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(54) **5-2-5 MATRIX DECODER SYSTEM**

5-2-5 MATRIX DEKODIERUNGSSYSTEM

SYSTEME DE DECODAGE A MATRICE 5-2-5

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(73) Proprietor: **HARMAN INTERNATIONAL INDUSTRIES, INCORPORATED**
Northridge, CA 91329 (US)

(72) Inventor: **GRIESINGER, David**
Cambridge, MA 02140 (US)

(74) Representative: **Grünecker, Kinkeldey, Stockmair & Schwanhäusser**
Leopoldstrasse 4
80802 München (DE)

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Description

Field of the Invention

[0001] This invention relates to sound reproduction systems involving the decoding of a stereophonic pair of input audio signals into a multiplicity of output signals for reproduction after suitable amplification through a like plurality of loudspeakers arranged to surround a listener.

Background of the Invention

[0002] The present invention concerns an improved set of design criteria and their solution to create a decoding matrix having optimum psychoacoustic performance in reproducing encoded multichannel material as well as standard two channel material, that includes maintaining high separation between left and right components of stereo signals under all conditions, even when there is a net forward or rearward bias to the input signals, or when there is a strong sound component in a particular direction, while maintaining high separation between the various outputs for signals with a defined direction, and while maintaining non-directionally encoded components at a constant acoustic level regardless of the direction of directionally encoded components of the input audio signals, including frequency dependent circuitry that improves the balance between front and rear signals, provides smooth sound motion around a seven channel version of the system, and makes the sound of a five channel version closer to that of a seven channel version.

[0003] The present invention is part of a continuing effort to refine means for separating two channels into the multichannel signals from which they were derived. One of the goals of this decode process is to recreate the original signals as perceptually identical to the originals as possible. Another important goal of the decoder is to extract five or more separate channels from a two channel source that was not encoded from a five channel original. The resulting five channel presentation must be at least as musically tasteful and enjoyable as the original two channel presentation.

[0004] The present invention is concerned with improvements to the derivation of suitable variable matrix coefficients. To assist understanding of these improvements, this disclosure makes reference to Griesinger's U.S. Patent No. 4,862,502 (1989) which will be referred to as the '89 patent: U.S. Patent No. 5,136,650 (1992), which will be referred to as the '92 patent; the July 1996 Griesinger U.S. Patent Application No. 08/684,948 referenced as the July '96 application; and the November 1996 Griesinger U.S. Patent Application No. 08/742,460, referred to as the November '96 application. Commercial versions of the decoder based upon this last application will be referred to as Version 1.11 (or V1.11). Some further improvements were disclosed in the Provisional Patent Application 60/058,169 filed September 1997, which will be referred to as Version 2.01 (or V2.01). These versions V1.11 and V2.01, and the present invention, will be referred to collectively as "Logic 7" decoders.

[0005] Additional technical references cited are: [1] "Multichannel Matrix Surround Decoders for Two-Eared Listeners," David Griesinger. AES preprint # 4402, October, 1996, and [2] "Progress in 5-2-5 Matrix Systems," David Griesinger, AES preprint # 4625, September, 1997.

Summary of the Invention

[0006] The means employed to realize the two goals of recreating the original signals encoded from five to two channels and of creating a perceptually satisfactory reproduction of two channel material in a five channel format have evolved as we continue to better understand the physics and the psychoacoustics involved. The previous patents and patent applications referred to above presented a design philosophy that produced useful decoder devices.

[0007] The present invention is concerned with the realization of an active matrix having certain properties that maximize its psychoacoustic performance. In another aspect it discloses frequency dependent modification of certain of the outputs from the active matrix. The present invention is defined in claim 1.

[0008] The invention is in part an active matrix decoder having matrix elements that vary depending on the directional component of the incoming signals. The matrix elements vary in such a way as to reduce the loudness of directionally encoded signals in outputs that are not involved in the intended direction, while enhancing the loudness of these signals in directions that are involved in reproducing the intended direction, while preserving at all times the left/right separation of other signals that may be present at the same time at the inputs. Moreover, matrix elements in accordance with the invention restore the left/right separation of decorrelated two channel material that has been directionally encoded by increasing or decreasing the blend between the two inputs - for example with a stereo width control. In addition matrix elements in accordance with the invention are designed to preserve as much as possible the energy balance between various components of the input signal, such that the balance between vocals and accompaniment is preserved in the decoder outputs. As a consequence matrix elements in accordance with the invention preserve both the loudness of non directionally encoded elements of the input sound and the left/right separation of these elements.

[0009] Additionally, decoders in accordance with preferred embodiments of the invention include frequency dependent circuits that improve the compatibility of the decoder outputs when standard two channel material is played, that convert the surround outputs from two for a five channel decoder to four for a seven channel decoder, and that modify the spectrum of the rear channels in a five channel decoder so that the sound direction appears to be more like the sound direction from a seven channel decoder.

Brief Description of the Drawing

[0010] The novel features believed characteristic of the present invention are set forth in the appended claims. The invention itself, as well as other features and advantages thereof, will best be understood by reference to the following detailed description of an illustrative embodiment when read in conjunction with the accompanying drawing figures, wherein:

FIG. 1 is a block schematic of a direction detection section and a two to five channel matrix section of a decoder according to the present invention, but not including aspects further shown in FIGs. 2 and 3;
FIG. 2 is a block schematic of a five-channel frequency- dependent active signal processor circuit which may be connected between the outputs of the matrix section of FIG. I and the decoder outputs;
FIG. 3 is a block schematic of a five-to-seven channel frequency-dependent active signal processor which may alternatively be connected between the outputs of the matrix section of FIG. I and the decoder outputs;
FIG. 4 is a block schematic of an active five-channel to two-channel encoder according to an example which is useful for understanding the invention;
FIG. 5 is a three-dimensional graphical representation of the prior art Left Front Left (LFL) matrix element from the ’89 patent and Dolby Pro-Logic, scaled so the maximum value is one;
FIG. 6 is a three-dimensional graphical representation of the prior art Left Front Right (LFR) matrix element from the ’89 patent and Dolby Pro-Logic, scaled by.71 so the minimum value is -0.5 and the maximum value is + 0.5;
FIG. 7 is a three-dimensional graphical representation of the square root of the sum of the squares of the prior art LFL and-LFR from the ’89 patent, scaled so the maximum value is one;
FIG. 8 is a three -dimensional graphical representation of the square root of the sum of the LFL and LFR matrix elements from application No. 08/742,460, scaled so the max value is 1;
FIG. 9 is a three-dimensional graphical representation of the Left Front Left (LFL) matrix element in V1.11;
FIG. 10 is a three-dimensional graphical representation of the partially completed Left Front Left matrix element in the present invention:
FIG. 11 is a graph showing the behavior of LFL and LFR in the present invention along the rear boundary between left and full rear;
FIG. 12 is a three-dimensional graphical representation of the fully completed Left Front Left (LFL) matrix element in the present invention as viewed from the left rear;
FIG. 13 is a three-dimensional graphical representation of the fully completed Left Front Right (LFR) matrix element in the present invention:
FIG. 14 is a three-dimensional graphical representation of the root mean squared sum of LFL and LFR according to the present invention:
FIG. 15 is a three-dimensional graphical representation of the square root of the sum of the squares of LFL and LFR according to the present invention, including the correction to the rear level, viewed from the left rear;
FIG. 16 is a graph showing with the solid curve the center matrix elements that should be used in a prior art Dolby Pro-Logic decoder as a function of CS in dB, and with the dotted curve the actual value of the center matrix elements in the Dolby Pro-Logic decoder;
FIG. 17 is a graph showing with the solid curve the ideal value of the center matrix elements, and with the dotted curve the actual values of the center matrix elements in prior art Dolby Pro-Logic;
FIG. 18 is a three-dimensional graphical representation of the square root of the sum of the squares of LRL and LRR, using the prior art elements of V1.11;
FIG. 19 is a graphical representation of the numerical solution for *GS(lr)* and GR(lr) for constant power level along the *cs*=0 axis, and zero output along the boundary between left and center;
FIG. 20 is a three-dimensional graphical representation of the square root of the sum of the squares of LRL and LRR using the values for GR and GS according to the present invention;
FIG. 21 is a three-dimensional graphical representation of the prior art Center Left (CL) matrix element of the ’89 patent four channel decoder (and the Dolby Pro-Logic decoder) which can also represent the Center Right (CR) matrix element with left and right interchanged:
FIG. 22 is a three-dimensional graphical representation of the Center Left (CL) matrix element in Logic 7 V1.11 decoders:
FIG. 23 is a graph showing with the solid curve the center Output channel attenuation needed for the new LFL and

LFR, and with the dotted curve the center attenuation for a standard prior art Dolby Pro-Logic decoder;

FIG. 24 is a graph showing with the solid curve the ideal center attenuation for the "film" strategy, according to the present invention, with the dashed curve a value that works significantly better, and with the dotted curve the center attenuation for the standard Dolby decoder for comparison;

FIG. 25 shows the center attenuation used in the "music" strategy in the present invention;

FIG. 26 is a graph showing with the solid curve the value of GF needed for constant energy ratios with the "music" center attenuation GC of the present invention, with the dashed curve the previous LFR element sin(cs)*corr1, and with the dotted curve the value of sin(*cs*);

FIG. 27 is a three-dimensional graphical representation of the Left Front Right (LFR) matrix element in the new invention, including the correction for center level along the *lr*= 0 axis;

FIG. 28 is a three-dimensional graphical representation of the Center Left (CL) matrix element with the new center boost function; and

FIG. 29 is a graph which plots the output level from the left front output (dotted curve) and the center output (solid curve) as a strong signal pans from center to left.

Detailed Description of the Preferred Embodiments

[0011] The design presented here preserves much of the design philosophy of the previous decoders. but the actual design has changed in many ways. It is not possible within the bounds of a document of readable length to completely describe the evolution of this design. To keep the document coherent we will present here the most important elements of the design philosophy, show the mathematical solutions to the problems presented, and make claims on the solutions that are original with this filing. It may be useful to consult our previous filings on this subject, but it should not be essential.

[0012] Experience with the decoder as described in the July ’96 and November ’96 applications and the September ’97 provisional patent application has led to additional improvements that have not yet been disclosed. This application will present the most essential features of the improved decoder of the present invention, and will make claims on the novel features that have been added since U. S. Patent Application No. 08/742,460.

1. General description of the decoder

[0013] The decoder in this application will be described as consisting of two separate parts. The first part is a matrix that splits the two input channels into five output channels, which are usually identified as center, left front, right front, left rear and right rear. The second part consists of a series of delays and filters that modify the spectrum and the levels of the two rear outputs. One of the functions of the second part is to derive an additional pair of outputs, left side and right side, when a seven channel version of the decoder is desired. In application No. 08/742,460 the second part was not explicit - the two additional channels were derived from an additional pair of matrix elements in the original matrix.

[0014] In mathematical equations describing the decoder we will use the standard typographical conventions for most variables, simple variables being shown in italics, while vector quantities are in bold lower case type and matrixes are represented by bold upper case type. Matrix elements that are coefficients from a named output channel resulting from a named input channel will be shown in normal upper case type. Some simple variables such as *lr* and cs are described by two-letter names which do not represent products of two separate simple variables. Other variables *l/r* and *c/s* do in some sense represent the values of left-right and center-surround ratios, but in terms of control signal voltages derived from these ratios. These conventions have also been used in the previous U. S. Patents and Patent Applications cited in this document. Program segments in the Matlab language will also be distinguished by the use of a different type face and point size and by indenting these lines. Equations will be numbered, to distinguish them from Matlab assignment statements, and to provide a reference for specific features described herein.

[0015] FIG. 1, which is identical to FIG. 4 in U. S. Patent Application No. 08/742,460, shows a block diagram of the first part of the decoder, a two channel to five channel matrix **90.** The left half of FIG. 1 partitioned by a vertical dashed line shows a means for deriving the two steering voltages *l/r* and *c/s.* These voltages represent the degree to which the input signals have an inherent or encoded directional component in the left/right or front/back directions respectively. This part of the FIG. will not be explicitly discussed in this application as it has been fully described in the above patent application.

[0016] In FIG. 1 the directional detection means of decoder **90** comprising elements **92** through **138** -is followed by a 5 x 2 matrix to the right of the vertical dashed line. The elements of this matrix **140** through **158** determine the amount of each input channel that is linearly combined with the other input channel to form each output channel. These matrix elements are assumed to be real. (The case of complex matrix elements was described in U. S. Patent Application No. 08/742,460 and will not be discussed here.) The matrix elements are functions of the two steering voltages *l/r* and *c/s.* U. S. PatentApplication No. 08/742,460 presented mathematical formulae for these functions. Part of the novelty in this application lies in improvements to these formulae. We present these formulae graphically and give an explanation for why they take the shape they do.

2. A brief description of the steering voltages

[0017] As shown in FIG. 1, the steering voltages *c/s* and *l/r* are derived from the logarithm of the ratio of the left input amplitude at terminal 92 to the right input amplitude at terminal 94, and the logarithm of the ratio of the sum amplitude to the difference amplitude. In describing the matrix elements it is convenient to express *l/r* and c/s as angles that vary from +45 degrees to --45 degrees. In the decoders of V1.11 and V2.01 these voltages have the units of decibels. We can convert the steering parameters to angles, where

$$lr = 90 - \arctan\left(10 \wedge ((l/r)/20)\right) \quad \text{...(1a)}$$

$$cs = 90 - \arctan\left(10 \wedge ((c/s)/20)\right) \quad \text{...(1b)}$$

[0018] The angles *lr* and cs determine the degree to which the input signals have a directional component. For example, when the inputs to the decoder are decorrelated, both *lr* and cs are zero. For a signal that comes from the center only, *lr* is zero, and cs has the value 45 degrees. For a signal that comes from the rear, *lr* is zero, and cs is -45 degrees. Similarly, a signal that comes from the left has an *lr* value of 45 degrees and a cs value of zero, and a signal from the right has an *lr* value of -45 degrees, and a cs value of zero. We will assume in our design that the encoder that creates the encoded signal has the properties that when we encode a left rear signal *lr* = 22.5 degrees, and cs = 22.5 degrees. Similarly, a signal applied to the right rear input to the encoder produces values of *lr*=22.5 degrees, and *cs*=22.5 degrees.

[0019] From the definitions of *l/r* and *c/s* and the derivation of *lr* and *cs,* it can be seen that the stun of the absolute value of *lr* and cs cannot be greater than 45 degrees. The allowed values of *lr* and cs form a surface bounded by the locus of abs(*lr*)-abs(cs)=45 degrees. Any input signal that produces values of *lr* and cs that lie along the boundary of this surface is fully localized - that is it consists of a single sound that has been encoded to come from a particular direction.

[0020] In this application we will make extensive use of graphs of the matrix elements as functions over this two dimensional surface. In general the derivation of the matrix elements will be different in the four quadrants of this surface. In other words the matrix elements are differently described depending on whether the steering is forward or rearward, and whether it is to the left or the right. Considerable work is devoted to insuring that the surface is continuous across the boundaries between quadrants. The occasional lack of such continuity is one of the problems with the decoder of V1.11 that this application will address.

3. Frequency dependent elements

[0021] The matrix elements shown in FIG. 1 are real and thus frequency independent. All signals in the inputs will be directed to the outputs depending on the derived angles *lr* and cs. (In the current art low frequencies and very high frequencies are attenuated in the derivation of *lr* and cs from the input signals by filters not shown in FIG. 1. However, the matrix itself is broadband.)

[0022] We have found in practice that there are several advantages to applying frequency dependent circuits to the signals after the matrix. One of these frequency dependent circuits - the phase shift network **170** at the right side output **180** in FIG. 1 - has been described in U. S. Patent Application No. 08/742,460 and not be further discussed here.

[0023] FIG. 2 shows a five channel version of the additional frequency dependent circuits. These circuits do not have fixed parameters. The frequency and level behavior is dependent on the steering values *lr* and cs. The circuits accomplish several purposes. First, in both a five channel and a seven channel decoder, the additional elements allow the apparent loudness of the rear channels to be adjusted when the steering is neutral *(lr* and cs 0) or toward the front (*cs* > 0). In U. S. Patent Application No. 08/742,460 this attenuation was performed as part of the matrix itself. and was frequency independent. We have found with theoretical studies and listening tests that it is highly desirable for the low frequencies to be reproduced from the sides of the listener. Thus in the decoder presented here only the high frequencies are attenuated by variable low pass filters **182, 184, 188,** and **190.**

[0024] This is accomplished by attenuating the frequencies above 500Hz in the rear channels by means of elements **188, 190,** and above 4kHz by means of elements **182, 184,** when the steering is nearly always neutral or forward, using a background control signal 186 to be defined later in this application. Occasional presence of sounds that are steered rearwards reduces the attenuation, a feature that automatically distinguishes surround encoded material from ordinary two channel material.

[0025] Further elements **192, 194,** in the five channel version modify the spectrum of the sound when the steering is toward the rear (*cs* < 0) using the *c/s* signal 196 such that the loudspeakers seem to be located behind the listener, even if their actual position is to the side. The modified left surround and right surround signals appear at terminals **198** and **200,** respectively. Additional details of this circuit will be given in a later section of this disclosure.

[0026] FIG. 3 shows the seven channel version of the frequency dependent elements. As before the first set of filters **182, 184, 188, 190,** attenuate the upper frequencies of the side and rear outputs when the steering is neutral or forward, again controlled by the background control signal **186.** This attenuation also results in a more forward sound image, and can be adjusted to the listener's taste. As the steering represented by the c/s signal **196** moves to the rear, additional circuits **202, 204, 206, 208,** act to differentiate the side outputs from the rear outputs. As steering moves rearward the attenuation mentioned above in the side speakers is first removed by elements **204** and **206** to produce a side oriented sound. As steering moves further to the rear the attenuation of elements **204** and **206** is then reinstated and increased. The result is that the sound moves smoothly from the front loudspeakers to the side loudspeaker(s). and then to the rear loudspeakers, which have a delay of about 10 ms, produced by the delay elements **202,** and **208.** Since the low frequencies are not affected by these circuits, the low frequency loudness in the side speakers (which is responsible for the perception of spaciousness) is not affected by the motion of the sound. Again, a later section of this application will provide additional details of the circuitry in FIG. 3.

4. General description of the encoder

[0027] FIG. 4 shows a block diagram of an encoder according to an example, which is useful for understanding the invention. The encoder is designed to automatically mix five input channels into two output channels. The architecture is quite different from the encoder described in U. S. Patent Application No. 08/742,460. The object of the new design is to preserve the musical balances of the five channel original, while providing phase/amplitude cues that allow the original five channels to be extracted by the decoder. The previous encoder had similar goals, but there have been improvements in the methods used to achieve these goals. The preservation of musical balances is highly important in the encoder. One of the primary purposes of the encoder is to automatically create a two channel mix of a five channel recording, that will play in an ordinary two channel system with the same artistic quality as the five channel original. The new encoder design includes active elements to ensure that musical balance is preserved.

[0028] Unlike the encoder of the November '97 application the new design allows input signals to be panned between any of the five inputs of the encoder. For example, a sound may be panned from the left front input to the right rear input. When the resulting two channel signal is decoded by the decoder described in this application the result will be quite close to the original sound. Decoding through an earlier surround decoder will also be similar to the original.

[0029] A detailed description of the encoder will follow in a later section.

5. Design goals for the decoder active matrix elements

[0030] The most basic goals of the current invention are identical to the goals of our previous decoders, particularly the one described in U. S. Patent Application No. 08/742,460 - "The invention is a surround sound decoder having variable matrix values so constructed as to reduce directionally encoded audio components in outputs which are not directly involved in reproducing them in the intended direction; enhance directionally encoded audio components in the outputs which are directly involved in reproducing them in the intended direction so as to maintain constant total power for such signals; while preserving high separation between the left and right channel components of non-directional signals regardless of the steering signals; and maintaining the loudness defined as the total audio power level of non-directional signals effectively constant whether or not directionally encoded signals are present and regardless of their intended direction if present."

[0031] Most of these goals are ostensibly shared by all matrix decoders. The novelty in this application lies in part with knowing how to implement the above rule more accurately, and in part with knowing when not to apply the above rule. However much of the methodology of U. S. Patent Application No. 08/742,460 is retained. One of the most important of the previous goals is the explicit maintenance of high separation between the left and right channels of the decoder under all conditions. All previous four channel decoders are unable to maintain separation in the rear, because they provide only a single rear channel. Five channel decoders from other manufacturers compromise separation in many ways. The decoder described in this application meets this goal in a similar way as that of V1.11- but it meets additional goals as well.

[0032] U. S. Patent Application No. 08/742,460 also describes many smaller improvements to the design, such as circuits to improve the accuracy of the steering signals, and a variable phase shift network to switch the phase of one of the rear channels during strong rear steering. These features of the decoder V1.11 are retained in the new design, but will not be covered in this document.

[0033] In FIG. 4 the front input signals L, C and R are applied to input terminals **50, 52** and **54** respectively. L and R go directly to the adders **278** and **282** respectively, while the C signal is first attenuated by a factor *fcn* in attenuator **372** before being applied to inputs of both adders **2 78** and **282.** The low frequency effects signal LFE is passed through a gain of 2.0 in element **374** and then applied to both adders **278** and **282.**

[0034] The surround input signals LS and RS are applied through two input terminals **62** and **64** respectively each to two separate paths: for the LS signal, the path through attenuator **378** has gain *fs(l,ls)* and the RS signal passes through a corresponding attenuator **380** with gain *fs(r,rs).* The outputs of these are passed into cross-coupling elements **384** and **386** having a gain factor of -*crx*, where *crx* is nominally 0.383. The cross-coupled signals from these elements are fed to summers **392** and **394** which also receive the attenuated LS and RS signals from 0.91 attenuators **388** and **392.** The outputs of summers **392, 394,** are applied to inputs of the adders **278, 282.** This positions the elements respectively at 45 degrees left and right of center rear in the decoded space.

[0035] The other signal branch passes the LS and RS signals through attenuators **376** with *gain fc(l,ls)* and **382** with gain *fc*(*r,rs*) respectively, and then through a similar arrangement of cross-coupling elements **396, 398, 402, 404, 406,** and **408,** the summers **406** and **408** having outputs representing left rear and right rear inputs at 45 degrees left and right of center rear, as before. However, these signals now each pass through phase shifter elements **234** and **246** respectively, while the left and right signals from adders **278** and **282** pass through phase shifter elements **286 and 288** respectively. Each of these phase shifter elements is an all-pass filter, the phase response being $\varphi(f)$ for elements **286** and **288,** and $\varphi(f)$-90° for elements **234** and **246.** Calculation of the component values required in these filters is well known in the art, and will not be discussed further here. The result is that the outputs of summers **406** and **408** are caused to lag those of adders **278** and **282** by 90 degrees at all frequencies after passage through the all-pass filter networks as shown in FIG. 4. The outputs of all-pass networks 234 and 286 are now combined in summer **276** to produce the A (or left) output signal at terminal **44,** while the outputs of filters **246** and **288** are combined by summer **280** to produce the B (or right) output signal at terminal **46.**

[0036] The gain functions *fs* and *fc* are designed to allow strong surround signals to be presented in phase with the other sounds while weak surround signals pass through the 90 degree phase-shifted path to retain constant power for decorrelated "music" signals. The value of *crx* can also change, and varies the angle from which the surround signals are heard.

<u>6. Design improvements since the application No. 08/742,460</u>

[0037] One of the most noticeable improvements in the present invention relative to that of U. S. Patent Application No. 08/742,460 is the change in the center matrix elements and left and right front matrix elements when a signal is steered in the center direction. We learned that there were two problems with the center channel as previously encoded and decoded. The most obvious problem is that in a five channel matrix system the use of a center channel is inherently in conflict with the goal of maintaining as much left/right separation as possible. If the matrix is to produce a sensible output from conventional two channel stereo material, when the two input channels have no left/right component the center channel must be driven with the sum of the left and right input channels. Thus both the left decoder Input and the right decoder input will be reproduced by the center speaker, and sounds that were originally only in the left (or right) channel will also be reproduced from the center. The result must be that the apparent positions of these sounds will be drawn to the middle of the room. The degree to which this occurs depends on the loudness of the center channel.

[0038] U. S. Patents Nos. 4,862,502 and 5,136,650 used matrix elements that had a minimum value of 3dB compared to the left and right channels. When the inputs to the decoder were decorrelated the loudness of the center channel was equal to the loudness of the left and right channels. As steering moved forward the center matrix elements increased another 3dB. The effect of this high loudness is to strongly reduce the width of the front image. Instruments that should have sounded at the left and the right of the sound image are always drawn toward the center of the sound image.

[0039] U. S. Patent Application No. 08/742,460 used center matrix elements that had a minimum value 4.5dB less than the earlier values. This minimum value was chosen on the basis of listening tests. This attenuation causes a pleasing spread to the front image when the input material is uncorrelated - as with orchestral music. The front image is not seriously narrowed. In U. S Patent Application No. 08/742,460 as the steering moved forward these matrix elements increased, ultimately reaching the values used in the Dolby matrix.

[0040] Experience with the V1.11 decoder showed that although the reduction in center channel loudness solved the spatial problem, the power balance in the input signals was not preserved through the matrix. Mathematical analysis revealed that not only was V1.11 in error, but the Dolby, decoder and our previous decoders were also in error. Paradoxically, although the center channel was too strong from the standpoint of reproducing the width of the front image, it was too weak to preserve power balance. The problem was particularly severe for the decoder of Mandel - the standard Dolby decoder. In the standard Dolby decoder the rear channels are stronger than in our decoder of patent No. 4,862,502. As a result the center channel must be stronger to preserve power balance. The lack of power balance 'in the center channel has been a continual problem for the Dolby decoder. Dolby recommends that the sound mix engineer should always listen to the balance through the matrix, so the lack of power balance in the matrix can be compensated during the mixing process. Unfortunately modern films are mixed for five-channel release, and automatic encoding to two channels can lead to problems with the dialog level.

