(12) **UK Patent Application** (19) **GB** (11) **2 073 556**   **A**

(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 $=$ $\dfrac{k \; mr \, I,}{\sqrt{2}}$

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

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

I being the 2×2 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.

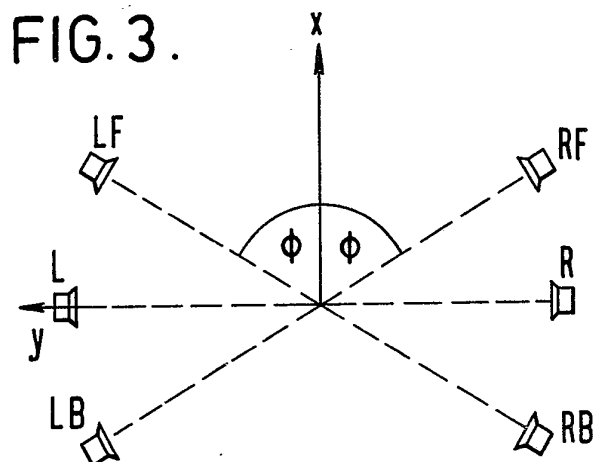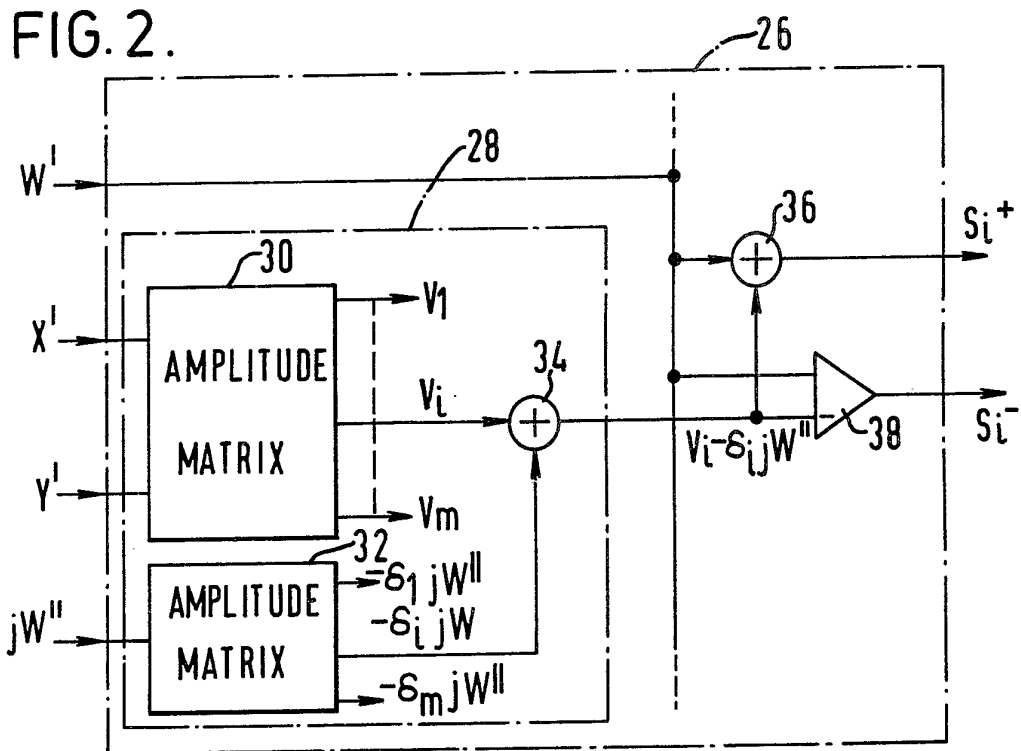GB 2 073 556 A

FIG.1.

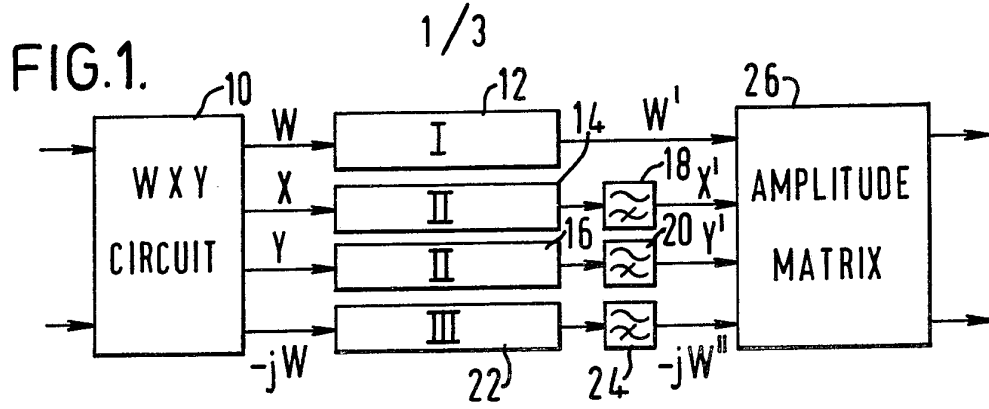## FIG.2.

