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Takahashi et al.

[11] **3,872,249**

[45] **Mar. 18, 1975**

[54] **ENCODING METHOD FOR CONVERTING MULTI-CHANNEL SOUND SIGNALS INTO 2-CHANNEL COMPOSITE SIGNALS**

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[30] **Foreign Application Priority Data**

Sept. 18, 1973 Japan 48-10517

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

[51] **Int. Cl.** **H04r 5/00**

[58] **Field of Search** .... 179/1 GQ, 1 G, 100.4 ST, 179/100.4 K

[56] **References Cited**

UNITED STATES PATENTS

3,787,622	1/1974	Itoh et al.	179/1 GQ
3,825,684	7/1974	Ito et al.	179/1 GQ

OTHER PUBLICATIONS

Journal of the Audio Engineering Society, Benjamin B. Bauer, Jan./Feb. 1973, Volume 21, Number 1, P. 19.

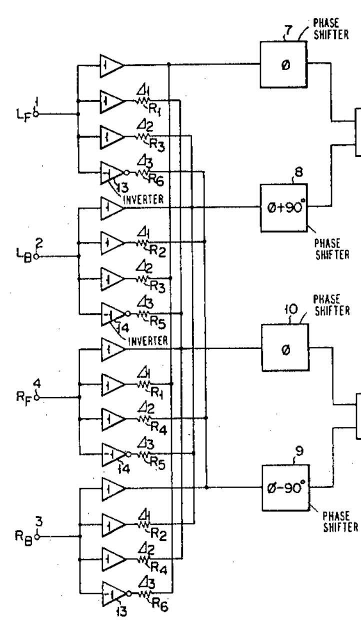
Audio, The Sansui Q S System, October 1971, P. 42.

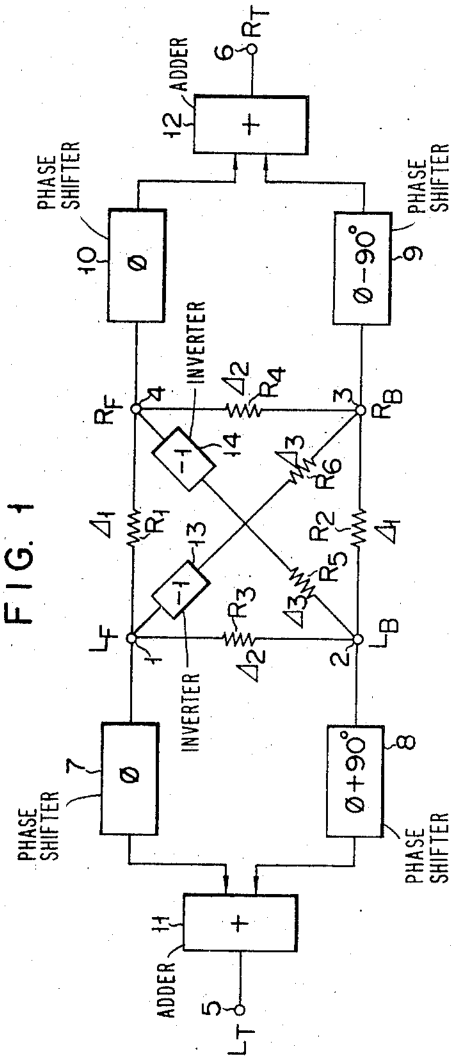
Primary Examiner—William C. Cooper
Assistant Examiner—Tommy P. Chin
Attorney, Agent, or Firm—Harris, Kern, Wallen & Tinsley

[57] **ABSTRACT**

An encoding method for use in a four-channel matrix system in which a front sound signal in one channel signal consists of a relatively large amplitude portion and a relatively small amplitude portion in phase quadrature with each other and the front sound signal in another channel signal is substantially in phase with the front sound signal in the one channel signal and consists of relatively small amplitude portions in phase quadrature with each other, and a back sound signal in one channel signal consists of a relatively large amplitude portion and a relatively small amplitude portion in phase quadrature with each other and the back sound signal in another channel signal is substantially in opposite phase with the back sound signal in the one channel signal and consists of relatively small amplitude portions in phase quadrature with each other. The present encoding method can improve the separation characteristic between the two composite signals and reduce the deterioration of sound quality particularly in two-channel reproduction of the two composite signals.

7 Claims, 9 Drawing Figures

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PHASE SHIFTER
PHASE SHIFTER
PHASE SHIFTER
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ADDER
INVERTER
INVERTER
L_T
R_T
L_F
R_F
L_B
R_B
ϕ
ϕ
$\phi+90°$
$\phi-90°$
Δ_1
Δ_2
Δ_3
R_1
R_2
R_3
R_4
R_5
R_6