[0041] Additional analysis and listening tests showed that films and music require different solutions to the balance problem. For films we found that it was most useful to preserve the left and right front matrix elements from U. S. Patent Application No. 08/742,460. These elements eliminate the center channel information from the left and right front channels as much as possible. This minimizes dialog leakage into the front left and right channels. In the new "film" design the power balance is corrected by changing the center matrix elements so the center channel loudness increases more rapidly than the standard decoder as the steering moves forward (as *cs* becomes greater than zero.) In practice it is not necessary that the final value of the center matrix elements be higher than those in the standard decoder, as this condition is reached when only the center channel is active. It is only necessary that the center level be stronger than the standard decoder when there are approximately equal levels in the center channel and the left and right channels.

[0042] With the "film" strategy the center channel loudness is increased to preserve the power balance in the input signals, while minimizing the center channel component in all the other outputs. This strategy seems to be ideal for films, where the major use of the center channel is for dialog, and dialog from positions other than the center is not expected. The major disadvantage of this strategy is that anytime there is significant center steering - such as occurs in many types of popular music - the front image is narrowed. However the advantages for film - minimum dialog leakage into the front channels and excellent power balance - outweigh this disadvantage.

[0043] For music we adopt another strategy. In this case we allow the center channel loudness to increase at the same rate as in U. S. Patent Application No. 08/742,460, up to a middle value of the steering, where cs $\geq$ 22.5 degrees. To restore the musical balance, we alter the left and right front matrix elements such that the center component of the input signals is not entirely removed. The amount of the center channel component in the left and right front is adjusted so that the sound power from all the outputs of the decoder matches the sound power in the input signals - without excessive loudness in the center.

[0044] With this strategy all three front speakers reproduce center channel information present in the original encoded material. The most useful version of this strategy limits the steering action at the point that the center component of the input is 6dB stronger in the center output than in either of the two other front outputs. This is done by simply limiting the positive value of cs.

[0045] This new strategy - allowing the center channel component to come from all three front speakers, and limiting the steering action when the center is 6dB louder than the front left and right, turns out to be excellent for all types of music. Both encoded five-channel mixes and ordinary two-channel mixes decode with a stable center, and adequate separation between the center channel and the left and right. Note that unlike previous decoders the separation between center and left and right is deliberately not complete. A signal intended to come from the left is eliminated from the center channel, but not the other way around. For music the high lateral separation and stable front image that this strategy offers outweighs this lack of complete separation. Listening tests with this setting on films reveal that although there is some dialog coming from the left and right front speakers the stability of the resulting sound image is quite good. The result is pleasant and not distracting. For this listener hearing a film with the decoder set for music does not detract from the artistic quality of the film. Listening to a music recording, with the decoder set for film is more problematic.

[0046] Possibly the next most obvious of the improvements in this application is the increase in separation between the front channels and the rear channels when a signal is steered to the left front or the left rear directions. The decoder of V1.11 used the matrix elements of U. S. Patent No. 4,862,502 for the front channels under these conditions. These matrix elements do not fully eliminate a rear steered signal unless it was steered to the full rear position - half way between left rear and right rear. When steering was to left rear or right rear (not full rear) the left or right front output had an output 9dB less than the corresponding rear output. In the present invention the front matrix elements are modified to eliminate sound from the front when steering is anywhere between left rear and right rear.

7. Improvements to the rear matrix elements

[0047] The improvements to the rear matrix elements are not immediately obvious to a typical listener. These improvements correct various errors in the continuity of the matrix elements across the boundaries between quadrants. They also improve the power balance between steered signals and unsteered signals under various conditions. The mathematical description of the matrix elements that will be given later includes these improvements.

8. Detailed description of the active matrix elements

The Matlab Language

[0048] The math used to describe the matrix elements is not based on continuous functions of the variables cs and *lr*. In general there are conditionals, absolute values, and other non-linear modifications to the formulae. For this reason we will describe the matrix elements using a programming language. The Matlab language provides a simple method of checking the formulation graphically. Matlab is very similar to Fortran or C. The major difference is that variables in Matlab can be vectors - that is each variable can represent an array of numbers in sequence. For example we can define the variable x in the following way:

```
x = 1:10;
```

[0049] This specification in Matlab creates a string of ten numbers with the values of one to ten. The variable x includes all ten values. It is described a vector, a 1 by 10 matrix. An individual number within each vector can be accessed or manipulated. For example, the expression

```
x(4) = 4;
```

will set the fourth member of the vector x to the value 4. Variables can also represent a two dimensional matrix. Individual elements in the matrix can be assigned in a similar way:

```
X(2,3) =10;
```

will assign the value 10 to the second row and third column of the matrix X.

[0050] The detailed description of the matrix elements that follows is nearly identical to the description published in reference [2]. The text has been somewhat improved. The major differences are:

1. Reference [2] includes the "tv matrix" feature. This feature reduces the level of the rear outputs when the steering is forward or neutral. In this application this function is achieved through the frequency dependent circuits that follow the matrix. Therefore we have left out the "tv matrix" correction.
2. The section on the center matrix elements has been modified to include references to the "film" strategy, the "music" strategy, and a strategy that limits the action of the "music" setting. Reference [21 described only the "music" setting, without the limit.

<u>9. Matrix decoders in equations and graphics</u>

[0051] In reference [1] we presented the design of a matrix decoder that can be described by the elements of a *n* x 2 matrix, where *n* is the number of output channels. Each output can be seen as a linear combination of the two inputs, where the coefficient of the linear combination are given by the elements in the matrix. In this paper the elements are identified by a simple combination of letters. Reference [1] described a five-channel and a seven-channel decoder. The conversion from five channels to seven channels is now done in the frequency dependent part of the decoder, so here we will describe a five-channel decoder only.

[0052] It is obvious from symmetry that we need to describe the behavior of only six elements - the center elements, the two left front elements, and the two left rear elements. The right elements can found from the left by simply switching the identity of left and right. These elements are:

CL: The matrix element for the Left input channel to the Center output
CR: The matrix element for the Right input channel to the Center output
LFL: The Left input channel to the Left Front output
LFR: The Right input channel to the Left Front output
LRL: The Left input channel to the Left Rear output
LRR: The Right input channel to the Left Rear output

[0053] These elements axe not constant. Their value varies as a two dimensional function of the apparent direction of the input sounds. Most phase/amplitude decoders determine the apparent direction of the input by comparing the ratio of the amplitudes of the input signals. For example, the degree of steering in the right/left direction is determined from the ratio of the left input channel amplitude to the right input channel amplitude. In a similar way, the degree of steering in the front/back direction is determined from the ratio of the amplitudes of the sum and the difference of the input channels. We will not discuss the method for determining these steering directions here, although Logic 7 decoders differ from standard decoders significantly in how this is done. We assume that the steering directions have been determined. Here we will represent these directions as angles - one angle for the left/right direction (*lr*), and one for the front/back (center/surround) direction (*cs*). The two steering directions are signed variables. When both *lr* and cs are zero the input signals are unsteered -that is, the two input channels are uncorrelated.

[0054] When the input consists of a single signal which has been directionally encoded the two steering directions have their maximum value. However under these conditions they are not independent. The advantage to representing the steering values as angles is that when there is only a single signal the absolute value of the two steering values must sum to 45 degrees. When the input includes some decorrelated material along with a strongly steered signal, the sum of the absolute values of the steering values must be less than 45 degrees.

$$|lr| + |cs| \leq 45 \qquad \ldots(2)$$

If we plot the values of the matrix elements over a two-dimensional plane formed by the steering values, the center of the plane will have the value (0, 0) and the legal values for the sum of the steering values will not exceed 45 . In practice, due to the behavior of the non-linear filters it is possible for the sum to exceed 45 - the application No. 08/742,460 claimed a circuit that limited the lessor of *lr* or *cs* so their sum did not exceed 45 degrees. This claim will not be further discussed here. We will assume the mathematics for the matrix elements is well behaved during overruns. When we graph the matrix elements we arbitrarily zero the values when the legal sum of the input variables is exceeded. This allows us to directly view the behavior of the element along the boundary trajectory - the trajectory followed by a strongly steered signal. The graphics were created by Matlab. In the Matlab language the unsteered position is (46, 46), because Matlab requires the angle variable to be 1 more than the actual angle value. Hopefully this will not be overly confusing.

[0055] Previous designs for matrix decoders tend to consider only the behavior of the matrix to a strongly steered signal - that is the behavior of the matrix elements around the boundary of our surface. This is a fundamental error in outlook. When you study real signals - either film or music - you find that the boundary of the surface is very seldom reached. For the most part signals wobble around the middle of the plane - slightly forward of the center. The behavior of the matrix under these conditions is of vital importance to the sound. When you compare our elements to previous elements you can see a striking increase in the complexity of the surface in the middle regions. It is this complexity which is responsible for the improvement in the sound.

[0056] Such complexity has a price. Our original 1987 design - see the 1989 patent-was simple to implement with analog components. The new elements are designed to be almost entirely described by one-dimensional lookup tables, which are trivial in a digital implementation. Designing an analog version with similar performance is possible, but not trivial.

[0057] In this application we contrast the several different versions of the matrix elements. The earliest are elements from our 1989 patent. These elements were used in our first surround processor, and are identical to the elements of a standard (Dolby) surround processor in the left, center, and right channels (but not identical in the surround channels.) In our design the surround channel is treated symmetrically to the center channel. In the standard (Dolby) decoder the surround channel is treated differently, and this issue will be discussed at length later in this application.

[0058] The elements presented here are not always correctly scaled. In general they are presented so that the unsteered value of the of the non-zero matrix elements are for any given channel is one. In practice the elements are usually scaled so the maximum value of each element is one or less. In any case, in a final product the scaling of the elements is additionally varied in the calibration procedure. The matrix elements presented here should be assumed to be scalable by appropriate constants.

<u>10. The left front matrix elements in our'89 patent</u>

[0059] Assume that cs and lr are the steering directions in degrees in the center/surround and left/right axis respectively.

[0060] In the '89 patent the equations for the front matrix elements are given as:

In the left front quadrant:

$$LFL = 1 - 0.\text{-}5 * G(cs) + 0.41 * G(lr) \qquad \ldots(3a)$$

$$LFR = -0.5{*}G(cs) \qquad ...(3b)$$

In the right front quadrant:

$$LFL = 1 - 0.5{*}G(cs) \qquad ...(3c)$$

$$LFR = -0.5{*}G(cs) \qquad ...(3d)$$

In the left rear quadrant (remember es is negative):

$$LFL = 1 - 0.5{*}G(cs) + 0.41{*}G(lr) \qquad ...(3e)$$

$$LFR = -0.5{*}G(cs) \qquad ...(3f)$$

In the right rear quadrant:

$$LFL = 1 - 0.5{*}G(cs) \qquad ...(3g)$$

$$LFR = -0.5{*}G(cs) \qquad ...(3h)$$

[0061] The function G(*x*) was determined experimentally in the 1989 patent, and is specified mathematically in the '91 patent. It varies from 0 to 1 as *x* varies from 0 to 45 degrees. When steering is in the left front quadrant (*lr* and cs are both positive) G(*x*) can be shown to be equal to 1-|*r*|/|*l*| where |*r*| and |*l*| are the right and left input amplitudes. G(*x*) can also be described in terms of the steering angles using various formulae. One of these is given in the '91 patent, and another will be given later in this document. See FIG. 5 and FIG. 6 for graphical representations of the LFL and LFR matrix elements plotted three-dimensionally against the *lr* and cs axes.

[0062] In reference [1] these elements were improved by adding the requirement that the loudness of unsteered material should be constant regardless of the direction of the steering. Mathematically this means that the root mean square sum of the LFL and LFR matrix elements should be a constant. It was pointed out in the paper that this goal should be altered in the direction of the steering - that is, when the steering is full left, the sum of the squares of these matrix elements should rise by 3dB. FIG. 7 shows the sum of the squares of these elements, demonstrating that the above matrix elements do not meet the requirement of constant loudness. In FIG 7, notice that the value is constant at . 71 along the axis from unsteered to right. The unsteered to left rises 3dB to the value one, and the unsteered to center or to rear falls by 3dB to the value 0.5. This part of the graph is hidden by the peak at left. The rear direction level is identical to that at the center direction.

[0063] In the application No. 08/742,460 and Reference [1], we corrected the amplitude errors in FIG. 7 by replacing the function G(x) in the matrix equations, with sines and cosines: See FIG. 8 for the resulting graph of the sum of the squares of the corrected elements LFL and LFR, which are described by the equations (4a) - (4h) below.

[0064] Note the constant value of .71 in the entire right half of the plane, and the gentle rise to one toward the left vertex.

For the left front quadrant:

$$LFL = \cos(cs) + 0.41{*}G(lr) \qquad ...(4a)$$

$$LFR = -\sin(cs) \qquad ...(4b)$$

For the right front quadrant:

$$\mathrm{LFL} = \cos(cs) \qquad \text{...(4c)}$$

$$\mathrm{LFR} = -\sin(cs) \qquad \text{...(4d)}$$

For the left rear quadrant:

$$\mathrm{LFL} = \cos(-cs) + 0.41^{*}\mathrm{G}(lr) \qquad \text{...(4e)}$$

$$\mathrm{LFR} = \sin(-cs) \qquad \text{...(4f)}$$

For the right rear quadrant:

$$\mathrm{LFL} = \cos(-cs) \qquad \text{...(4g)}$$

$$\mathrm{LFR} = \sin(-cs) \qquad \text{...(4h)}$$

11. Improvements to the left front matrix elements

[0065] In March of 1996 we made several changes to these matrix elements. We kept the basic functional dependence, but added an additional boost along the cs axis in the front, and added a cut along the cs axis in the rear. The reason for the boost was to improve the performance with stereo music that was panned forward. The purpose of the cut in the rear was to increase the separation between the front channels and the rear channels when stereo music was panned to the rear.

For the front left quadrant:

$$\mathrm{LFL} = (\cos(cs) - 0.41^{*}\mathrm{G}(lr))^{*}\mathrm{boost1}(cs) \qquad \text{...(5a)}$$

$$\mathrm{LFR} = (-\sin(cs))\ ^{*}\mathrm{boost\ 1}(cs) \qquad \text{...(5b)}$$

For the right front quadrant:

$$\mathrm{LFL} = (\cos(cs)\ )^{*}\mathrm{boostl}(cs) \qquad \text{...(5c)}$$

$$\mathrm{LFR} = (-\sin\ (cs))\ ^{*}\mathrm{boost\ 1}\ (cs) \qquad \text{...(5d)}$$

For the left rear quadrant:

$$\mathrm{LFL} = (\cos(-cs) + 0.41^{*}\mathrm{G}(lr))/\mathrm{boost}(cs) \qquad \text{...(5e)}$$

$$\mathbf{LFR} = (\sin(cs))/\mathrm{boost}(cs) \quad \text{...(5f)}$$

For the right rear quadrant:

$$\mathbf{LFL} = (\cos(cs))1\mathrm{boost}(cs) \quad \text{...(5g)}$$

$$\mathbf{LFR} = (\sin(cs))/\mathrm{boost}(cs) \quad \text{...(5h)}$$

[0066] The function G(x) is the same as the one in the '89 patent. When expressed with angles as an input, it can be shown to be equal to:

$$G(x) = 1 - \tan(45 - x) \quad \text{... (6)}$$

[0067] The function boost 1 (cs) as used in March 1997 was a linear boost of 3dB total applied over the first 22.5 degrees of steering, decreasing back to 0dB in the next 22.5 degrees. Boost(*cs*) is given by corr(*x*) in the Matlab code below (comment lines are preceded by the percent symbol %).

```
% calculate a boost function of +3dB at 22.5 degrees
% corr(x) goes up 3dB and stays up. corrl(x) goes up then down again
for x = 1:24; % x has values of 1 to 24 representing 0 to 23 degrees
  corr(x) = 10 ^ (3*(x-1)/(23*20)); % go up 3dB over this range
  corr1(x) = corr(x),
 end
 for x = 25:46 % go back down for corr1 over this range 24 to 45 degrees
  corr(x) = 1.41;
  corr(x) = corr(48-x);
 end
```

[0068] See FIG. 9 for the plot of LFL resulting from equations (5a)-(5h). Note that the boost as the steering moves toward center is applied both along the *lr* = 0 axis, and along the left to center boundary. Note also the reduction in level as the steering moves to the rear.

[0069] The performance of the March 1997 circuit can be improved. The first problem is in the behavior of the steering along the boundaries between left and center, and between right and center. As a strong single signal pans from the left to the center, it can be seen in FIG. 9 that the value of the LFL matrix element increases to a maximum half-way between left and center. This increase in value is an unintended consequence of the deliberate increase in level for the left and right main outputs as a center signal is added to stereo music.

[0070] When a stereo signal is panned forward it is desirable that the left and right front outputs should rise in level to compensate for the removal by the matrix of the correlated component from these outputs. However the method used to increase level under these conditions should only occur when the *lr* component of the inputs is minimal - that is when there is no net left or right steering. The method chosen to implement this increase in March of 1997 was independent of the value of *lr,* and resulted in an increase in level when a strong signal panned across the boundary.

[0071] The boost is only needed along the *lr* = 0 axis. When *lr* is non zero the matrix element should not be boosted. This problem can be solved by using, an additive term to the matrix elements, instead of a multiply. We define a new steering index, the boundary limited cs value with the following Matlab code:

Assume both lr and cs > 0 - we are in the left front quadrant (assume cs and lr follow the Matlab conventions of varying from 1 to 46)

```
% find the bounded c/s
  if (cs < 24)
    bcs = cs-(lr-1);
    if (bcs< 1) % this limits the maximum value
      bcs = 1;
    end
```

```
else
  bcs = 47-cs-(lr-1);
  if (bcs < 1)
    bcs = 1;
  end
end
```

[0072] If *cs* < 22.5 and *lr* =0, (in Matlab convention *cs* < 24 and lr = 1) bcs is equal to cs. However as *lr* increases bcs will decrease to zero. If *cs* > 22.5, as *lr* increases *bcs* also decreases.

[0073] Now to find the correction function needed, we find the difference between the boosted matrix elements and the non-boosted ones, along the *lr*=0 axis. We call this difference **cos_tbl_plus** and **sin_tbl_plus.** Using Matlab code,
a = 0:45; % define a vector in one degree steps. a has the values of 0 to 45 degrees
a1 = 2*pi*a/360: % convert to radians
% now define the sine and cosine tables, as well as the boost tables for the front
sin_tb1 = sin(**a1**);
cos_tb1 = cos(**a1**);
cos_tb1_plus = cos(**a1**).*corr1(**a+1**);
cos_tb1_plus = **cos_tb1_plus - cos_tb1;** % this is the one we use
cos_tb1_minus = cos(**a1**)./corr(**a+1**);
sin_tb1_plus = sin(**s1**).*corr1(**a+1**);
sin_tb1_plus = **sin_tb1_plus - sin_tb1;** % this is the one we use
sin_tb1_minus = sin(**a1**)./corr(a+1);

[0074] The vectors **sin_tb1_plus** and **cos_tb1_plus** are the difference between a plain sine and cosine, and the boosted sine and cosine. We now define

$$\mathbf{LFL} = \cos(cs) + 0.41{*}G(lr) + \mathbf{cos\_tbl\_plus}(bcs) \qquad \ldots(7a)$$

$$\mathbf{LFR} = -\sin(cs) - \mathbf{sin\_tbl\_plus}(bcs) \qquad \ldots(7b)$$

[0075] LFL and LFR in the front right quadrant are similar, but without the +0.41*G term. These new definitions lead to the matrix element shown graphically in FIG. 10.

[0076] In FIG 10, notice that the new element has the correct amplitude along the left to center boundary, as well as along the center to right boundary.

[0077] The steering in the rear quadrant is not optimal either. When the steering is toward the rear the above matrix elements are given by:

$$\mathbf{LFL} = \mathbf{cos\_tbl\_minus}(-cs) + 0.41{*}G(-cs) \qquad \ldots(8a)$$

$$\mathbf{LFR} = \mathbf{sin\_tbl\_minus}(-cs) \qquad \ldots(8b)$$

[0078] These matrix elements are very nearly identical to the elements in the '89 patent. Consider the case when a strong signal pans from left to rear. The '89 elements were designed so that there is complete cancellation of the output from the front left output only when this signal is fully to the rear (*cs* = -45, *lr* = 0). However in a Logic 7 decoder it would be desirable that the output from the left front output should be zero when the encoded signal reaches the left rear direction (*cs* = -22.5 and *lr* = 22.5). The left front output should remain at zero as the signal pans further to full rear. The matrix elements used in March 1997 - the ones above - result in the output in the front left channel being about -9dB when a signal is panned to the left rear position. This level difference is sufficient for good performance of the matrix, but it is not as good as it could be.

[0079] This performance can be improved by altering the LFL and LFR matrix elements in the left rear quadrant. Notice that here we are concerned with how the matrix elements vary along the boundary between left and rear. The mathematical method given in reference [1] can be used to find the behavior of the elements along the boundary. Let us assume that the amplitude of the left front output should decrease with the function F(*t*) as *t* varies from 0 (left) to -22.5 degrees (left rear). The method gives the matrix elements:

$$\mathbf{LFL = cos}(t)\ \mathbf{{}^*F}(t)\ \mathbf{-/+\ sin}(t)\ \mathbf{{}^*(sqrt(1-F}(t)\ \mathbf{{}^\wedge 2))} \quad \text{...(9a)}$$

$$\mathbf{LFR = (sin}(t)\mathbf{{}^*F}(t)\ \mathbf{+/-\ cos}(t)\mathbf{{}^*(sqrt(1-F}(t)\mathbf{{}^\wedge 2)))} \quad \text{...(9b)}$$

If we choose F(t) = cos(4*t) and choose the correct sign, these simplify to

$$\mathbf{LFL = cos}(t)\mathbf{{}^*cos(4{}^*}t\mathbf{)+sin}(t)\mathbf{{}^*sin(4{}^*}t\mathbf{)} \quad \text{...(9c)}$$

$$\mathbf{LFR = (sin}(t)\mathbf{{}^*cos(4{}^*}t\mathbf{)-cos}(t)\mathbf{{}^*sin(4{}^*}t\mathbf{)} \quad \text{...(9d)}$$

[0080] See FIG 11 for the plot of these coefficients LFL (solid curve) and LFR (dotted curve) against t. (The slight glitch in the middle is due to the absence of a point at 22.5 degrees, since all angles in Matlab are integers.)

[0081] These elements work well - the front left output is reduced smoothly to zero as t varies from 0 to 22.5 degrees. We want the output to remain at zero as the steering continues from 22.5 degrees to 45 degrees (full rear.) Along this part of the boundary,

$$\mathbf{LFL = -sin}(t) \quad \text{...(10a)}$$

$$\mathbf{LFR = cos}(t) \quad \text{...(10b)}$$

[0082] Note that these matrix elements are a far cry from the matrix elements along the *lr* = 0 boundary, where in reference [1] the values were

$$\mathbf{LFL = cos}(cs) \quad \text{...(10c)}$$

$$\mathbf{LFR = sin}(cs) \quad \text{...(10d)}$$

[0083] Note that these matrix elements are designed to behave properly with a strongly steered signal - where both cs and *lr* have maximum values. The previous matrix elements were successful for signals where *lr* is near zero - that is stereo signals that have been panned to the rear. We need a method of smoothly transforming the earlier matrix elements into the newer matrix elements as *lr* and cs approach the boundary. A linear interpolation could be used. In the processor used in Lexicon products, where multiplies are expensive, a better strategy is to define a new variable - the minimum of *lr* and cs, as defined by the Matlab segment below:

```
% new - find the boundary parameter
 bp = x;
 if (bp > y)
   bp = y;
 end
 and a new correction function which depends on bp:
for x = 1:24
 ax = 2*pi*(46-x) 360;
 front_boundary_tb1(x) = (cos(ax)-sin(ax))/(cos(ax)+sln(ax));
end
for x = 25:46
```

```
 ax = 2*pi*(x-1)/360;
 front_boundary_tbl(x) = (cos(ax)-sin (ax)) /(cos (ax) +sin (ax));
end
```

[0084] We then define LFL and LFR in this quadrant as:

$$\mathbf{LFL = \cos(\mathit{cs})/(\cos(\mathit{cs})+\sin(\mathit{cs})) - front\_boundary\_tbl(\mathit{bp}) + 0.41*G(\mathit{lr})} \quad ...(11a)$$

$$\mathbf{LFR = \sin(\mathit{cs})/(\cos(\mathit{cs})+\sin(\mathit{cs})) + front\_boundary\_tbl(\mathit{bp})} \quad ...(11b)$$

[0085] Note the correction of cos(cs)+sin(cs). When we divide cos(cs) by this factor we get the function 1- 0.5*G(cs), which is the same as the Dolby matrix in this quadrant. When we divide sin(cs) by this factor, we get the earlier function +0.5*G(cs).
Similarly in the right rear quadrant:

$$\mathbf{LFL = \cos(\mathit{cs})/(\cos(\mathit{cs})+\sin(\mathit{cs})) = 1 - 0.5*G(\mathit{cs})} \quad ...(12a)$$

$$\mathbf{LFR = \sin(\mathit{cs})/(\cos(\mathit{cs})+\sin(\mathit{cs})) = 0.5*G(\mathit{cs})} \quad ...(12b)$$

[0086] See FIGs. 12 and 13 for a graphical display of these values.

