

(19)



(11)

EP 1 964 443 B1

(12)

EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
21.09.2011 Bulletin 2011/38

(21) Application number: **06837740.7**

(22) Date of filing: **16.11.2006**

(51) Int Cl.:
H04S 3/02 (2006.01)

(86) International application number:
PCT/US2006/044447

(87) International publication number:
WO 2007/067320 (14.06.2007 Gazette 2007/24)

(54) **LOW-COMPLEXITY AUDIO MATRIX DECODER**

AUDIOMATRIX-DECODER MIT VERRINGERTER KOMPLEXITÄT

DECODEUR AUDIO MATRICIEL DE FAIBLE COMPLEXITE

(84) Designated Contracting States:
DE FR GB

(30) Priority: **02.12.2005 US 741567 P**

(43) Date of publication of application:
03.09.2008 Bulletin 2008/36

(73) Proprietor: **DOLBY LABORATORIES LICENSING CORPORATION**
California 94103-4813 (US)

(72) Inventors:
• **CHEN, Ching-wei**
San Francisco, CA 94103-4813 (US)
• **CHABANNE, Christophe**
San Francisco, CA 94103-4813 (US)

(74) Representative: **MERH-IP**
Matias Erny Reichl Hoffmann
Paul-Heyse-Strasse 29
80336 München (DE)

(56) References cited:
EP-A1- 0 630 168 US-A- 3 864 516
US-A- 4 799 260

- **DOLBY PRO LOGIC SURROUND DECODER PRINCIPLES OF OPERATION:[Online]**
XP002421954 Retrieved from the Internet: URL:
<http://web.archive.org/web/19961018090112/www.dolby.com/ht/ds&pl/whtppr.html>
[retrieved on 1996-10-18]

Note: Within nine months of the publication of the mention of the grant of the European patent in the European Patent Bulletin, any person may give notice to the European Patent Office of opposition to that patent, in accordance with the Implementing Regulations. Notice of opposition shall not be deemed to have been filed until the opposition fee has been paid. (Art. 99(1) European Patent Convention).

EP 1 964 443 B1

Description**Technical Field**

5 [0001] The invention relates to audio signal processing. More particularly, the invention relates to a low-complexity adaptive audio matrix decoder or decoding process usable for decoding both encoded and non-encoded input signals. Although usable as a stand-alone decoder or decoding process, the decoder or decoding process may be advantageously used in combination with a "virtualizer" or "virtualization" process such that the decoder or decoding process provides multichannel inputs to the virtualizer or virtualization process. The invention also relates to computer programs, stored
10 on a computer-readable medium, for causing a computer to perform a decoding process or decoding and virtualization process according to aspects of the invention.

Background Art

15 [0002] "Virtual headphone" and "virtual loudspeaker" audio processors ("virtualizers") typically encode multichannel audio signals, each associated with a direction, into two encoded channels so that, when the encoded channels are applied to a pair of transducers such as a pair of headphones or a pair of loudspeakers, a listener suitably located with respect to the transducers perceives the audio signals as coming from locations that may be different from the location of the transducers, desirably the directions associated with the directions of the multichannel audio signals. Headphone
20 virtualizers typically result in a listener perceiving that the sounds are "out-of-head" rather than inside the head. Both virtual headphone and virtual loudspeaker processors involve the application of head-related-transfer-functions (HRTFs) to multichannel audio signals applied to them. Virtual headphone and virtual loudspeaker processors are well known in the art and are similar to each other (a virtual loudspeaker processor may differ from a virtual headphone processor, for example, by including a "crosstalk canceller").

25 [0003] Examples of headphone and loudspeaker virtualizers include virtualizers sold under the trademarks "Dolby Headphone" and "Dolby Virtual Speaker." "Dolby", "Dolby Headphone", and "Dolby Virtual Speaker" are trademarks of Dolby Laboratories Licensing Corporation.. Patents and an application relating to Dolby Headphone and Dolby Virtual Speaker include U.S. Patents 6,370,256; 6,574,649; and 6,741,706 and published International Application WO 99/14983. Other "virtualizers" include, for example, those described in U.S. Patent 6,449,368 and published International
30 Patent Application WO 2003/053099.

[0004] Dolby Headphone and Dolby Virtual Speaker provide, respectively, the impression of multichannel surround sound using a pair of standard headphones or a pair of standard loudspeakers. Recently, low-complexity versions of Dolby Headphone and Dolby Virtual Speaker were introduced that are useful, for example, in a wide variety of new, low-cost products, such as multimedia mobile phones, portable media players, portable game consoles, and low-cost tele-
35 vision sets. However, such low-cost products typically are two-channel stereophonic ("stereo") devices; whereas a virtualizer requires a multichannel surround sound input.

[0005] A method according to the pre-characterizing portion of claim 1 is known from DOLBY PRO LOGIC SURROUND DECODER PRINCIPLES OF OPERATION: [Online] XP002421954 Retrieved from the Internet: URL: <http://web.archive.org/web/19961018090112/www.dolby.com/html/ds&pl/whthppr.html> > [retrieved on 1996-10-18]

40 [0006] Although existing matrix decoders, for example Dolby Pro Logic II and its predecessor Pro Logic, are useful in matching the two-channel stereo audio output of low-cost devices to the multichannel surround sound input of a Dolby Headphone virtualizer, existing matrix decoders typically may be more complex and resource intensive than desirable for use with some low-cost devices. "Dolby Pro Logic" and "Dolby Pro Logic II" are a trademarks of Dolby Laboratories Licensing Corporation. Aspects of Dolby Pro Logic II are set forth in U.S. Patents 6,920,223 and 6,970,567 and in published International Patent Application WO 2002/019768. Aspects of Dolby Pro Logic are set forth in U.S. Patents
45 4,799,260; 4,941,177; and 5,046,098.

[0007] Thus, it is an object of the present invention to provide a low-complexity matrix decoding method and apparatus, in particular one intended and optimized for use with virtualizers, particularly virtualizers such as Dolby Headphone and Dolby Virtual Speaker. Ideally, such a new matrix decoding method and apparatus should minimize complexity in every
50 stage of the process, while obtaining performance similar to a Dolby Pro Logic II decoder.

Disclosure of the Invention

55 [0008] This object is achieved by a method as claimed in claim 1, an apparatus as claimed in claim 23 and a computer program as claimed in claim 24. Preferred embodiments of the invention are defined in the dependent claims.

[0009] Although aspects of the present invention are usable with other types of matrix decoders, in an exemplary embodiment a fixed-matrix-variable-gains approach is employed because of its low complexity compared to the variable-matrix approach. The excellent isolation of single sound sources occurring with use of a variable gains decoder may be

acceptable, if not preferable, for game audio where single audio events may be common.

[0010] When working with virtualizers, it is desirable to reduce inter-channel leakage as much as possible, because of the interactions and cancellations between and among the Head Related Transfer Functions (HRTFs) of different channels. The variable gains approach allows turning off certain channels completely, keeping inter-channel leakage to a minimum.

[0011] Furthermore, "pumping" side-effects that may occur under certain signal conditions when using a variable gains decoder are not as objectionable when used in conjunction with a virtualizer. This is because of the nature of virtualizers to produce two channels of output for every one input channel. Although a variable gains matrix decoder may cause certain speakers to turn off completely, neither of the two outputs of a virtualizer turn off completely as long as at least one of its inputs is active.

[0012] As explained further below, optimizations may be made to deal with another known disadvantage of the variable-gains approach - the loss of non-dominant signals, resulting in a decoder with the best of both worlds.

[0013] Also, because one use of the matrix decoder according to aspects of the invention is to derive multichannel content for virtualizers, the number of outputs may be restricted to four: Left, Right, Left Surround, and Right Surround. Indeed, the main goal of virtualizers is to convey a good sense directionality all around the listener; this may be achieved using only four channels, omitting the center channel, the inclusion of which would have significantly increased the processing execution time, while marginally enhancing the perception of directionality.

[0014] Because destructive interferences occur when Head Related Transfer Functions (HRTFs) are summed together; it is preferable to avoid correlations between and among channels. In other words, virtualizers perform better when sources are steered as much as possible toward one speaker at a time. However, achieving such a result should be balanced against compromising the overall soundstage.

Description of the Drawings

[0015]

FIG. 1 is a schematic functional block diagram showing an example of a processor or process according to aspects of the present invention for deriving pairs of intermediate control signals from a plurality of audio input signals, the pairs of intermediate controls signals representing signal strength in opposing directions along a directional axis. In this example, which may be designated "Stage 1," there are two audio input signals, Lin and Rin and there are two pairs of intermediate control signals, L-R and F-B.

FIG. 2 is a schematic functional block diagram showing an example of a processor or process according to aspects of the present invention for deriving a plurality of directional dominance signals, at least one such signal for every pair of intermediate control signals. In this example, which may be designated "Stage 2," there are two pairs of intermediate control signals, L-R and F-B and two directional dominance signals, LR and FB.

FIG. 3 shows an example of a notional or theoretical directional dominance vector in a two-dimensional plane based on orthogonal LR and FB axes.

FIG. 4 is an idealized plot of signal amplitude versus time showing the absolute values L and R, respectively, of a two-channel stereo signal in which the Left input-channel (Lin) before taking its absolute value is a 50 Hz sine wave with a peak amplitude of 0.4, and the Right input channel (Rin) before taking its absolute value is a sine wave with frequency of $(50 * \sqrt{2})$ Hz and peak amplitude of 1.0. The frequencies of the sine waves are uncorrelated, while the level of the Left channel is 0.4 times the level of the Right channel.

FIG. 5 is an idealized plot of signal amplitude versus time showing both the result of subtracting L from R and the result of multiplying the difference and then clipping at -1.0 and +1.0 to provide a quasi-rectangular wave.

FIG. 6 is an idealized plot of signal amplitude versus time showing a smoothed LR intermediate control signal resulting from feeding the quasi-rectangular wave of FIG. 5 through a smoother filter, illustrating that, for substantially non-correlated signal inputs, the directional dominance signal approaches a value close to a value that would result from a ratio-based comparison of signal strengths along the directional axis to which the LR intermediate control signal relates.

FIG. 7 is a schematic functional block diagram showing an example of a modification of the processor or process according to aspects of the present invention shown in FIG. 2. In this example, which may also be designated "Stage 2," the amplified and clipped FB difference is limited to values less than zero in order to bias the FB dominance signal towards the back.

FIG. 8 is an idealized plot of gain versus angle in radians showing a common pan-law between Left (L) and Right (R) audio channels, a sine/cosine pan-law where $L = \cos(x) * \text{input}$, and $R = \sin(x) * \text{input}$, with x varying from 0 to $\pi/2$.

FIG. 9a is an idealized plot of gain versus directional dominant signal level for panL and panR when the same Sine/Cosine pan-law of FIG. 8 is applied to the LR axis, panL and panR representing the gain contribution, respectively, from Left and Right.

FIG. 9b is an idealized plot of gain versus directional dominant signal level for panB and panF when the same Sine/Cosine pan-law of FIG. 8 is applied to the FB axis, panB and panF representing the gain contribution, respectively, from Back and Front.

FIG. 10 is an idealized plot showing a quasi-3-dimensional representation of the LGain equation (the axes being normalized gain, and the values of FB and LR).

FIG. 11 is an idealized plot showing a quasi-3-dimensional representation of the LGain, RGain, LsGain and RsGain equations (the axes being normalized gain, and the values of FB and LR).