FIG. 1

FIG. 2

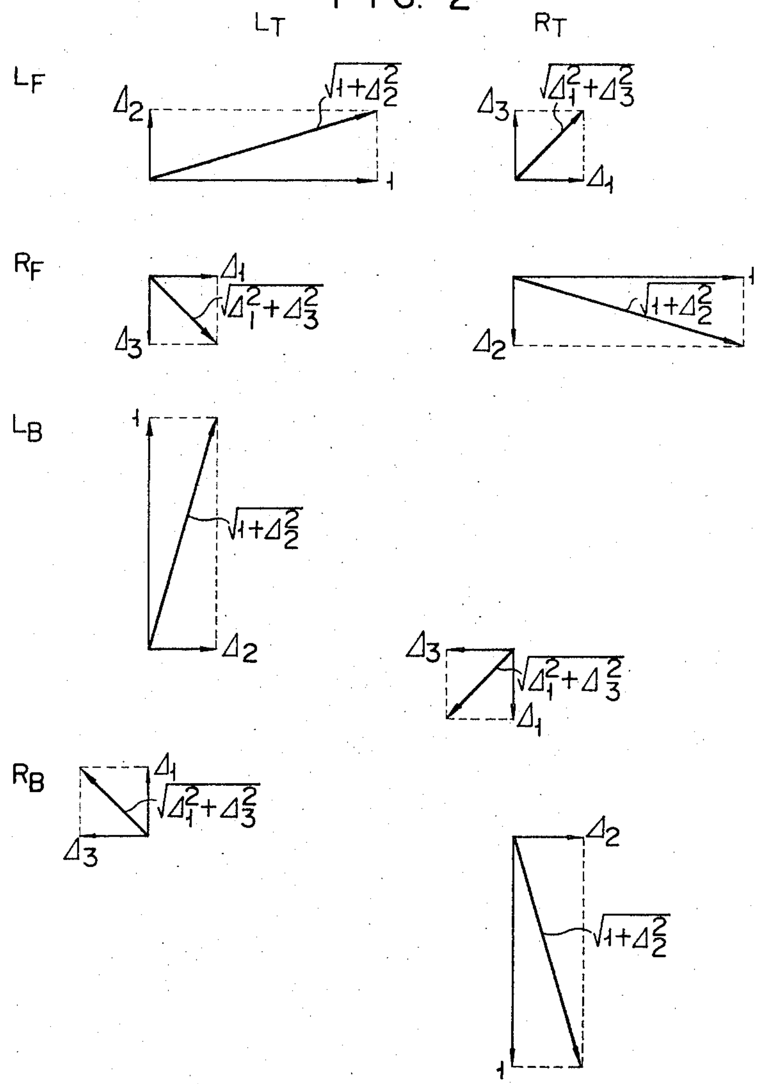

L_T
R_T
L_F
R_F
L_B
R_B
Δ_1
Δ_2
Δ_3
1
$\sqrt{1+\Delta_2^2}$
$\sqrt{\Delta_1^2+\Delta_3^2}$

FIG. 3

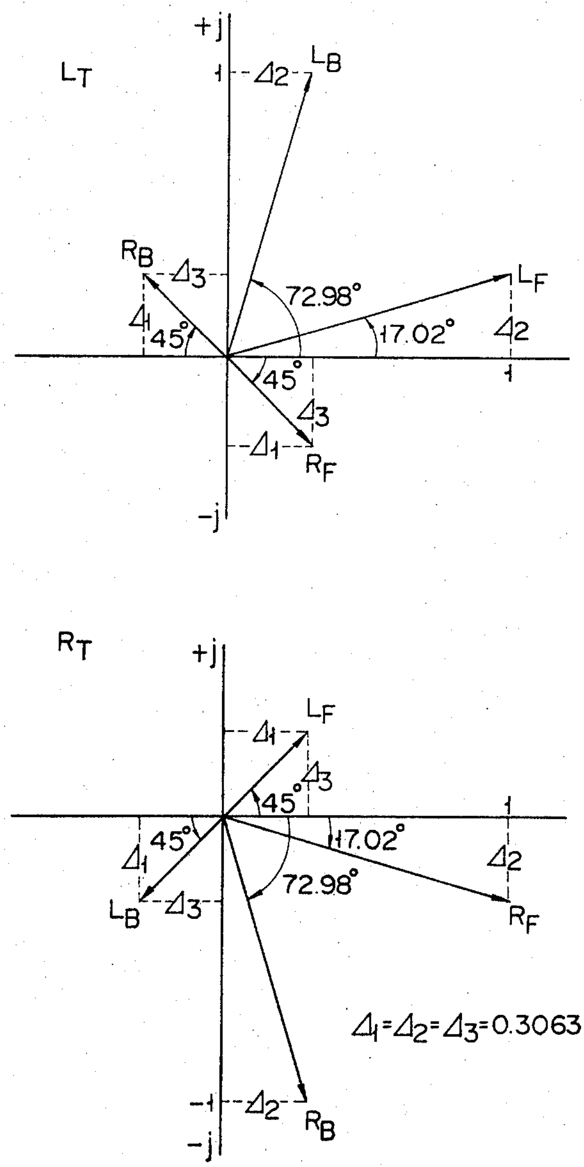

+j
L_T
L_B
Δ_2
R_B
Δ_3
72.98°
L_F
Δ_1
45°
17.02°
Δ_2
45°
Δ_3
$-\Delta_1$
R_F
-j
R_T
+j
L_F
Δ_1
Δ_3
45°
45°
17.02°
Δ_1
Δ_2
72.98°
L_B
Δ_3
R_F
$\Delta_1 = \Delta_2 = \Delta_3 = 0.3063$
-1
Δ_2
R_B
-j

FIG. 4A

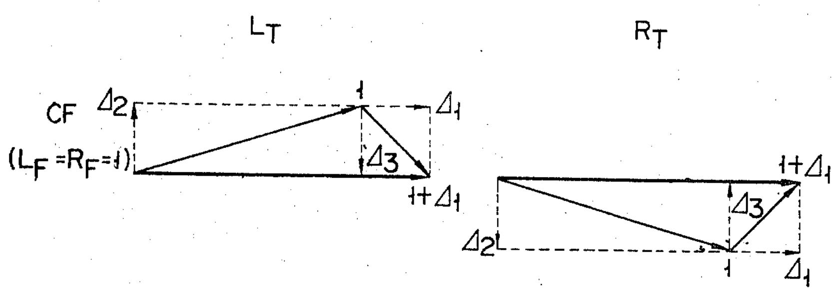

L_T
R_T
CF
$(L_F = R_F = 1)$
Δ_2
1
Δ_1
Δ_3
$1+\Delta_1$
$1+\Delta_1$
Δ_3
Δ_2
1
Δ_1