FIG.3.

FIG. 4.
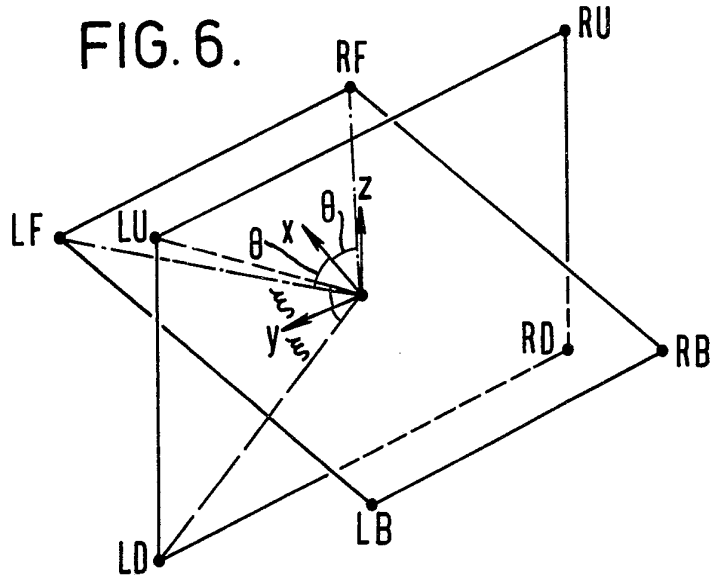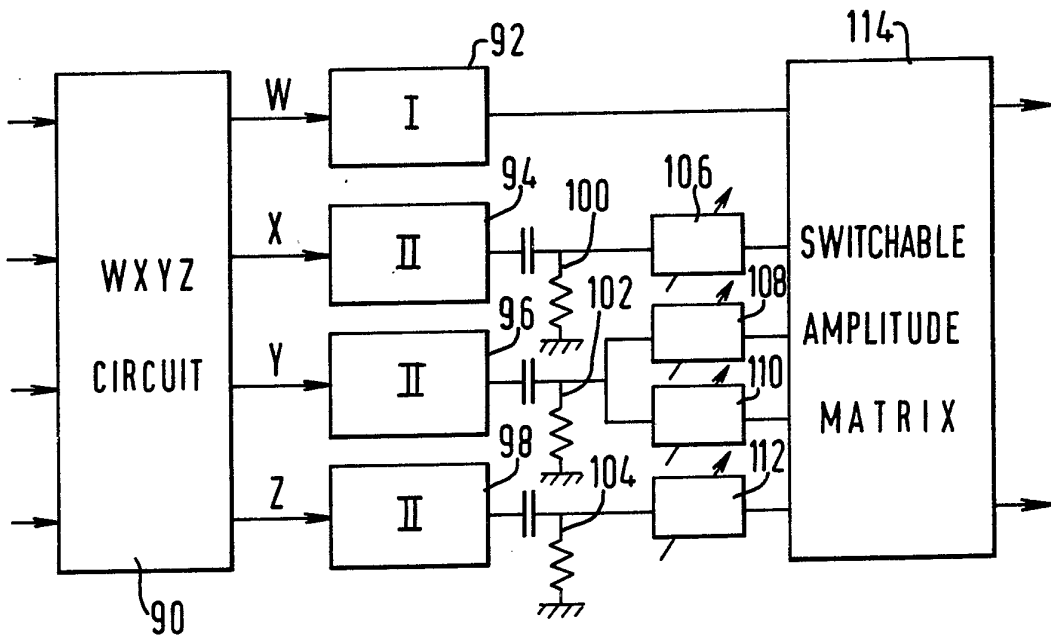
