X', Y' and Z' before they are fed into a fixed matrix circuit.

21. A decoder according to claim 20 when appended to claim 19, wherein a first variable gain circuit is provided for multiplying the signal X' by the gain coefficient α_1, a second variable gain circuit is provided for multiplying the signal Y' by the gain coefficient β_2 and β_3, and a third variable gain circuit is provided for multiplying the signal Z' by the gain coefficients γ_2 and γ_3.

22. A decoder according to claim 9 or 18, wherein four power amplifiers having one output terminal in common are provided for receiving signals S_1^+, S_2^+, S_3^+ and $2W' = S_1^+ + S_1^- = S_2^+ + S_2^- = S_3^+ + S_3^-$, the power amplifiers being connected to the six loudspeakers such that each of the loudspeakers requiring signals S_1^+, S_2^+ and S_3^+ is driven by a respective amplifier and each of the diametrically opposite loudspeakers requiring signals S_1^-, S_2^- and S_3^- is driven by having one terminal of the loudspeaker coupled to the non-common output terminal of a respective amplifier and the other terminal of the loudspeaker coupled to the non-common output terminal of the amplifier provided for receiving the signal 2W'.

23. A decoder according to claim 13, for feeding respective signals S_1^+, S_1^-, S_2^+, S_2^-, S_3^+, S_3^-, S_4^+ and S_4^- to an irregular arrangement of eight loudspeakers placed at the vertices of a rectangle in the xy-plane and at the vertices of a rectangle in the yz-plane at the respective cartesian coordinates:

$(x_1^+, y_1^+, z_1^+) = (r\cos \emptyset, r\sin \emptyset, 0)$,
$(x_1^-, y_1^-, z_1^-) = (-r\cos \emptyset, -r\sin \emptyset, 0)$,
$(x_2^+, y_2^+, z_2^+) = (r\cos \emptyset, -r\sin \emptyset, 0)$,
$(x_2^-, y_2^-, z_2^-) = (-r\cos \emptyset, r\sin \emptyset, 0)$,
$(x_3^+, y_3^+, z_3^+) = (0, r\cos\xi, r\sin\xi)$,
$(x_3^-, y_3^-, z_3^-) = (0, -r\cos\xi, -r\sin\xi)$,
$(x_4^+, y_4^+, z_4^+) = (0, -r\cos\xi, r\sin\xi)$,
$(x_4^-, y_4^-, z_4^-) = (0, r\cos\xi, -r\sin\xi)$,

where

$\gamma_1 = \gamma_2 = \alpha_3 = \alpha_4 = 0$,
$\alpha_1 = \alpha_2 = \sqrt{2}\, k/\cos \emptyset$,
$\beta_1 = -\beta_2 = \sqrt{2}\, k \sin \emptyset \,/\, (\sin^2 \emptyset + \cos^2 \emptyset + \cos^2\xi)$,
$\beta_3 = -\beta_4 = \sqrt{2}\, k \cos\xi / (\sin^2 \emptyset + \cos^2\xi)$,
$\gamma_3 = \gamma_4 = \sqrt{2}\, k/\sin\xi$.

24. A decoder according to claim 23, adjustable for a range of values of the angles Ø and ξ, wherein the amplitude matrix circuit comprises adjustment means for matching a range of loudspeaker arrangements by adjusting the gain of the signals X', Y' and Z' before they are fed into a fixed matrix circuit, and wherein a first variable gain circuit is provided for multiplying the signal X' by the gain coefficients α_1 and α_2, a second variable gain circuit is provided for multiplying the signal Y' by the gain coefficients β_1 and β_2, a third variable gain circuit is provided for multiplying the signal Y' by the gain coefficients β_3 and β_4, and a fourth variable gain circuit is provided for multiplying the signal Z' by the gain coefficients γ_3 and γ_4.

25. A decoder for feeding an irregular array (as hereinbefore defined) of *m* (being 3 or more) pairs of diametrically opposite loudspeakers, substantially as hereinbefore described with reference to the accompanying drawings.