FIG. 4B

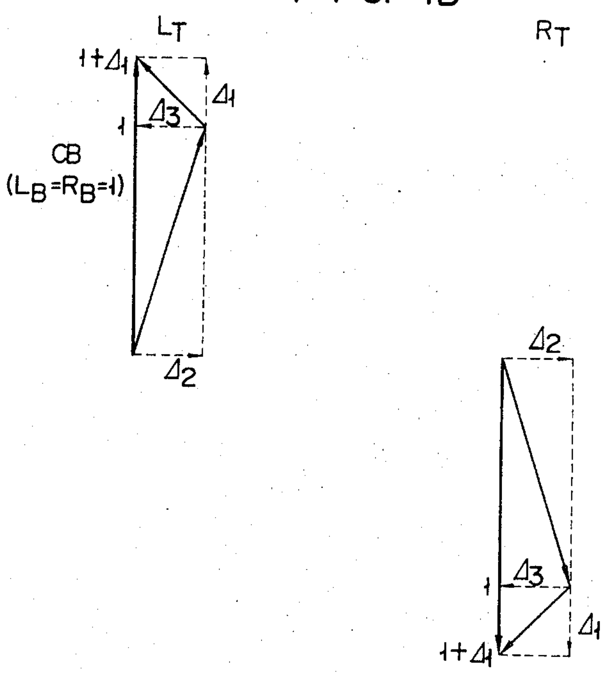
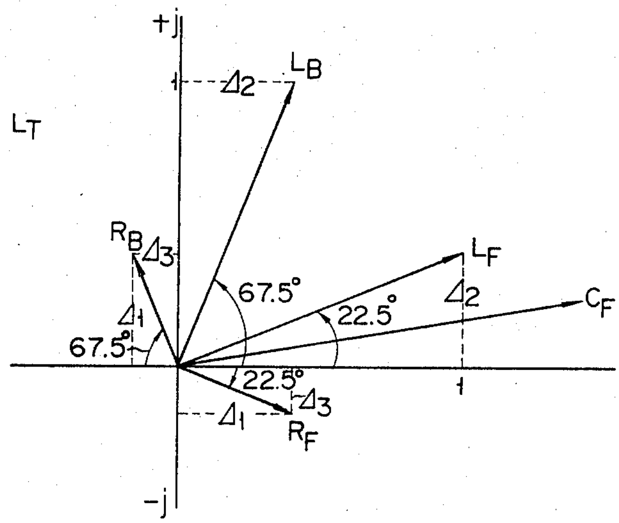
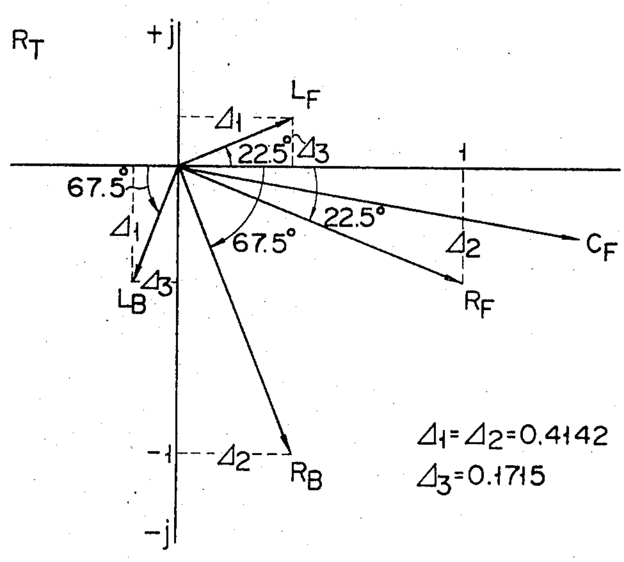

L_T
R_T
$1+\Delta_1$
Δ_1
Δ_3
1
CB
$(L_B = R_B = 1)$
Δ_2
Δ_2
Δ_3
1
Δ_1
$1+\Delta_1$

FIG. 5

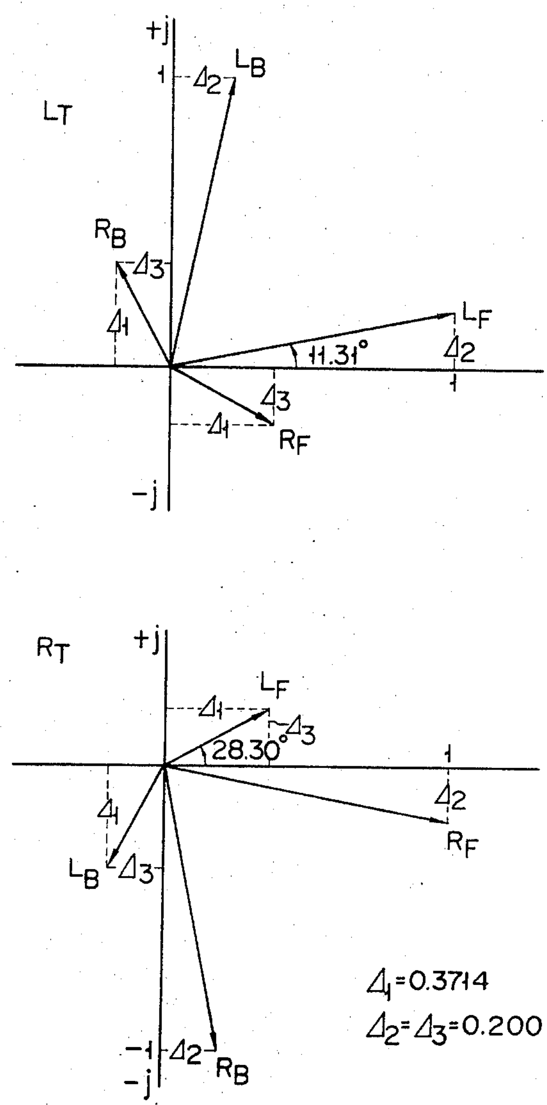


+j
L_B
Δ_2
L_T
R_B
Δ_3
67.5°
L_F
22.5°
Δ_2
C_F
Δ_1
67.5°
22.5°
Δ_3
Δ_1
R_F
-j



R_T
+j
L_F
Δ_1
22.5°
Δ_3
67.5°
22.5°
Δ_1
67.5°
Δ_2
C_F
Δ_3
L_B
R_F
-1
Δ_2
R_B
$\Delta_1 = \Delta_2 = 0.4142$
$\Delta_3 = 0.1715$
-j