FIG. 12 is an idealized plot showing a cosine curve and a second-order polynomial approximation of a cosine curve between 0 and $\pi/2$, showing that the approximation, $y = (1 - x^2)$ is reasonably close to $y = \cos(x * \pi/2)$ within the range $0 < x < 1$. The lower curve is the approximation.

FIG. 13 is an idealized plot showing a quasi-3-dimensional representation of a modification to the LGain, RGain, LsGain and RsGain equations (the axes being normalized gain, and the values of FB and LR) in which the LR panning component is not employed when calculating LGain and RGain.

FIG. 14 is a schematic functional block diagram showing an example of a processor or process according to aspects of the present invention for deriving a plurality of control signals from the plurality of directional dominance signals. In this example, which may be designated "Stage 3," four control signals LGain, RGain, LsGain and RsGain are derived from two directional dominance signals LR and FB.

FIG. 15 is a schematic functional block diagram showing an example of an adaptive matrix processor or process according to aspects of the present invention for deriving a plurality of audio output signals from the input audio signals and a plurality of control signals. In this example, which may be designated "Stage 4," a pair of audio input signals Lin and Rin are applied to a passive matrix and the level of each matrix output is controlled by a respective one of the four control signals LGain, RGain, LsGain and RsGain to produce four audio output signals LOut, ROut, LsOut and RsOut.

FIG. 16 is a schematic functional block diagram showing an overview of all four Stages of the example, indicating their inter-relationships.

Best Mode for Carrying Out the Invention

[0016] Aspects of the present invention may be better understood in connection with an exemplary embodiment, which embodiment may be broken into four "stages" for convenience in description. The overall relationship of the four stages in the context of an adaptive matrix audio decoder or decoding process receiving m input audio signals, two signals, Lin and Rin in this example, and outputting n audio signals, four signals, LOut (left out), ROut (right out), LsOut (left-surround out), and RsOut (right-surround out), in this example, is shown in FIG. 16. The decoder or decoding process has a control path that includes Stages 1, 2 and 3 and a signal path that includes an adaptive matrix or matrixing process in Stage 4. A plurality of time-varying control signals, four control signals in this example, are generated by the control path and are applied to the adaptive matrix or matrixing process.

Stage 1

[0017] Turning first to Stage 1, shown in FIG. 1, m audio input signals, Lin and Rin in this example, are applied to a processor or process that derives pairs of signals in response to the m audio input signals, a first pair of signals, L and R in this example, representing signal strength in opposing directions along a first directional axis, an L-R or Left-Right axis in this example, and a second pair of signals, F and B in this example, representing signal strength in opposing directions along a second directional axis, an F-B or Front-Back axis in this example. Although this example employs two directional axes that are orthogonal, there may be more than two directional axes (and, hence, more than two pairs of signals representing signal strength in opposing directions along respective ones of additional directional axes) and the axes need not be orthogonal (see, e.g., said U.S. Patent 6,970,567). The processor or process of Stage 1 may be viewed as a passive matrix or matrixing process. In this example, a simple passive matrix computes Left, Right, Sum and Difference signals, and their absolute values are used as intermediate control signals L, R, F, and B. More specifically, the passive matrix or passive matrixing process of this example may be characterized by the following equations:

$$L = |Lin|$$

$$R = |Rin|$$

$$F = | (0.5 * Lin) + (0.5 * Rin) |$$

$$B = | (0.5 * Lin) - (0.5 * Rin) |$$

Stage 2

[0018] Turning next to Stage 2, shown in FIG. 2, the plurality of pairs of signals, each pair representing a signal strength in opposing directions along a directional axis, are applied to a processor or process that produces a plurality of directional dominance signals. In this example, there are two pairs of signals L-R and F-B applied to Stage 2 and two directional dominance signals, LR and FB are produced by Stage 2. In principle, as mentioned above, there may be more than two directional axes (and, hence, more than two pairs of signals and more than two directional dominance signals). It is also possible to produce more directional dominance signals than there are pairs of signals and related axes. This may be accomplished by processing a pair of applied signals in more than one way so as to produce multiple directional dominance signals in response to a particular pair of applied signals. Before turning to details of a Stage 2 example, it is useful to explain the operational rationale of Stage 2.

[0019] Having obtained a measure of the signal's strength in each of the four directions (L, R, F, B), one would like to compare the strength in one direction against the strength in the opposite direction (L against R, and F against B) to provide a measure of the dominance along that directional axis. Because the four directions of this example provide two directional axes at 90-degrees to each other (orthogonal axes), such a pair of dominances may be interpreted as a single dominance vector on a 2-dimensional LR/FB plane. Such a notional or theoretical dominance vector may be shown as in the example of FIG. 3. Although such a dominance vector is implicit in the operation of a matrix decoder or decoding process in accordance with aspects of the invention, such a dominance vector need not be explicitly calculated.

[0020] A negative value along the LR axis may indicate dominance towards the Left, while a positive LR value may indicate dominance towards the Right. Similarly, a negative FB value may indicate dominance towards the Back, while a positive FB value may indicate dominance towards the Front. Interpreting the two dominance values as components of a 2D vector, one may visualize the dominance of a signal as lying anywhere on the LR/FB plane.

[0021] In most modem matrix decoders, including Dolby Pro Logic and Dolby Pro Logic II, the dominance in the LR direction is computed using the ratio of L and R, and the dominance in the FB direction is computed using the ratio of F and B. Because a ratio is independent of the magnitude of the two signals being compared, it provides a steady dominant direction throughout the natural amplitude variations found in real audio signals. Unfortunately, if implemented by a computer program controlling a digital signal processor ("DSP"), such an approach requires case statements in the program to choose the numerator and denominator, as well as to assign the sign to the dominance value. More importantly, common methods of deriving a ratio, such as a division, or a subtraction in the log domain, require significant computational resources. A more simplistic approach of subtracting the two numbers in the linear amplitude domain (e.g., not the logarithmic-domain) is certainly more efficient to compute, but such subtraction produces dominance signals that change rapidly with natural variances in signal amplitudes.

[0022] To reduce the complexity of implementation, aspects of the present invention retain much of the amplitude independence of the ratio-based comparison, but require much less computation.

[0023] The processor or process of Stage 2 produces a plurality of directional dominance signals using linear-amplitude-domain subtractors or subtraction processes that obtain a positive or negative difference between the magnitudes of each pair of applied signals. Such subtraction may be implemented with very low computation resources. The result of each subtraction is amplified by an amplifier or amplification process and the amplified difference is applied to a clipper or clipping process that limits each of the amplified differences substantially at a positive clipping level and a negative clipping level. Alternatively, the order of the amplifier/amplification process and the clipper/clipping process may be reversed, using appropriate clipping levels in order to produce an equivalent result. A smoother or smoothing process may time average each of the amplified and limited differences to provide a directional dominance signal.

[0024] The relationship between the amplification factor of the amplifier or amplification process and the clipping level at which the clipper or clipping function limits the amplified difference constitutes a positive and negative threshold for magnitudes below which the limited and amplified difference has an amplitude between zero and substantially the clipping level, and above which the limited and amplified difference has an amplitude substantially at the clipping level. Although the particular transfer function is not critical and may take many forms, a transfer function in which the limited and amplified difference with respect to the difference is substantially linear between the thresholds has very low computational requirements and is suitable.

[0025] The processor or process of Stage 2 may include modifications to an amplified and limited difference signal

prior to or after smoothing during its processing so that the derived directional dominance signal is "biased" along the axis to which the directional dominance signal relates. The bias may be fixed or adaptive. For example, a difference signal after amplification and clipping may be scaled in amplitude and/or shifted in amplitude (*i.e.*, offset) and/or restricted in amplitude or sign in a fixed manner or, for example, as a function of the magnitude, sign, or magnitude and sign of the amplified and clipped difference signal. The result, for example, may include the application of less bias to non-dominant signals than to dominant signals (dominance and non-dominance are explained further below). An example of applying "bias" to a directional dominance is described below in connection with FIG. 7.

[0026] In the Stage 2 example of FIG. 2, two pair of signals, L-R and F-B, are applied in order to produce two directional dominance signals LR and FB. Given the four intermediate directionality signals (L, R, F, B), as described above, one would like to derive two dominance signal components, LR and FB, by comparing the directionality along each axis. According to aspects of this invention, this is accomplished by subtracting R from L, and B from F (or vice-versa in each case), to provide a magnitude difference signal along each axis. Heavy gain is applied to the difference signals, and the amplified difference is clipped (hard limited) to -1.0 and +1.0. The clipped difference signal is then applied to a time-smoothing filter.

[0027] By applying heavy gain and clipping to the difference signals, essentially any amount of dominance in a direction is treated as an absolute dominance in that direction. For signals where instantaneous directionality changes from one polarity to another, the result of this operation is similar to a rectangular wave with varying frequency and duty-cycle. The time-smoothing filter averages out the mostly rectangular wave to provide a continuous curve that approximates a ratio of the original directionality signals to one another. Although the exact filter used is a design choice, the filter may be implemented efficiently, for example, as a first order digital IIR lowpass filter having a time constant of about 40 ms.

[0028] In addition to detecting the dominant direction along each axis, it may be advantageous to represent "non-dominance." For example, a purely Left-steered input signal should exhibit a strong dominance on the Left-Right axis, but should have absolutely no dominance along the Front-Back axis. Another example is for extremely low level signals such as background noise, which one would prefer not to cause any steering effects. In accordance with aspects of the invention, a general approach to doing this is to choose a threshold value, and assign differences with a magnitude greater than the threshold a value of -1.0 or 1.0 (depending on the sign of the difference), and assign differences with magnitudes smaller than this threshold some value in between the two extremes. One possibility is to assign a value of 0.0 to all difference values below the threshold. To implement this in a program-controlled DSP would require some case statements and numerical comparisons. A better approach from the standpoint of low complexity is to amplify the difference by a large gain such that the output of values below the threshold follow a linear function from -1.0 to +1.0. The gain is the inverse of the threshold. This approach is very efficient - both the gain and clipping stages may be implemented in a program-controlled DSP as an arithmetic left shift (for gains that are a power of 2) with the DSP's "saturation logic" set (*i.e.*, set a control register/bit in the DSP so that when the ALU overflows, the result is set to the maximum positive value or minimum negative value represented by the platform, depending on the sign). Gains that are not a power of two may be implemented with only a slight increase in processing complexity.

[0029] A three-regioned dominance signal (negative dominance, positive dominance, and non-dominance) permits distinguishing between dominance and non-dominance along a directional axis before smoothing. Distinguishing dominance and non-dominance facilitates the adaptive application of "bias" to a directional dominance signal, as mentioned above and an example of which is given below in connection with FIG. 7. For example, as shown below, it is useful in aspects of the present invention for distinguishing, before smoothing, a solely Left-steered signal from a Left Surround-steered signal, and a solely Right-steered signal from a Right Surround-steered signal.

[0030] In a practical embodiment of the invention, to determine the minimum gain necessary to distinguish a side (Left or Right)-steered signal from a Surround (Left Surround or Right Surround)-steered signal, musical material encoded with a Dolby Pro Logic II matrix encoder was decoded. The average (F-B) difference signal was measured for a Left Surround- or Right Surround-steered input and this was used as an estimate of the maximum threshold (minimum gain) that would maintain a clear distinction between Left and Left Surround (or Right and Right Surround). In this practical embodiment of a decoder according to aspects of the invention, a gain factor of 1024 was used, equivalent to a threshold of approximately 0.001 for signals normalized to [-1 +1]. Thresholds smaller than 0.001 produce marginal audible improvement, while larger thresholds reduce the separation between the sides (Left and Right) and the surrounds (Left Surround and Right Surround) to unacceptable levels. In general, the threshold level is not critical.