[0087] In FIG 12, which views the coefficient graph from left rear, note the large correction along the left-rear boundary. This causes the front left output to go to zero when steering goes from left to left rear. The output remains zero as the steering progresses to full rear. Along the *lr* = 0 axis and in the right rear quadrant the function is identical to the Dolby matrix.

[0088] In FIG 13, note the large peak in the left to rear boundary. This works in conjunction with the LFL matrix element to keep the front output at zero along this boundary as steering goes from left rear to full rear. Once again in the rear direction along the *lr* = 0 axis and in the rear right quadrant the element is identical to the Dolby matrix.

[0089] One of the major design goals of the design of the Logic 7 matrix is that the loudness in any given output of unsteered material presented to the inputs of the decoder should be constant, regardless of the direction of a steered signal which is present at the same time. As explained previously, this means that the sum of the squares of the matrix elements for each output should be one, regardless of the steering direction. As explained before, this requirement must be altered when there is strong steering in the direction of the output in question. That is, if we are looking at the left front output, the sum of the squares of the matrix elements must increase by 3dB when the steering goes full left. The above elements also alter the requirement somewhat when the steering moves forward and backward along the *lr* = 0 axis.

[0090] However we can still test the success of our design by plotting the square root of the sum of the squares of the matrix elements. See FIGs. 14 and FIG. 15 for these plots for the revised design.

[0091] In FIG. 14, note the 3dB peak in the left direction, and the somewhat lesser peak as a signal goes from unsteered to 22.5 degrees in the center direction. (For this plot we deleted the 1/(sin(cs)+cos(cs)) correction in the rear quadrant, so we could see how accurately the resulting sum came to unity.) This peak is a result of the deliberate boost of the left and right outputs during half-front steering. Note that in the other quadrants the rms sum is very close to one, as was the design intent. The value in the rear left quadrant is not quite equal to one, as the method used to produce the elements is an approximation, but the match is pretty good.

[0092] In FIG. 15, the unsteered (middle) to right axis has the value one, the center vertex has the value 0.71, the rear vertex has the value 0.5, and the left vertex has the value 1.41. Note the peak along the middle to center axis.

<u>12. Rear matrix elements during front steering</u>

[0093] The rear matrix elements in the ’89 patent (except that we have introduced a scaling by 0.71 here to show the effect of the standard calibration procedure) are given by:

For the front left quadrant

$$LRL = 0.71*(1 - G(lr)) \qquad ...(13a)$$

$$LRR = 0.71*(-1) \qquad ...(13b)$$

For the rear left quadrant

$$LRL = 0.71*(1 - G(lr) + .41*G(-cs)) \qquad ...(13c)$$

$$LRR = -0.71*(l + 0.41*G(-cs)) \qquad ...(13d)$$

(the right half of the plane is identical but switches LRL and LRR.)

[0094] The rear matrix elements in the Dolby Pro-Logic are (after similar calibration):

For the front left quadrant

$$LRL = 1 - G(lr) \qquad ...(14a)$$

$$LRR = -1 \qquad ...(14b)$$

For the rear left

$$LRL = 1 - G(lr) \qquad ...(14c)$$

$$LRR = -1 \qquad ...(14d)$$

(the right half of the plane is identical but switches LRL and LRR.)

[0095] Note that the Dolby elements and the '89 elements are calibrated to be equal in the rear left quadrant when *cs* = -45 degrees.

<u>13. A brief digression on the surround level in Dolby Pro-Logic</u>

[0096] The Dolby elements are similar to our '89 patent elements, but without the boost dependent on cs in the rear. This difference is in fact quite important, as after the standard calibration procedure the elements have quite different values for unsteered signals. In general our description of the matrix elements does not consider the calibration procedure for these decoders. We derive all the matrix elements with a relatively arbitrary scaling. In most cases the elements are presented as if they had a maximum value of 1.41. In fact, for technical reasons the matrix elements are all eventually scaled so they have a maximum value of less than one. In addition, when the decoder is finally put to use, the gain of each output to the loudspeaker is adjusted. To adjust them, a signal which has been encoded from the four major directions - left, center, right, and surround - with equal sound power is played, and the gain of each output is adjusted until the sound power is equal in the listening position. In practice this means that the actual level of the matrix elements is scaled so the four outputs of the decoder are equal under conditions of full steering. We have explicitly included this calibration in the equations for the rear elements above.

[0097] The 3dB difference in the elements in the forward steered or unsteered condition is not trivial. During unsteered conditions the elements from the '89 patent have the value 0.71, and the sum of the squares of the elements has the value of one. This is not true of the Dolby rear elements when calibrated. LRL has the unsteered value of one, and the sum of the squares is 2, or 3dB higher than the '89 outputs. Note that the calibration procedure results in a matrix that does not correspond to a "Dolby Surround" passive matrix when the matrix is unsteered. The Dolby Surround passive matrix specifies that the rear output should have the value of $.71*(A_{in} - B_{in})$, and the Pro-Logic matrix does not meet this specification. A result is that when the A and B inputs are decorrelated the rear output will be 3dB stronger than the others. If there are two speakers sharing the rear output each will be adjusted 3dB softer than a single rear speaker, which will make all five speakers have approximately equal sound power when the decoder inputs are uncorrelated. When the matrix elements from the '89 patent are used, the same calibration procedure results in 3dB less sound power from the rear when the decoder inputs are uncorrelated.

[0098] The issue of how loud the rear channels should be when the inputs are decorrelated ends up as a matter of taste. When a surround encoded recording is being played one would like to reproduce the balance the producer heard when the recording was mixed. Achieving this balance is a design goal for our decoder and encoder as a combination. However with standard stereo material the goal is to reproduce the power balance in the original recording, while generating a tasteful and unobtrusive surround. The problem with the Dolby matrix elements is that the power balance in a conventional two channel recording is not preserved through the matrix. The surround channels are too strong, and the center channel is too weak.

[0099] To see the importance of this issue, consider what happens when we have an input to the decoder that consists of three components, an uncorrelated left and right component, and a separate and uncorrelated center component.

$$A_{in} = L_{in} + .71*C_{in} \qquad \text{...(15a)}$$

$$B_{in} = R_{in} + .71*C_{in} \qquad \text{...(15b)}$$

[0100] When A_{in} and B_{in} are played through a conventional stereo system, the sound power in the room will be proportional to $L_{in}^2 + R_{in}^2 + C_{in}^2$. If all three components have roughly equal amplitudes, the power ratio of the center component to the left plus right component will be 1:2.

[0101] We would like our decoder to reproduce sound power in the room with approximately the same power ratio as stereo, regardless of the power ratio of C_{in} to L_{in} and R_{in}. We can express this mathematically. Essentially the equal power ratio requirement will specify the functional form of the center matrix elements along the cs axis, if all the other matrix elements are taken as given. If we assume the Dolby matrix elements, calibrated such that the rear sound power is 3dB less than the other three outputs when the matrix is fully steered - i.e. 3dB less than the standard calibration, then the center matrix elements should have the shape shown in FIG. 16. We can do the same thing for the standard calibration, and the results in FIG. 17 emerge.

[0102] In FIG 16, assuming the power ratios in the decoder outputs should be identical to the power ratios in stereo, and with the rear Dolby matrix elements calibrated 3dB lower in level that is typically used, notice that while the actual values give reasonable results for an unsteered signal and a fully steered signal, they are about 1.5dB too low in the middle.

[0103] In FIG 17, assuming equal power ratios to stereo given the matrix elements and calibration actually used in Dolby Pro-Logic (dotted curve), notice that the actual values are more than 3dB too low for all values of cs.

[0104] These two figures show something mix engineers are often aware of - namely that a mix prepared for playback on a Dolby Pro-Logic system can need more center loudness than a mix prepared for playback in stereo. Conversely, a mix prepared for stereo will lose vocal clarity when played over a Pro-Logic decoder. Ironically, this is not true of a passive Dolby Surround decoder. We will address this issue again when we discuss the center matrix elements.

14. Creating two independent rear outputs

[0105] The major problem with both the '89 elements and the Dolby elements is that there is only a single rear output. The '91 patent disclosed a method for creating two independent side outputs, and the math in that patent was incorporated in the front left quadrant in reference [1] of 1996 and the application No. 08/742,460. The goal of the elements in this quadrant was to eliminate the output of a signal steered from left to center, while maintaining some output from the left rear channel for unsteered material present at the same time. To achieve this goal we assumed that the LRL matrix element would have the following form:

For the left front quadrant

$$LRL = 1 - GS(lr) - 0.5*G(cs) \qquad \text{...(16a)}$$

$$\mathrm{LRR} = -0.5^{*}\mathrm{G}(cs) - \mathrm{G}(lr) \qquad \text{...(16b)}$$

[0106] As can be seen, these matrix elements are very similar to the '89 elements, but with the addition of a G(lr) term in LRR, and a GS term in LRL. G(*lr*) was included to add signals from the B input channel of the decoder to the left rear output, to provide some unsteered signal power as the steered signal was being removed. We then solved for the function GS(lr), the criterion, that there should be no signal output with a fully steered signal moving from left to center. The formula for GS(lr) turned out to be equal to G^2(lr), although a more complicated representation of the formula is given in the '91 patent. The two representations can be shown to be identical.

[0107] In reference [1] these elements are corrected by being given a boost of (sin(cs)+cos(cs)) to make them closer to constant loudness for unsteered material. While completely successful in the right front quadrant, the correction is not very successful in the left front quadrant. See FIG. 18. (For the right front quadrant the matrix elements are identical to the LRL and LRR elements in the '89 patent.)

[0108] In FIG 18, notice that in the front left quadrant there is a 3dB dip along the line from the middle to the left vertex, and nearly a 3dB boost in the level along the boundary between left and center. The mountain range in the rear quadrant will be discussed later. For this drawing the "tv matrix" correction in V1.11 has been removed, to allow better comparison to the present invention in FIG. 20.

[0109] FIG. 18 shows several problems with the sound power. First consider the dip in the sum of the squares along the cs = 0 axis. This dip exists because of the functional shape of G(*lr*) in LRR is not optimal. The choice of G(*lr*) was arbitrary - in the earlier design this function already existed in the decoder, and its implementation in analog circuitry is easy.

[0110] Ideally we would like to have a function GR(*lr*) in this equation, and choose GS(*lr*) and GR(*lr*) in such a way as to keep the sum of the squares of LRL and LRR constant along the *cs* = 0 axis, and keep the output zero along the boundary between left and center. This can be done. We would also like to be sure the matrix elements are identical to the matrix elements in the right front quadrant along the *lr* = 0 axis. Thus we assume:

$$\mathrm{LRL} = \cos(cs) - \mathrm{GS}(lr) \qquad \text{...(17a)}$$

$$\mathrm{LRR} = -\sin(cs) - \mathrm{GR}(lr) \qquad \text{...(17b)}$$

We want the sum of the squares to be one along the cs=O axis,,

$$(1 - \mathrm{GS}\,(lr))^2 + (\mathrm{GR}\,(lr))^2 = 1 \qquad \text{...(18)}$$

and the output to be zero to a steered signal, or as t varies from zero to 45 degrees,

$$\mathrm{LRL}^{*}\cos(t) + \mathrm{LRR}^{*}\sin(t) = 0 \qquad \text{...(19)}$$

Equations (18) and (19) result in a messy quadratic equation for GR and GS, which is solved numerically and graphed in FIG. 19. Use of GS and GR as shown results in a large improvement in the power sum along the *cs* = 0 axis, as intended. However, the peak in the sum of the squares along the boundary between left and center remains.

[0111] In a practical design it is probably not very important to compensate for this error, but we decided to do so heuristically with the following strategy. We will divide both matrix elements by a factor that depends on a new combined variable based on *lr* and cs. Call the new variable *xymin.* (In practice we don't use the divide, but multiply by the inverse of the factor described below.) In Matlab notation,

```
% find the minimum of x or y
 xymin = x;
 if (xymin > y)
    xymin = y;
   end
   if (xymin > 23)
    xymin = 23.
   end
```

```
% note that xymin varies from zero to 22.5 degrees.
```

[0112] We then find the correction to the matrix elements along the boundary using *xymin.*

In the front left quadrant

$$\mathrm{LRL} = (\cos(cs) - \mathrm{GS}(lr))/(1 + .29*\sin(4*xymin)) \quad \text{...(20a)}$$

$$\mathrm{LRR} = (-\sin(cs) - \mathrm{GR}(lr))/(1 + .29*\sin(4*xymin)) \quad \text{..(20b)}$$

In the front right quadrant,

$$\mathrm{LRL} = \cos(cs) \quad \text{...(20c)}$$

$$\mathrm{LRR} = -\sin(cs) \quad \text{...(20d)}$$

[0113] In reference [2] these elements are additionally multiplied by the "tv matrix" correction. FIG. 20 in this application shows the matrix elements without the "tv matrix" correction. In this application this correction is handled by the frequency dependent circuitry that follows the matrix and will be described later.

[0114] In FIG. 20, note that the sum of the squares is close to one and continuous, except for the deliberate rise in level in the rear.

15. The rear matrix elements during rear steering

[0115] The rear matrix elements given in the '91 patent were not appropriate to a five-channel decoder, and were modified heuristically in our CP-3 product. Reference [1] and U. S. Patent Application No. 08/742,460 presented a mathematical method to derive these elements along the boundary of the left rear quadrant. The method worked along the boundary, but resulted in discontinuities along the *lr* = 0 axis, and along the *cs* = 0 axis. In March of 1997 these discontinuities were repaired (mostly) by additional corrections to the matrix elements, which preserved their behavior along the steering boundaries.

[0116] For the elements described in this application these errors have been corrected by interpolation. The first interpolation fixes discontinuities along the *cs* = 0 boundary for LRL. The interpolation causes the value to match the value of GS(*lr*) when cs is zero, and allows the value to rise smoothly to the value given by the previous math as cs increases negatively toward the rear. The second interpolation causes LRR to interpolate along the cs = 0 axis to the value of GR(*lr*).

16. Left side/rear outputs during rear steering from Right to Right Rear

[0117] Consider first the Left Rear Left and Left Rear Right matrix elements when the steering is neutral or anywhere between full right and right rear. That is, *lr* can vary from 0 to -45 degrees, and cs can vary from 0 to -22.5 degrees.

[0118] Under these conditions the steered component of the input should be removed from the left outputs - there should be no output from the rear left channel when the steering is toward the right or right rear.

[0119] The matrix elements given in the '91 patent achieve this goal. They are essentially the same as the rear matrix elements in the 4 channel decoder, with the addition of the sin(*cs*)+ cos(*cs*) correction for the unsteered loudness. When this is done the matrix elements are simple sines and cosines:

$$\mathrm{LRL} = \cos(-cs) = \mathrm{sri}(-cs) \quad \text{...(21a)}$$

$$LRR = \sin(-cs) = sric(-cs) \quad \text{...(21b)}$$

[0120] Notice that we have defined a new function sric(*x*), which is equal to sin(*x*) over the range of 0 to 22.5 degrees, and sri(*x*), which is equal to cox(*x*). We will use these functions again in defining the Left Rear matrix elements during Left steering.

17. Left side/rear outputs during rear steering from Right Rear to Rear

[0121] Now consider the same matrix elements as cs becomes greater than -22.5 degrees. As we said in reference [1] and the two patent applications, LRL should rise to one or more over this range, and LRR should decrease to zero. Simple functions fulfill this (remember cs is negative and varies from -22.5 to -45 in these equations):

$$LRL = (\cos(45+cs) + rboost(-cs)) = (sri(-cs) + rboost(-cs)) \quad \text{...(22a)}$$

$$LRR = \sin(45+cs) = sric(-cs) \quad \text{...(22b)}$$

[0122] Rboost(cs) is defined in reference [1] and application No. 08/742,460. It is closely equivalent to the function 0.41*G(cs) in the earlier matrix elements, except that rboost(*cs*) is zero for 0 > *cs* > -22.5, and varies from zero to 0.41 as cs varies from -22.5 degrees to -45 degrees. Its exact functional shape is determined by the desire to keep the loudness of the rear output constant as sound is panned from left rear to full rear.

[0123] The Left Rear matrix elements during right steering are now complete.

18. The Left Rear elements during steering from left to left rear

[0124] The behavior of the Left Rear Left and Left Rear Right elements is much more complex. The Left Rear Left element must quickly rise from zero to near maximum as *lr* decreases from 4.5 to 22.5, or to zero. The matrix elements given in reference [1] perform this, but as we showed earlier, there are problems with continuity at the *cs* = 0 boundary.

[0125] For the March 1997 release a solution was found which uses functions of one variable and several conditionals. In reference [1] the problem at the *cs* = 0 boundary arises because on the forward side of the boundary (*cs* ≥ 0) the LRL matrix element is given by GS(lr). On the rear side (*cs* < 0) the function given by reference [1] has the same end points, but is different when *lr* is not zero or 45 degrees.

[0126] The mathematical method in reference [1] provides the following equations for the Left Rear matrix elements over the range 22.55 < *lr* < 45: (When transcribing these equations from reference [1] remember that t = 45-lr).

$$\begin{aligned} LRL &= \cos(45-lr)*\sin(4*(45-lr))-\sin(45-lr)*\cos(4*(45-lr)) \\ &= sra(lr) \end{aligned} \quad \text{...(23a)}$$

$$\begin{aligned} LRR &= -(\sin(45-lr).*\sin(4*(45-lr)) + \cos(45-lr).*\cos(4*(45-lr))) \\ &= -srac(lr) \end{aligned} \quad \text{...(23b)}$$

[0127] Note that we have defined two new functions, sra(*lr*) and srac(lr) over this range.

[0128] *If cs* ≥ 22.5, *lr* can still vary from 0 to 45. Reference [1] defines LRL and LRR (when *lr* has the range 0 < *lr* < 22.5) (see FIG. 6 in reference [1]) as:

$$LRL = \cos(lr) = sra(lr) \quad \text{...(23c)}$$

$$LRR = -\sin(lr) = -srac(lr) \quad \text{...(23d)}$$

[0129] The two functions sra(x) and srac(x) - are now defined for $0 < lr < 45$.

19. March 1997 version

[0130] The March 1997 version uses an interpolation technique to correct LRR along the boundaries. Here there are two discontinuities. Along the cs = 0 boundary, LRR in the rear must match the LRR for the forward direction, which shows LRR = -G(lr) along the cs = 0 boundary.

[0131] The choice used in March 1997 - although somewhat computationally intensive - is to employ an interpolation based on the value of cs over the range of 0 to 15 degrees. In other words, when cs is zero we employ G(lr) to find LRR. As cs increases to 15 degrees we interpolate to the value of srac(lr).

[0132] There is also the possibility of a discontinuity along the lr = 0 axis. In March 1997 this discontinuity was corrected (somewhat) by adding a term to LRR, which is found by using a new variable, cs_bounded. The correction term becomes simply sric(cs_bounded). This term will insure continuity across the lr = 0 axis.

[0133] First define cs_bounded in the following Matlab notation:

```
cs_bounded = lr – cs:
if (cs_bounded < 1) % this limits the maximum value
    cs_bounded = 0;
end
if (45-|lr| < cs_bounded) % use the smaller of the two values
    cs_bounded = 45-lr;
end
for cs = 0 to 15
    LRR = (-(srac(lr) + (srac(lr)-G(lr))*(15-cs)/15) + sric(cs_bounded));
for cs = 15 to 22.5
  LRR = (-srac(lr) – sric(cs_bounded));
```

20. LRL as implemented in Logic 7 as of August ’97

[0134] In the present invention, LRL is computed with interpolation, just as for LRR. In Matlab notation:

```
for cs = 0 to 15
 LRL = ((sra(lr) ÷ (sra(lr-GS(lr))*(15-cs)/15) + sri(-cs));
for cs = 15 to 22.5
 LRL = (sra(lr) + sri(-cs));
```

21. Rear outputs during steering from Left Rear to Full Rear

[0135] As the steering goes from left rear to full rear the elements follow the ones given in reference [1], with the addition of the corrections for rear loudness. In Matlab notation,

```
For cs > 22.5, lr < 22.5
 LRL = (sra(lr)-sri(cs) + rboost(cs))
 LRR = -srac(lr) - sric(cs_bounded)
```

[0136] This completes the LRL and LRR matrix elements during left steering. The values for right steering can be found by swapping left and right in the definitions.

22. Center matrix elements

[0137] The ’89 patent and Dolby Pro-Logic both have the following matrix elements:

For front steering:

$$CL = 1 - G(lr) + 0.41*G(cs) \quad ...(24a)$$

$$CR = 1 + 0.41*G(cs) \quad ...(24b)$$

For rear steering:

$$CL = 1 - G(lr) \quad ...(24c)$$

$$CR = 1 \quad ...(24d)$$

[0138] Since the matrix elements have symmetry about the left right axis, the values of CL and CR for right steering can be found by swapping CL and CR. See FIG. 21 for a graphical representation of this element.

[0139] In FIG 21, the middle of the graph, and the right and rear vertices have the value 1. The center vertex has the value 1.41. In practice this element is scaled so the maximum value is one.

[0140] In application No. 08/742,460 and reference [1] these elements are replaced by sines and cosines.

For front steering:

$$CL = \cos(45-lr)*\sin(2*(45-lr)) - \sin(45-lr)*\cos(2*(45-lr)) + 0.41*G(cs) \quad ...(25a)$$

$$CR = \sin(45-lr)*\sin(2*(45-lr)) + \cos(45-lr)*\cos(2*(45-lr)) + 0.41*G(cs) \quad ...(25b)$$

[0141] These equations were never implemented. The March 1997 product used the elements in the'89 patent, but with a different scaling, and a different boost function than G(*cs*). We found that it was important to reduce the unsteered level of the center output, and a value 4.5dB less than the Pro-Logic level was chosen. The boost function (0.41*G(*cs*)) was changed to increase the value of the matrix elements back to the Pro-Logic value as cs increases toward center. The boost function in the March 1997 version was chosen heuristically through listening tests.

[0142] In the version of March 1997 the boost function of cs starts at zero as before, and rises with cs in such a way that CL and CR increase 4.5dB as cs goes from zero to 22.5 degrees. The increase is a constant number of dB for each dB of increase in cs. The boost function then changes slope, such that in the next 20 degrees the matrix elements rise another 3dB, and then hold constant. Thus when the steering is "half front" (8dB or 23 degrees) the new matrix elements are equal to the neutral values of the old matrix elements. As the steering continues to move forward, the new and the old matrix elements become equal. The output of the center channel is thus 4.5dB less than the old output when steering in neutral, but rises to the old value when the steering is fully to the center. See FIG. 22 for a plot in three dimensions of this element.

[0143] In FIG 22, note that the middle value and the right and rear vertices have been reduced by 4.5dB. As cs increases the center rises to the value of 1.41 in two slopes.

[0144] We have since found that the center elements used in March 1997 are not optimal. Considerable experience with the decoder in practice has shown that the center portion of popular music recordings, and the dialog in some films, can tend to get lost when you switch between stereo (two channel) reproduction, and reproduction through the matrix. In addition, as the center channel changes in level a listener who is not equidistant from the front speakers can notice the apparent position of a center voice moving. This problem was extensively analyzed in developing the new center matrix elements presented here. As we will see later, there is also a problem when a signal pans from left to center or from right to center along the boundary. The matrix elements in application No. 08/742,460 give too low an output from the center speaker when the pan is half way between.

23. Center channel in the new design

[0145] While it is possible to remove a strongly steered signal from the center channel output using matrix techniques, any time the steering is frontal but not biased either left or right, the center channel must reproduce the sum of the A and B inputs with some gain factor. In other words it is not possible to remove uncorrelated left and right material from the center channel. Our only option is to regulate the loudness of the center speaker. How loud should it be?

[0146] This question depends on the behavior of the left and right main outputs. The matrix values presented above for LFL and LFR are designed to remove the center component of the input signals as the steering moves forward. We can show that if the input signal has been encoded to come from the forward direction by using a cross mixer, such as a stereo width control, the matrix elements given above (the '89 elements, the 1996 AES paper elements, the March 1997 elements, and the ones presented earlier in this paper,) all completely restore the original separation.

[0147] However, if the input to the decoder consists of uncorrelated left and right channels to which an unrelated center channel has been added, i.e.

$$A_{in} = L_{in} + .71^{*}C_{in} \qquad \text{...(26a)}$$

$$B_{in} = R_{in} + .71^{*}C_{in} \qquad \text{...(26b)}$$

then as the level of C_{in} increases relative to L_{in} and R_{in} the C component of the L and R front outputs of the decoder is not completely eliminated unless C_{in} is large compared to L_{in} and R_{in}. In general there is a bit of C_{in} left in the L and R front outputs. What does a listener hear?

[0148] There are two ways of calculating what a listener hears. If a listener is exactly equidistant from the Left, Right, and Center speakers they will hear the sum of the sound pressures from each speaker. This is equivalent to summing the three front outputs. Under these conditions it is easy to show that any reduction of the center component of the left and right speakers will result in a net loss of sound pressure from the center component, regardless of the amplitude of the center speaker. This is because the center speaker is always derived from the sum of the A and B inputs, and as its amplitude is raised the amplitude of the L_{in} and R_{in} signals must rise along with the amplitude of the C_{in} signal.