FIG. 5.
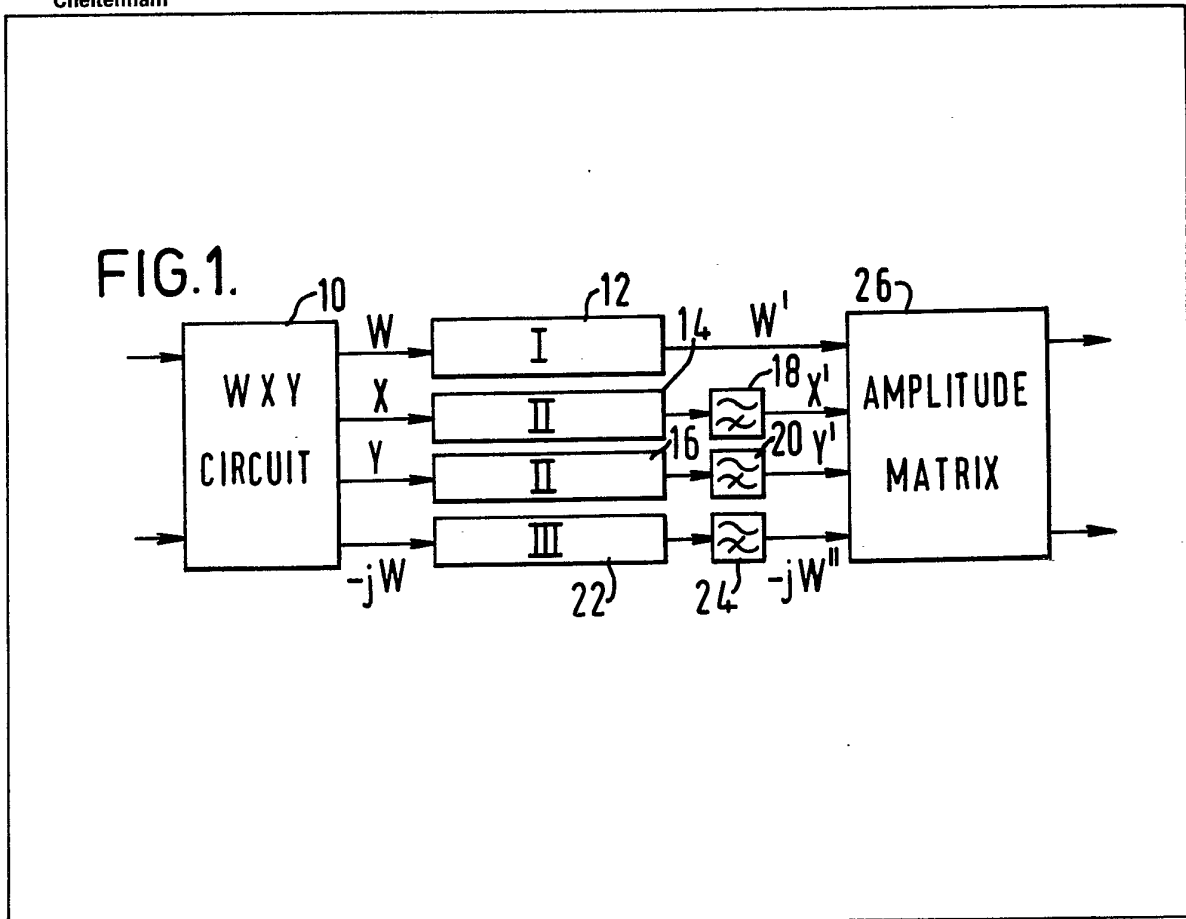
FIG. 6.