[0031] To illustrate this technique, consider a two-channel stereo signal where the Left input channel (Lin) is a 50 Hz sine wave with a peak amplitude of 0.4, and the Right input channel (Rin) is a sine wave with frequency of $(50 * \sqrt{2})$ Hz and peak amplitude of 1.0. Such signals are shown in FIG. 4. The frequencies of the sine waves are uncorrelated, while the level of the Left channel is 0.4 times the level of the Right channel. Using a ratio-based comparison as described above, this provides a dominance in the Right direction (defined as positive here) of 0.6. As shown in Stage 1, the L and R intermediate signals are the magnitudes of the input signals Lin and Rin.

[0032] After subtracting L from R, the difference is multiplied, for example, by 1024 (implemented as an arithmetic left shift of 10 bits), and then clipped at -1.0 and +1.0 to provide a quasi-rectangular wave. FIG. 5 shows the difference

signal before and after clipping.

[0033] Feeding the quasi-rectangular wave through a smoother filter that provides the LR directional dominance signal. In this example in which the input signals have constant levels, the directional dominance signal eventually reaches and oscillates around a value of 0.65, as shown in FIG. 6, close to the dominance value computed using a ratio-based comparison. The smoothness of the oscillation is a function of the order and characteristics of the smoother filter.

[0034] This example is representative of audio material that has significant amounts of uncorrelated signals in each input, such as un-encoded two-channel stereo music, where the polarity of the clipped amplified difference signal is inverted very often. Under these input conditions, the subtract/amplify/clip derived dominance control signal produces results close to those obtained from a ratio-based comparison.

[0035] However, for material with common (*i.e.*, correlated) signals in both channels, such as a steered mono sound source contained in matrix-encoded content, the clipped difference signal does not contain many zero crossings. In such cases, even the smoothed control signal tends to "lock" to one of the two extremes (*i.e.*, +1.0 and -1.0), with a smoothed transition across to the other extreme if and when the polarity of the difference signal eventually inverts. Such "locking" of one dominance component may be thought of as pulling a 2-dimensional dominance vector out along the edges of the LR/FB plane. When both components are "locked", the dominance vector is pulled to one of the four corners of the LR/FB plane. According to aspects of the present invention, such hard-panning improves the spatial imaging of matrix-encoded content, by providing a more discrete, single channel of input to a virtualizer.

Front-Back Dominance Bias

[0036] A shortcoming of the variable gain approach is that non-dominant signals may be lost in the decoded output. This is apparent in musical sound sources, where there are a large number of sound sources mixed together with many different level and phase differences. Often, there are a few main instruments and vocals mixed equally in both Left and Right, while there are still many other less dominant, out-of-phase sounds that add to the overall space and ambience of the soundfield. Because the decoder uses only the direction of the most dominant sound component, a traditional variable gains approach on such material may result in almost no output of the out-of-phase material from the rear decoder outputs (the Left Surround and Right Surround outputs in the example).

[0037] According to an aspect of the present invention, this problem is mitigated by biasing the FB dominance signal towards the back, assuring that out-of-phase material is not completely removed from the surround outputs. One way to accomplish this is to limit the FB signal to negative values before the smoother filter. This is shown in the example of FIG. 7. For a pure rectangular wave between -1.0 and 1.0, this is equivalent to scaling down by half the output of the smoother filter followed by a fixed offset of -0.5. Thus, such a modification may be imposed either before or after the smoother filter. However, the clipped difference signal may not be a pure rectangular wave. Rather, it may contain in-between values when the difference signal falls below the threshold value, indicating non-dominance along a particular axis. When the magnitude of the clipped difference signal is smaller than 1.0, the process of limiting FB to negative values results in a smaller to negligible effective bias after smoothing. Thus, being able to distinguish non-dominance from positive or negative dominance before smoothing in this way allows pure Left and Right-steered signals to maintain high separation from the surrounds, while giving most other signals a significant bias towards the back.

Stage 3

[0038] The processor or process of Stage 3 produces control signals for controlling the adaptive matrix or matrixing process in response to the plurality of directional dominance signals by applying one or more panning functions (a panning function is a transfer function representing an interchannel "panning" characteristic) to each of the directional dominance signals. One or more of the panning functions may implement one or more of:

- a trigonometric transfer function (such as a sine or cosine transfer function),
- a logarithmic transfer function,
- a linear transfer function, and
- a mathematically simplified approximation of a trigonometric transfer function.

[0039] The goal of Stage 3, in the example, is to take the LR and FB dominance signals computed in the previous Stage, and derive the gain factors that are applied to the outputs of the passive matrix to produce the decoded outputs.

[0040] The general approach for the matrix decoder or decoding process according to aspects of the present invention is this: having detected a certain dominant directionality in the input, emphasize the output channels closest to that dominant location, and de-emphasize the outputs furthest from the dominant location. Between the two outputs closest to the dominant location, the problem may be reduced to a pair-wise pan, which may be expressed as a panning function.

Sine/Cosine Pan-law

[0041] The most common pan-law between two channels is the sine/cosine pan-law where $L = \cos(x) \cdot \text{input}$, and $R = \sin(x) \cdot \text{input}$, with x varying from 0 to $\pi/2$. See FIG. 8.

2-Dimensional Sine/Cosine Pan-law

[0042] The gain for each decoder output channel must be expressed as a function of LR and FB:

$$LGain = f_L(LR, FB)$$

$$RGain = f_R(LR, FB)$$

$$LsGain = f_{Ls}(LR, FB)$$

$$RsGain = f_{Rs}(LR, FB)$$

[0043] One may apply to the LR and FB axes the same Sine/Cosine pan-law described above, and obtain the panning curves shown in FIGS. 9a and 9b, where panL, panR, panB, and panF represent the gain contribution from respectively Left, Right, Back and Front.

[0044] Recognizing that the sine function is a cosine with a phase shift, one may obtain the following panning equations using only cosine functions:

$$panL = \cos((LR + 1) / 2 * \pi/2)$$

$$panR = \sin((LR + 1) / 2 * \pi/2) = \cos((LR - 1) / 2 * \pi/2)$$

$$panB = \cos((FB + 1) * \pi/2)$$

$$panF = \sin((FB + 1) * \pi/2) = \cos(FB * \pi/2)$$

[0045] By the nature of the left channel location on the LR/FB plane (see FIG. 3), LGain should be maximum only when both panL and panF are maximum, and should decrease as the dominance gets farther away on both, or either of the axis. This may be achieved by multiplying panL with panF. The same principle may be applied to RGain, LsGain and RsGain, and the final equations for all gains become:

$$LGain = panL * panF$$

$$RGain = panR * panF$$

$$LsGain = panL * panB$$

$$RsGain = panR * panB$$

[0046] The use of a multiplication may also be seen as a mutual scaling of the two Sine/Cosine amplitude-panning functions, where the smallest value of the two components becomes the largest value that the overall gain can reach.

[0047] FIG. 10 shows the 3-dimensional representation of the LGain equation, and FIG. 11 the 3-dimensional representation of all four gains superimposed.

Polynomial Approximation of a Cosine Function

[0048] As shown in FIG. 8, the pan-law is composed of two curves: $\cos(x)$ and $\sin(x)$. (The sin function can be replaced by a cos function with the appropriate phase shift.) In order to avoid complex computations or using large lookup tables, in accordance with an aspect of the present invention, a second-order polynomial approximation of a cosine curve between 0 and $\pi/2$ may be used instead. The equation $y = (1 - x^2)$ is reasonably close to $y = \cos(x * \pi/2)$ within the range $0 < x < 1$. (see FIG. 12 in which the lower curve is the approximation). There may be little to no audible difference resulting from the use of this approximation.

Front Panning Adjustment

[0049] Because the anticipated audio input source is two-channel stereo, which is already mixed to pan naturally between L and R, it is an aspect of the present invention not to consider the LR panning component when calculating LGain and RGain. The additional left-right panning in the variable gains would not significantly improve separation in this case, since L and R are already well separated. In addition to saving some computation, it also allows a more stable soundfield in the front, by avoiding unnecessary gain riding. Removing the LR component, one arrives at these equations:

$$LGain = panF$$

$$RGain = panF$$

$$LsGain = panL * panB$$

$$RsGain = panR * panB$$

[0050] The 3-dimensional representation of these new equations is shown in FIG. 13.

[0051] Note that a similar simplification may be applied to the Ls gain and Rs gain equations, whereby no additional LR panning is used, and the natural panning within the source signal is used to create separation between the two surround channels. However, in such a case, the Ls and Rs separation is limited by the performance of the passive decoding taking place in Stage 4. A passive decoding matrix or matrixing process, such as forms part of aspects of the present invention, can only achieve a 3 dB separation between Ls and Rs, therefore making this simplification unacceptable from a channel separation standpoint. In order to maintain a higher degree of separation, the LR component in the equations of LsGain and RsGain is retained.

Final Gain Equations

[0052] Substituting the polynomial approximation for the cosine in each panning term, one may derive the final equation for each gain factor:

$$\text{LGain} = 1 - \text{FB}^2$$

$$\text{When FB} = 0 \rightarrow \text{LGain} = 1$$

$$\text{When FB} = -1 \rightarrow \text{LGain} = 0$$

$$\text{RGain} = 1 - \text{FB}^2$$

$$\text{When FB} = 0 \rightarrow \text{RGain} = 1$$

$$\text{When FB} = -1 \rightarrow \text{RGain} = 0$$

$$\text{LsGain} = [1 - ((\text{LR} + 1) / 2)^2] * [1 - (\text{FB} + 1)^2]$$

$$\text{When FB} = 0 \rightarrow \text{LsGain} = 0$$

$$\text{When FB} = -1 \text{ and LR} = -1 \rightarrow \text{LsGain} = 1$$

$$\text{When FB} = -1 \text{ and LR} = 1 \rightarrow \text{LsGain} = 0$$

$$\text{RsGain} = [1 - ((\text{LR} - 1) / 2)^2] * [1 - (\text{FB} + 1)^2]$$

$$\text{When FB} = 0 \rightarrow \text{Rs gain} = 0$$

$$\text{When FB} = -1 \text{ and LR} = -1 \rightarrow \text{RsGain} = 0$$

$$\text{When FB} = -1 \text{ and LR} = 1 \rightarrow \text{RsGain} = 1$$

[0053] Referring to FIG. 14, the control signals LGain, RGain, LsGain, and RsGain are derived from the application of a panning function to a directional dominance signal, and/or the product of the application of a panning function to one directional dominance signal and the application of a panning function to another directional dominance signal, wherein each panning function may be different from ones or all of the other panning functions. The panning functions are panning functions that are not inherent in the n input audio signals. In this example, one of the directional axes is a left/right axis and the panning functions are panning functions that do not include a left/right panning component. The following applies to this example. The LR directional dominance signal is applied to a panL panning function and to a panR panning function. The FB directional dominance signal (either without biasing as in FIG. 2 or with biasing as in FIG. 7) is applied to a panF panning function and to a panB panning function. The result of applying the panF function to the FB dominance signal is applied as both the LGain and as the RGain to the Stage 4 passive decoder or decoding process. The result of applying the panB function to the FB dominance signal is multiplied by the result of applying the panL function to the LR dominance signal and is applied as the LsGain to the Stage 4 passive decoder or decoding process. The result of applying the panR function to the LR dominance signals is multiplied by the result of applying the panB function to the FB dominance signal and is applied as the RsGain to the Stage 4 passive decoder or decoding process.