[0149] However if the listener is not equidistant from each speaker, the listener is much more likely to hear the sum of the sound power from each speaker, which is equivalent to the sum of the squares of the three front outputs. In fact, extensive listening has shown that in fact the sum of the powers of all the speakers is actually what is important, so we must consider the sum of the squares of all the outputs of the decoder, including the rear outputs.

[0150] If we want to design the matrix so the ratio of the amplitudes of L_{in}, R_{in} and C_{in} are preserved when switching between stereo reproduction and matrix reproduction, the sound power of the C_{in} component from the center output must rise in exact proportion to the reduction in its sound power from the left and right outputs, and its reduction in the rear outputs. An additional complication is that the left and right front outputs have the level boost of up to 3dB described above. This will cause the center to need to be somewhat louder to keep the ratios constant. We can write this requirement as a set of equations for the sound power. These equations can be solved for the gain function we need for the center speaker.

[0151] We previously gave graphs showing the energy relations for a Dolby Pro-Logic decoder under various conditions. The Pro-Logic decoder is not optimal. We can do the same for our new decoder.

[0152] FIG. 23 shows the center gain needed (solid curve) if the energy of the center component of the input signal is to be preserved in the front three channels as steering increases toward the front. As can be seen, the needed rise in the level of the center channel is quite steep - the rise is many dB of amplitude per dB of steering value. Also shown is the gain in a standard decoder (dotted curve).

[0153] As we previously mentioned, there are two solutions to this problem. We will describe the "film" solution first. The solution here is not entirely mathematical. We found in practice that the function shown in FIG. 23 rises too steeply. The change in level of the center channel is too obvious. We decided to relax the power requirement slightly - to about 1dB less center than ideal. If we recalculate the center values we get the result shown in the solid line of FIG. 24. In practice we can substitute a linear rise for the early part of the curve, shown in the dashed line in FIG. 24. In practice results with these center values have been excellent for films.

[0154] Referring to FIG 24, in practice the solid curve rises too steeply. The linear slope given by the dashed line works better.

[0155] Music requires a different solution. The center attenuation shown in FIGs. 23 and 24 is derived assuming the matrix elements previously given for LFL and LFR. What if we used different elements? Specifically, do we need to be aggressive about removing the center component from the left and right front outputs?

[0156] Listening tests show that the previous left and right front matrix elements are needlessly aggressive about removing the center component during music playback. Acoustically there is no need that they should do so. The energy removed from the left and right front must be given to the center loudspeaker. If we don't remove this energy it comes from the left and right front speakers, and the center speaker need not be as strong. The sound power in the room is the same. The trick is to put just enough energy into the center speaker to create a convincing front image for an off-axis listener while minimizing the reduction of stereo width for a listener who is equidistant from the front left and right speakers.

[0157] As we did in application No. 08/742,460 we can find the optimal center loudness by trial and error. We can then solve for the matrix elements we need in the front left and right to preserve the power of the C_{in} component in the room. As before, we assume that the center channel is reduced in level by 4.5 dB below the level in our '89 patent decoder, or -7.5dB total attenuation. -7.5dB equals 0.42. The matrix elements for the center can be multiplied by this factor, and a new center boost function (GQ can be defined.

For front steering

$$\mathrm{CL} = 0.42*(1 - \mathrm{G}(lr)) + \mathrm{GC}(cs) \qquad \text{...(27a)}$$

$$\mathrm{CR} = 0.42 + \mathrm{GC}(cs) \qquad \text{...(27b)}$$

For rear steering

$$\mathrm{CL} = 0.42*(1 - \mathrm{G}(lr)) \qquad \text{...(27c)}$$

$$\mathrm{CR} = 0.42 \qquad \text{...(27d)}$$

[0158] Several functions were tried for GC(cs). The one given below may not be ideal, but seems good enough. It is specified in terms of the angle cs in degrees, and was obtained by some trial and error.

[0159] In MATLAB notation:

```
center_max = 0.65;
center_rate = 0.75;
center_max2 = 1;
center_rate2 = 0.3;
center_rate3 = 0. 1;
If (cs < 12)
   gc(cs-1) = 0.42*10 (db*center_rate/(20));
   tmp = gc(cs + 1);
elseif (cs < 30)
   gc(cs-1) = tmp*10^((cs-11)*center_rate3/(20));
   if (gc(cs-1) > center_max)
     gc(cs÷1) = center_max;
   end
else
   gc(cs÷1)=center_max*10^((cs-29)*center_rate2/(20));
   if (gc(cs÷1) > center_max2)
     gc(cs÷1) =center_max2;
   end
end
```

[0160] The function (0.42 + GC(*cs*)) is plotted in FIG. 2.5. Note the quick rise from the value 0.42 (45dB lower than Dolby surround), followed by a gentle rise, followed finally by a steep rise to the value 1..

[0161] We can solve for the needed function for LFR if we assume functions for LFL, LRL, and LRR. We want to solve for the rate that the C_{in} component in the left and right outputs should decrease, and then design matrix elements, which provide this rate of decrease. These matrix elements should also provide some boost of the L_{in} and R_{in} components, and should have the current shape at the left to center boundary, as well as the right to center boundary.

[0162] We assume that:

$$\mathrm{LFL} = \mathrm{GP}(cs) \quad \text{...(28a)}$$

$$\mathrm{LFR} = \mathrm{GF}(cs) \quad \text{...(28b)}$$

$$\mathrm{CL} = 0.42*(1 - \mathrm{G}(lr) + \mathrm{GC}(cs) \quad \text{...(28c)}$$

$$\mathrm{CR} = 0.42 + \mathrm{GC}(cs) \quad \text{...(28d)}$$

Power from the front left and right can then be computed as follows:

$$\mathrm{PLR} = (\mathrm{GP}^2+\mathrm{GF}^2)*(\mathrm{L_{in}}+\mathrm{R_{in}}) + (\mathrm{GP}-\mathrm{GF})^{2}*\mathrm{C_{in}}^2 \quad \text{...(29a)}$$

Power from the center is:

$$\mathrm{PC} = \mathrm{GC}^{2}* (\mathrm{L_{in}}^2+\mathrm{R_{in}}^2) + 2 * \mathrm{GC}^{2}*\mathrm{C_{in}}^2 \quad \text{...(29b)}$$

[0163] Power from the rear depends on the matrix elements we use. We will assume the rear channels are attenuated by 3dB during forward steering, and that LRL is cos(cs) and LRR is sin(cs). From a single speaker,

$$\mathrm{PREAR} = (0.71*(\cos(cs)*(\mathrm{L_{in}} + 0.71*\mathrm{R_{in}}) - \sin(cs)*(\mathrm{R_{in}} + 0.\ 71*\mathrm{C_{in}})))^2 \quad \text{...(29c)}$$

[0164] If we assume that $L_{in}^2 \approx R_{in}^2$ then, for two speakers,

$$\mathrm{PREAR} = 0.5*\mathrm{C_{in}}^{2}*((\cos(cs)-\sin(cs))^2) + \mathrm{L_{in}}^2 \quad \text{...(29d)}$$

The total power from all three speakers is PLR + PC + PREAR:

$$\mathrm{PT} = (\mathrm{GP}^2+ \mathrm{GF}^2+ \mathrm{GC}^2)*(\mathrm{L_{in}}^2+\mathrm{R_{in}}^2) + ((\mathrm{GP} - \mathrm{GF})^2 + 2*\mathrm{GC}^2)*\mathrm{C_i}^{n2} + \mathrm{PREAR} \quad \text{...(30)}$$

The ratio of C_{in} power to L_{in} and R_{in} power is: (assume $L_{in}^2 = R_{in}^2$)

$$\text{RATIO} = (((GP(cs)\text{-}GF(cs))^2 + 2 * (GC(cs)^2 + 0.5 * (\cos(cs)\text{-}\sin(cs))^2)) * C_{in}^2 / ((2*(GP(cs)^2 + GC(cs)^2 + GF(cs)^2) + 1) * L_{in}^2 \quad ...(31a)$$

$$\text{RATIO} = (C_{in}^2/ L_{in}^2) *((GP(cs)\text{–}GF(cs))^2 + 2*(GC(cs)^2) + 0.5*(\cos(cs) - \sin(cs))^2 / (2*(GP(cs)^2 + GC(cs)^2 + GF(cs)^2 + 1) \quad ...(31b)$$

[0165] For normal stereo, GC=0, GP=1, and GF=0. The center to LR power ratio is then:

$$\text{RATIO} = (C_{in}^2/ L_{in}^2)*0.5 \quad ...(32)$$

[0166] If this ratio is to be constant regardless of the value of (C_{in}^2/ L_{in}^2) for our active matrix, then

$$((GP(cs)\text{–}GF(cs))^2 + 2*(GC(cs)^2 + 0.5*(\cos(cs)\text{–}\sin(cs))^2) = ((GP(cs)^2 + GC(cs)^2 + GF(cs)^2 + 0.5) \quad ...(33)$$

[0167] The equation above can be solved numerically. If we assume the GC above, and GP = LFL as before, we can see the result in FIG. 26.

[0168] In FIG. 26 the solid curve is the graph of GF needed for constant energy ratios with the new "music" center attenuation GC. The dashed curve is the LFR element of March '97, sin(*cs*)*corr1. The dotted curve is sin(cs), the LFR element without the correction term corr1. Note that GF is close to zero until cs reaches 30 degrees, and then increases sharply. We have found in practice it is best to limit the value of cs at about 33 degrees. In practice LFR derived from these curves has a negative sign.

[0169] GF gives the shape of the LFR matrix element along the *lr* = 0 axis, as cs increases from zero to center. We need a method of blending this behavior to that of the previous LFR element, which must be preserved along the boundary between left and center, as well as from right to center. A method of doing this when *cs* ≤ 22.5 degrees is to define a difference function between GF and sin(cs). We then limit this function in various ways. In Matlab notation:

```
gf_diff = sin(cs) - gf(cs):
  for cs = 0:45;
   if (gf_diff(cs) > sin(cs))
     gf_diff(cs) = sin(cs);
     end
     if (gf_diff(cs) < 0)
      gf_diff(cs) = 0:
     end
   end
   % find the bounded c/s
     if (y < 24)
      bcs = y-(x-1);
      if (bcs<1) % this limits the maximum value
       bcs = 1;
      end
    else
      bcs = 47-y-(x-1);
      if(bcs <1) %> 46)
       bcs = 1; %46;
      end
    end
```

[0170] The LFR element can now be written in Matlab notation: % this neat trick does an interpolation to the boundary

```
% the cost, of course, is a divide!!!
    if (y < 23) % this is the easy way for half the region
      lfr3d(47-x, 47-y) = -sin_tbl(y)+gf_diff(bcs);
    else
      tmp = ((47-y-x)/(47-y))*gf_diff(y);
      lfr3d(47-x,47-y) = -sin_tbl(y)+tmp;
    end
```

[0171] Note that the sign of gf_diff is positive in the equation above. Thus gf_diff cancels the value of sin(*cs*), reducing the value of the element to zero along the first part of the *lr* = 0 axis. See FIG. 27.

[0172] In FIG. 27, note that the value is zero in the middle of the plane (no steering) and remains zero as *cs* increases to ~30 degrees along the *lr* = 0 axis. The value then falls off to match the previous value along the boundary from left to center and from right to center.

24. Panning error in the center output

[0173] As it turns out, the new center function, if we write it this way,

$$\mathbf{CL = 0.42{*}(1 - G(\mathit{lr})) + GC(\mathit{cs})} \qquad \text{...(34a)}$$

$$\mathbf{CR = 0.42 + GC(\mathit{cs})} \qquad \text{...(34b)}$$

works well along the *lr* = 0 axis, but causes a panning error along the boundary between left and center, and between right and center. The values in reference [1] of 1996 (which were never implemented) give a smooth function of cos (2**cs*) along the left boundary. These values create smooth panning between left and center. We would like our new center function to have similar behavior along this boundary.

[0174] We can make a correction to the matrix element that will do the job by adding an additional function *of xymin* (in Matlab notation):

```
center_fix_tbl = .8*(corr1-1);
```

Then,

$$\mathbf{CL = 0.42 - 0.42{*}G(\mathit{lr}) + GC(\mathit{cs}) + center\_fix\_table(\mathit{xymin})} \qquad \text{...(35a)}$$

$$\mathbf{CR = 0.42 + GC(\mathit{cs}) + center\_fix\_table(\mathit{xymin})} \qquad \text{...(35b)}$$

[0175] See FIG. 28 for a three-dimensional representation of the CL matrix element. While not perfect, this correction works well in practice.

[0176] In FIG. 28, note the correction for panning along the boundary between left and center, which is fairly smooth.

[0177] In FIG 29, which is a graph of the left front (dotted curve) and center (solid curve) outputs, note that center steering is at the left of the plot, and full left is at the right. In the "music" strategy we currently limit the value of cs to about 33 degrees, (about 13 on the axis as labeled) where the center is about 6dB stronger than the left.

25. Technical details of the encoder

[0178] No protection is sought for the encoder in the present application.

[0179] There are two major goals of the Logic 7 encoder. Firstly, it should be able to encode a 5.1 channel tape in a way that allows the encoded version to be decoded by a Logic 71 decoder with minimal subjective change. Secondly, the encoded output should be stereo compatible - that is, it should sound as close as possible to a manual two channel mix of the same material. One factor in this stereo compatibility should be that the output of the encoder, when played on a standard stereo system, should give identical perceived loudness for each sound source in an original 5 channel mix. The apparent position of the sound source in stereo should also be as close as possible to the apparent position in the 5 channel original.

[0180] In discussions with the Institute for Broadcast Technique (IRT) in Munich, it became apparent that the goal of stereo compatibility of the stereo signal as described above cannot be met by a passive encoder. A five channel recording where all channels have equal foreground importance must be encoded as described above. This encoding requires that surround channels be mixed into the output of the encoder in such a way as the energy is preserved. That is, the total energy the output of the encoder should be the same, regardless of which input is being driven. This constant energy setting will be necessary for most film sources and for five channel music sources where instruments have been assigned equally to all 5 loudspeakers. Although such music sources are not common at the present time, it is the author's opinion that they will become common in the future. Music recordings where the foreground instruments are placed in the front three channels, with primarily reverberation in the rear channels, require a different encoding.

[0181] After a series of tests (at the IRT and elsewhere) it was determined that music recordings of this type were successfully encoded in a stereo compatible form when the surround channels were mixed with 3dB less power than the other channels. This -3dB level has been adopted as a standard for surround encoding in Europe, but the standard specifies that other surround levels can be used for special purposes. The new encoder contains active circuits which detect strong signals in the surround channels. When such signals are occasionally present, the encoder uses full surround level. If the surround inputs are consistently -6dB or less compared to the front channels, the surround gain is gradually lowered 3dB, to correspond to the European standard.

[0182] These active circuits were also present in the encoder in application No. 08/742,460. However tests with the earlier encoder at the Institute for Broadcast Technique (IRT) in Munich I found that direction of some sound sources encoded incorrectly. A new architecture was developed to solve these problems. The new encoder is clearly superior in its performance on a wide variety of difficult material. The original encoder was developed first as a passive encoder. The new encoder will also work in a passive mode, but is primarily intended to work as an active encoder. The active circuitry corrects several small errors inherent in the design. However even without the active correction the performance is better than the previous encoder.

[0183] With extensive listening several other small problems with the first encoder were discovered. Many (but not all) of these problems have been addressed in the new encoder. For example, when stereo signals are applied to both the front and the rear terminals of the encoder at the same time, the resulting encoder output is biased too far to the front. The new encoder compensates for this effect by increasing the rear bias slightly. Likewise, we have found that when a film is encoded with substantial surround content, dialog can sometimes get lost. This problem was greatly improved by the changes to the power balance described above, but the encoder is also intended to be used with a standard (Dolby) decoder. The new encoder compensates for this effect by raising the center channel input to the encoder slightly under these conditions.

<u>26. Explanation of the design</u>

[0184] The new encoder handles the left, center, and right signals identically to the previous design and identically to the Dolby encoder, providing the center attenuation function *fcn* is equal to 0.71, or -3dB.

[0185] The surround channels look more complicated than they are. The functions *fc*() and *fs*() direct the surround channels either to a path with a 90 degree phase shift relative to the front channels, or to a path with no phase shift. In the basic operation of the encoder *fc* is one, and *fs* is zero - that is, only the path which uses the 90 degree phase shifts is active.

[0186] The value *crx* is typically 0.38. It controls the amount of negative cross feed for each surround channel. As in the previous encoder, when there is only an input to one of the surround channels the A and B outputs have an amplitude ratio of -.38/.91, which results in a steering angle of 22.5 degrees to the rear. As usual, the total power in the two output channels is unity - that is the sum of the squares of .91 and .38 is one.

[0187] While the output of this encoder is relatively simple when only one channel is driven, it becomes problematic when both surround inputs are driven at the same time. If we drive the LS and the RS input with the same signal (a common occurrence in film), all the signals at the summing nodes are in phase, so the total level in each output channel is .38 + .91 or 1.29. This output is too strong by the factor of 1.29, or 2.2 dB. Active circuitry is included in the encoder to reduce the value of the function *fc* by up to 2.2 dB when the two surround channels are similar in level and phase.

[0188] Another error occurs when the two surround channels are similar in level and out of phase. In this case the two attenuation factors subtract, so the A and B outputs have equal amplitude and phase, and a level of .91-.38, or .53. This signal will decode as a center direction signal. This error is severe. The previous encoder design produced an unsteered signal under these conditions, which is reasonable. It is not reasonable that signals applied to the rear input terminals should result in a center oriented signal. Thus active circuitry is supplied which increases the value of *fs* when the two rear channels are similar in level and antiphase. The result of mixing both the real path and the phase shifted path for the rear channels is a 90 degree phase difference between the output channels A and B. This results in an unsteered signal, which is what we want.

[0189] As previously mentioned I discovered during discussions at the IRT in Munich that there is a European standard surround encoder. This encoder simply attenuates the two surround channels by 3dB, and adds them into the front channels. Thus the left rear channel is attenuated and added to the left front channel. This encoder has many disadvantages when encoding multichannel film sound, or recordings which have specific instruments in the surround channels. Both the loudness and the direction of these instruments will be incorrectly encoded. However this encoder works rather well with classical music, where the two surround channels are primarily reverberation. The 3dB attenuation was carefully chosen through listening tests to produce a stereo compatible encoding. I decided that our encoder should include this 3dB attenuation when classical music was being encoded, and that one could detect this condition through the relative levels of the front channels and the surround channels in the encoder.

[0190] A major function of the function *fc* in the surround channels is to reduce the level of the surround channels in the output mix by 3dB when the surround channels are much softer than the front channels. Circuitry is provided to compare the front and rear levels, and when the rear is less by 3dB, the value of *fc* is reduced to a maximum of 3dB. The maximum attenuation is reached when the rear channels are 8dB less strong than the front channels. This active circuit appears to work well. It makes the new encoder compatible with the European standard encoder for classical music. The action of the active circuits causes instruments which are intended to be strong in the rear channels to be encoded with full level.

[0191] There is another function of the real coefficient mixing path *fs* for the surround channels. Then a sound is moving from the left front input to the left rear input active circuitry detects that these two inputs are similar in level and in phase. Under these condition *fc* is reduced to zero and *fs* is increased to one. This change to real coefficients in the encoding results in a more precise decoding of this type of pan. In practice this function is probably not essential, but it seems an elegant refinement.

[0192] There is an additional active circuit that has not yet been released in a product. Level detecting circuits look at the phase relationship between the center channel and the front left and right. Some popular music recordings that use five channels mix the vocals into all three front channels. When there is a strong signal in all three inputs the encoder output will have excessive vocal power, since the three front channels will add together in phase. When this occurs, active circuits increase the attenuation in the center channel by 3dB to restore the power balance in the encoder output.

[0193] To summarize: Active circuits are provided to

1. Reduce the level of the surround channels by 2.2dB when the two channels are in phase
2. Increase the real coefficient mixing path for the rear channels sufficiently to create an unsteered condition when the two rear channels are out of phase.
3. Decrease the level of the surround channels by up to 3dB when the surround level is much less than the front levels.
4. Increase the level and negative phase of the rear channels when their level is similar to the front channels.
5. Make the surround channel mix use real coefficients when a sound source is panning from a front input to the corresponding rear input.

[0194] Increase the level of the center channel in the encoder when the center level and the level of the front and surround inputs are approximately equal.

[0195] Decrease the level of the center channel in the encoder when a there is a common signal in all three front inputs.

[0196] Future improvements to the encoder are likely to include a feature similar to feature 2 above for the front channels. In the current encoder when the two front channels are out of phase the encoding will cause the decoder to place the sound in the rear. We intend to detect this condition and make the resulting output unsteered.

<u>27.Frequency dependent circuits in the decoder</u>

[0197] FIG. 2 shows a block diagram of the frequency dependent circuits that follow the matrix in a five channel version of the decoder. There are three sections: a variable low pass filter, a variable shelf filter, and a HRTF (Head Related Transfer Function) filter. The HRTF filter changes its characteristics depending on the value of the rear steering voltage c/s. The first two filters change their characteristics in response to a signal that is intended to represent the average direction of the input signals to the decoder during pauses between strongly steered signals. This signal is called the background control signal.

<u>28. The background control signal</u>

[0198] One of the major goals of the current decoder is to be able to optimally create a five channel surround signal from an ordinary two channel stereo signal. It is also highly desirable that the decoder should recreate a five channel surround recording that was encoded into two channels by the encoder described as part of this application. These two applications differ in the way the surround channels are perceived. With an ordinary stereo input the majority of the sound needs to be in front of the listener. The surround speakers should contribute a pleasant sense of envelopment and ambience, but should not draw attention to themselves. An encoded surround recording needs the surround speakers to be stronger and more aggressive.

[0199] To play both types of input optimally without any adjustment from the user it is necessary to discriminate between a two channel recording and an encoded five channel recording. The background control signal is designed to make this discrimination. The background control signal (BCS) is similar to and derived from the rear steering signal cs. BCS represents the negative peak value of cs. That is, when cs more negative than BCS, then BCS is made to equal cs. When cs is more positive than BCS the value of BCS slowly decays. However the decay of BCS involves a further calculation.

[0200] Music of many types consists of a series of strong foreground notes - or in the case of a song, sung words. In between the foreground notes there is a background. The background may consist of other instruments playing other notes, or it may consist of reverberation. The circuit that derives the, BCS signal keeps track of the peak level of the foreground notes. When the current level is -7dB less than the peak level of the foreground, the level of cs is measured. The value of cs during these gaps between foreground peaks is used to control the decay of BCS. If the material in the gaps between notes is reverberation, it may tend to have a net rearward bias in a recording that was made by encoding a five channel original. This is because the reverberation on the rear channels of the original will be encoded with a rearward bias. The reverberation in an ordinary two channel recording will have no net rearward bias. Cs for this reverberation will be zero or slightly forward.

[0201] BCS derived in this way tends to reflect the type of recording. Any time there is significant rear steered material BCS will always be strongly negative. However BCS can be negative even in the absence of strong steering to the rear if there the reverberation in the recording has a net rearward bias. We can use BCS to adjust the filters that optimize the decoder for stereo vs surround inputs.

<u>29. Frequency dependent circuits: five channel version</u>

[0202] The first of the filters in FIG. 2 is a simple 6dB per octave low pass filter, with an adjustable cutoff frequency. When BCS is positive or zero this filter is set to a value that is user adjustable, but is typically about 4kHz. As BCS becomes negative the cutoff frequency is raised, until when BCS is more rearward than 22 degrees the filter is not active. This low frequency filter makes the rear outputs less obtrusive when ordinary stereo material is played. The filter has been a part of the decoder at least since V1.11, but in the earlier decoders it was controlled by *cs*, and not by BCS.

[0203] The second filter is a variable shelf filter. The low frequency section (the pole) of this filter is fixed, at 500 Hz. The high frequency section (the zero) varies depending on user adjustment and on BCS. This filter implements the "soundstage" control in the current decoder. In the application No. 08/742,460 "soundstage" is implemented through the matrix elements, using the "tv matrix" correction. The earlier decoders based on this work reduced the overall level of the rear channels when the steering was neutral or forward. In the new decoder presented here the matrix elements do not include the "tv matrix" correction.

[0204] In the new decoders when the soundstage control is set to "rear" the high frequency section of the shelf filter is set equal to the low frequency section - in other words the shelf has no attenuation. and the filter has flat response.

[0205] When the soundstage control is set to "neutral" the setting of the high frequency zero varies. When BCS is positive or zero the zero moves to 710Hz - resulting in a 3dB attenuation of higher frequencies. For the high frequencies the result is the same as the earlier decoders. There is a 3dB attenuation when the steering is neutral or forward. However the low frequencies are not attenuated. They come from the sides of the room with full level. The result is greater low frequency richness and envelopment, without distracting high frequencies in the rear. As BCS becomes negative the high frequency zero moves toward the pole, so that when BCS is about 22 degrees to the rear the shelf filter has on attenuation.

[0206] When the soundstage control is set to "front" the action is similar, but the zero moves to 1kHz when BCS is zero or positive. This gives the high frequencies an attenuation of 6dB. Once again, as BCS goes negative the attenuation is removed.

[0207] The third filter is controlled by *c/s* and not by BCS. This filter is designed to emulate the frequency responses of the human head and pinnae when a sound source is approximately 150 degrees in azimuth from the front of the listener. This type of frequency response curve is called a "Head Related Transfer Function" or HRTF. These frequency response functions have been measured for many angles for many different people. In general when a sound source is about 150 degrees from the front there is a strong notch in the frequency response at about 5kHz. A similar notch exists when a sound source is in front of a listener - only in this case the notch is at about 8kHz. Sound sources to the side of the listener do not produce these notches. The human brain uses the presence of the notch at 5kHz as one of the ways it detects that a sound source is behind the listener.