FIG. 6



L_T
+j
1
Δ_2
L_B
R_B
Δ_3
Δ_1
L_F
11.31°
Δ_2
1
Δ_3
Δ_1
R_F
-j
R_T
+j
L_F
Δ_1
Δ_3
28.30°
1
Δ_2
R_F
Δ_1
L_B
Δ_3
-1
Δ_2
R_B
-j
Δ_1=0.3714
Δ_2=Δ_3=0.200

FIG. 7

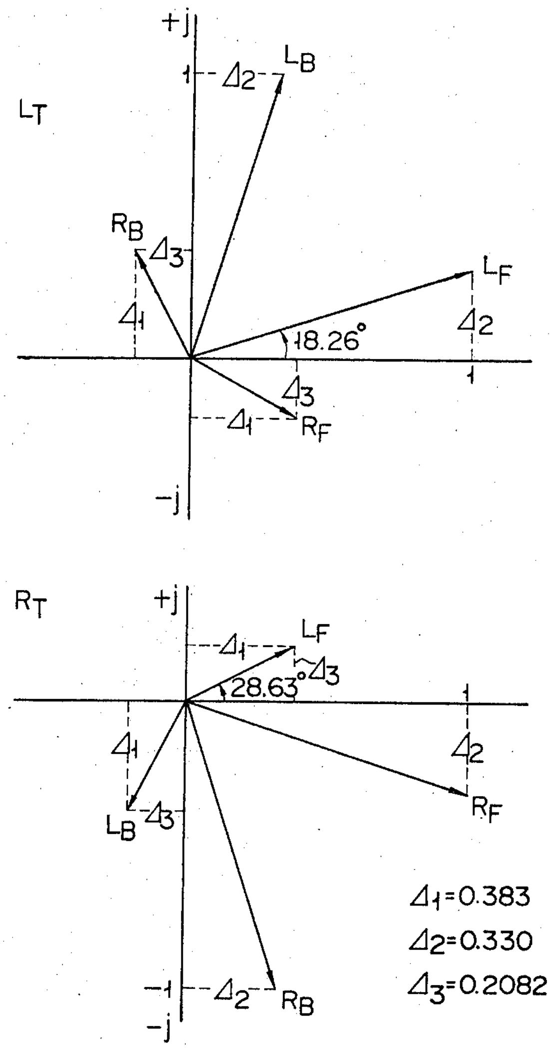
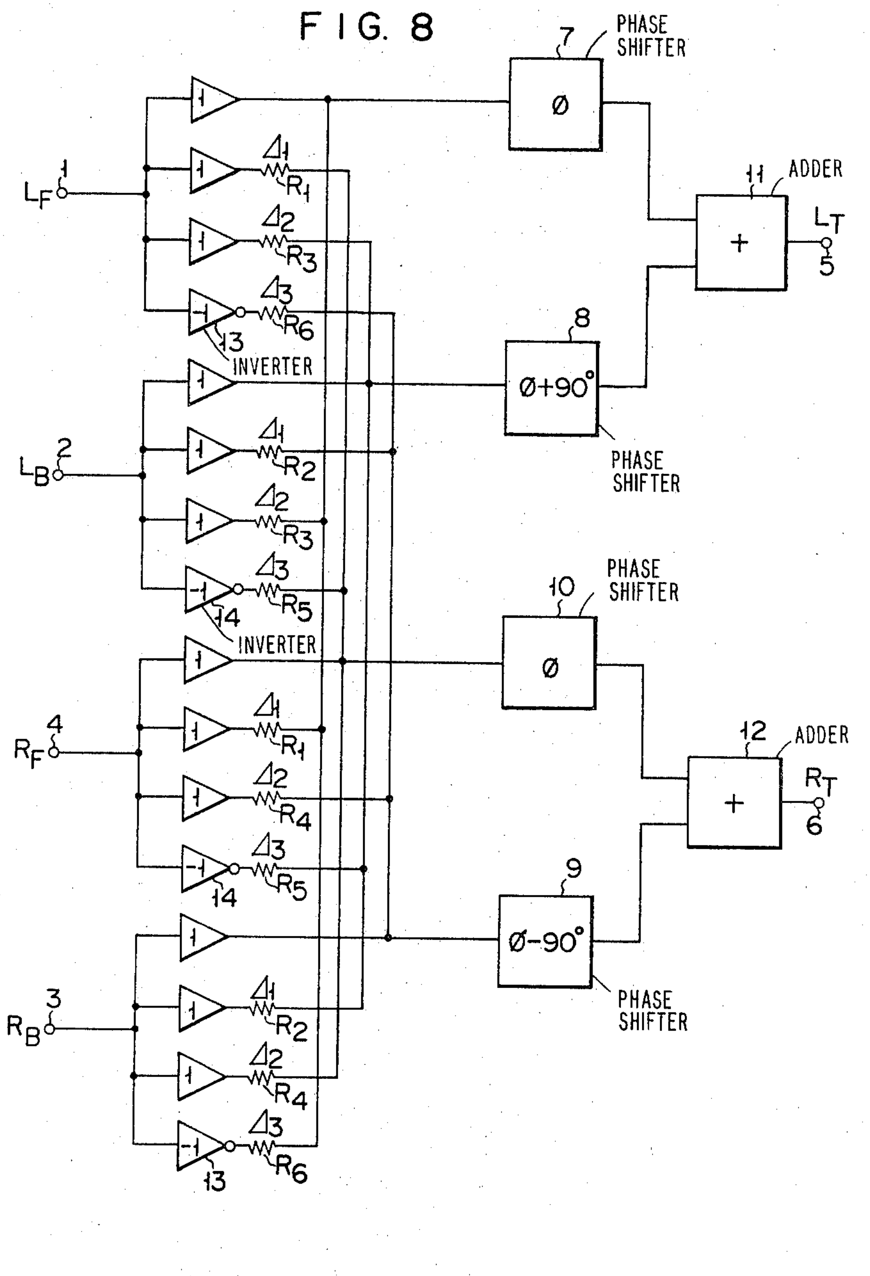