Stage 4

[0054] FIG. 15 shows a passive matrix or matrixing process that produces n audio signals in response to m audio signals, and amplitude scalers or amplitude scaling processes, each of which amplitude scales one of the audio signals produced by the passive matrix or matrixing process in response to a time-varying amplitude-scale-factor control signal to produce the n audio output signals, wherein the plurality of time-varying control signals are n time-varying amplitude scale factor control signals, one for amplitude scaling each of the audio signals produced by the passive matrix or matrixing process. In the example of FIG. 14, there are two input audio signals, Lin and Rin , four audio output signals $LOut$, $ROut$, $LsOut$ and $RsOut$, and four scale-factor control signals $LGain$, $RGain$, $LsGain$, and $RsGain$ (from Stage 3).

[0055] In the example of FIG. 15, four audio output signals may be characterized by the following equations:

$$LOut = LGain * (a * Lin + b * Rin)$$

$$ROut = RGain * (c * Lin + d * Rin)$$

$$LsOut = LsGain * (e * Lin + f * Rin)$$

$$RsOut = RsGain * (g * Rin + h * Lin)$$

where a through h are matrix coefficients, as indicated in FIG. 15. The coefficients a through h may be chosen to match those used in the Dolby Pro Logic II encode/decode system, where:

$$a = 1.0, b = 0.0,$$

$$c = 0.0, d = 1.0,$$

$$e = 0.8710, f = -0.4898,$$

$$g = -0.4898, h = 0.8710$$

[0056] This provides the final equations:

$$LOut = LGain * Lin$$

$$ROut = RGain * Rin$$

$$LsOut = LsGain * (0.8710 * Lin - 0.4898 * Rin)$$

$$RsOut = RsGain * (0.8710 * Rin - 0.4898 * Lin)$$

5 [0057] FIG. 16 shows an overview of all four Stages of the example, indicating their inter-relationships.

Implementation

10 [0058] The invention may be implemented in hardware or software, or a combination of both (e.g., programmable logic arrays). Unless otherwise specified, the algorithms included as part of the invention are not inherently related to any particular computer or other apparatus. In particular, various general-purpose machines, such as digital signal processors, may be used with programs written in accordance with the teachings herein, or it may be more convenient to construct more specialized apparatus (e.g., integrated circuits) to perform the required method steps. Thus, the invention may be implemented in one or more computer programs executing on one or more programmable computer systems each comprising at least one processor, at least one data storage system (including volatile and non-volatile memory and/or storage elements), at least one input device or port, and at least one output device or port. Program code is applied to input data to perform the functions described herein and generate output information. The output information is applied to one or more output devices, in known fashion.

20 [0059] Each such program may be implemented in any desired computer language (including machine, assembly, or high level procedural, logical, or object oriented programming languages) to communicate with a computer system. In any case, the language may be a compiled or interpreted language.

25 [0060] Each such computer program is preferably stored on or downloaded to a storage media or device (e.g., solid state memory or media, or magnetic or optical media) readable by a general or special purpose programmable computer, for configuring and operating the computer when the storage media or device is read by the computer system to perform the procedures described herein. The inventive system may also be considered to be implemented as a computer-readable storage medium, configured with a computer program, where the storage medium so configured causes a computer system to operate in a specific and predefined manner to perform the functions described herein.

30 [0061] A practical embodiment of the present invention embodied in a computer program suitable for controlling a digital signal processor has been implemented with under 30 lines of C code, running at an estimated 3 MIPS, and using virtually no memory. This is approximately 15% of the estimated MIPS usage of a Dolby Pro Logic II decoder. Processing may remain entirely in the time domain and be performed on a sample per sample basis (no block processing). In order to minimize the execution time for every sample, implementations may avoid the use of branches and mathematical functions such as square root, sine, cosine, and divide. Implementations may also avoid the use of lookup tables and look-ahead delays, which increase memory requirements and increase execution time. Thus aspects of the invention may be implemented with very simple computer programs and very basic digital signal processors. Particularly in view of their simplicity, aspects of the present invention may also be implemented using analog circuitry.

35 [0062] A number of embodiments of the invention have been described. Nevertheless, it will be understood that various modifications may be made without departing from the scope of the invention.

Claims

40 1. A method for processing audio signals, comprising
 deriving n audio output signals from m audio input signals, where m and n are positive whole integers and the n
 45 audio output signals are derived using an adaptive matrix or matrixing process responsive to n time-varying control signals, which matrix or matrixing process produces n audio signals in response to m audio signals (4),
 deriving said n control signals from said m audio input signals (1-3), said deriving using

50 a passive matrix or matrixing process that produces pairs of signals in response to the m audio input signals,
 a first pair of signals representing signal strength in opposing directions along a first directional axis and a
 second pair of signals representing signal strength in opposing directions along a second directional axis (1)

55 a first processor or process that produces a plurality of directional dominance signals in response to said
 pairs of input signals, at least one directional dominance signal relating to a first directional axis and at least
 one other directional dominance signal relating to a second directional axis (2), and
 a second processor or process that produces said control signals in response to said directional dominance
 signals (3),

characterized in that:

the first processor or process that produces a plurality of directional dominance signals uses linear amplitude domain subtractors or subtraction processes that obtain a positive or negative difference between the magnitudes of each pair of signals, a clipper or clipping process that limits each of the differences substantially at a positive clipping level and a negative clipping level, an amplifier or amplification process that either a) amplifies each of said differences such that the clipper or clipping process limits each of the amplified differences or b) amplifies each of said limited differences, and a smoother or smoothing process that time-averages each of the amplified and limited or limited and amplified differences (2).

2. A method according to claim 1 wherein for uncorrelated audio input signals each directional dominance signal approximates a directional dominance signal based on a ratio of signal pairs and for correlated audio input signals the directional dominance signal tends toward the negative or positive clipping level.
3. A method according to claim 2 wherein a difference above the positive clipping level indicates a positive dominance along a directional axis, a difference below the negative clipping level indicates a negative dominance along a directional axis, and a difference between the positive and negative clipping levels indicates non-dominance along a directional axis.
4. A method according to claim 3 wherein the processor or process that produces a plurality of directional dominance signals modifies the amplified and limited or limited and amplified difference signal differently when there is non-dominance along a directional axis than when there is positive or negative dominance.
5. A method according to any one of claims 1 or 2 wherein the processor or process that produces a plurality of directional dominance signals also restricts either the positive or negative magnitude of the output of a clipper or clipping process prior to a smoother or smoothing process.
6. A method according to claim 5 wherein the processor or process that produces a plurality of directional dominance signals restricts the positive magnitude of the output of at least one of the clippers or clipping processes prior to the smoother or smoothing process.
7. A method according to claim 6 wherein the first directional axis is a front/back axis and the processor or process that produces a plurality of directional dominance signals restricts the positive magnitude of the output of the clipper or clipping process that processes a front/back axis directional dominance signal.
8. A method according to any one of claims 1-7 wherein said second processor or process that produces said control signals in response to said plurality of directional dominance signals applies at least one panning function to each of said plurality of directional dominance signals.
9. A method according to claim 8 wherein one or more of the panning functions implement a trigonometric transfer function.
10. A method according to claim 8 wherein one or more of the panning functions implement a logarithmic transfer function.
11. A method according to claim 8 wherein one or more of the panning functions implement a linear transfer function.
12. A method according to claim 8 wherein one or more of the panning functions implement a mathematically simplified approximation of a trigonometric transfer function.
13. A method according to any one of claims 8-12 wherein the control signals are derived from the application of a panning function to a directional dominance signal, and/or the product of the application of a panning function to one directional dominance signal and the application of a panning function to another directional dominance signal, wherein each panning function may be different from ones or all of the other panning functions.
14. A method according to any one of claims 8-12 wherein the panning functions are panning functions that are not inherent in the n input audio signals.

15. A method according to claim 14 wherein one of the directional axes is a left/right axis and the panning functions are panning functions that do not include a left/right panning component.
16. A method according to any one of claims 8-15 wherein at least some of said n time-varying scale factor signals are derived from the application of a single panning function to a directional dominance signal and others of said n time-varying scale factor signals are derived from the products of the application of a single panning function to a directional dominance signal and the application of another single panning function to another directional dominance signal.
17. A method according to claim 16 wherein the directional axis of one of the directional dominance signals is a left/right axis and the directional axis of another of the directional dominance signals is a front/back axis, wherein at least some of said n time-varying scale factor signals are derived from the application of a single panning function to the front/back directional dominance signal and at least some of said n time-varying scale factor signals are derived from the products of the application of a single panning function to the left/right directional dominance signal and the application of another single panning function to the front/back directional dominance signal.
18. A method according to any one of claims 1-17 further comprising deriving p audio signals from said n audio output signals, wherein p is two and said p audio signals are derived from said n audio signals using a virtualizer or virtualization process such that, when the p audio signals are applied to a pair of transducers, a listener suitably located with respect to the transducers perceives the n audio signals as coming from locations that may be different from the location of the transducers.
19. A method according to 18 wherein the virtualizer or virtualization process includes the application of one or more head-related-transfer-functions to ones of said n audio output signals.
20. A method according to claim 18 or claim 19 wherein the pair of transducers is a pair of headphones.
21. A method according to claim 18 or claim 19 wherein the pair of transducers is a pair of loudspeakers.
22. A method according to any of claims 1 through 21 wherein m is 2 and n is 4 or 5.
23. Apparatus comprising means adapted to perform the steps of the method of any one of claims 1 through 21.
24. A computer program, stored on a computer-readable medium for causing a computer to perform the steps of the method of any one of claims 1 through 21.

Patentansprüche

1. Verfahren zur Verarbeitung von Audiosignalen, das aufweist
Ableiten von n Audioausgangssignalen von m Audioeingangssignalen, wobei m und n positive Ganzzahlen sind und die n Audioausgangssignale abgeleitet werden unter Verwendung einer adaptiven Matrix oder eines Matrixing-Prozesses in Reaktion auf n zeitvariiierende Steuerungssignale, wobei die Matrix oder der Matrixing-Prozess n Audiosignale in Reaktion auf m Audiosignale erzeugt (4),
Ableiten der n Steuerungssignale von den m Audioeingangssignalen (1-3), wobei das Ableiten verwendet eine passive Matrix oder einen Matrixing-Prozess, der Paare von Signalen in Reaktion auf die m Audioeingangssignale erzeugt, wobei ein erstes Paar von Signalen eine Signalstärke in entgegengesetzten Richtungen entlang einer ersten Richtungsachse repräsentiert und ein zweites Paar von Signalen eine Signalstärke in entgegengesetzten Richtungen entlang einer zweiten Richtungsachse repräsentiert (1)
einen ersten Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt in Reaktion auf die Paare von Eingangssignalen, wobei zumindest ein Richtungsdominanzsignal eine erste Richtungsachse betrifft und zumindest ein anderes Richtungsdominanzsignal eine zweite Richtungsachse betrifft (2), und
einen zweiten Prozessor oder Prozess, der die Steuerungssignale in Reaktion auf die Richtungsdominanzsignale erzeugt (3),
dadurch gekennzeichnet, dass:

der erste Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt, lineare Amplitude-Domäne-Subtrahierer oder Subtraktions-Prozesse verwendet, die eine positive oder negative Differenz zwischen den Größen jedes Paares von Signalen erlangen, einen Clipper oder Clipping-Prozess, der jede der

Differenzen im Wesentlichen an einem positiven Clipping-Pegel und einem negativen Clipping-Pegel begrenzt, einen Verstärker oder Verstärkungs-Prozess, der entweder a) jede der Differenzen derart verstärkt, dass der Clipper oder Clipping-Prozess jede der verstärkten Differenzen begrenzt, oder b) jede der begrenzten Differenzen verstärkt, und einen Glätter oder Glättungs-Prozess, der jede der verstärkten und begrenzten oder begrenzten und verstärkten Differenzen zeitlich mittelt (2).