[0208] The current standard for five channel sound reproduction recommends that the two rear speakers be placed slightly behind the listener, at +/-110 or 120 degrees from the front. This speaker position supplies good envelopment at low frequencies. However a sound from the side a listener does not produce the same level of excitement as a sound that is fully behind a listener. Very often a film director wants a sound effect to come from behind the listener, and not from the side.

[0209] It is also often the case that a listening room does not have a size or shape that is appropriate to place the loudspeakers fully behind the listener, and a side position is the best that can be achieved.

[0210] The HRTF filter in the decoder adds the frequency notches of a rear sound source, so that a listener hears the sound as further behind that the actual position of the loudspeaker. The filter is designed to vary with cs. When *cs* is positive or zero, the filter is maximum. This causes ambient sounds and reverberation to seem to be more behind the listener. As cs becomes negative the filter is reduced. When cs is approximately -15 degrees the filter is completely removed, and the sound source appears to come fully from the side. As cs goes further negative the filter is once again applied, so the sound source appears to go behind the listener. When cs is fully to the rear the filter is slightly modified to correspond to the HRTF function for a sound fully to the rear.

30. Frequency dependent circuits: the seven channel version

[0211] FIG. 3 shows the frequency dependent circuits the seven channel version of the decoder. These are shown as consisting of three sections - although in an actual implementation the second two sections can be combined into one circuit.

[0212] The first two sections are identical to the two sections in the five channel decoder, and perform the same function. The third section is unique to the seven channel decoder. In V1.11 and application No. 08/742,460 the side and the rear channels had separate matrix elements. The action of the elements was such that when cs was positive or neutral the side and the rear outputs were identical except for delay. The two outputs stayed identical until cs was more negative than 22 degrees. As the steering moved further to the rear the side outputs were attenuated by 6dB, and the rear outputs were boosted by 2dB. This caused the sound to appear to move from the sides of the listener to the rear of the listener.

[0213] In the present decoder the differentiation between the side output and the rear output is achieved by a variable shelf filter in the side output. The third shelf filter in FIG. 3 has no attenuation when cs is forward or zero. When cs becomes more negative than 22 degrees the zero in the shelf filter moves rapidly toward 1100Hz, resulting in an attenuation of the high frequencies of about 7dB. Although this shelf filter has been described as a separate filter from the shelf filter that provides the "soundstage" function, the action of the two shelf filters can be combined into a single shelf through suitable control circuitry.

[0214] While the preferred embodiments of the invention have been described and illustrated herein, many other possible embodiments exist, and these and other modifications and variations will be apparent to those skilled in the art.

Claims

1. A surround sound decoder (90) for redistributing a pair of left and right audio input signals including directionally encoded and non-directional components into a plurality of output channels (172, 174, 176, 178, 180) for reproduction through loudspeakers surrounding a listening area, and incorporating means for determining the directional content of said left and right audio signals and generating therefrom at least a left/right steering signal and center/surround steering signal, the decoder comprising:

left and right input terminals (92, 94) for receiving said corresponding left and right audio input signals;
left and right delay means (96, 118) for producing delayed left and right audio signals from said left and right audio input signals:

a plurality of multiplier means (140, 142, 144, 146, 148, 150, 152, 154, 156, 158) equal to twice the number of said plurality of output channels, organized in pairs, a first element of each said pair receiving said delayed left audio signal and a second element receiving said delayed right audio signal, each of said multiplier means multiplying its input audio signal by a variable matrix coefficient to provide an output signal:
said variable matrix coefficient being controlled by one or both of said steering signals; and
a plurality of summing means (160, 162, 164, 166, 168) one for each of said plurality of output channels each said summing means receiving the output signals of a pair of said multiplier means and producing at its output one of said plurality of output signals,

the decoder having said variable matrix values so constructed as to reduce directionally encoded audio com-

ponents in outputs which are not directly involved in reproducing them in the intended direction and enhance directionally encoded audio components in the outputs which are directly involved in reproducing them in the intended direction so as to maintain constant total power for such signals, while preserving separation between the left and right channel components of non-directional signals regardless of the said steering signals, and maintaining the loudness defined as the total audio power level of non-directional signals effectively constant whether or not directionally encoded signals are present and regardless of their intended direction if present,
characterized in that
at least two different modes of operation are provided and the matrix coefficients are controlled differently by said steering signals in the different modes of operation,
wherein a film mode of operation is optimized for reproduction of surround-encoded audio signals derived from film soundtracks and other video sources, and a music mode of operation is optimized for reproduction of musical recordings or broadcasts,
wherein for decoding in the film mode the matrix values for the left and right front outputs are so constructed to eliminate or attenuate the center component of the input signals, and the matrix values for the center output are so constructed that the attenuation of the center output starts at least 4dB greater than previously standard Dolby Pro-Logic decoders, and reduces as the center/surround steering signal becomes more positive, the center/surround signal varying between -45 and 45 degrees, wherein a value of -45 degrees corresponds to a signal from the rear and a value of 45 degrees corresponds to a signal from the center, the intermediate matrix values being determined by the requirement of keeping the power ratio of the center component to the uncorrelated component of the input signals identical at the outputs of the decoder.

2. The decoder of claim 1 wherein said plurality of output channels is five, identified as left front, center, right front, left surround and right surround.

3. The decoder of claim 2, further comprising frequency-dependent variable filter means (182, 184, 188, 190) following said left surround and right surround outputs so as to vary the frequency response and phase response of the outputs in a prescribed manner said variation being controlled by a number of control signals responding to the presence of surround or background ambience components detected in said left and right audio input signals.

4. The decoder of claim 2 further comprising frequency-dependent variable filter means (182, 184, 188, 190) and additional delay means (202, 208) following said left and right surround outputs for providing from each said surround output a side and a rear output channel such as to vary the frequency and phase responses of the several outputs in a prescribed manner said variation being controlled by a number of controls signal responding to the presence of surround or background ambience components detected in said left and right audio input signals.

5. The decoder of claim 3 or 4 wherein said control signals are:

   a center-surround control signal responsive to the ratio of in-phase center signal components to surround or antiphase signal components contained in said left and right audio input signals; and
   a background control signal responsive to the presence of antiphase signal component contained in said left and right audio input signals during periods when no strongly steered signals are present.

6. The decoder of claim 1, wherein for decoding music sources the matrix values for the center output are so constructed that the center attenuation starts at least 4dB greater than a standard Dolby Pro-Logic decoder, and reduces gradually to the maximum value for a standard Dolby Pro-Logic decoder, a value reached at a center/surround steering signal value of about 20 degrees, the attenuation then holding relatively constant as the steering value increases, and where the left and right front matrix values are so constructed that the center component of the input signals is not maximally removed from these outputs, but is deliberately adjusted to preserve at the output of the decoder the power ratio of the center component to the uncorrelated component of the input signals, the action of the center and left and right front elements being additionally limited at the center/surround steering value that results in an approximately 6dB difference in level between the center output and either the left or right front output.

7. The decoder of claim 1 where the left and right front matrix elements are so constructed that an input signal encoded to the rear, such that the direction lies between the left rear direction and right rear direction, produces no output from the front outputs.

8. The decoder of claim 1 where the left and right front matrix elements are so constructed that there is a level boost of about 3dB for signals that have no net left/right component, but have a center/surround steering value of about

22 degrees, said level boost reducing to zero as the center/surround steering value decreases to zero, increases to 45 degrees, or as the left/right steering value increases from zero to +/-45 degrees.

9. The decoder of claim 4 or 5 further comprising additional circuitry that creates a background control signal by detecting the direction of the background sound between notes or syllables in the input material, said background control signal rapidly being set to the value of the said center-surround steering signal when the center-surround steering signal is negative, and said background control signal being slowly set positive when the direction of the background sound between notes and syllables is in the forward direction, said background control signal tending to hold a negative value when surround encoded material is played, and a positive or zero value when standard two channel material is played.

10. The decoder of claim 9, where the said background control signal is used to control the relative loudness of the front and the rear outputs, such that the loudness of the rear outputs is reduced when the background between notes is either neutral or positive in direction.

11. The decoder of claim 9, where the said background control signal controls a variable low pass filter in the rear outputs such that the cutoff frequency is set to a user adjustable value when the background direction signal is positive or zero, and to rises to a high value when the background direction signal is negative, thus making the surround outputs less obtrusive when ordinary two channel material is played.

12. The decoder of claim 9, wherein the said background control signal controls a variable shelf filter, such that when the background control signal is positive or zero frequencies above 500Hz in the rear outputs are attenuated by a user adjustable value, and when the background control signal is negative this attenuation is reduced to zero, thus making the surround outputs less obtrusive when ordinary two channel material is played.

13. The decoder of claim 5 where the rear outputs of the matrix are split into a side output and a rear output by a combination of additional delay in the rear output, and a variable low pass filter in the side output, said low pass filter being set to a high frequency when the said center-surround steering signal is more positive than -22 degrees, and as the center-surround steering signal becomes more negative than -22 degrees the low pass frequency rapidly being reduced, to a final value of 500Hz when the center-surround steering signal reaches its minimum value of -45 degrees.

14. The decoder of claim 4 where the left and right surround outputs of a five channel version of the decoder are additionally supplied with a variable filter that emulates the frequency response of the human head/pinnae system for sound sources that are more than 150 degrees in azimuth from the front, such that when the said center-surround steering become further negative the filter once again acts maximally, and then modifies itself slightly to correspond to the frequency response of the human head-pinnae system for sound sources fully to the rear as the center-surround steering signal reaches its minimum value of -45 degrees.

Patentansprüche

1. Raumklang-Decoder (90) zum Umverteilen eines Paars linker und rechter Audioeingangssignale, die richtungsabhängig codierte und richtungsunabhängige Komponenten umfassen, auf eine Mehrzahl von Ausgangskanälen (172, 174, 176, 178, 180) zur Wiedergabe über Lautsprecher, die einen Hörbereich umgeben, und umfassend Mittel zum Bestimmen des richtungsabhängigen Gehalts der linken und rechten Audiosignale und zum Erzeugen daraus mindestens eines linken/rechten Lenksignals und eines mittleren/Raumklang-Lenksignals, wobei der Decoder Folgendes umfasst:

linke und rechte Eingangsanschlüsse (92, 94) zum Empfangen der entsprechenden linken und rechten Audioeingangssignale;
linke und rechte Verzögerungsmittel (96, 118) zum Erzeugen verzögerter linker und rechter Audiosignale aus den linken und rechten Audioeingangssignalen;
eine Mehrzahl von Multiplikatormitteln (140, 142, 144, 146, 148, 150, 152, 154, 156, 158) gleich zweimal der Anzahl der Mehrzahl von Ausgangskanälen, die paarweise organisiert sind, wobei ein erstes Element jedes Paars das verzögerte linke Audiosignal empfängt und ein zweites Element das verzögerte rechte Audiosignal empfängt, wobei jedes der Multiplikatormittel sein Eingangsaudiosignal mit einem variablen Matrixkoeffizienten multipliziert, um ein Ausgangssignal bereitzustellen;

wobei der variable Matrixkoeffizient von einem der Lenksignale oder von beiden gesteuert wird; und
eine Mehrzahl von Additionsmitteln (160, 162, 164, 166, 168), und zwar eins für jeden der Mehrzahl von Ausgangskanälen, wobei jedes Additionsmittel die Ausgangssignale eines Paars der Multiplikatormittel empfängt und an seinem Ausgang eines der Mehrzahl von Ausgangssignalen erzeugt,
wobei die variablen Matrixwerte des Decoders derart ausgebildet sind, dass sie richtungsabhängig codierte Audiokomponenten in Ausgängen reduzieren, die nicht direkt daran beteiligt sind, diese in der beabsichtigten Richtung wiederzugeben, und richtungsabhängig codierte Audiokomponenten in den Ausgängen verstärken, die direkt daran beteiligt sind, diese in der beabsichtigten Richtung wiederzugeben, um eine konstante Gesamtleistung für diese Signale beizubehalten und dabei die Trennung zwischen den linken und rechten Kanalkomponenten von richtungsunabhängigen Signalen unabhängig von den Lenksignalen zu bewahren und die Lautstärke, die als der Gesamtaudioleistungspegel der richtungsunabhängigen Signale definiert ist, effektiv konstant zu halten, ob richtungsabhängig codierte Signale vorliegen oder nicht und unabhängig von ihrer gegebenenfalls vorliegenden beabsichtigten Richtung,
dadurch gekennzeichnet, dass
mindestens zwei verschiedene Betriebsarten bereitgestellt werden und die Matrixkoeffizienten von den Lenksignalen in den verschiedenen Betriebsarten unterschiedlich gesteuert werden,
wobei eine Filmbetriebsart für die Wiedergabe von raumklangcodierten Audiosignalen, die von Filmtonspuren und anderen Videoquellen abgeleitet werden, optimiert ist, und eine Musikbetriebsart für die Wiedergabe von Musikaufzeichnungen oder Musiksendungen optimiert ist,
wobei zum Decodieren in der Filmbetriebsart die Matrixwerte für die linken und rechten vorderen Ausgänge derart ausgebildet sind, dass sie die mittlere Komponente der Eingangssignale eliminieren oder abschwächen, und die Matrixwerte für den mittleren Ausgang derart ausgebildet sind, dass die Abschwächung des mittleren Ausgangs bei mindestens 4 dB größer startet als bei früheren standardmäßigen Dolby Pro Logic-Decodern und sich reduziert, wenn das mittlere/Raumklang- Lenksignal positiver wird, wobei das mittlere/Raumklang-Signal zwischen -45 und 45 Grad variiert, wobei ein Wert von -45 Grad einem Signal von hinten entspricht und ein Wert von 45 Grad einem Signal aus der Mitte entspricht, wobei die dazwischen liegenden Matrixwerte durch die Anforderung bestimmt werden, das Leistungsverhältnis der mittleren Komponente zu der unkorrelierten Komponente der Eingangssignale mit den Ausgängen des Decoders identisch zu halten.

2. Decoder nach Anspruch 1, wobei die Mehrzahl von Ausgangskanälen gleich fünf ist, die als vorne links, Mitte, vorne rechts, Raumklang links und Raumklang rechts identifiziert sind.

3. Decoder nach Anspruch 2, ferner umfassend frequenzabhängige variable Filtermittel (182, 184, 188, 190), die den linken und rechten Raumklangausgängen folgen, um den Frequenzgang und den Phasengang der Ausgänge in vorgeschriebener Art und Weise zu variieren, wobei die Variation durch eine Anzahl von Steuersignalen gesteuert wird, die auf das Vorliegen von Raumklang- oder Hintergrund-Umgebungskomponenten ansprechen, die in den linken und rechten Audioeingangssignalen erkannt werden.

4. Decoder nach Anspruch 2, ferner umfassend frequenzabhängige variable Filtermittel (182, 184, 188, 190) und zusätzliche Verzögerungsmittel (202, 208), die den linken und rechten Raumklangausgängen folgen, um aus jedem Raumklangausgang einen seitlichen und einen hinteren Ausgangskanal bereitzustellen, um die Frequenz- und Phasenausgänge der mehreren Ausgänge in vorgeschriebener Art und Weise zu variieren, wobei die Variation durch eine Anzahl von Steuersignalen gesteuert wird, die auf das Vorliegen von Raumklang- oder Hintergrund-Umgebungskomponenten ansprechen, die in den linken und rechten Audioeingangssignalen erkannt werden.

5. Decoder nach Anspruch 3 oder 4, wobei die Steuersignale folgende sind:

ein Mitte-Raumklang-Steuersignal, das auf das Verhältnis der phasengleichen Mitte-Signalkomponenten zu Raumklang- oder gegenphasigen Signalkomponenten, die in den linken und rechten Audioeingangssignalen enthalten sind, anspricht; und
ein Hintergrund-Steuersignal, das auf das Vorliegen einer gegenphasigen Signalkomponente anspricht, die in den linken und rechten Audioeingangssignalen während Zeiträumen, in denen keine stark gelenkten Signale vorliegen, enthalten ist.

6. Decoder nach Anspruch 1, wobei zum Decodieren von Musikquellen die Matrixwerte für den Mittenausgang derart ausgebildet sind, dass die Mitte-Abschwächung bei mindestens 4 dB größer startet als ein standardmäßiger Dolby Pro Logic-Decoder und sich allmählich auf den Höchstwert für einen standardmäßigen Dolby Pro Logic-Decoder reduziert, wobei dieser Wert bei einem Mitte/Raumklang- Lenksignalwert von ungefähr 20 Grad erreicht wird, wobei die Abschwächung dann relativ konstant bleibt, während der Lenkwert zunimmt, und wobei die linken und rechten vorderen Matrixwerte derart ausgebildet sind, dass die Mitte-Komponente der Eingangssignale nicht maximal aus diesen Ausgängen entfernt wird, sondern bewusst angepasst wird, um am Ausgang des Decoders das Leistungsverhältnis der Mitte-Komponente zu der unkorrelierten Komponente der Eingangssignale zu bewahren, wobei die Wirkung der mittleren und linken und rechten vorderen Elemente zusätzlich auf den Mitte/Raumklang- Lenkwert beschränkt ist, der zu einem Pegelunterschied von ungefähr 6 dB zwischen der mittleren Ausgabe und entweder der linken oder rechten vorderen Ausgabe führt.

7. Decoder nach Anspruch 1, wobei die linken und rechten vorderen Matrixelemente derart ausgebildet sind, dass ein Eingangssignal, das nach hinten codiert ist, so dass die Richtung zwischen der linken hinteren Richtung und der rechten hinteren Richtung liegt, keine Ausgabe aus den vorderen Ausgängen erzeugt.

8. Decoder nach Anspruch 1, wobei die linken und rechten vorderen Matrixelemente derart ausgebildet sind, dass es eine Pegelverstärkung von ungefähr 3 dB für Signale gibt, die keine linke/rechte Nettokomponente aufweisen, sondern einen Mitte/Raumklang- Lenkwert von ungefähr 22 Grad aufweisen, wobei sich die Pegelverstärkung auf Null reduziert, während der Mitte/Raumklang- Lenkwert auf Null absinkt, auf 45 Grad ansteigt oder während der linke/rechte Lenkwert von Null auf +/-45 Grad ansteigt.

9. Decoder nach Anspruch 4 oder 5, ferner umfassend zusätzliche Schaltungen, die ein Hintergrund-Steuersignal erstellen, indem sie die Richtung des Hintergrundklangs zwischen Noten oder Silben in dem Eingangsmaterial erkennen, wobei das Hintergrund-Steuersignal schnell auf den Wert des Mitte/Raumklang- Lenksignals eingestellt wird, wenn das Mitte/Raumklang- Lenksignal negativ ist, und wobei das Hintergrund-Steuersignal langsam positiv eingestellt wird, wenn die Richtung des Hintergrundklangs zwischen Noten und Silben nach vorne geht, wobei das Hintergrund-Steuersignal dazu neigt, einen negativen Wert zu halten, wenn raumklangcodiertes Material abgespielt wird, und einen positiven oder Nullwert zu halten, wenn standardmäßiges Zweikanalmaterial abgespielt wird.

10. Decoder nach Anspruch 9, wobei das Hintergrund-Steuersignal verwendet wird, um die relative Lautstärke der vorderen und der hinteren Ausgänge derart zu steuern, dass die Lautstärke der hinteren Ausgänge reduziert wird, wenn der Hintergrund zwischen Noten entweder neutraler oder positiver Richtung ist.

11. Decoder nach Anspruch 9, wobei das Hintergrund-Steuersignal ein variables Tiefpassfilter in den hinteren Ausgängen steuert, so dass die Grenzfrequenz auf einen benutzeranpassbaren Wert eingestellt wird, wenn das Hintergrund-Richtungssignal positiv oder gleich Null ist, und auf einen hohen Wert ansteigt, wenn das Hintergrund-Richtungssignal negativ ist, wodurch die Raumklangausgaben weniger auffällig werden, wenn normales Zweikanalmaterial abgespielt wird.

12. Decoder nach Anspruch 9, wobei das Hintergrund-Steuersignal ein variables Filter mit Kuhschwanz-Charakteristik derart steuert, dass wenn das Hintergrund-Steuersignal positiv oder gleich Null ist, Frequenzen oberhalb von 500 Hz in den hinteren Ausgängen durch einen benutzeranpassbaren Wert abgeschwächt werden, und wenn das Hintergrund-Steuersignal negativ ist, diese Abschwächung auf Null reduziert wird, wodurch die Raumklangausgänge weniger auffällig werden, wenn normales Zweikanalmaterial abgespielt wird.

13. Decoder nach Anspruch 5, wobei die hinteren Ausgänge der Matrix durch eine Kombination einer zusätzlichen Verzögerung in dem hinteren Ausgang und eines variablen Tiefpassfilters in dem seitlichen Ausgang in einen seitlichen Ausgang und einen hinteren Ausgang geteilt werden, wobei das Tiefpassfilter auf eine hohe Frequenz eingestellt wird, wenn das Mitte/Raumklang- Lenksignal positiver als -22 Grad ist, und wenn das Mitte/Raumklang-Lenksignal negativer als -22 Grad wird, die Frequenz schnell auf einen endgültigen Wert von 500 Hz reduziert wird, wenn das Mitte/Raumklang- Lenksignal seinen Mindestwert von -45 Grad erreicht.

14. Decoder nach Anspruch 4, wobei die linken und rechten Raumklangausgänge einer Fünfkanalversion des Decoders zusätzlich mit einem variablen Filter versehen sind, das für Klangquellen, die sich auf mehr als 150 Grad Azimut von vorne befinden, den Frequenzgang des menschlichen Kopf-/Ohrmuschelsystems emuliert, so dass, wenn das Mitte/Raumklang- Lenken negativer wird, das Filter wieder maximal wirkt, und sich dann geringfügig ändert, um für Klangquellen ganz nach hinten dem Frequenzgang des menschlichen Kopf-/Ohrmuschelsystems zu entsprechen, wenn das Mitte/Raumklang- Lenksignal seinen Mindestwert von -45 Grad erreicht.

Revendications

1. Décodeur (90) de son ambiophonique permettant de redistribuer une paire de signaux d'entrée audio gauche et droit incluant des composantes codées de manière directionnelle et non directionnelle dans une pluralité de canaux de sortie (172, 174, 176, 178, 180) pour la reproduction à travers des haut-parleurs entourant une zone d'écoute, et incorporant des moyens pour déterminer la contenu directionnel desdits signaux audio gauche et droit et pour produire à partir de celui-ci au moins un signal de direction gauche/droit et un signal de direction central/ambiophonique, le décodeur comprenant :

des bornes d'entrée (92, 94) gauche et droite pour recevoir lesdits signaux d'entrée audio gauche et droit correspondants ;
des moyens de retard gauche et droit (96, 118) pour produire des signaux audio gauche et droit retardés à partir desdits signaux d'entrée audio gauche et droit :

une pluralité de moyens multiplicateurs (140, 142, 144, 146, 148, 150, 152, 154, 156, 158) égaux à deux fois le nombre de ladite pluralité de canaux de sortie, organisés par paires, un premier élément parmi chacune desdites paires recevant ledit signal audio gauche retardé et un deuxième élément recevant ledit signal audio droit retardé, chacun desdits moyens multiplicateurs multipliant son signal audio d'entrée par un coefficient de matrice variable pour fournir un signal de sortie :

ledit coefficient de matrice variable étant régulé par l'un ou par les deux desdits signaux de direction ; et une pluralité de moyens de sommation (160, 162, 164, 166, 168), un pour chacun de ladite pluralité de canaux de sortie, chacun desdits moyens de sommation recevant les signaux de sortie d'une paire desdits moyens multiplicateurs et produisant à sa sortie l'un parmi ladite pluralité de signaux de sortie,

le décodeur ayant lesdites valeurs de matrice variables construites de manière à réduire les composantes audio codées de manière directionnelle dans des sorties qui ne sont pas directement impliquées dans leur reproduction dans la direction voulue, et à augmenter les composantes audio codées de manière directionnelle dans les sorties qui sont directement impliquées dans leur reproduction dans la direction voulue, de manière à maintenir une puissance totale constante pour ces signaux, tout en préservant la séparation entre les composantes de canaux gauche et droite des signaux non directionnels, quels que soient lesdits signaux de direction, et à maintenir le correcteur physiologique, défini comme le niveau de puissance audio totale de signaux non directionnels effectivement constants, que des signaux à codage directionnel soient présents ou non et quelle que soit leur direction voulue, le cas échéant,
caractérisé en ce que
au moins deux modes de fonctionnement différents sont prévus et **en ce que** les coefficients de matrice sont régulés différemment par lesdits signaux de direction dans les différents modes de fonctionnement,
dans lequel un mode de fonctionnement film est optimisé pour la reproduction de signaux audio codés en ambiophonie dérivant de bandes son de films et d'autres sources vidéo, et un mode de fonctionnement musique est optimisé pour la reproduction d'enregistrements musicaux ou d'émissions,
dans lequel pour décoder en mode film, les valeurs de matrice pour les sorties avant gauche et droite sont construites de manière à éliminer ou atténuer la composante centrale des signaux d'entrée, et les valeurs de matrice pour la sortie centrale sont construites de manière à ce que l'atténuation de la sortie centrale commence au moins 4 dB au-dessus des décodeurs Dolby Pro-Logic standard antérieurs, et diminue quand le signal de direction central/ambiophonique devient plus positif, le signal central/ambiophonique variant entre -45 et 45 degrés, une valeur de -45 degrés correspondant à un signal provenant de l'arrière et une valeur de 45 degrés correspondant à un signal provenant du centre, les valeurs de matrice intermédiaires étant déterminées par l'exigence de maintenir le rapport de puissance de la composante centrale à la composante non corrélée des signaux d'entrée identique au niveau des sorties du décodeur.