L_T
+j
L_B
Δ_2
R_B
Δ_3
Δ_1
L_F
Δ_2
18.26°
Δ_3
Δ_1
R_F
−j
R_T
+j
Δ_1
L_F
Δ_3
28.63°
Δ_1
Δ_2
R_F
L_B
Δ_3
$\Delta_1=0.383$
$\Delta_2=0.330$
$\Delta_3=0.2082$
−1
Δ_2
R_B
−j

FIG. 8


PHASE SHIFTER
7
Ø
ADDER
11
+
L_T
5
L_F
1
$\Delta 1$
R_1
$\Delta 2$
R_3
$\Delta 3$
R_6
13
INVERTER
8
Ø+90°
PHASE SHIFTER
L_B
2
$\Delta 1$
R_2
$\Delta 2$
R_3
$\Delta 3$
R_5
14
INVERTER
PHASE SHIFTER
10
Ø
12
ADDER
+
R_T
6
R_F
4
$\Delta 1$
R_1
$\Delta 2$
R_4
$\Delta 3$
R_5
14
9
Ø-90°
PHASE SHIFTER
R_B
3
$\Delta 1$
R_2
$\Delta 2$
R_4
$\Delta 3$
R_6
13

ENCODING METHOD FOR CONVERTING MULTI-CHANNEL SOUND SIGNALS INTO 2-CHANNEL COMPOSITE SIGNALS

This invention relates to a 4-channel matrix encoding system for converting multi-channel sound signals into 2-channel signals.

There is conventionally known an encoding system in which sound signals from sound sources located in front of the left-center and right-center of a sound field are respectively coupled with the same phase to a right and left channel transmission system and sound signals from sound sources positioned behind the left-center and right-center of the sound field are respectively coupled with an opposite phase to a right and left channel transmission system. In this case, it is preferred that a left-center signal be coupled only to a left transmission system and a righ-center signal be coupled only to a right transmission system.

As one example of the above-mentioned encoding system, there is known an encoding system adapted to synthesize a left channel signal L_T and right channel signal R_T which are shown in the following expression:-

$$\left.\begin{aligned} L_T &= L_F + \Delta R_F + jL_B + j\Delta R_B \\ R_T &= R_F + \Delta L_F - jR_B - j\Delta L_B \end{aligned}\right\} \quad (1)$$

in which Δ denotes a blend quantity whose representative value is 0.414 (= sin 22.5°/cos 22.5°).

As shown in the expression (1), the front-left signal L_F and back-left signal L_B included in the left channel signal L_T are greater in amplitude level than the front-left signal L_F and back-left signal L_B included in the right channel signal R_T. On the other hand, the front-right signal R_F and back-right signal R_B included in the right channel signal R_T are greater in amplitude level than the front-right signal R_F and back-right signal R_B included in the left channel signal L_T. The front signals L_F and R_F included in the right and left channel signals L_T and R_T, respectively, are phase shifted by a reference angle, and the back signals L_B and R_B included in the left channel signal L_l are phase shifted by an angle equal to a reference angle plus 90° while the back signals R_B and L_B included in the right channel signal R_T are phase shifted by an angle equal to a reference angle minus 90°.

The encoding system as shown in the above expression (1) has the following advantages. Since the front signals are coupled with the same phase to the first and second channels and the back signals are coupled with an opposite phase to the first and second channels, 4-channel reproducing signals can be obtained merely by effecting the addition and subtraction of the first and second channel signals in a decoder device. This renders the decoder simpler in construction. Furthermore, even when a 4-channel matrix stereophonic record is reproduced using a stereophonic sound system, a clear image localization is obtained from the front signals. However, the encoding system suffers from the following disadvantages. With the encoder adapted to be supplied with only four sound signals L_F, R_F, L_B and R_B, where a left-center signal L_C, for example, is encoded, it will be sufficient if the abovementioned expression is given as $L_F = L_B$. Thus, right and left signals L_T and R_T are represented as follows:

$$\left.\begin{aligned} L_T &= L_F(1+j) = \sqrt{2}\,L_F\angle +45° \\ R_T &= L_F(\Delta - j\Delta) = \sqrt{2}\cdot\Delta\cdot L_F\angle -45° \end{aligned}\right\} \quad (2)$$

As will be evident from the above expression, the amplitude ratio between the signals L_T and R_T is 1:Δ with respect to the left-center signal L_C. For this reason, a cross-talk (−7.7 db) of the quantity corresponding to a blend quantity Δ(=0.414) is produced at the right channel. The same is true with respect to the right-center signal R_C and a cross-talk quantity (−7.7 db) is produced at the left channel. This means that the basic object of transmitting the left-center signal and right-center signal respectively to one of the channels is not fully realized. With the above encoding system, since the sound signals L_F and R_F in the composite signals are blended with the same phase as will be clear from the expression (1), ambience components happening to be included in the sound signals L_F and R_F with an opposite phase are cancelled during the encoding process, thus presenting a problem. For this reason, when particularly a 4-channel matrix stereophonic record is reproduced using a conventional stereophonic reproduction system, a sound quality is deteriorated with the attendant disadvantage.