2. Verfahren gemäß Anspruch 1, wobei für nicht-korrelierte Audioeingangssignale jedes Richtungsdominanzsignal ein Richtungsdominanzsignal basierend auf einem Verhältnis von Signalkpaaren approximiert und für korrelierte Audioeingangssignale das Richtungsdominanzsignal zu dem negativen oder positiven Clipping-Pegel tendiert.
3. Verfahren gemäß Anspruch 2, wobei eine Differenz über dem positiven Clipping-Pegel eine positive Dominanz entlang einer Richtungsachse anzeigt, eine Differenz unter dem negativen Clipping-Pegel eine negative Dominanz entlang einer Richtungsachse anzeigt und eine Differenz zwischen den positiven und negativen Clipping-Pegeln eine Nicht-Dominanz entlang einer Richtungsachse anzeigt.
4. Verfahren gemäß Anspruch 3, wobei der Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt, das verstärkte und begrenzte oder begrenzte und verstärkte Differenzsignal anders modifiziert, wenn es eine Nicht-Dominanz entlang einer Richtungsachse gibt, als wenn es eine positive oder negative Dominanz gibt.
5. Verfahren gemäß einem der Ansprüche 1 oder 2, wobei der Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt, auch entweder die positive oder negative Größe der Ausgabe eines Clippers oder Clipping-Prozesses vor einem Glätter oder Glättungs-Prozess beschränkt.
6. Verfahren gemäß Anspruch 5, wobei der Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt, die positive Größe der Ausgabe von zumindest einem der Clipper oder Clipping-Prozesse vor einem Glätter oder Glättungs-Prozess beschränkt.
7. Verfahren gemäß Anspruch 6, wobei die erste Richtungsachse eine vorne/hinten-Achse ist und der Prozessor oder Prozess, der eine Vielzahl von Richtungsdominanzsignalen erzeugt, die positive Größe der Ausgabe des Clippers oder Clipping-Prozesses, der ein vorne/hinten-Achse-Richtungsdominanzsignal verarbeitet, beschränkt.
8. Verfahren gemäß einem der Ansprüche 1 - 7, wobei der zweite Prozessor oder Prozess, der die Steuerungssignale in Reaktion auf die Vielzahl von Richtungsdominanzsignalen erzeugt, zumindest eine Panning-Funktion auf jedes der Vielzahl von Richtungsdominanzsignalen anwendet,
9. Verfahren gemäß Anspruch 8, wobei eine oder mehrere der Panning-Funktionen eine trigonometrische Übertragungsfunktion implementiert/implementieren.
10. Verfahren gemäß Anspruch 8, wobei eine oder mehrere der Panning-Funktionen eine logarithmische Übertragungsfunktion implementiert/implementieren.
11. Verfahren gemäß Anspruch 8, wobei eine oder mehrere der Panning-Funktionen eine lineare Übertragungsfunktion implementiert/implementieren.
12. Verfahren gemäß Anspruch 8, wobei eine oder mehrere der Panning-Funktionen eine mathematisch vereinfachte Approximation einer trigonometrischen Übertragungsfunktion implementiert/implementieren.
13. Verfahren gemäß einem der Ansprüche 8-12, wobei die Steuerungssignale abgeleitet sind von der Anwendung einer Panning-Funktion auf ein Richtungsdominanzsignal, und/oder dem Produkt der Anwendung einer Panning-Funktion auf ein Richtungsdominanzsignal und der Anwendung einer Panning-Funktion auf ein anderes Richtungsdominanzsignal, wobei jede Panning-Funktion von einer oder allen der anderen Panning-Funktionen verschieden sein kann.
14. Verfahren gemäß einem der Ansprüche 8-12, wobei die Panning-Funktionen Panning-Funktionen sind, die in den n Eingangsaudiosignalen nicht inhärent sind.
15. Verfahren gemäß Anspruch 14, wobei eine der Richtungsachsen eine links/rechts-Achse ist und die Panning-Funktionen Panning-Funktionen sind, die keine links/rechts-Panning-Komponente umfassen.

16. Verfahren gemäß einem der Ansprüche 8-15, wobei zumindest einige der n zeitvariierenden Skalierungsfaktor-Signale abgeleitet werden aus der Anwendung einer einzelnen Panning-Funktion auf ein Richtungsdominanzsignal und andere der n zeitvariierenden Skalierungsfaktor-Signale abgeleitet werden aus den Produkten der Anwendung einer einzelnen Panning-Funktion auf ein Richtungsdominanzsignal und der Anwendung einer anderen einzelnen Panning-Funktion auf ein anderes Richtungsdominanzsignal.
17. Verfahren gemäß Anspruch 16, wobei die Richtungsachse von einem der Richtungsdominanzsignale eine links/rechts-Achse ist und die Richtungsachse eines anderen der Richtungsdominanzsignale eine vorne/hinten-Achse ist, wobei zumindest einige der n zeitvariierenden Skalierungsfaktor-Signale abgeleitet werden von der Anwendung einer einzelnen Panning-Funktion auf das vorne/hinten-Richtungsdominanzsignal und zumindest einige der n zeitvariierenden Skalierungsfaktor-Signale abgeleitet werden von den Produkten der Anwendung einer einzelnen Panning-Funktion auf das links/rechts-Richtungsdominanzsignal und der Anwendung einer anderen einzelnen Panning-Funktion auf das vorne/hinten-Richtungsdominanzsignal.
18. Verfahren gemäß einem der Ansprüche 1-17, das weiter aufweist ein Ableiten von p Audiosignalen von den n Audioausgangssignalen, wobei p zwei ist und die p Audiosignale von den n Audioausgangssignalen abgeleitet werden unter Verwendung eines Virtualizers oder eines Virtualisierungs-Prozesses derart, dass, wenn die p Audiosignale auf ein Paar von Transducern angewendet werden, ein Zuhörer, der in Bezug auf die Transducer geeignet positioniert ist, die n Audiosignale als von Positionen kommend wahrnimmt, die von den Positionen der Transducer abweichen können.
19. Verfahren gemäß Anspruch 18, wobei der Virtualizer oder Virtualisierungs-Prozess die Anwendung von einer oder mehreren Kopf-bezogenen-Übertragungsfunktionen auf einzelne der n Audioausgangssignale umfasst.
20. Verfahren gemäß Anspruch 18 oder Anspruch 19, wobei das Paar von Transducern ein Paar von Kopfhörern ist.
21. Verfahren gemäß Anspruch 18 oder Anspruch 19, wobei das Paar von Transducern ein Paar von Lautsprechern ist.
22. Verfahren gemäß einem der Ansprüche 1 bis 21, wobei m 2 ist und n 4 oder 5 ist.
23. Vorrichtung, die Mittel aufweist, die ausgebildet sind zum Durchführen der Schritte des Verfahrens gemäß einem der Ansprüche 1 bis 21.
24. Computerprogramm, das auf einem computerlesbaren Medium gespeichert ist, um einen Computer zu veranlassen, die Schritte des Verfahrens gemäß einem der Ansprüche 1 bis 21 durchzuführen.

Revendications

1. Procédé permettant de traiter des signaux audio, comprenant :

la dérivation de n signaux de sortie audio à partir de m signaux d'entrée audio, où m et n sont des nombres entiers positifs et les n signaux de sortie audio sont dérivés en utilisant une matrice adaptative ou un processus de matriçage adaptatif en réponse à des signaux de commande présentant des variations temporelles, laquelle matrice ou lequel processus de matriçage produit n signaux audio en réponse à m signaux audio (4), la dérivation desdits n signaux de commande à partir desdits m signaux d'entrée audio (1 - 3), ladite dérivation utilisant

une matrice passive ou un processus de matriçage passif qui produit des paires de signaux en réponse aux m signaux d'entrée audio une première paire de signaux représentant une force de signal dans des directions opposées le long d'un premier axe directionnel et une seconde paire de signaux représentant la force de signal dans des directions opposées le long d'un second axe directionnel (1),

un premier processeur ou processus qui produit une pluralité de signaux à dominance directionnelle en réponse auxdites paires de signaux d'entrée, au moins un signal à dominance directionnelle lié à un premier axe directionnel et au moins un autre signal à dominance directionnelle lié à un second axe directionnel (2), et

un second processeur ou processus qui produit lesdits signaux de commande en réponse auxdits

signaux à dominance directionnelle (3),

caractérisé en ce que :

- 5 le premier processeur ou processus qui produit une pluralité de signaux à dominance directionnelle utilise des soustracteurs ou des processus de soustraction de domaine d'amplitude linéaire qui obtiennent une différence positive ou négative entre les amplitudes de chaque paire de signaux, un écrêteur ou un processus d'écrtage qui limite chacune des différences sensiblement à un niveau d'écrtage positif et un niveau d'écrtage négatif, un amplificateur ou un processus d'amplification qui soit a) amplifie chacune desdites différences de telle sorte
10 que l'écrtageur ou le processus d'écrtage limite chacune des différences amplifiées soit b) amplifie chacune des différences limitées, et un lisseur ou un processus de lissage qui fait la moyenne temporelle de chacune desdites différences amplifiées et limitées ou limitées et amplifiées (2).
- 15 2. Procédé selon la revendication 1, dans lequel pour des signaux d'entrée audio non corrélés, chaque signal à dominance directionnelle approche un signal à dominance directionnelle fondé sur un ratio de paires de signaux et pour des signaux d'entrée audio corrélés, le signal à dominance directionnelle tend vers le niveau d'écrtage négatif ou positif.
- 20 3. Procédé selon la revendication 2, dans lequel une différence supérieure au niveau d'écrtage positif indique une dominance positive le long d'un axe directionnel, une différence inférieure au niveau d'écrtage négatif indique une dominance négative le long d'un axe directionnel, et une différence entre les niveaux d'écrtage positif et négatif indique une non dominance le long d'un axe directionnel.
- 25 4. Procédé selon la revendication 3, dans lequel le processeur ou le processus qui produit une pluralité de signaux à dominance directionnelle modifie le signal différentiel amplifié et limité ou limité et amplifié d'une manière différente, lorsqu'il existe une non dominance le long d'un axe directionnel, de lorsqu'il existe une dominance positive ou négative.
- 30 5. Procédé selon l'une quelconque des revendications 1 ou 2, dans lequel le processeur ou le processus qui produit une pluralité de signaux à dominance directionnelle restreint également l'amplitude soit positive soit négative de la sortie d'un écrêteur ou du processus d'écrtage avant le lisseur ou le processus de lissage,
- 35 6. Procédé selon la revendication 5, dans lequel le processeur ou processus qui produit une pluralité de signaux de dominance directionnelle restreint l'amplitude positive de la sortie d'au moins l'un des écrtageurs ou processus d'écrtage avant le lisseur ou le processus de lissage.
- 40 7. Procédé selon la revendication 6, dans lequel le premier axe directionnel est un axe avant/arrière et le processeur ou le processus qui produit une pluralité de signaux de dominance directionnelle restreint l'amplitude positive de la sortie de l'écrtageur ou du processus d'écrtage qui traite un signal à dominance directionnelle d'axe avant/arrière.
- 45 8. Procédé selon l'une quelconque des revendications 1 à 7, dans lequel ledit second processeur ou processus qui produit lesdits signaux de commande en réponse à ladite pluralité de signaux à dominance directionnelle applique au moins une fonction de panoramique à chacun de ladite pluralité des signaux à dominance directionnelle.
- 50 9. Procédé selon la revendication 8, dans lequel une ou plusieurs des fonctions de panoramique met/mettent en oeuvre une fonction de transfert trigonométrique.
10. Procédé selon la revendication 8, dans lequel une ou plusieurs des fonctions de panoramique met/mettent en oeuvre une fonction de transfert logarithmique.
- 55 11. Procédé selon la revendication 8, dans lequel une ou plusieurs des fonctions de panoramique met/mettent en oeuvre une fonction de transfert linéaire.
12. Procédé selon la revendication 8, dans lequel une ou plusieurs des fonctions de panoramique met/mettent en oeuvre une approximation simplifiée mathématiquement d'une fonction de transfert trigonométrique.
13. Procédé selon l'une quelconque des revendications 8 à 12, dans lequel les signaux de commande sont dérivés de l'application d'une fonction de panoramique à un signal à dominance directionnelle, et/ou le produit de l'application