2. Décodeur selon la revendication 1, dans lequel ladite pluralité de canaux de sortie comprend cinq éléments, identifiés comme avant gauche, central, avant droit, gauche ambiophonique et droit ambiophonique.

3. Décodeur selon la revendication 2, comprenant en outre des moyens (182, 184, 188, 190) de filtrage variables dépendant de la fréquence suivant lesdites sorties gauche ambiophonique et droite ambiophonique, de manière à faire varier la réponse en fréquence et la réponse en phase des sorties de manière prescrite, ladite variation étant régulée par un nombre de signaux de régulation répondant à la présence de composantes ambiophoniques ou d'ambiance de fond détectées dans lesdits signaux d'entrée audio gauche et droit.

4. Décodeur selon la revendication 2, comprenant en outre des moyens (182, 184, 188, 190) de filtrage variables dépendant de la fréquence et des moyens (202, 208) de retard supplémentaires suivant lesdites sorties ambiophoniques gauche et droite, afin de fournir, à partir de chacune desdites sorties ambiophoniques, un canal de sortie latéral et un canal de sortie arrière de manière à faire varier les réponses en fréquence et en phase des diverses sorties de manière prescrite, ladite variation étant régulée par un nombre de signaux de régulation répondant à la présence de composantes ambiophoniques ou d'ambiance de fond détectées dans lesdits signaux d'entrée audio gauche et droit.

5. Décodeur selon la revendication 3 ou 4, dans lequel lesdits signaux de régulation sont :

un signal de régulation central ambiophonique répondant au rapport des composantes de signaux centraux en phase aux composantes de signaux ambiophoniques ou d'antiphase contenus dans lesdits signaux d'entrée audio gauche et droit ; et
un signal de régulation d'arrière-plan répondant à la présence d'une composante de signal d'antiphase contenue dans lesdits signaux d'entrée audio gauche et droit pendant des périodes où aucun signal de direction fort n'est présent.

6. Décodeur selon la revendication 1, dans lequel pour le décodage de sources de musique, les valeurs de matrice pour la sortie centrale sont construites de manière à ce que l'atténuation centrale commence au moins 4 dB au-dessus d'un décodeur Dolby Pro-Logic standard, et diminue progressivement jusqu'à la valeur maximale pour un décodeur Dolby Pro-Logic standard, une valeur atteinte à une valeur de signal de direction central/ambiophonique d'environ 20 degrés, l'atténuation restant alors relativement constante lors de l'augmentation de la valeur de direction, et où les valeurs de matrice avant gauche et droite sont construites de manière à ce que la composante centrale des signaux d'entrée n'est pas supprimée au maximum de ces sorties, mais est délibérément ajustée pour préserver, à la sortie du décodeur, le rapport de puissance de la composante centrale à la composante non corrélée des signaux d'entrée, l'action des éléments avant centre et gauche et droit étant par ailleurs limitée à la valeur de direction centrale/ambiophonique qui résulte d'une différence d'environ 6dB de niveau entre la sortie centrale et la sortie avant gauche ou droite.

7. Décodeur selon la revendication 1, où les éléments de matrice avant gauche et droit sont construits de manière à ce qu'un signal d'entrée codé à l'arrière, tel que la direction réside entre la direction arrière gauche et la direction arrière droite, ne produise aucune sortie à partir des sorties avant.

8. Décodeur selon la revendication 1, dans lequel les éléments de matrice avant gauche et droit sont construits de manière à ce qu'il existe une amplification de niveau d'environ 3dB pour des signaux n'ayant pas de composante gauche/droite nette, mais ayant une valeur de direction centrale/ambiophonique d'environ 22 degrés, ladite amplification de niveau se réduisant à zéro quand la valeur de direction centrale/ambiophonique diminue jusqu'à à zéro, augmente à 45 degrés, ou quand la valeur de direction gauche/droite augmente de 0 à $\pm$45 degrés.

9. Décodeur selon la revendication 4 ou 5, comprenant en outre des circuits supplémentaires qui créent un signal de régulation d'arrière-plan par détection de la direction du son d'arrière-plan entre les notes ou les syllabes dans le matériel d'entrée, ledit signal de régulation d'arrière-plan étant rapidement fixé à la valeur dudit signal de direction central/ambiophonique quand le signal de direction central/ambiophonique est négatif, et ledit signal de régulation d'arrière-plan étant fixé lentement comme positif quand la direction du son d'arrière-plan entre les notes et les syllabes est dans le sens d'avancée, ledit signal de régulation d'arrière-plan tendant à maintenir une valeur négative lorsque du matériel codée ambiophonique est lu, et une valeur positive ou nulle quand du matériel à deux canaux standard est lu.

10. Décodeur selon la revendication 9, dans lequel ledit signal de régulation d'arrière-plan sert à réguler le correcteur physiologique relatif des sorties avant et arrière, si bien que le correcteur physiologique des sorties arrière est réduit quand la direction de l'arrière-plan entre les notes est soit neutre soit positive.

11. Décodeur selon la revendication 9, dans lequel ledit signal de régulation d'arrière-plan régule un filtre passe-bas variable dans les sorties arrière, si bien que la fréquence de coupure est fixée à une valeur ajustable par l'utilisateur lorsque le signal de direction d'arrière-plan est positif ou nul, et augmente à une valeur élevée quand le signal de direction d'arrière-plan est négatif, rendant ainsi les sorties ambiophoniques plus discrètes quand du matériel ordinaire à deux canaux est lu.

12. Décodeur selon la revendication 9, dans lequel ledit signal de régulation d'arrière-plan régule un filtre d'étagère variable, si bien que quand le signal de régulation d'arrière-plan est positif ou nul, les fréquences supérieures à 500 Hz des sorties arrière sont atténuées par une valeur ajustable par l'utilisateur, et que quand le signal de régulation d'arrière-plan est négatif, cette atténuation est réduite à zéro, rendant ainsi les sorties ambiophoniques plus discrètes quand du matériel ordinaire à deux canaux est lu.

13. Décodeur selon la revendication 5, dans lequel les sorties arrière de la matrice sont divisées en une sortie latérale et en une sortie arrière par une combinaison de retard supplémentaire dans la sortie arrière et de filtre passe-bas variable dans la sortie latérale, ledit filtre passe-bas étant réglé à une fréquence élevée quand le signal de direction ambiophonique central est plus positif que -22 degrés, et alors que le signal de direction central ambiophonique devient plus négatif que -22 degrés, la fréquence passe-bas est rapidement réduite à une valeur finale de 500 Hz quand le signal de direction ambiophonique central atteint sa valeur minimale de -45 degrés.

14. Décodeur selon la revendication 4, où les sorties ambiophoniques gauche et droite d'une version à cinq canaux du décodeur sont en outre dotées d'un filtre variable qui simule la réponse en fréquence du système tête/pavillons humain pour des sources sonores dépassant 150 degrés en azimut à partir de l'avant, si bien que quand ledit ordre de direction ambiophonique central devient plus négatif, le filtre agit une fois encore au maximum, puis se modifie légèrement pour correspondre à la réponse en fréquence du système tête-pavillons humain pour des sources sonores entièrement à l'arrière alors que le signal de direction ambiophonique central atteint sa valeur minimale de -45 degrés.

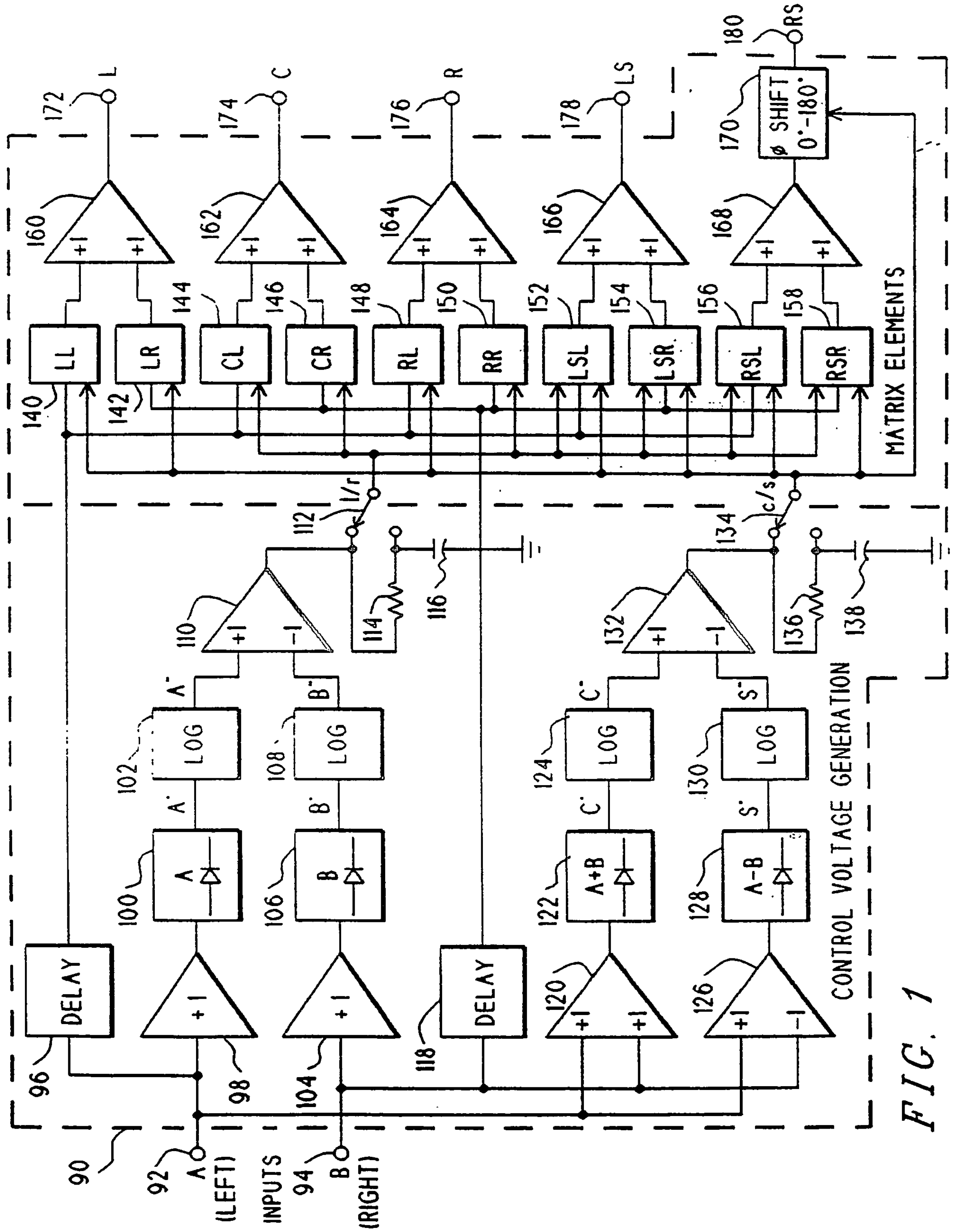

90
92
A (LEFT)
INPUTS
94
B (RIGHT)
96
DELAY
98
104
118
DELAY
100
A
106
B
102
LOG
108
LOG
A'
B'
110
112
l/r
114
116
120
126
122
A+B
128
A-B
124
LOG
130
LOG
C'
S'
132
134
c/s
136
138
CONTROL VOLTAGE GENERATION
140
LL
142
LR
144
CL
146
CR
148
RL
150
RR
152
LSL
154
LSR
156
RSL
158
RSR
MATRIX ELEMENTS
160
162
164
166
168
170
φ SHIFT 0°-180°
172
L
174
C
176
R
178
LS
180
RS

FIG. 1

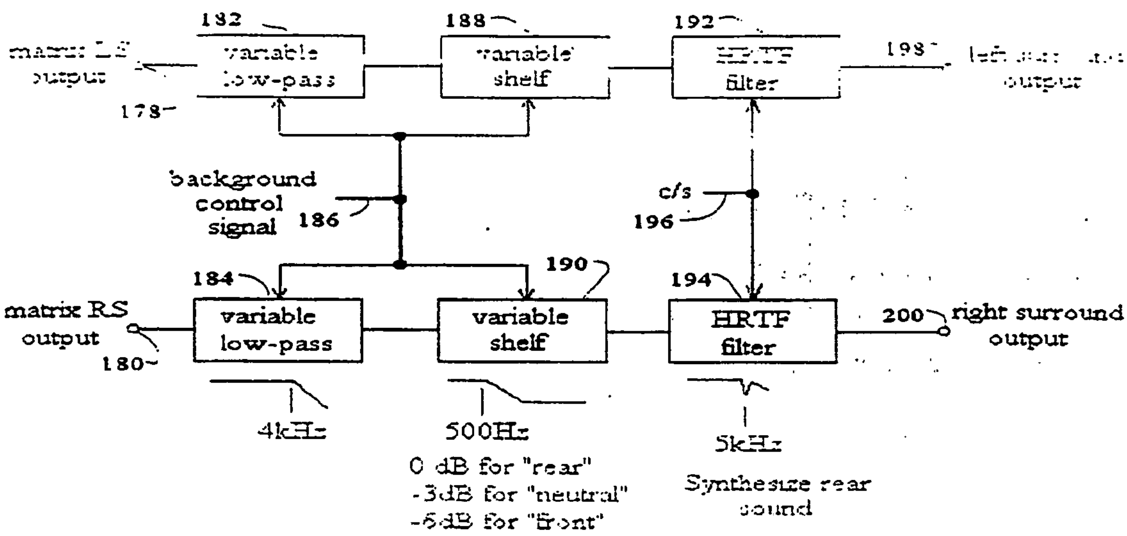

182
variable
low-pass
188
variable
shelf
192
filter
198
output
178
background
control
signal
186
c/s
196
184
190
194
matrix RS
output
180
variable
low-pass
variable
shelf
HRTF
filter
200
right surround
output
4kHz
500Hz
0 dB for "rear"
-3dB for "neutral"
-6dB for "front"
5kHz
Synthesize rear
sound

FIG. 2

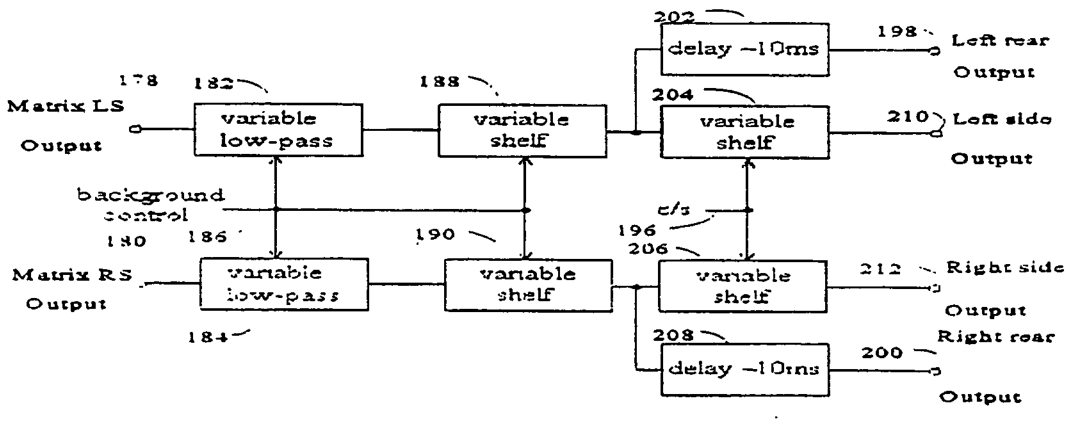

202
delay ~10ms
198
Left rear
Output
178
182
188
204
Matrix LS
Output
variable
low-pass
variable
shelf
variable
shelf
210
Left side
Output
background
control
180
186
190
c/s
196
206
Matrix RS
Output
variable
low-pass
variable
shelf
variable
shelf
212
Right side
Output
184
208
Right rear
delay ~10ms
200
Output

FIG. 3

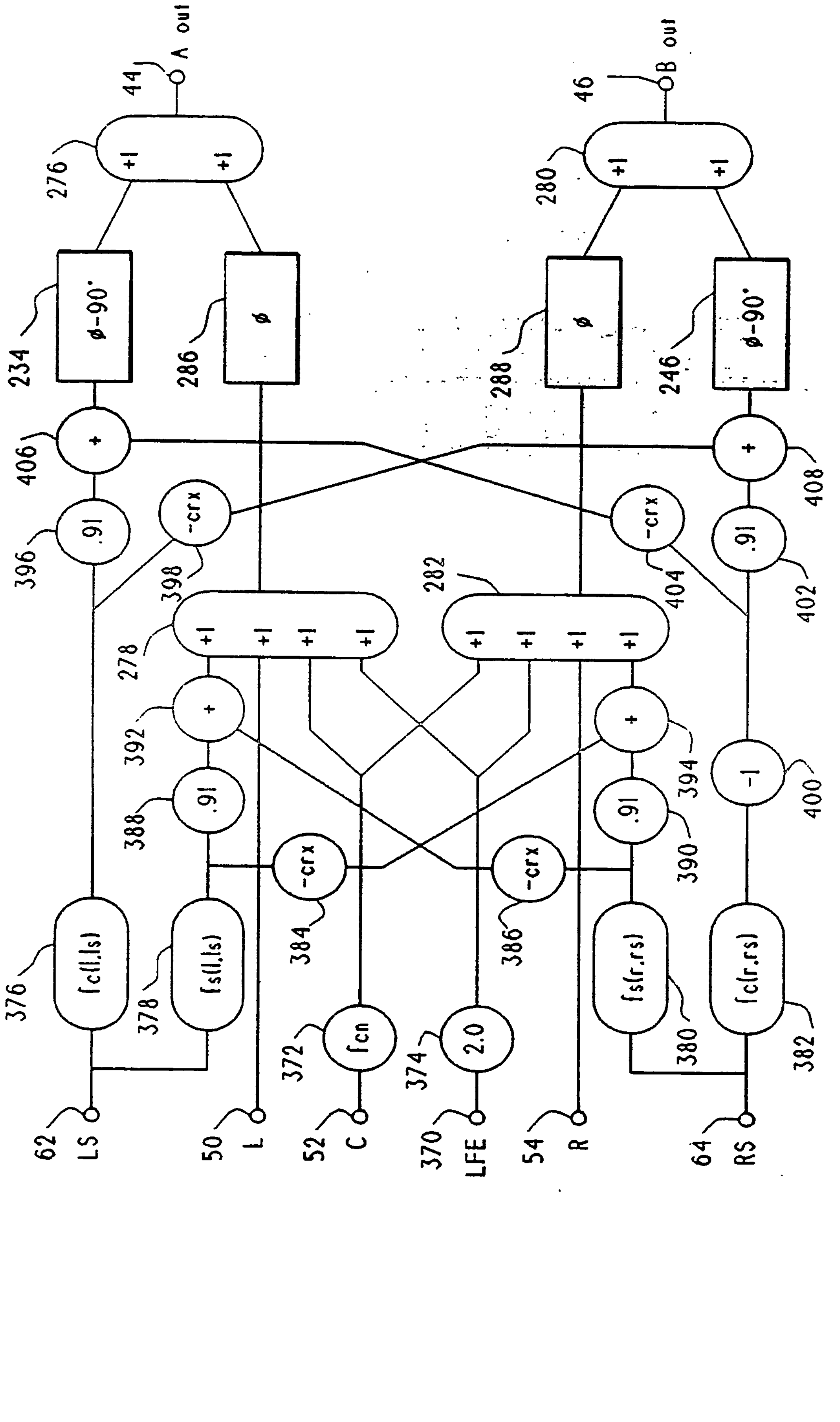

A out
B out
44
46
276
280
+1
234
286
288
246
ϕ-90°
ϕ
406
408
396
398
402
404
.91
-crx
+
278
282
392
394
388
390
384
386
400
-1
376
378
380
382
fc(l,ls)
fs(l,ls)
fs(r,rs)
fc(r,rs)
372
374
fcn
2.0
62
LS
50
L
52
C
370
LFE
54
R
64
RS

FIG. 4

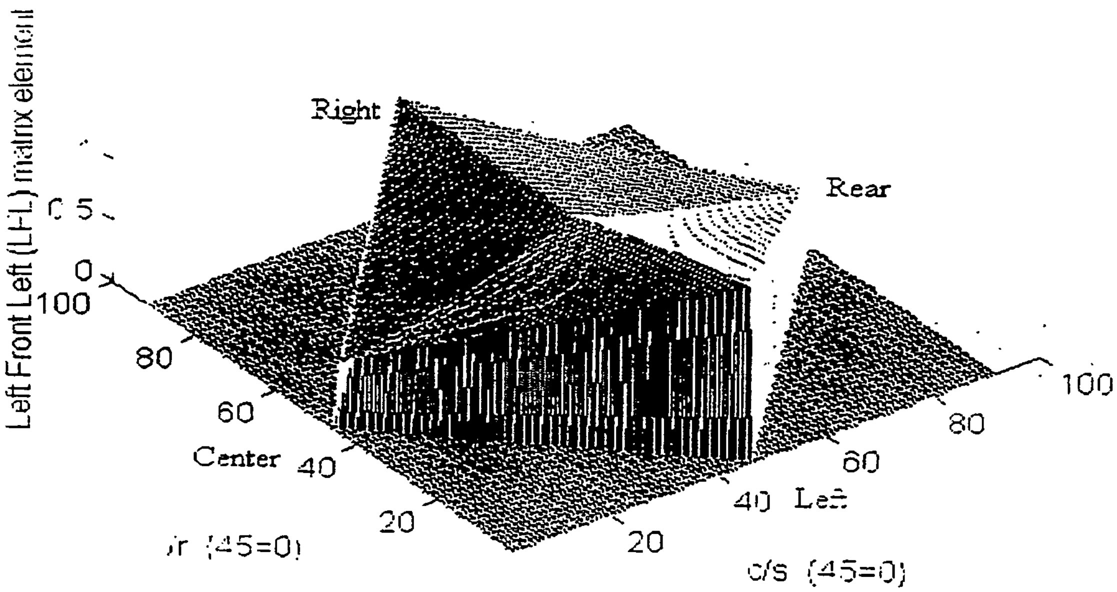

Left Front Left (LFL) matrix element
Right
Rear
0.5
0
100
80
60
Center
40
20
l/r (45=0)
20
40 Left
60
80
100
c/s (45=0)

FIG. 5

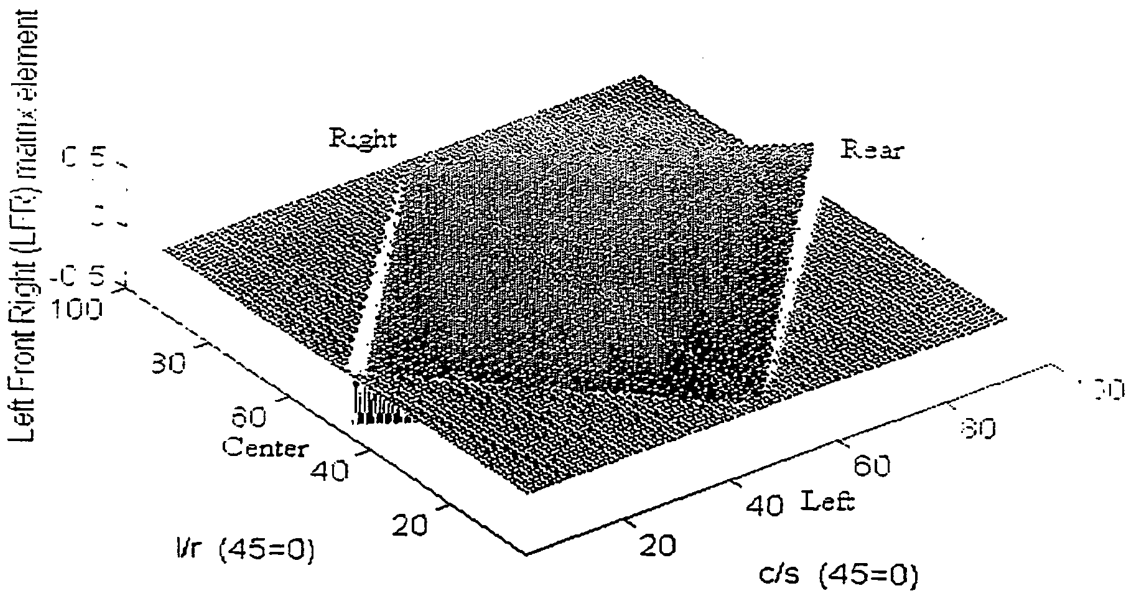

Left Front Right (LFR) matrix element
Right
Rear
0.5
-0.5
100
80
60
Center
40
20
l/r (45=0)
20
40 Left
60
80
100
c/s (45=0)