It is accordingly the object of this invention to provide a simple encoding system capable of reducing a cross-talk between the left and right channels with respect to the left-center signal and right-center signal and capable of reducing a sound quality deterioration even when reproducing is effected by the conventional stereophonic reproduction system.

According to the present invention there is provided an encoding method in which, in coupling to respective first and second channels at least first and second sound input signals associated with front channels and at least third and fourth sound input signals associated with back channels to generate first and second channel signals, said first, second, third and fourth sound input signals are coupled to the first and second channels with such an amplitude relation that the amplitude levels of said first and third sound input signals included in said first channel signal are greater than those of the first and third sound input signals included in the second channel signal, and the amplitude levels of the second and fourth sound input signals included in said second channel are greater than those of second and fourth sound input signals included in the first channel signal, and in such a phase relation that the first and second sound input signals included in said first channel signal are in a substantially in-phase relation to the first and second sound input signals included in said second channel signal respectively and the third and fourth sound input signals included in said first channel signal is in a substantially opposite relation to the third and fourth sound input signals included in said second channel signal respectively: said encoding method comprising the steps of coupling said first sound input signal to said first channel at a relatively large amplitude level and at a reference phase shift angle and to said first channel at a relatively small amplitude level and at a phase shift angle corresponding to the reference angle plus 90°; coupling said first sound input signal to said second channel at a relatively small amplitude level and at the reference phase shift angle and to said second channel at a relatively small amplitude level and at a phase shift angle corresponding to the reference angle plus 90°; coupling said second input signal to said first channel at a relatively small amplitude level and at the reference phase shift angle and to said first channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90°; coupling said second sound input signal to said second channel at a relatively large amplitude level and at the reference phase shift angle and to said second channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90°; coupling said third sound input signal to said first channel at a relatively large amplitude level and at a phase shift angle corresponding to the reference angle plus 90° and to said first channel at a relatively small amplitude level and at the reference phase shift angle; coupling said third sound input signal to said second channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90° and to said second channel at a relatively small amplitude level and at a phase shift angle displaced 180° from said reference angle; coupling said fourth sound input signal to said first channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle plus 90° and to said first channel at a relatively small amplitude level and at the phase shift angle displaced 180° from said reference angle; and coupling said fourth sound input signal to said second channel at a relatively large amplitude level and at the phase shift angle corresponding to the reference angle minus 90° and to said second channel at a relatively small amplitude level and the reference phase shift angle.

This invention can be more fully understood from the following detailed description when taken in conjunction with the accompanying drawings, in which:

FIG. **1** shows a model diagram for explaining an encoding system according to this invention;

FIG. **2** shows vector diagrams of left and right channel signals L_T and R_T obtained by the encoding system of this invention with respect to respective input sound signals;

FIG. **3** shows vector diagrams of the left and right channel signals obtained under a certain blend condition;

FIGS. **4A 4B** show vector diagrams of the left and right channel signals, respectively, of front-center and back center signals;

FIGS. **5** to **7** show vector diagrams of the left and right channel signals obtained under respective blend conditions; and

FIG. **8** shows a concrete example of the encoding system embodying this invention.

FIG. **1** shows a model of an encoding system according to this invention. in the drawing reference numerals **1** to **4** denote encoding input terminals to which sound signals L_F, L_B, R_B and R_F are respectively applied, and reference numerals **5** and **6** represent encoding output terminals from which a left channel composite signal L_T and right channel composite signal R_T are respectively derived out. The sound signal L_F is coupled to the output terminal **5** through a phase shifter **7** for phase shifting an input signal by a reference angle ϕ at a relatively large amplitude level and an adder **11.** The sound signal R_F has its amplitude multiplied by a blend quantity Δ_1 (<1) due to the action of a blend resistor R_1 i.e. is coupled, at a relatively small amplitude level to the output terminal **5** through the phase shifter **7** and adder **11.** The sound signal L_B has a relatively large amplitude level and is coupled to the output terminal **5** through a phase shifter **8** for phase shifting an input signal by an angle equal to the reference angle plus 90° and the adder **11.** The sound signal R_B has its amplitude multiplied by a blend quantity Δ_1 through a blend resistor R_2 i.e. is coupled, at a relatively small amplitude level to the output terminal **5** through the phase shifter **8** and adder **11.** The sound signal R_f has a relatively large amplitude level and is coupled to the output terminal **6** through a phase shifter **10** having a phase shift characteristic similar to that of the phase shifter **7** and adder **12,** while the sound signal L_F has its amplitude multiplied by a blend quantity Δ_1 and is coupled to the output terminal **6** through phase shifter **10** and adder **12.** The sound signal R_B is coupled to the output terminal **6** through a phase shifter **9** for phase shifting an input signal by an angle equal to a reference signal ϕ minus 90° and adder **12,** while the sound signal L_B has its amplitude multiplied by a blend quantity Δ_1 and is coupled to the output terminal **6** through phase shifter **9** and adder **12.** Though the construction of the abovementioned encoder is the same as the conventional encoder, the encoding system of this invention has blend quantites Δ_2 and Δ_3 in addition to the blend quantity Δ_1. More particularly, the signal L_F is coupled to the output terminal **5** through the phase shifter **8** with the blend quantity Δ_2 due to a resistor R_3 and also coupled to the output terminal **6** through an inverter **13** and the phase shifter **9** with the blend quantity Δ_3 due to a resistor R_6. The signal L_B is coupled through the phase shifter **7** to the output terminal **5** with the blend quantity Δ_2 and also coupled to the output terminal **6** through an inverter **14** and phase shifter **10** with the blend quantity Δ_3 due to a reisitor **5.** The signal R_B is coupled through phase shifter **10** to the output terminal **6** with the blend quantity Δ_2 and also coupled to the output terminal **5** through the inverter **13** and phase shifter **7** with the blend quantity Δ_3. The signal R_F is coupled through phase shifter **9** to the output terminal **6** with the blend quantity Δ_2 and also coupled through inverter **14** and phase shifter **8** to the output terminal **5** with the blend quantity Δ_3. In the model diagram of FIG. **1** the inverters **13** and **14** are shown as of bidirectional type.