FIG. 7.

SPECIFICATION

Sound reproduction systems

5   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 sys-
10 tems. 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
15 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 louds-
20 peakers 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
25 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.
30   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.
35 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
40 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-
45 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:

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

$$X_0 = \text{Re}\left\{ \left(\sum_{i=1}^{n} g_i x_i \right) / rw \right\} \qquad \ldots(2)$$

$$y_0 = \text{Re}\left\{ \left(\sum_{i=1}^{n} g_i y_i \right) / rw \right\} \qquad \ldots(3)$$

$$Z_0 = \text{Re}\left\{ \left(\sum_{i=1}^{n} g_i z_i \right) / rw \right\} \qquad \ldots(4)$$

where "Re" means "the real part of" and (x_0, y_0, z_0) is
50 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 local-
55 isation 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
60 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
65 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 louds-
70 peaker 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
75 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,
$$S_i^+ = W' + \alpha_i X' + \beta_i Y' + \gamma_i Z' - \delta_i j W_i''$$
80 $$S_i^- = W' - \alpha_i X' - \beta_i Y' - \gamma_i Z' + \delta_i j W_i''$$
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 com-
85 ponents 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

α_i, β_i, γ_i, and δ_i are real gain coefficients such that
90 α_i, β_i, and γ_i substantially satisfy the following matrix equation:

$$KM = \frac{k \quad mr1}{\sqrt{2}}$$

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
5 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
10 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
15 such that the intended direction of sound localisation is an azimuth ø, measured anticlockwise from the x-axis, where:

cos ø: sinø = Re (X/W) : Re (Y/W) ....(8)

then a decoder in accordance with the invention for
20 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
25 $\quad S_i^+ = W' + \alpha_i X' + \beta_i Y' - \delta_i jW_i''$ ....(9)
$\quad S_i^- = W' - \alpha_i X' - \beta_i Y' + \delta_i jW_i''$ ....(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
30 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.
35 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 pro-
40 vided 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
45 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
50 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 louds-
55 peaker 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
60 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
65 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
70 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
75 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
80 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
85 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
90 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 amp-
95 litude matrix 26. Referring to Figure 2, the matrix 26 comprises a 3 × 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
100 a 2 × 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 × 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
105 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 cho-

sen to achieve the desired degree of directional bias-
ing. 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
5 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:

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

10 and forms a feed signal for one of the loudspeakers
of the ith 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
15 by:

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

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

It will be understood that, if no directional biasing
20 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 amp-
litude matrix producing outputs identical to those of
25 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 coef-
30 ficients such that:

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

that is

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

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

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

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

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

If we write the 2 × m matrix of the coefficients:

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

of the matrix 28 as M and the m × 2 matrix

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

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

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

where I is the 2 × 2 identity matrix and r is the dis-
tance of the loudspeakers from the reference listen-
ing position as before.

45 In practice, any positive real multiple k of the mat-
rix 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:

50 $$k^2\{(Re \, (X' /W'))^2 + (Re \, (Y' /W'))^2\} = 2 \quad ....(18)$$

is satisfied, since, if this condition is met, not only
the Makita theory, but also other low frequency
localisation theories are satisfied. This last men-
tioned condition is satisfied, for example, when W'
55 has unity gain for all sounds, and X' has gain $\sqrt{2}$ cos
Θ and Y' has gain $\sqrt{2}$ sin Θ 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
60 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 fre-
quency response may be applied to these signals,
provided that all the signals are affected equally by
65 these additional changes.

A convenient way of devising a matrix M satisfy-
ing the condition:

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

, and therefore giving correct localisation, is as fol-
lows.
70 For each pair of loudspeakers:

$$\begin{pmatrix} \alpha_i \\ \beta_i \end{pmatrix} = \frac{k}{\sqrt{2}} mr \begin{pmatrix} \sum_{h=1}^{m} \begin{pmatrix} x_h^2 & x_h y_h \\ x_h y_h & y_h^2 \end{pmatrix} \end{pmatrix}^{-1} \begin{pmatrix} x_i \\ y_i \end{pmatrix}$$

$$...(20)$$

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
75 due left loudspeaker L, a due right loudspeaker R and
four loudspeakers LB, LF, RF and RB placed at
respective azimuths $180° - \emptyset, \emptyset, -\emptyset,$ and $-180° + \emptyset$
measured anticlockwise from due front. Putting due
front as the x direction, due left as the y direction and
80 $S_1^+, S_2^+, S_3^+, S_1^-, S_2^-, S_3^-$ 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:

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

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

$$... (24)$$

From Equation 24:

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

$$...(25)$$

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

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

$$...(26)$$

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

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

$$...(28)$$

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

$$...(29)$$

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

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

$$...(31)$$

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
10 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 combina-
15 tions 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, mat-
rices may be combined or rearranged in the circuitry
wherever this is of design or contructional conveni-
20 ence 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
25 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 $\alpha_1 X' = -\alpha_3 X'$)
30 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 $\beta_2 Y'$) may be
implemented by a second variable gain circuit
placed in the Y' signal path, and the gains $\beta_1 = \beta_3$
35 (which result in a signal component $\beta_1 Y' = \beta_3 Y'$) may
be implemented by a third variable gain circuit
placed in the Y' signal path. Simultaneous adjust-
ment of these three variable gain circuits will then
permit the decoder to be adapted to loudspeaker
40 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
45 point. In the following discussion it is assumed that
the ith 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 respec-
50 tively of the desired pressure and x-axis, y-axis, and
z-axis components of velocity of sound at the refer-
ence listening position. Such signals may be sub-
jected to shelf filters having identical phase
responses and to RC high-pass filters compensating
55 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
60 Makita and energy vector localisation theories give
the same direction of sound provided that:

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

In addition, it is often desired that this localisation be
65 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:

$S_i^+ = W' + \alpha_i X' + \beta_i Y' + \gamma_i Z' - \delta_{ij}W_i''$ ....(33)

$S_i^- = W' - \alpha_i X' - \beta_i Y' - \gamma_i Z' + \delta_{ij}W_i''$ ....(34)

70 where $\alpha_i, \beta_i, \gamma_i$ 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 × m matrix:

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

satisfies the matrix equation:

$$KM = \frac{1}{\sqrt{2}} kmrl \qquad \ldots\ldots(36)$$

where k is a positive constant and

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

and l is the 3×3 identity matrix and where δ_i are the arbitrary real coefficients of directional biasing sign-
5 als.