FIG. 6

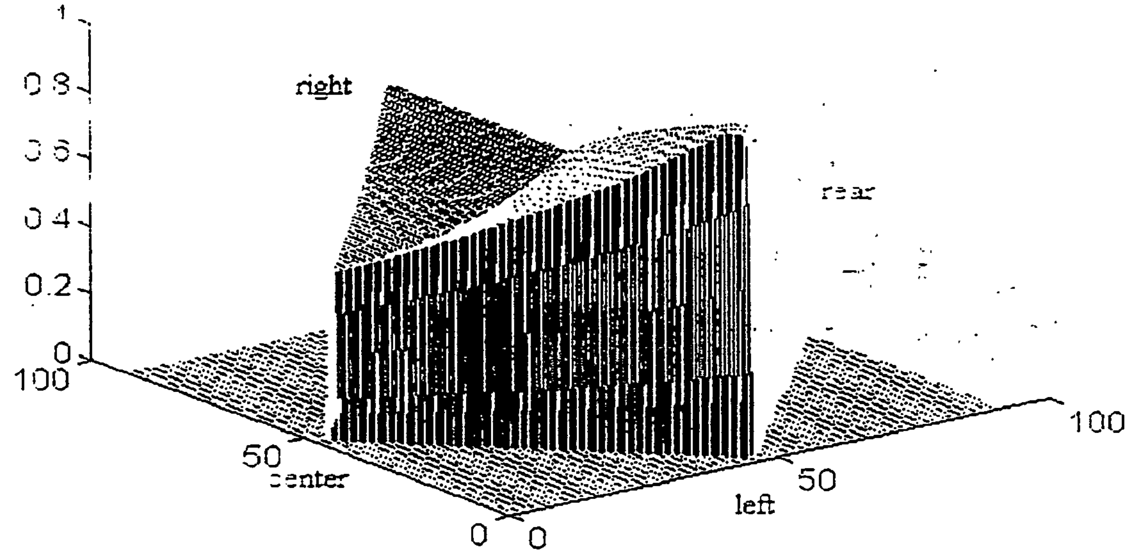

1
0.8
0.6
0.4
0.2
0
100
right
rear
50
center
0
0
left
50
100

FIG. 7

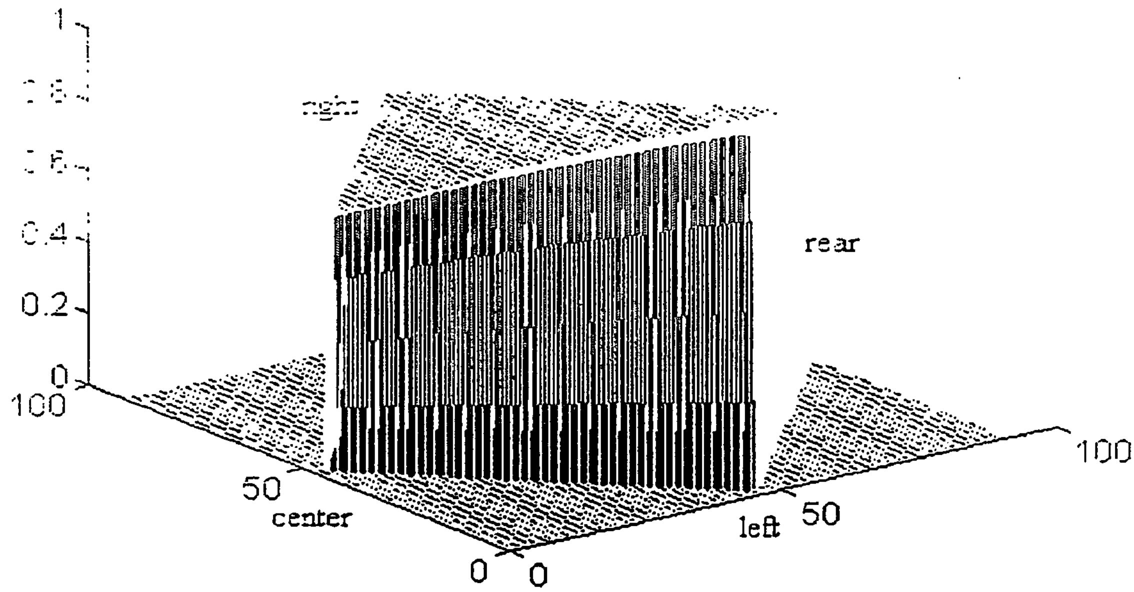

1
0.4
0.2
0
100
rear
50
center
0
0
left
50
100

FIG. 8

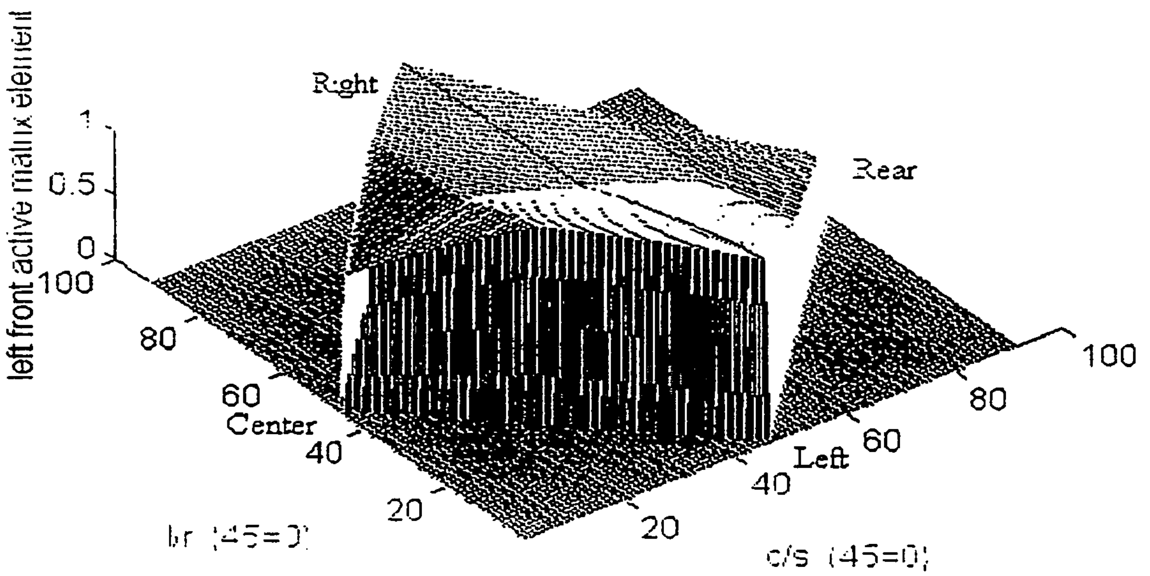

left front active matrix element
Right
Rear
1
0.5
0
100
80
60
Center
40
20
lr (45=0)
20
40
Left
60
80
100
c/s (45=0)

FIG. 9

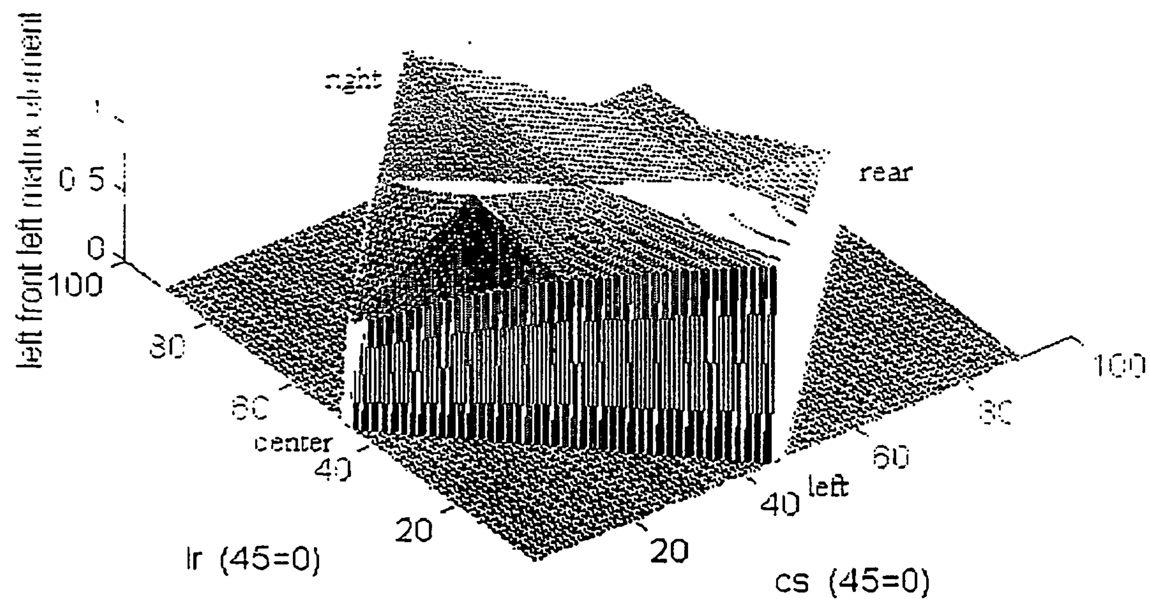

left front left matrix element
right
rear
0.5
0
100
80
60
center
40
20
lr (45=0)
20
40
left
60
80
100
cs (45=0)

FIG. 10

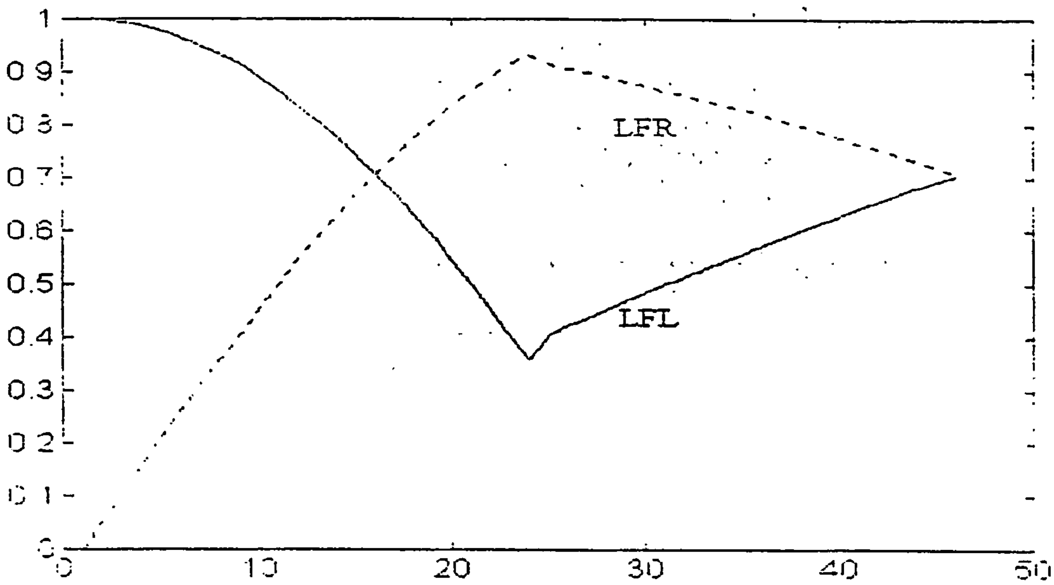

LFR
LFL

FIG. 11

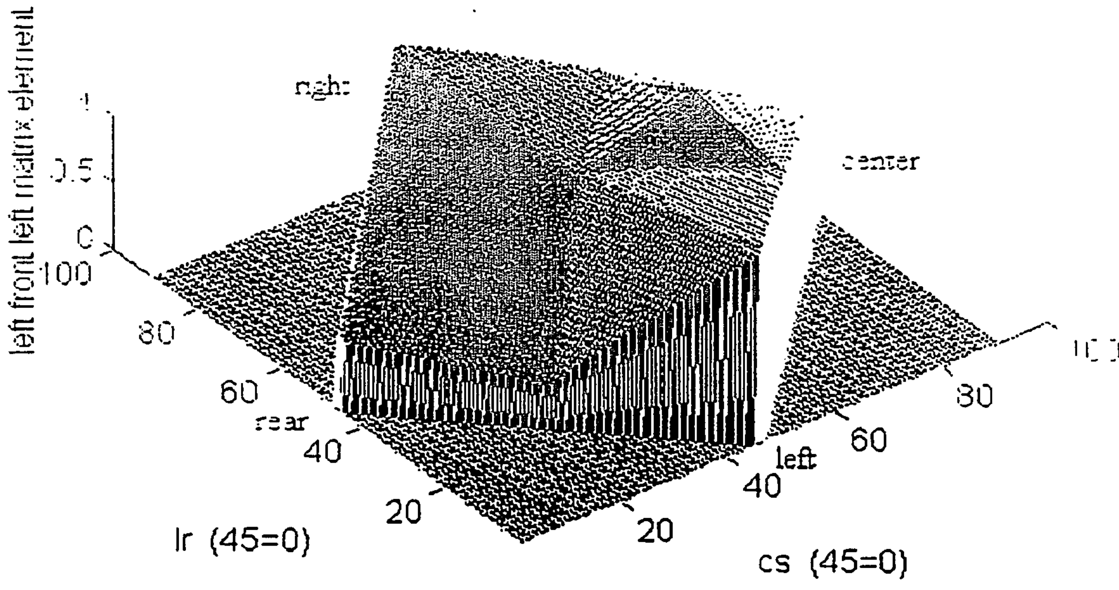

left front left matrix element
right
center
rear
left
lr (45=0)
cs (45=0)

FIG. 12

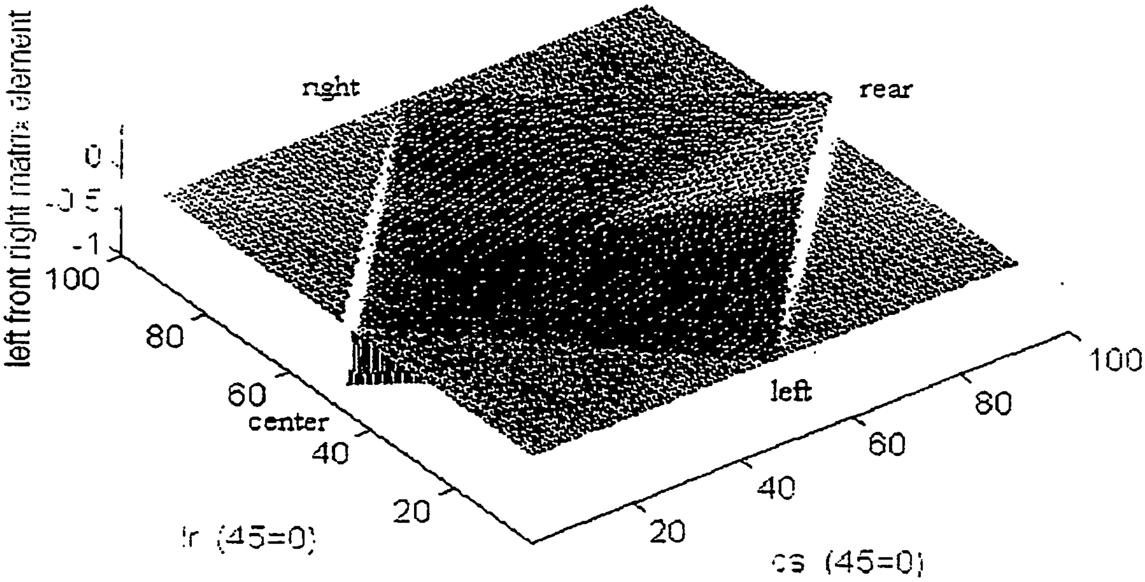

left front right matrix element
right
rear
0
-0.5
-1
100
80
60
center
40
20
lr (45=0)
left
100
80
60
40
20
cs (45=0)