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United States Patent [19]

Gerzon

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[54] **SOUND REPRODUCTION SYSTEMS**

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[52] **U.S. Cl.** .... **179/1 GQ**; 179/1 GP; 179/1 G

[58] **Field of Search** .... 179/1 GQ, 1 G, 1 GP, 179/100.1 TP, 100.4 ST, 1 GA, 1 VE

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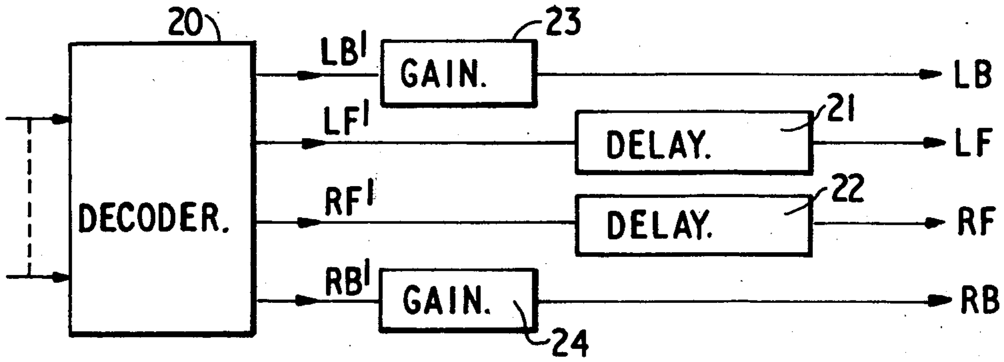
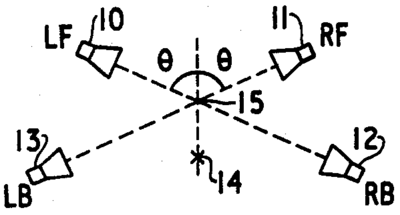
[57] **ABSTRACT**

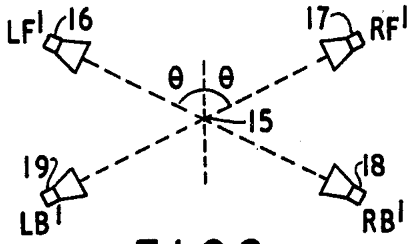
A basic decoder designed to feed signals to at least three loudspeakers disposed at respective azimuths around a reference point at equal distances therefrom is modified to make it suitable for feeding loudspeakers at the same azimuths but at non-uniform distances from the reference point. Each output of the basic decoder is subject to a time delay proportional to the difference of time of travel of sound from the first pair of loudspeakers to the reference point and the time of travel of sound from the second pair of loudspeakers to the reference point, divided by the sum of said times of travel to an adder connected between the first velocity signal input and the corresponding high-pass filter.