d'une fonction de panoramique à un signal à dominance directionnelle et l'application d'une fonction de panoramique à un autre signal à dominance directionnelle, dans lequel chaque fonction de panoramique peut être différente de certaines ou de toutes les autres fonctions de panoramique.

- 5 14. Procédé selon l'une quelconque des revendications 8 à 12, dans lequel les fonctions de panoramique sont des fonctions de panoramique qui ne sont pas inhérentes aux n signaux audio d'entrée.
15. Procédé selon la revendication 14, dans lequel l'un des axes directionnels est un axe gauche/droite et les fonctions de panoramique sont des fonctions de panoramique qui n'incluent pas un composant de panoramique gauche/droite.
- 10 16. Procédé selon l'une quelconque des revendications 8 à 15, dans lequel au moins certains desdits n signaux de facteur d'échelle présentant des variations temporelles sont dérivés de l'application d'une fonction de panoramique unique à un signal à dominance directionnelle et d'autres desdits n signaux de facteur d'échelle présentant des variations temporelles sont dérivés des produits de l'application d'une fonction de panoramique unique à un signal à dominance directionnelle et de l'application d'une autre fonction de panoramique unique à un autre signal à dominance directionnelle.
- 15 17. Procédé selon la revendication 16, dans lequel l'axe directionnel de l'un des signaux à dominance directionnelle est un axe gauche/droite et l'axe directionnel d'un autre des signaux à dominance directionnelle est un axe avant/arrière, dans lequel au moins certains desdits n signaux de facteur d'échelle présentant des variations temporelles sont dérivés de l'application d'une fonction de panoramique unique au signal à dominance directionnelle avant/arrière et au moins certains desdits n signaux de facteur d'échelle présentant des variations temporelles sont dérivés des produits de l'application d'une fonction de panoramique unique à un signal à dominance directionnelle gauche/droite et de l'application d'une autre fonction de panoramique unique au signal à dominance directionnelle avant/arrière.
- 20 18. Procédé selon l'une quelconque des revendications 1 à 17, comprenant en outre la dérivation de p signaux audio à partir desdits n signaux de sortie audio, dans lequel p est deux et lesdits p signaux audio sont dérivés desdits n signaux audio en utilisant un virtualiseur ou un processus de virtualisation qui, lorsque les p signaux audio sont appliqués à une paire de transducteurs, un auditeur placé de manière appropriée par rapport aux transducteurs, perçoit les n signaux audio comme venant d'emplacements qui peuvent être différents de l'emplacement des transducteurs.
- 25 19. Procédé selon la revendication 18, dans lequel le virtualiseur ou le processus de virtualisation inclut l'application d'une ou de plusieurs fonctions de transfert liées à la tête à certains desdits n signaux de sortie audio.
- 30 20. Procédé selon la revendication 18 ou la revendication 19, dans lequel la paire de transducteurs est une paire d'écouteurs.
- 35 21. Procédé selon la revendication 18 ou la revendication 19, dans lequel la paire de transducteurs et une paire de haut-parleurs.
22. Procédé selon l'une quelconque des revendications 1 à 21, dans lequel m est 2 et n est 4 ou 5.
- 40 23. Appareil comprenant un moyen conçu pour exécuter les étapes du procédé selon l'une quelconque des revendications 1 à 21.
- 45 24. Programme informatique, mémorisé sur un support lisible par ordinateur, permettant de faire exécuter les étapes du procédé selon l'une quelconque des revendications 1 à 21 par un ordinateur.
- 50
- 55