FIG. 13

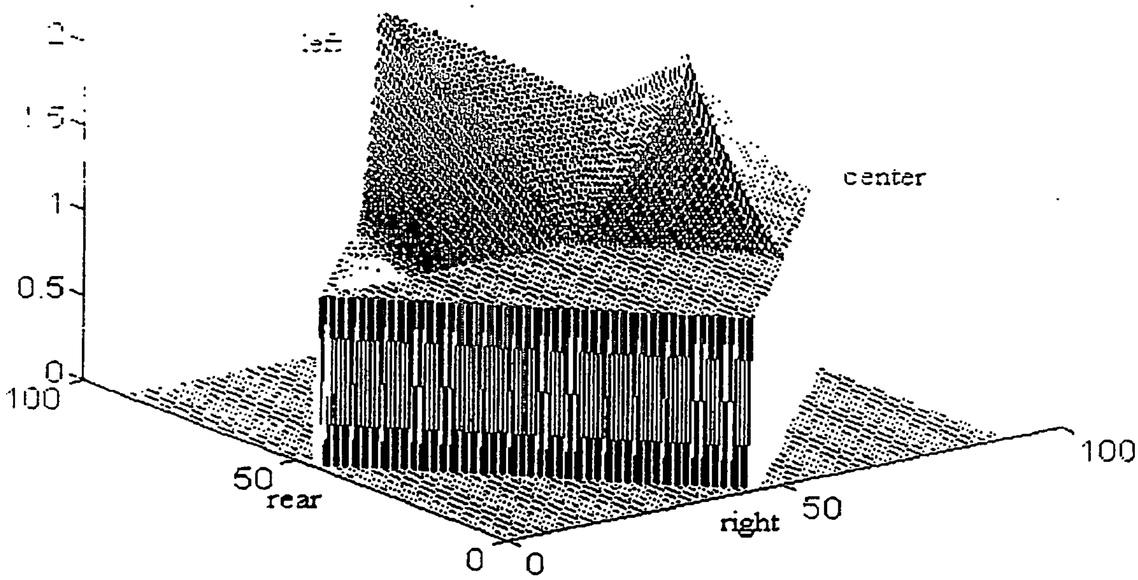

2
left
1.5
center
1
0.5
0
100
50
rear
0
0
right
50
100

FIG. 14

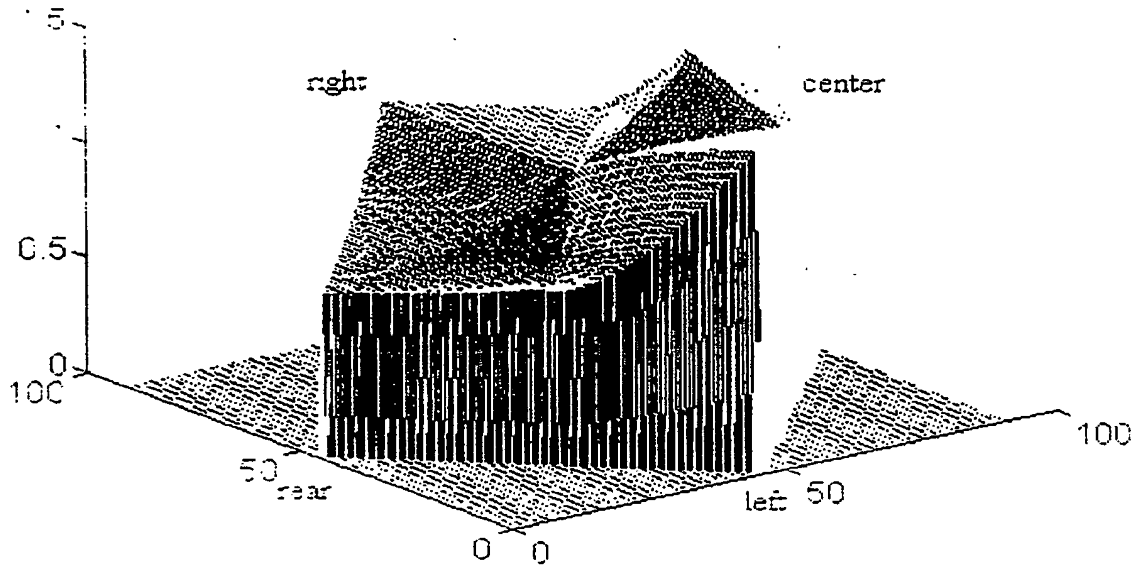

right
center
0.5
0
100
50
rear
0
0
left
50
100

FIG. 15

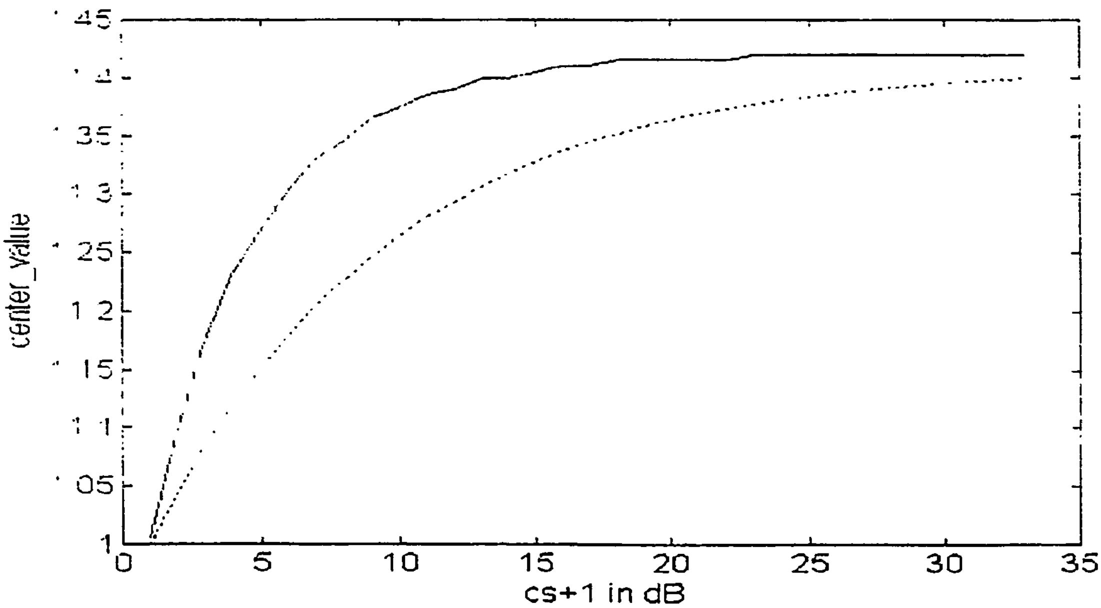

center_value
cs+1 in dB
0
5
10
15
20
25
30
35

FIG. 16

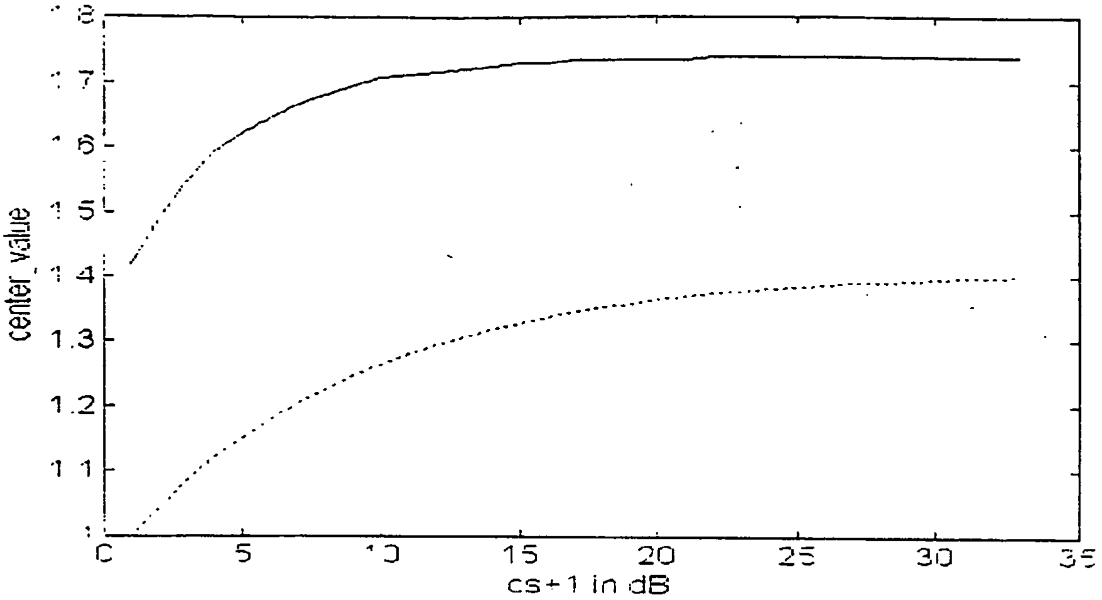

center_value
1.3
1.2
1.1
0
5
15
20
30
cs+1 in dB

FIG. 17

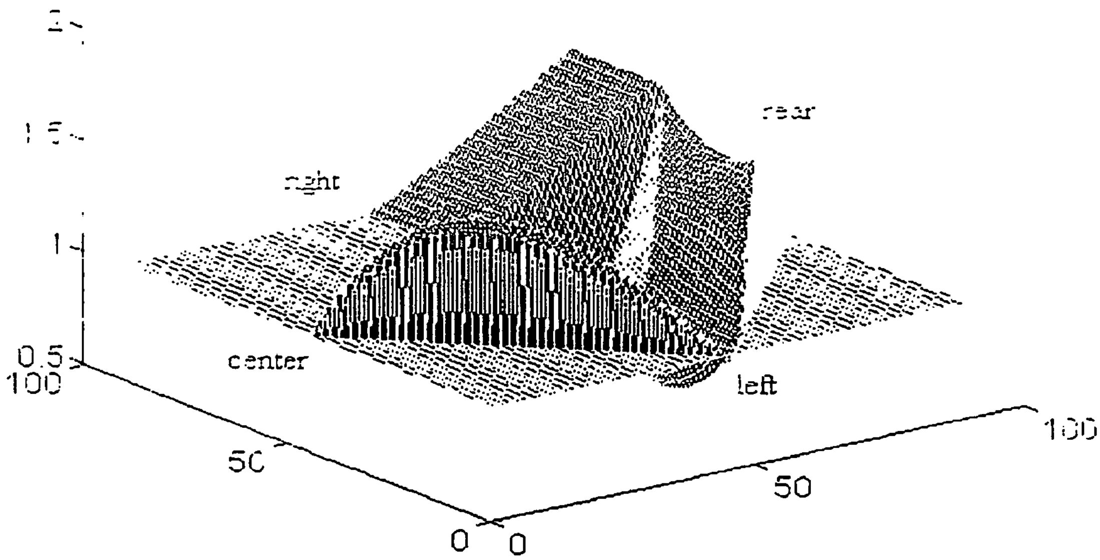
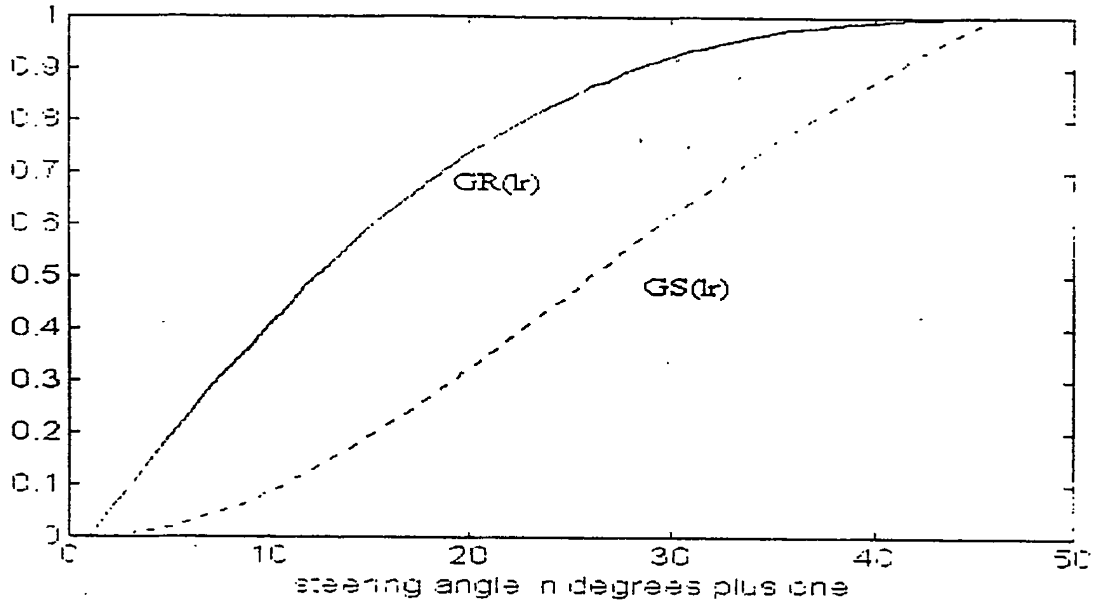
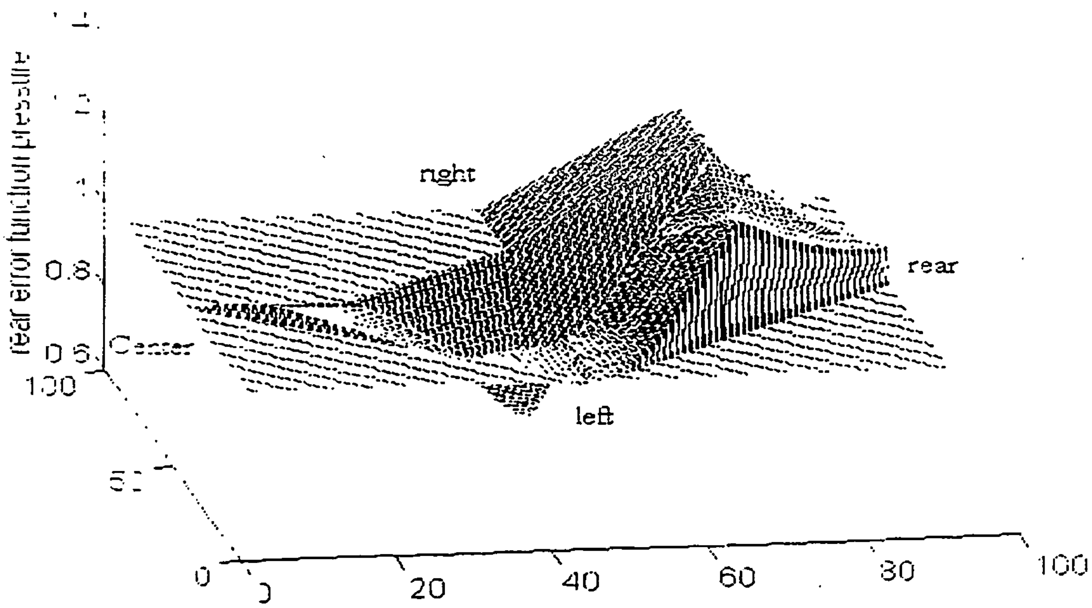

rear
center
left
0.5
100
50
0
0
50
100

FIG. 18



GR(lr)
GS(lr)
steering angle in degrees plus one

FIG. 19



rear error function pressure
right
rear
Center
left
0.8
0.6
100
50
0
20
40
60
80
100

FIG. 20

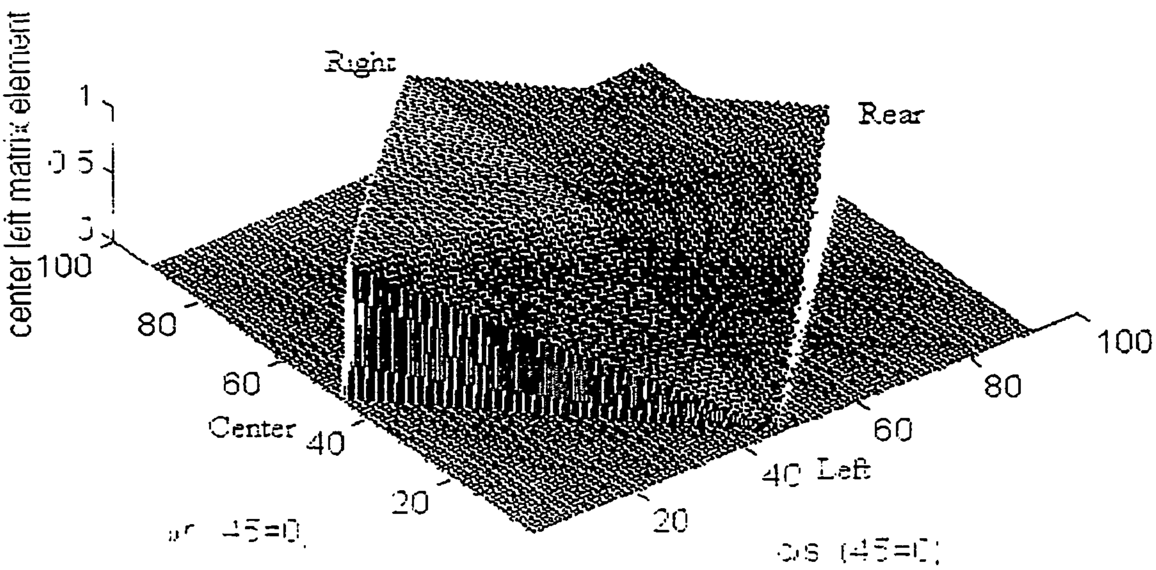

center left matrix element
1
100
80
60
Center
40
20
Right
Rear
100
80
60
40
Left
20

FIG. 21

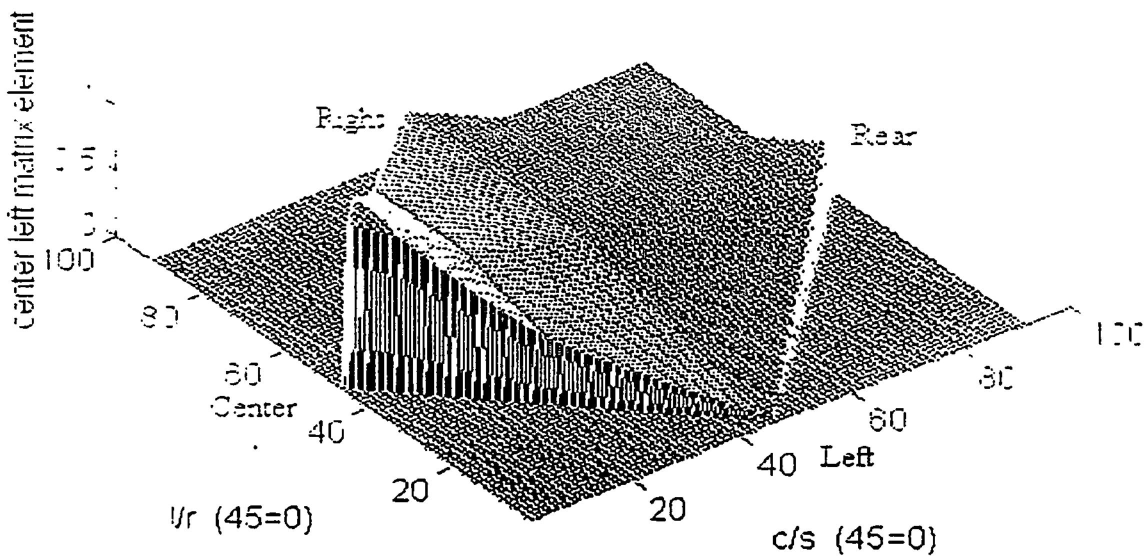

center left matrix element
100
60
Center
40
20
l/r (45=0)
Right
Rear
60
40
Left
20
c/s (45=0)

FIG. 22

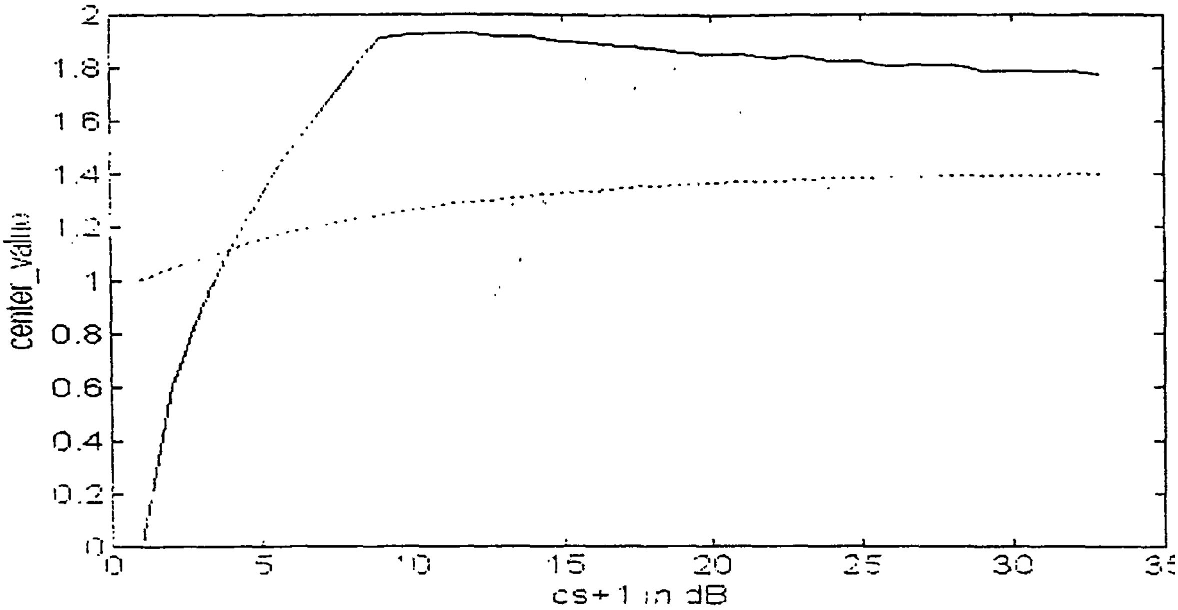

center_value
cs+1 in dB
0
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
5
10
15
20
25
30
35

FIG. 23

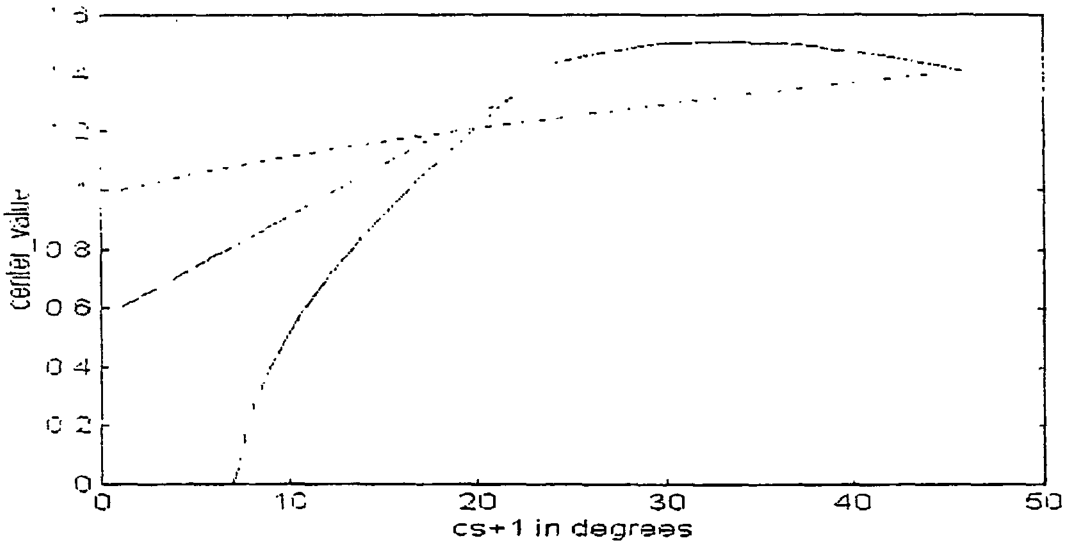

center_value
cs+1 in degrees
0
0.2
0.4
0.6
0.8
10
20
30
40
50

FIG. 24

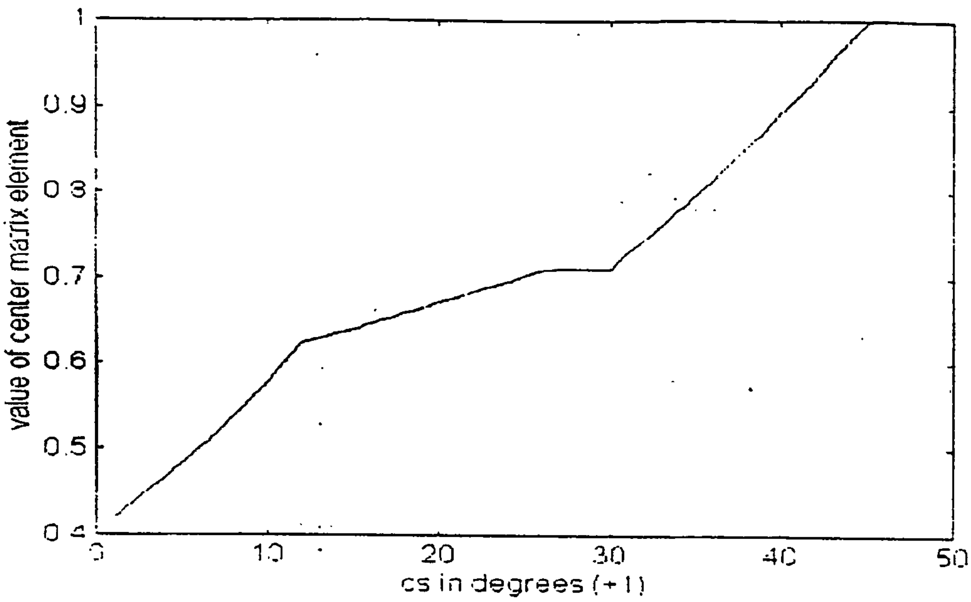

value of center matrix element
cs in degrees (+1)

FIG. 25

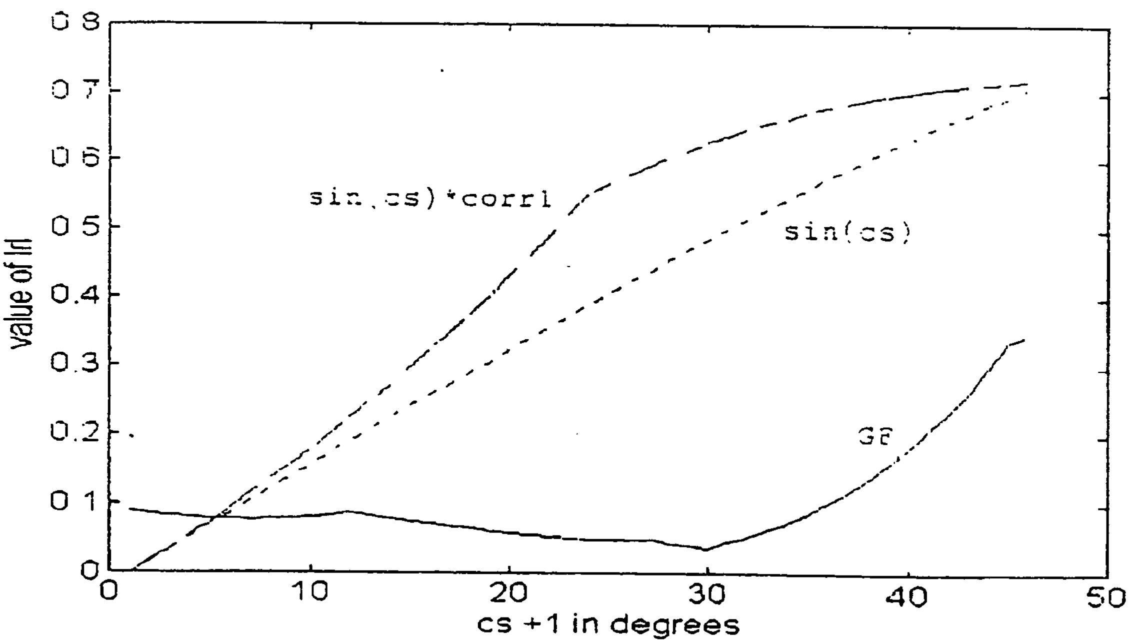

value of |r|
cs +1 in degrees
sin(cs)

FIG. 26

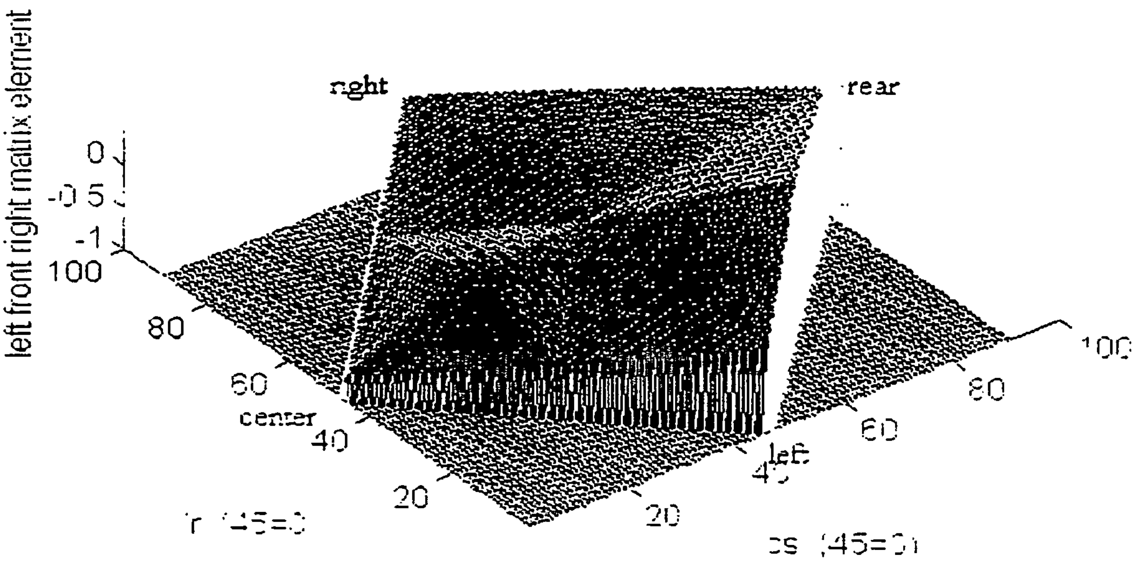

left front right matrix element
right
rear
0
-0.5
-1
100
80
60
center
40
20
20
40
left
60
80
100

FIG. 27

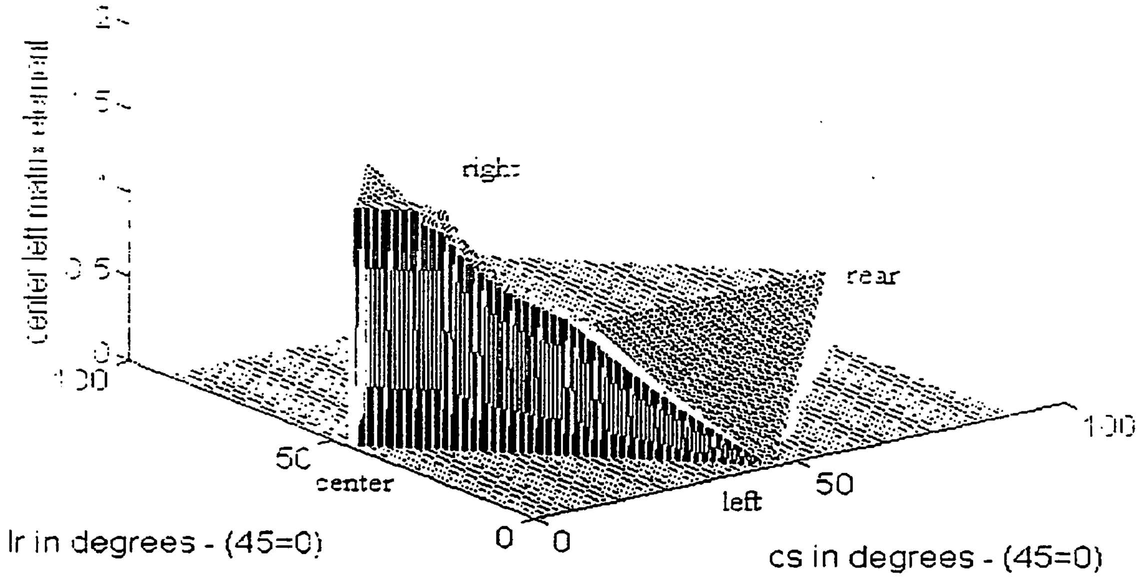

center left matrix element
right
rear
0.5
0
50
center
0
lr in degrees - (45=0)
0
left
50
100
cs in degrees - (45=0)

FIG. 28

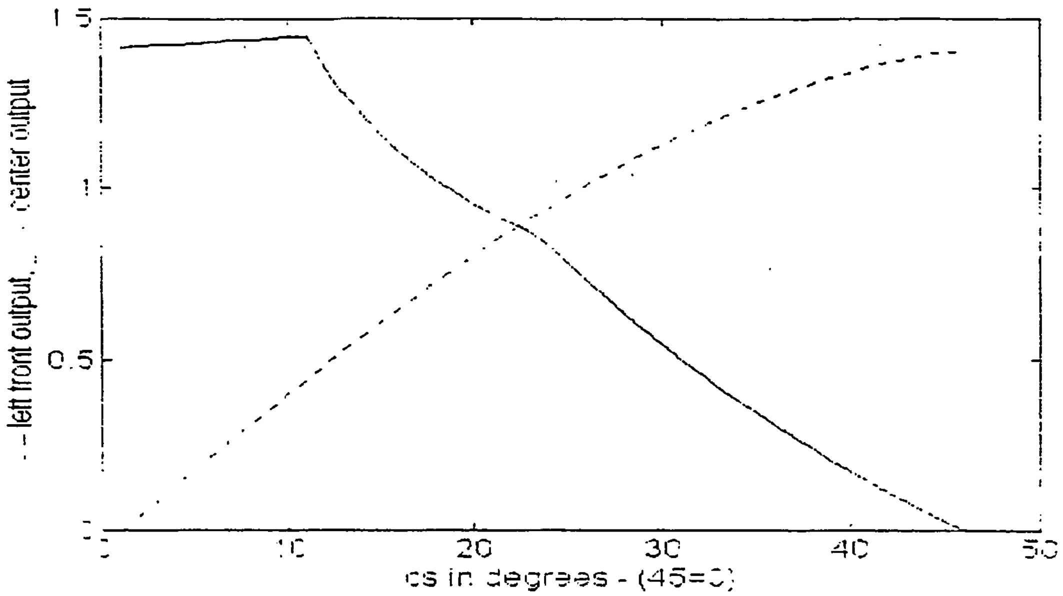

- - left front output, . . center output
1.5
1
0.5
0
0
10
20
30
40
50
cs in degrees - (45=0)

FIG. 29

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

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- US 5136650 A **[0004] [0038]**
- US 68494896 A, Griesinger **[0004]**
- US 74246096 A, Griesinger **[0004]**
- US 60058169 B **[0004]**
- US 742460 A **[0012] [0015] [0016] [0022] [0023] [0027] [0030] [0031] [0032] [0037] [0039] [0041] [0043] [0115]**
- US 08742460 B **[0013] [0016] [0063] [0105] [0122] [0144] [0157] [0182] [0203] [0212]**

Non-patent literature cited in the description

- **DAVID GRIESINGER.** Multichannel Matrix Surround Decoders for Two-Eared Listeners. *AES preprint # 4402,* October 1996 **[0005]**
- **DAVID GRIESINGER.** Progress in 5-2-5 Matrix Systems. *AES preprint # 4625,* September 1997 **[0005]**