The matrix representation of the above-mentioned encoding system is given below:-

$$\begin{bmatrix} L_T \\ R_T \end{bmatrix} = \begin{bmatrix} 1+j\Delta_2 & j+\Delta_2 & \Delta_1-j\Delta_3 & j\Delta_1-\Delta_3 \\ \Delta_1+j\Delta_3 & -j\Delta_1-\Delta_3 & 1-j\Delta_2 & -j+\Delta_2 \end{bmatrix} \cdot \begin{bmatrix} L_F \\ L_B \\ R_F \\ R_B \end{bmatrix}$$

FIG. **2** shows vector diagrams of respective signals included in the right and left channel signals L_T and R_T obtained by the above-mentioned encoding system. The amplitude ratio of respective input signals contained in the output signals L_T and R_T is, as will be clearly shown, $\sqrt{1+\Delta_2^2} : \sqrt{\Delta_1^2+\Delta_3^2}$. FIG. **3** shows the vector diagrams of the output signals L_T and R_T involved where $\Delta_1 = \Delta_2 = \Delta_3 = 0.3063$. In this case, the amplitude ratio of the respective input signals included in the output signals L_T and R_T is 1:0.414 as in the conventional system. With a left-center signal L_C ($L_F = L_B$) under this blend condition, as will be evident from the vector diagram of FIG. **3,** the input signals L_F and L_B

in the output signal R_T are cancelled with respect to each other with the result that no left-center signal L_C appears in the output signal R_T. With a right-center signal R_C ($R_F = R_B$) the input signals R_F and R_B in the output signal L_T are cancelled with respect to each other. As a result, no right-center signal R_C appears in the output signal L_T. More particularly the left-center signal L_C is transmitted only to a left channel transmission system and the right-center signal R_C is transmitted only to a right channel transmission system. Unlike the conventional encoding system, therefore, no cross-talk exists between left and right channel signals with respect to left and right-center signals.

FIGS. **4A** and **4B** show vector diagrams of the output signals L_T and R_T, respectively, involved when the front-center signal C_F ($L_F = R_F$) and back-center signal C_B ($L_B = R_B$) are encoded by the encoding system as described above. Like the conventional system, in the case of the front-center signal C_F, the output signals L_T and R_T have the same phase and the same level, while in the case of the back-center signal C_B, the output signals L_T and R_T have opposite phases and the same level. From the above it will be understood that the encoding system of this invention is suitable for use as a **4**-channel matrix encoding system.

As will be understood from the vector diagram of FIG. **3**, a phase difference of 27.98° exists between each of the front input signals L_F and R_F in the output signals L_T and R_T, while a phase difference of 152.02° exists between each of the rear input signals L_B and R_B in the output signals L_T and R_T Therefore, separation between diagonal channels is not made infinite and stays at 15.2 db. However, with the left and right-center signals, separation between the output signals L_T and R_T is made infinite and in the case of the front-center signal C_F the output signals L_T and R_T have the same phase. As a result, the output signals obtained by the above-mentioned encoding system represent a particularly good compatibility with the conventional two-channel stereophonic reproduction system. A phase difference of 28° present between the corresponding front input signals included in the above-mentioned output signals L_T and R_T may be safely taken as meaning that the respective front input signals are coupled, in substantially the same phase relation, to the left and right transmission systems. A phase difference of 152° present between the corresponding back signals may be safely taken as meaning that the respective input signals are coupled, in a substantially opposite phase relation, to the left and right transmission systems.

Where the blend quantities Δ_1, Δ_2 and Δ_3 are all equal to 0.3063, the respective front input signals included in the output signals L_T and R_T do not become entirely the same phase and the respective back input signals do not become entirely an opposite phase. In order to cause the respective front input signals to be entirely in phase and the respective back input signals to be entirely in opposite phase, it is only necessary to proportion the blend quantities to have a relation of $1:\Delta_2 = \Delta_1:\Delta_3$. FIG. **5** shows a vector diagram of the output signals L_T and R_T involved when the blend quantities are selected to be $\Delta_1=\Delta_2=0.4142$ and $\Delta_3=0.1715$ so as to meet the relation. As will be evident, the corresponding front input signals included in the output signals are caused to be entirely in phase and the corresponding back input signals to be entirely in opposite phase. Under this blend condition, with regard to the front center signal C_F a phase difference of about 19° exists between the output signals L_T and R_T and, therefore, the sound image localization of the front center signal becomes somewhat indistinct during **2**-channel stereophonic reproduction. With regard to the left and right-center signals L_C and R_C separation between the output signals L_T and R_T stays at 15.3 db. Generally, however, this separation is sufficient from the practical view point.

FIG. **6** shows the vector diagrams of the output signals L_T and R_T involved when $\Delta_1=0.3714$ and $\Delta_2=\Delta_3=0.2$. Under this blend condition, with regard to the front-center signal C_F the output signals L_T and R_T are in phase and a phase difference between the front input signals is held down to 17°. With respect to the left and right-center signals there is obtained separation of **16** db between output signals L_T and R_T, and separation between the diagonal channels is 19.57 db.