7 Claims, 5 Drawing Figures

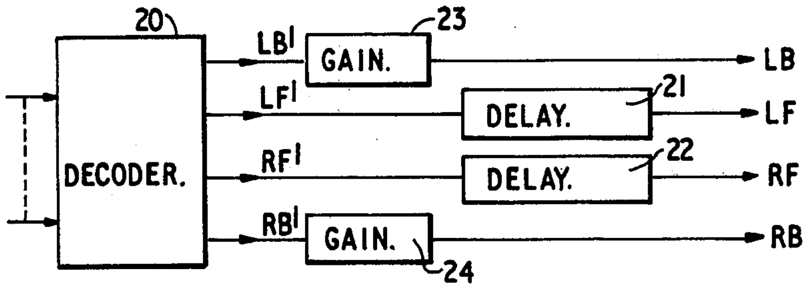




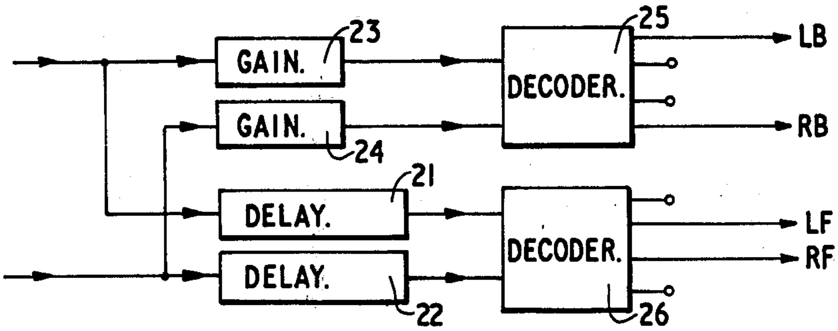
—FIG.1—



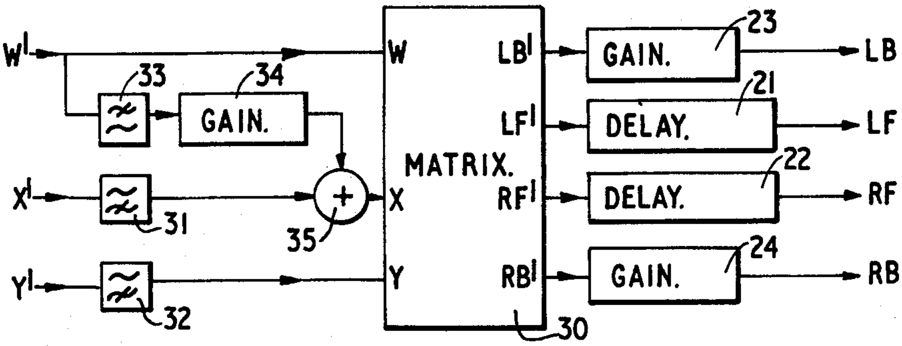
—FIG.2—



—FIG.3—



—FIG.4—



—FIG.5—

SOUND REPRODUCTION SYSTEMS

This invention relates to directional sound reproduction systems reproducing sound via three or more loudspeakers spaced round a listening position.

In designing decoders for sound reproduction systems of this type, it is customary to assume that all loudspeakers are at the same distance from a central listening point. The signals fed to the various loudspeakers are so adjusted as to produce a sound field in the immediate neighbourhood of this reference point which provides information to the ears of a listener at the point sufficient to create an impression of directionality. Provided that this information satisfies a sufficient number of the various mechanisms used by the human ear to localise sounds for listeners at the reference point, it is found in practice that good directional reproduction is obtained for listeners at other positions in the surrounding listening area.

The various relevant psychoacoustic criteria for a listener at the reference point are discussed in M. A. Gerzon, "Surround Sound Psychoacoustics", Wireless World, December, 1974, pages 483 to 486, U.K. Patent Specfication No. 1,494,751 and co-pending Application No. 46822/75 and corresponding U.S. Pat. Nos. 4,081,606 and 4,086,433. The latter two references disclose matrix circuitry for use in decoders for such systems.

It has been found that, even when all loudspeakers in a layout are equidistant from a reference point, there are some shapes of layout for which it is not possible to design decoders using matrix circuitry such that sufficiently many psychoacoustic criteria are satisfied for a listener at the reference point. In the following description, a loudspeaker layout for which it is possible to design a decoder using matrix circuitry will be referred to as a "solvable equidistant layout" and the decoder for such a layout will be described as a "solvable equidistant decoder".

According to the invention, there is provided a decoder for producing output signals which, if fed to at least three loudspeakers disposed at respective azimuths around a reference point at non-uniform distances therefrom, would produce a desired directional effect, comprising a basic decoder which, if fed to loudspeakers disposed at said azimuths at a uniform distance from the reference point would produce said desired directional effect, means for subjecting each output of the basic decoder to a relative time delay proportional to the difference between the distance from the reference point of the loudspeaker at the azimuth to which the output relates and the distance from the reference point of the most distant loudspeaker and to an amplitude gain proportional to the distance of the loudspeaker at said azimuth at the reference point.

The applied time delay is preferably so chosen as to be exactly equal to the difference in the time of travel of sound signals from each of the loudspeakers to the reference point. The constant of proportionality which determines the applied time delay is therefore at least approximately equal to the reciprocal of the speed of sound in air. The applied gain is chosen to compensate for the attenuation of sound intensity with increasing distance from a sound source.

It is already known to use delay lines to feed loudspeakers placed around a listener but these have been used to cause sounds other than those intended to come from loudspeakers in front of the listener to be heard by the listener with a delay relative to such sounds from loudspeakers in front of the listener of between 5 and 50 milliseconds. In accordance with the present invention, the sound from all loudspeakers arrive at the reference position at the same time from all loudspeakers. In addition, the invention requires the signals for the various loudspeakers to be produced by a solvable equidistant decoder for the same azimuth angles.

Embodiments of the invention will now be described by way of example with reference to the accompanying drawing, in which:

FIG. 1 is a schematic diagram of a loudspeaker layout suitable for use with a decoder in accordance with the invention,

FIG. 2 is a schematic diagram of a solvable equidistant loudspeaker layout having the same azimuths as the layout shown in FIG. 1,

FIG. 3 is a block diagram of a decoder in accordance with an embodiment of the invention,

FIG. 4 is block diagram of a decoder in accordance with another embodiment of the invention, and

FIG. 5 is a block diagram of a decoder similar to the decoder shown in FIG. 3 but having compensation for curvature of the sound field.

FIG. 1 shows a layout of four loudspeakers **10, 11, 12** and **13** equidistant from and surrounding a point **14**. The loudspeakers are disposed at corners of a trapezium. There is no known way of designing a solvable equidistant decoder for this layout using, matrix circuitry with the point **14** as a reference.