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:

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

$$\ldots\ldots(38)$$

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

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

for encoded sounds.

The constant k may be implemented by means of
15 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
20 these additional changes.

For horizontally encoded sounds, Z = O, in which
case the Z signal path may be omitted, and the sys-
tem reduces to that previously described with refer-
ence to Figures 1 and 2, except that the values of α_i
25 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
30 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)
35 where:

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

Use of the above-mentioned matrix Formula 38
gives:

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

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
40 when the signal W has unit gain for sounds encoded
from all directions in space, and where X, Y and Z
have respective gains $\sqrt{2}\cos\Theta\cos\eta$, $\sqrt{2}\sin\Theta\cos$
η and $\sqrt{2}\sin\eta$ for sounds having source azimuth Θ
measured anticlockwise from due front and source
45 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
50 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
5 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 Equa-
10 tions (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.
15   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:
20

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

25

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

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

The output signals in the Y and Z channels from
35 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
40 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 sign-
als is combined by a respective addition circuit 66, 68
and 70 to give the signals S_1^+, S_2^+, and S_3^+ which are
45 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
50 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 out-
55 put 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 louds-
peaker matrixing technique forms the subject of U.K.
Patent Specification No. 1,548,674 and may also be
60 applied to decoders for feeding horizontal louds-
peaker arrays in accordance with the present inven-
tion, such as the decoder illustrated in Figures 2 and
3.
It will be appreciated that other spatial orientations
65 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 com-
70 plex 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 car-
tesian co-ordinates (x_i, y_i, z_i) and $(-x_i, -y_i, -z_i)$ with
75 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:

$(x_1, y_1, z_1) = (r\cos\varnothing, r\sin\varnothing, o)$

$(x_2, y_2, z_2) = (r\cos\varnothing, -r\sin\varnothing, o)$

80    $(x_3, y_3, z_3) = (o, r\cos\xi, r\sin\xi)$

$(x_4, y_4, z_4) = (O, -r\cos\xi, r\sin\xi)$ ....(44)

This corresponds to a loudspeaker layout consist-
ing of a combination of a horizontal array of four
loudspeakers with an angle $2\varnothing$ subtended at the
85 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
90 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,
95 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 accor-
dance with this invention, including those described
100 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
105 and also high-pass filters 100, 102 and 104 to com-
pensate 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,
110 110 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
115 amplifiers permit adjustment of the angles $\varnothing$ 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
120 circuitry which serves to modify the output signals
from the decoder prior to feeding these to the louds-

peakers 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.

5   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

10 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

15 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

20 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 louds-

25 peakers 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

30 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

35 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

40 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,

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

45 $$S_i^- = W' - \alpha_i X' - \beta_i Y' - \gamma_i Z' + \delta_i jW_i''$$

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 com-

50 ponents 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

55 α_i, β_i and γ_i substantially satisfy the following matrix equation:

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

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:

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

or

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

and k is a positive real constant which may be fre-

60 quency 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

65 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,

70 $$S_i^+ = W' + \alpha_i X' + \beta_i Y' - \delta_i jW_i''$$
$$S_i^- = W' - \alpha_i X' - \beta_i Y' + \delta_i jW_i''$$

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

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

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

75 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 $\alpha_1 = \dfrac{-3k}{2\sqrt{2}\cos\varnothing}$,

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

$\alpha_2 = 0,$

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

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

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

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

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

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

and

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

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
5 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
10 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.

15 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 louds-
20 peakers 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
25 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
30 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:

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

where

$\gamma_1 = \gamma_2 = \alpha_3 = \alpha_4 = 0,$
$\alpha_1 = \alpha_2 = \sqrt{2}\,k/\cos\varnothing,$

$\beta_1 = -\beta_2 = \sqrt{2}\,k\sin\varnothing / (\sin^2\varnothing + \cos^2\varnothing + \cos^2\xi),$
$\beta_3 = -\beta_4 = \sqrt{2}\,k\cos\xi / (\sin^2\varnothing + \cos^2\xi),$
$\gamma_3 = \gamma_4 = \sqrt{2}\,k/\sin\xi.$

35 24. A decoder according to claim 23, adjustable for a range of values of the angles $\varnothing$ 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',
40 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 coeffi-
45 cients 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.

50 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.