FIG. **7** shows a vector diagram of the output signals L_T and R_T involved when $\Delta_1=0.383$, $\Delta_2=0.330$ and $\Delta_3 = 0.2082$. Under the blend condition thus far described, with regard to the respective front input signals or front-center signals, encoding is effected to cause the output signals L_T and R_T to be in phase. When with regard to the respective front input signals the output signals L_T and R_T are caused to be in phase, a phase difference between thee output signals becomes greater with respect to the front-center signal. When with regard to the front-center signal the output signals are caused to be in phase, a phase difference between the output signals becomes greater with respect to the respective front input signals. In the case of FIG. **7**, with regard to either the respective front input signals or the front-center input signal, a phase difference between the output signals L_T and R_T can be held down to about 10°. More particularly a phase difference between the respective front input signals is 10.27° and a phase difference between the front-center input signals is 10.07°. In this case, left and right separation with respect to left and right-center signals L_C and R_C is about 17 db and a better compatibility is obtained with regard to a **2**-channel stereophonic reproduction. Furthermore, separation between the diagonal channels is 25.35 db.

With the encoding system according to this invention, the blend quantity Δ_1 may be varied between the front and back directions; the blend quantity Δ_2 between the left and right directions and the blend quantity Δ_3 between the two diagonal directions. According to this invention, the blend quantities Δ_1, Δ_2 and Δ_3 may take any value below unity.

FIG. **8** shows one example of an encoding system according to this invention in which elements identical to those shown in FIG. **1** are designated by the same reference characters, so that any detailed explanation is believed unnecessary. In FIG. **8**, shifting by $\phi-90°$ under the action of a phase shifter **9** the phase of a signal produced by inverting a signal L_F, for example, under the action of an inverter **13** is equivalent to shift of the phase of the signal L_F by $\phi+90°$ without inverting this signal, and shifting by a reference angle ϕ under the action of a phase shifter **7** the phase of a signal produced by inverting a signal L_B under the action of an inverter **14** is equivalent to shift of the phase of signal L_B by $\phi\pm180°$ without inverting the signal.

With the encoding system according to this invention, it is possible to optionally vary a phase difference between the respective input signals by varying the blend condition, and therefore it is possible to produce left and right channel signals having a compatibility practically sufficient for the conventional 2-channel stereophonic reproduction in respect of separation between the left and right channels, image localization involved in the case of front-center input signals and sound quality. Thus, the encoding system can be of simple construction because phase shifters as used in the conventional encoding system can be used intact.

What we claim is:

1. An encoding method in which, in coupling to respective first and second channels at least first and second sound input signals associated with front channels and at least third and fourth sound input signals associated with back channels to generate first and second channel signals, said first, second, third, and fourth sound input signals are coupled to the first and second channels with such an amplitude relation that the amplitude levels of said first and third sound input signals included in said first channel signal are greater than those of the first and third sound input signals included in the second channel signal and the amplitude levels of the second and fourth sound input signals included in said second channel are greater than those of second and fourth sound input signals included in the first channel signal, and in such a phase relation that the first and second sound input signals included in said first channel signal are in a substantially in-phase relation to the first and second sound input signals included in said second channel signal respectively and the third and fourth sound input signals included in said first channel signal is in a substantially opposite relation to the third and fourth sound input signals included in said second channel signal respectively: said encoding method comprising the steps of coupling said first sound input signal to said first channel at a relatively large amplitude level and at a reference phase shift angle and to said first channel at a relatively small amplitude level and at a phase shift angle corresponding to the reference angle plus 90°; coupling said first sound input signal to said second channel at a relatively small amplitude level and at the reference phase shift angle and to said second channel at a relatively small amplitude level and at a phase shift angle corresponding to the reference angle plus 90°; coupling said second input signal to said first channel at a relatively small amplitude level and at the reference phase shift angle and to said first channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90°; coupling said second sound input signal to said second channel at a relatively large amplitude level and at the reference phase shift angle and to said second channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90°; coupling said third sound input signal to said first channel at a relatively large amplitude level and at a phase shift angle corresponding to the reference angle plus 90° and to said first channel at a relatively small amplitude level and at the reference phase shift angle; coupling said third sound input signal to said second channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle minus 90° and to said second channel at a relatively small amplitude level and at a phase shift angle displaced 180° from said reference angle; coupling said fourth sound input signal to said first channel at a relatively small amplitude level and at the phase shift angle corresponding to the reference angle plus 90° and to said first channel at a relatively small amplitude level and at the phase shift angle displaced 180° from said reference angle; and coupling said fourth sound input signal to said second channel at a relatively large amplitude level and at the phase shift angle corresponding to the reference angle minus 90° and to said second channel at a relatively small amplitude level and the reference phase shift angle.

2. An encoding method according to claim **1** in which said first and second channel signals L_T and R_T are represented by the following matrix

$$\begin{bmatrix} L_T \\ R_T \end{bmatrix} = \begin{bmatrix} 1+j\Delta_2 & j+\Delta_2 & \Delta_1-j\Delta_3 & j\Delta_1-\Delta_3 \\ \Delta_1+j\Delta_3 & -j\Delta_1-\Delta_3 & 1-j\Delta_2 & -j+\Delta_2 \end{bmatrix} \cdot \begin{bmatrix} L_F \\ L_B \\ R_F \\ R_B \end{bmatrix}$$

wherein L_F, R_F, L_B and R_B represent the first, second, third and fourth sound input signals, respectively; and Δ_1, Δ_2 and Δ_3 represent coefficients each having a value less than unity.

3. An encoding method according to claim **2** in which said coefficients have a relation of $\Delta_1 = \Delta_2 = \Delta_3$.

4. An encoding method according to claim **3** in which said coefficients have a relation of $\Delta_1 = \Delta_2 = \Delta_3 =$ about 0.3.

5. An encoding method according to claim **2** in which said coefficients have a relation of $1:\Delta_2 = \Delta_1:\Delta_3$.

6. An encoding method according to claim **5** in which said coefficients Δ_1 and Δ_2 each have a value of about 0.41 and said coefficient Δ_3 has a value of about 0.17.

7. An encoding method according to claim **2** in which said coefficient Δ_1 has a value of about 0.37 and said coefficients Δ_2 and Δ_3 each have a value of about 0.2.

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