The diagonal lines joining the loudspeakers **10** and **12** and loudspeakers **11** and **13** respectively intersect at a point **15**. The loudspeakers **10** and **11** are closer to this point than are the loudspeakers **12** and **13** but, referring to FIG. 2, the loudspeakers **16, 17, 18** and **19** which are located on these diagonal lines at equal distances from the reference point **15**, form a solvable equidistant layout. A decoder for this layout can be as described in either the above-mentioned patent specification or the above-mentioned co-pending application.

FIG. 3 shows a decoder in accordance with the invention. Two or more input signals are fed to a solvable equidistant decoder **20** which produces output signals LB', LF', RF' and RB' suitable for the loudspeakers **16, 17, 18** and **19** respectively of the layout shown in FIG. 2. The two output signals LF' and RF' for the front loudspeakers are fed to respective delay devices **21** and **22** which produce output signals LF and RF respectively for feeding to the loudspeakers **10** and **11** of the layout shown in FIG. 1. Similarly, the signals LB' and RB' are fed to respective amplifiers **23** and **24** which produce output signals LB and RB respectively for the loudspeakers **12** and **13**. The delay applied by the delay device **21** is equal to the difference between the distance of the furthest loudspeaker, i.e. the loudspeaker **12** or the loudspeaker **13**, from the reference point **15** and the distance of the loudspeaker **10** from the reference point **15** divided by the speed of sound in air. A similar delay is applied by the delay device **22**. The amplifiers **23** and **24** apply amplitude gains which are proportional to the distance of each of the loudspeakers **12** and **13** from the reference point **15**, the constant of proportionality being such that the equivalent gain for the distance of the loudspeakers **10** and **11** from the reference points **15** would be unity. More generally, for an array of loudspeakers in which the distance of the i'th loudspeaker from the reference point is r_i and the maximum loudspeaker distance from the reference point is r_{max} then i'th loudspeaker is fed from its associated solvable equidistant decoder output via an amplitude gain proportional to r_i and a time delay given by:

$$(r_{max}-r_i)/c$$

where c is the speed of sound in air. Thus, the signal for the most remote loudspeaker will be fed via an amplifier only, the signal for the closest loudspeaker to the reference point would be fed via a delay device only while the signals for intermediate loudspeakers will be fed via both respective amplifiers and respective delay devices.

If desired, the various gains and time delays may be applied to the input signals to solvable equidistant decoders rather than to the output signals. FIG. **4** illustrates a decoder of this type which is equivalent to the decoder of FIG. **3** when the latter is adapted to receive two input signals only. One of the input signals is applied to the amplifier **23** and the delay device **21** and the other input signal is applied both to the amplifier **24** and the delay device **22**. Two solvable equidistant decoders **25** and **26**, both of which as identical with the decoder **20**, are provided. The outputs from the two amplifiers **23** and **24** are applied to the inputs of the decoder **25**, two of the outputs of which comprise the signals LB and RB respectively. The other outputs are not used. Similarly, the outputs of the delay devices **21** and **22** are applied to the inputs of the decoder **26**, two of the outputs of which produce the signals LF and RF, the other two outputs not being used. In general, the delay devices and amplifiers may be incorporated in any part of the circuitry provided that the required output signals are produced.

Certain of the decoders described in the above-mentioned patent specification and co-pending application include so called "distance compensation" which compensates for the effect of the curvature of the sound field at the reference point due to the distance of the loudspeaker from the reference point being finite. Such compensation consists of an RC high-pass filter in all signals paths representative of reproduced velocity at the reference point with a -3dB point 54/r Hz, where r is the loudspeaker distance in meters. Decoders in accordance with the present invention are for use with loudspeaker layouts which do not have a single value for r. An economical, although not strictly correct, method of providing compensation of sound field curvature in decoders in accordance with the invention is to apply compensation for an average value of the loudspeaker distances involved.

It is in fact possible to compute the circuitry required to compensate for sound field curvature for layouts not equidistant from the reference point and this compensation can always be realised by a matrix using non-cascaded low-pass and high-pass RC networks acting on the signals representative of pressure and velocity to produce modified output signals representative of velocity.