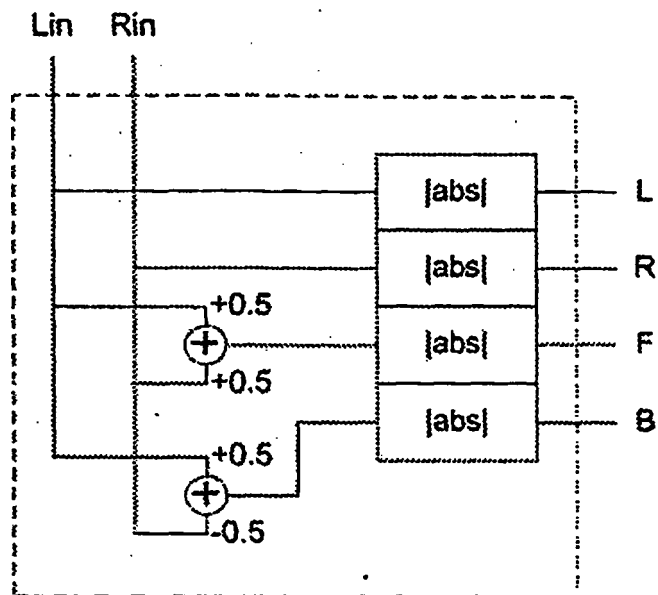


FIG. 1

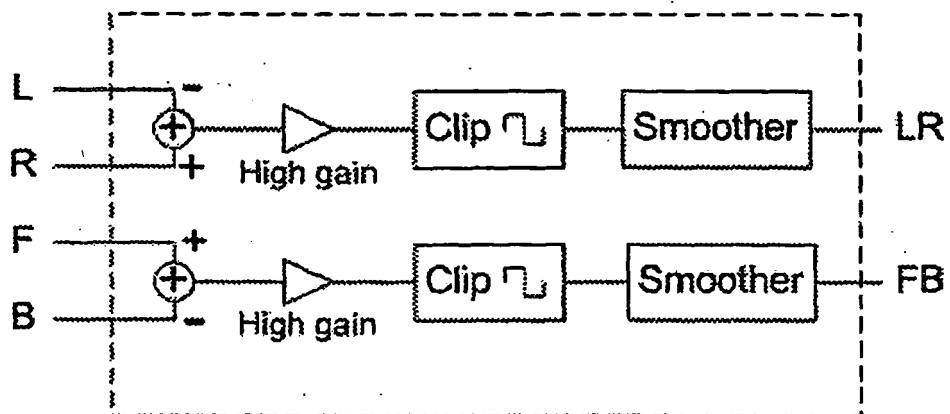


FIG. 2

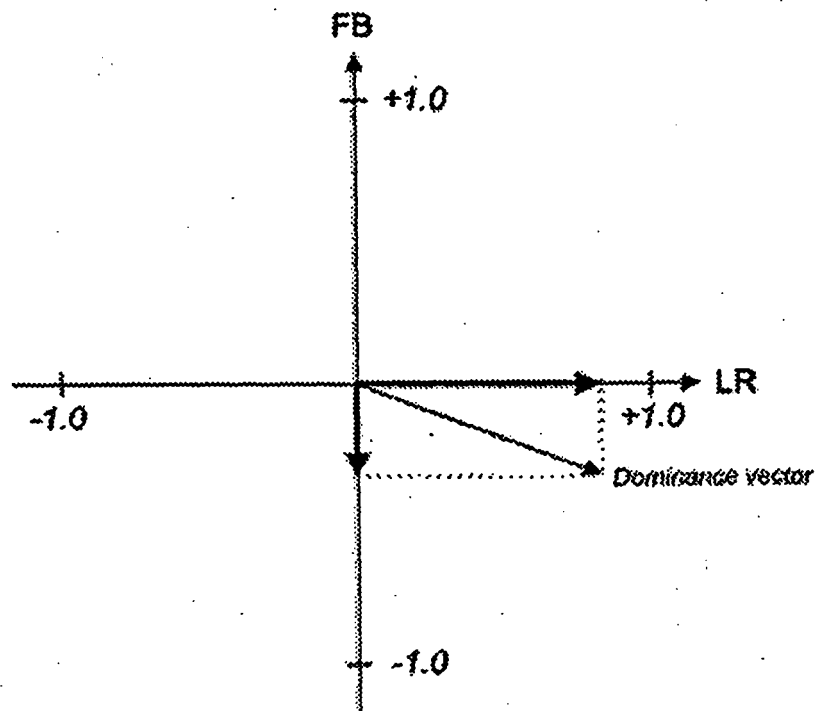


FIG. 3

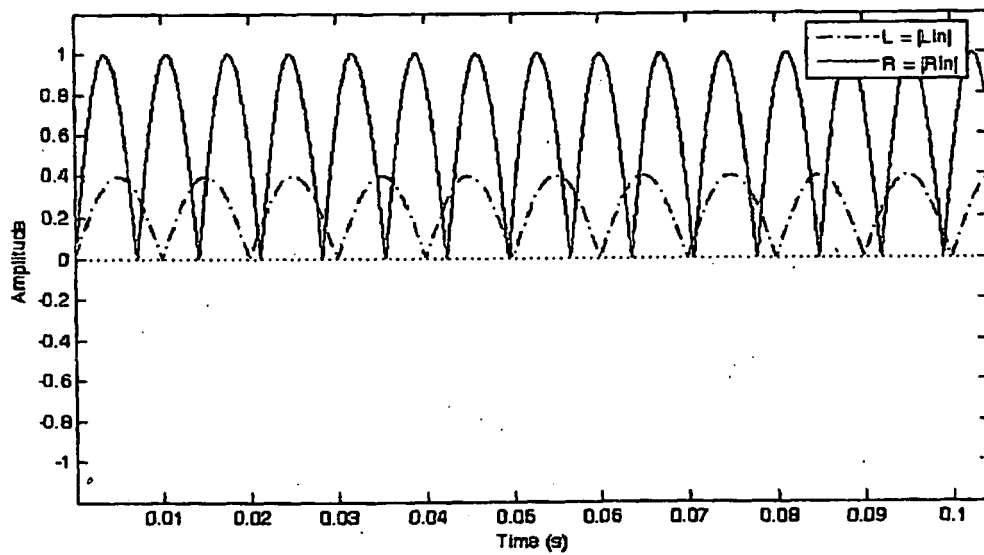


FIG. 4

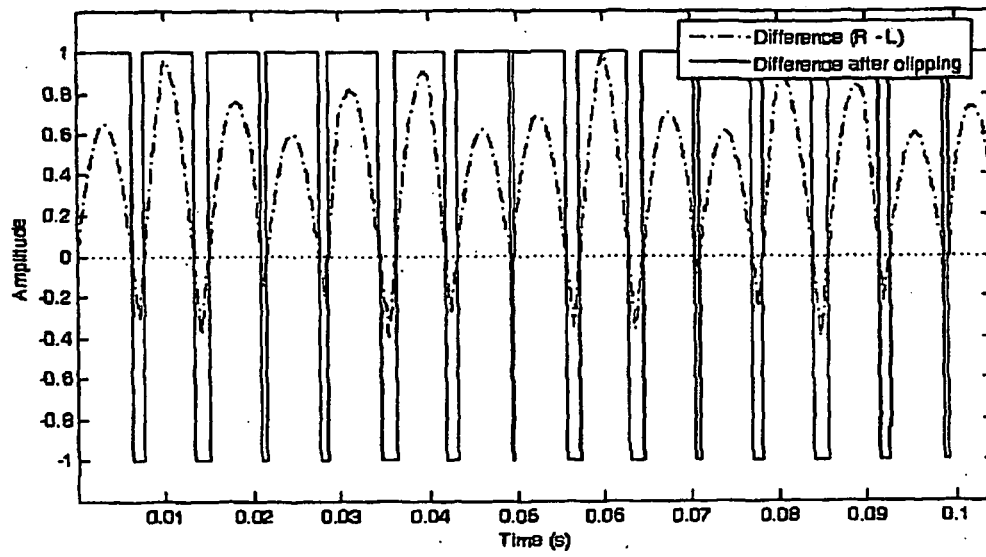


FIG. 5

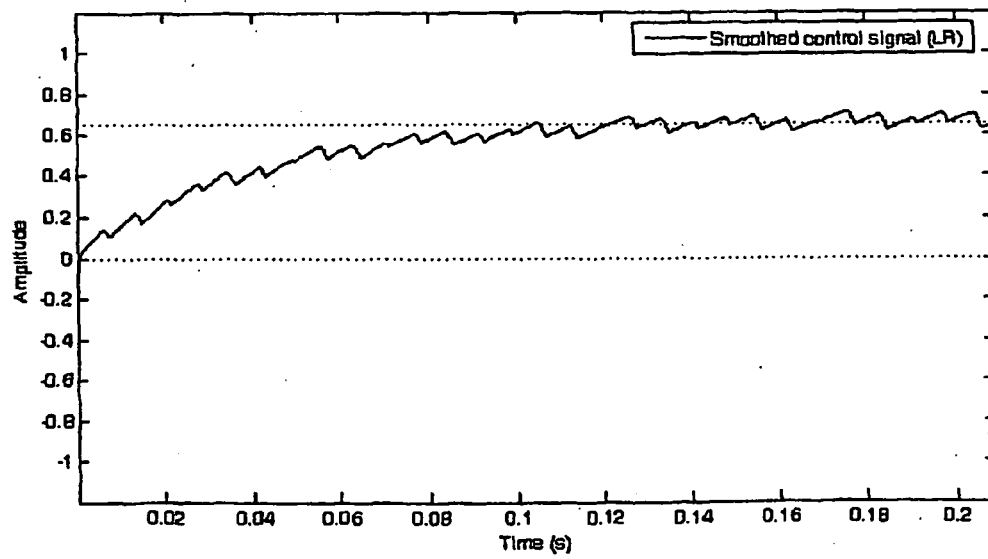


FIG. 6

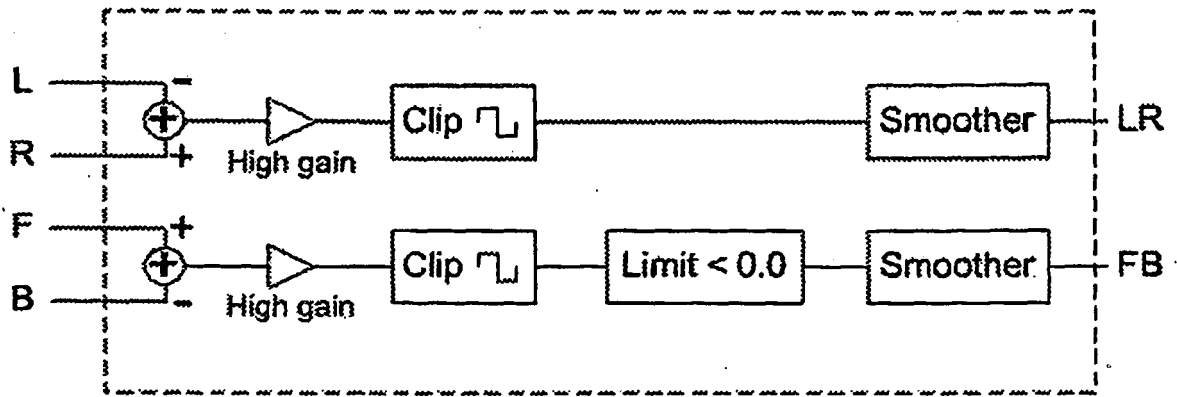


FIG. 7

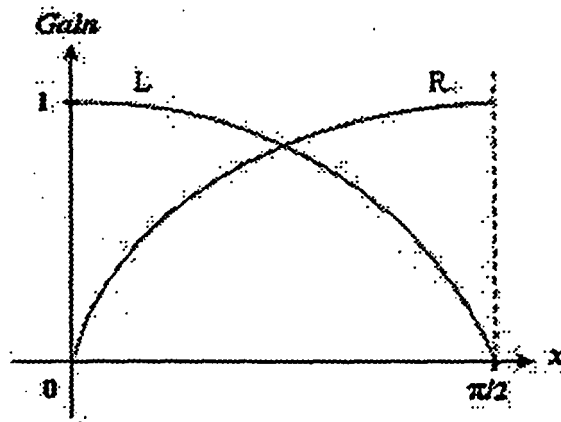
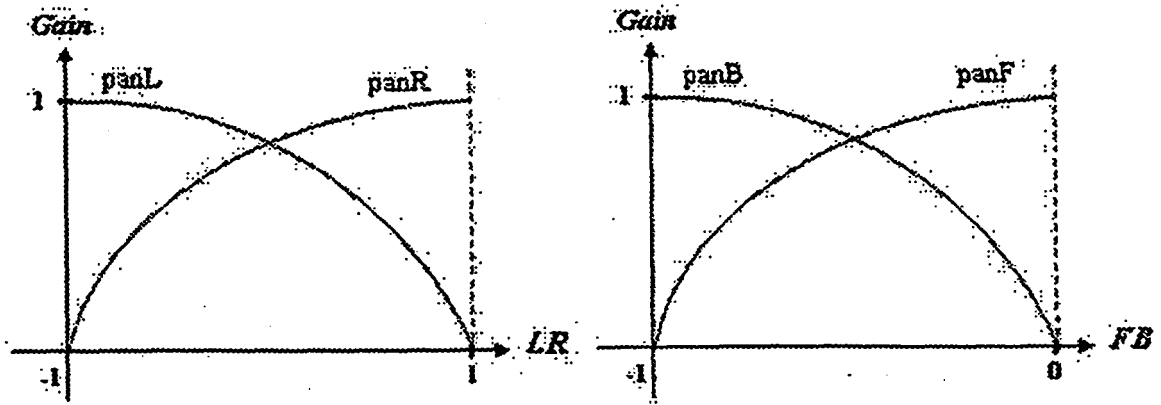