FIG. **5** illustrates the later stages of a decoder, similar to the decoder of FIG. **3**, with compensation for sound field curvature. Three input signals W', X' and Y' are representative respectively of the desired pressure, forward components of velocity and lateral components of velocity at the reference position. The distance compensated signals, W, X and Y are applied to the output matrix **30** of a solvable equidistant decoder for the layout of FIG. **2** which produces output signals as follows:

$$LB'=\tfrac{1}{2}(W-X+Y)$$
$$LF'=\tfrac{1}{2}(W+X+Y)$$
$$RF'=\tfrac{1}{2}(W+X-Y)$$
$$RB'=\tfrac{1}{2}(W-X-Y)$$

The output signals for the matrix **30** are applied to the delay devices **21** and **22** and the amplifier **23** and **24** to produce the signals LB, LF, RF and RB as described with reference to FIG. **2**.

To provide the required distance compensation, the two inputs signals X' and Y' representative of velocity are applied to the matrix **30** via respective high-pass filters **31** and **32** with time constant given by:

$$2(t_1^{-1}+t_2^{-1})$$

where t_1 and t_2 are the time sound takes to travel from the two front loudspeakers **10** and **11** to the reference point **15** and from the rear loudspeakers **12** and **13** to the reference point **15** respectively, so that the -3dB frequency is equal to the average of that associated with the front loudspeaker distance and that associated with the rear loudspeaker distance. In addition, the pressure signal W' is RC low-pass filtered by a filter **33** having the same time constant as the filters **31** and **32**, passed through an attenuator **34** having amplitude gain given by:

$$(t_2-t_1)/(t_2+t_1)$$

and added to the output of the filter **31** by an adder **35**.

When the loudspeaker layout departs only slightly from being equidistant from the reference point, the required delays are small and may be provided by cascaded RC all-pass networks. For example if the required delay corresponds to a 20 centimeter difference between r_i and r_{max} an approximation to the required delay may be provided by a cascaded pair of RC all-pass networks, each of time constant equal to a quarter of the required delay, i.e. a time constant of 0.147 msec. The pair of all-pass networks acts as a delay of the required time for frequencies up to about 1 kHz. At higher frequencies, it does not alter the polarity of signals. It is considered to be desirable for good high frequency localisation that the relative polarities of signals from all loudspeakers should be undisturbed by the delay circuitry.

I claim:

1. A decoder for producing output signals which, if fed to at least three loudspeakers disposed at respective azimuths around a reference point at non-uniform distances therefrom, would produce a desired directional effect, comprising a basic decoder which, if fed to loudspeakers disposed at said azimuths at a uniform distance from the reference point would produce said desired directional effect, means for subjecting each output of the basic decoder to a relative time delay proportional to the difference between the distance from the reference point of the loudspeaker at the azimuth to which the output relates and the distance from the reference point of the most distant loudspeaker and to an amplitude gain proportional to the distance of the loudspeaker at said azimuth at the reference point.

2. A decoder according to claim **1**, wherein the means for subjecting each output of the basic decoder to a time delay and an amplitude gain comprises means for applying such delay and gain directly to each output of the basic decoder.

3. A decoder according to claim 1, wherein the means for subjecting each output of the basic decoder to a time delay and an amplitude gain comprises means for applying respective time delays and gains to the inputs of the basic decoder.

4. A decoder according to claim 1, having means for providing outputs for four loudspeakers.

5. A decoder according to claim 4, having means for providing outputs for four loudspeakers disposed at respective corners of a trapezium with a first pair of loudspeakers at one distance from the reference point and a second pair of loudspeakers at another greater distance from the reference point.

6. A decoder according to claim 5, having an output matrix arranged to produce the output signals, a pressure signal input, a first velocity signal input for a signal representing the required velocity of sound in a first direction bisecting the angle subtended by the first pair of loudspeakers at the reference position and a second velocity input for a signal representing the velocity of sound in a direction perpendicular to said first direction, the velocity signal inputs being connected to the outputs of respective high-pass filters having time constant whose reciprocal equals the average of the reciprocals of the times taken by sound to travel from each of the loudspeakers to the reference point and a low-pass filter having the same time constant as said high-pass filters connected to apply the signal applied to the pressure signal input via means for applying a gain equal to the difference of time of travel of sound from the first pair of loudspeakers to the reference point and the time of travel of sound from the second pair of loudspeakers to the reference point, divided by the sum of said times of travel to an adder connected between the first velocity signal input and the corresponding high-pass filter.

7. A decoder according to claim 1, wherein the relative delay to which each output of the decoder is subject is equal to the difference between the distance of the corresponding loudspeaker from the reference point and the distance of the most remote loudspeaker to the reference point divided by the velocity of sound in air.

* * * * *