FIG. 8



FIGS. 9a and 9b

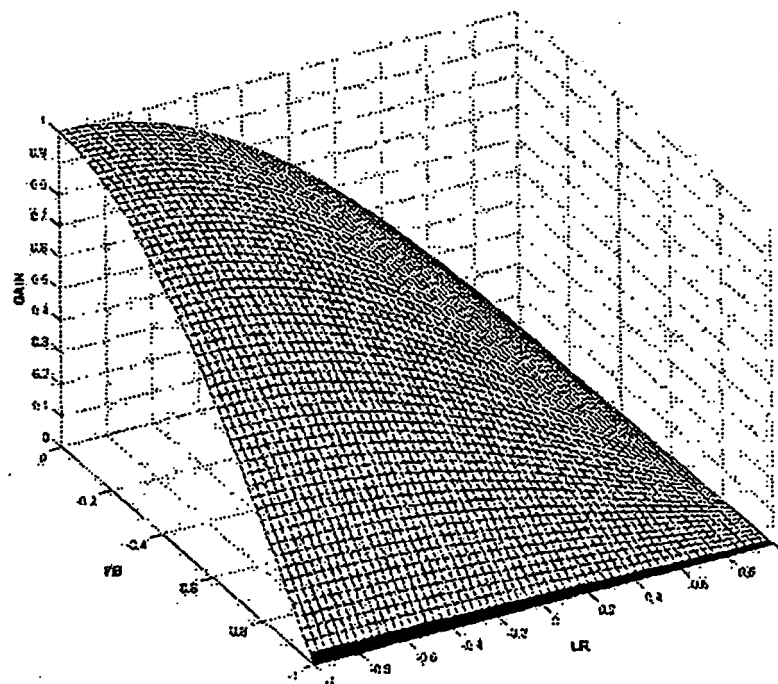


FIG. 10

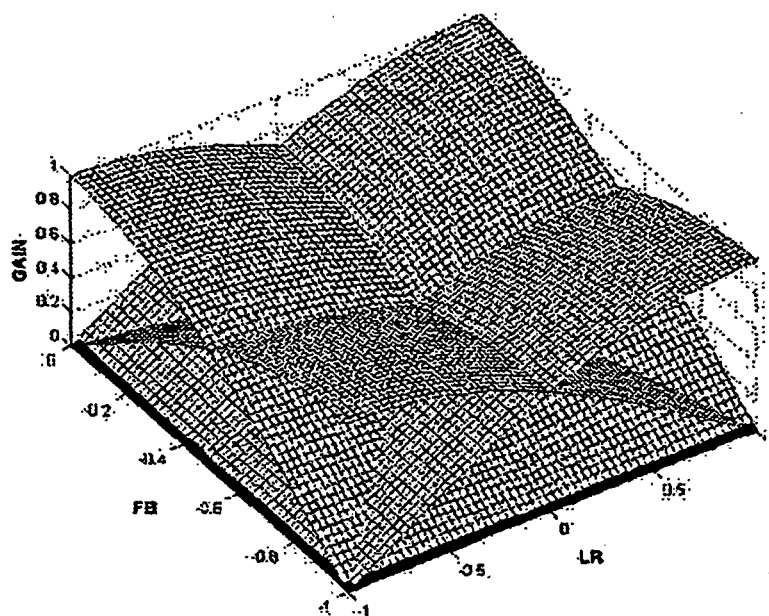


FIG. 11

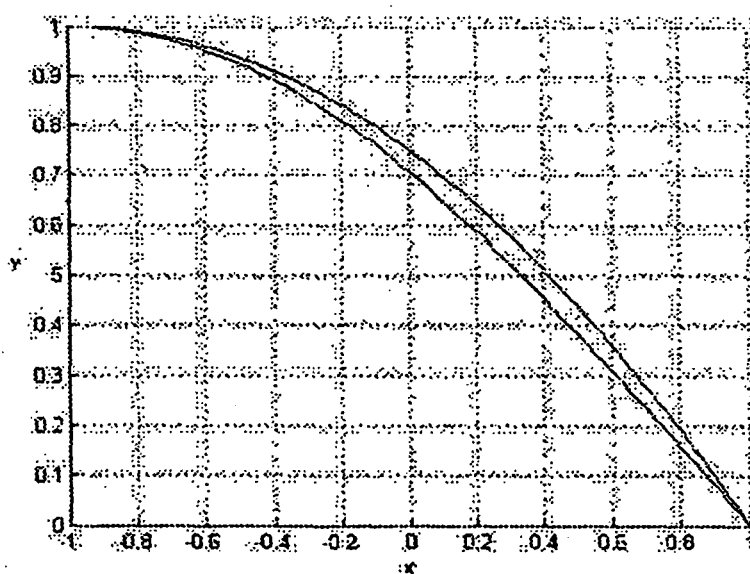


FIG. 12

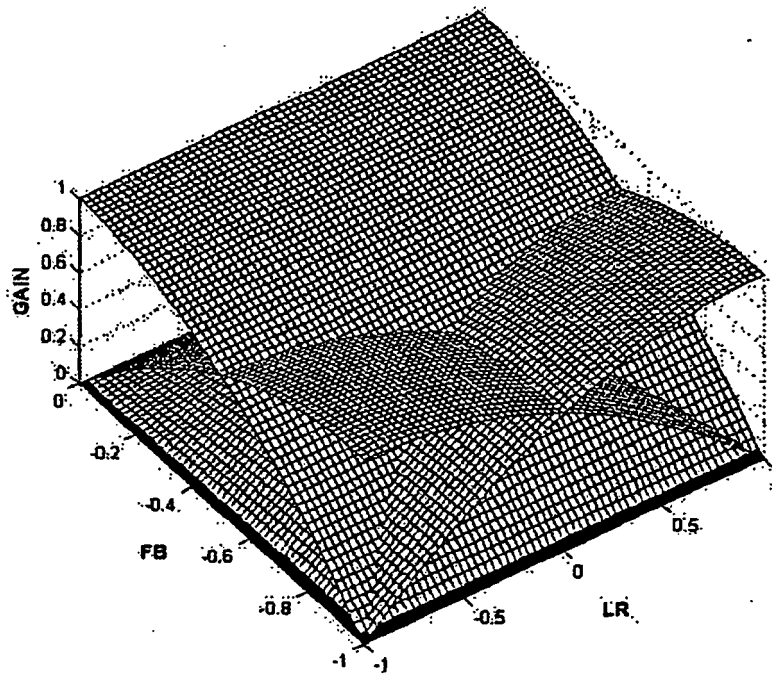


FIG. 13

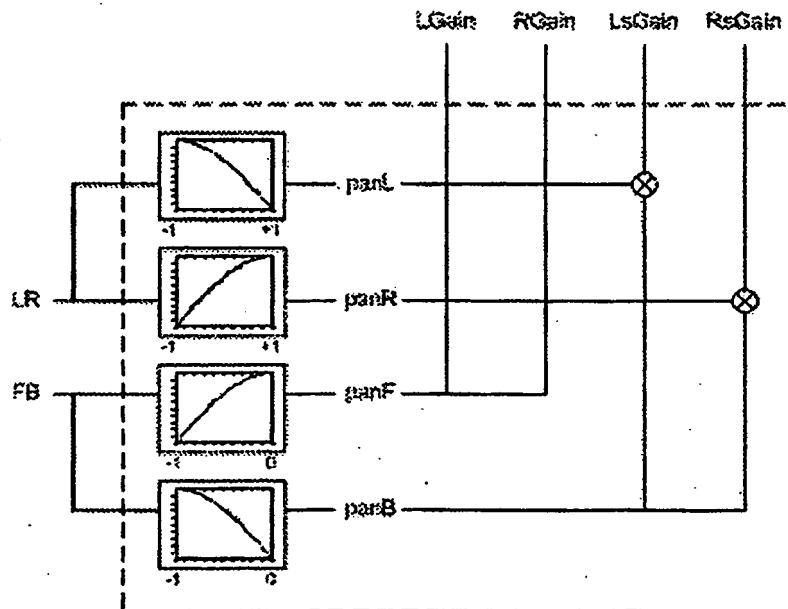


FIG. 14

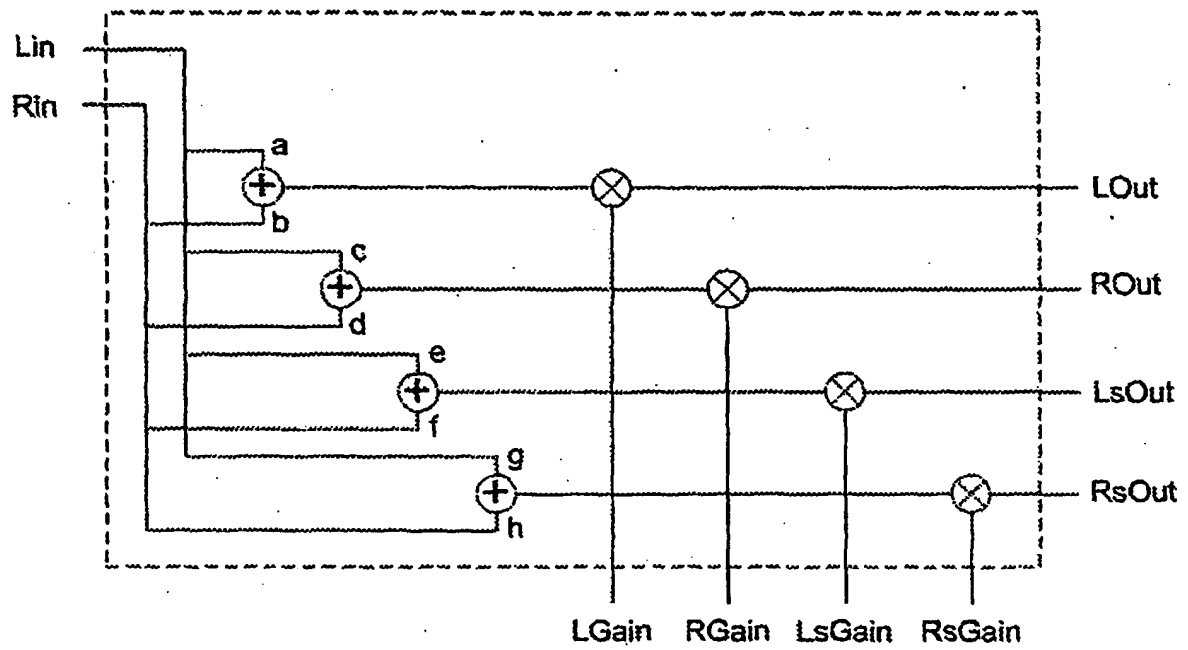


FIG. 15

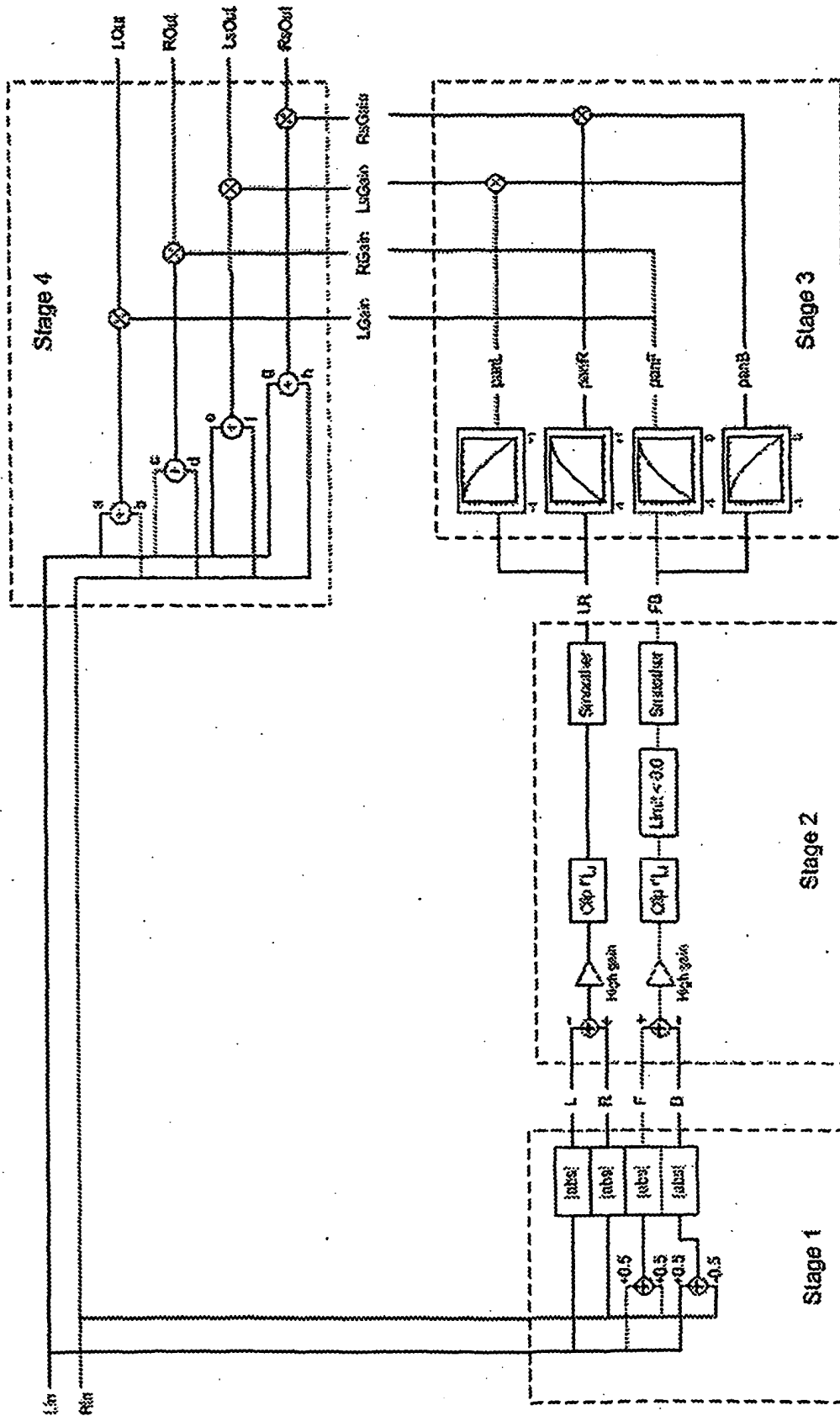


FIG. 16

REFERENCES CITED IN THE DESCRIPTION

This list of references cited by the applicant is for the reader's convenience only. It does not form part of the European patent document. Even though great care has been taken in compiling the references, errors or omissions cannot be excluded and the EPO disclaims all liability in this regard.

Patent documents cited in the description

- US 6370256 B [0003]
- US 6574649 B [0003]
- US 6741706 B [0003]
- WO 9914983 A [0003]
- US 6449368 B [0003]
- WO 2003053099 A [0003]
- US 6920223 B [0006]
- US 6970567 B [0006] [0017]
- WO 2002019768 A [0006]
- US 4799260 A [0006]
- US 4941177 A [0006]
- US 5046098 A [0006]